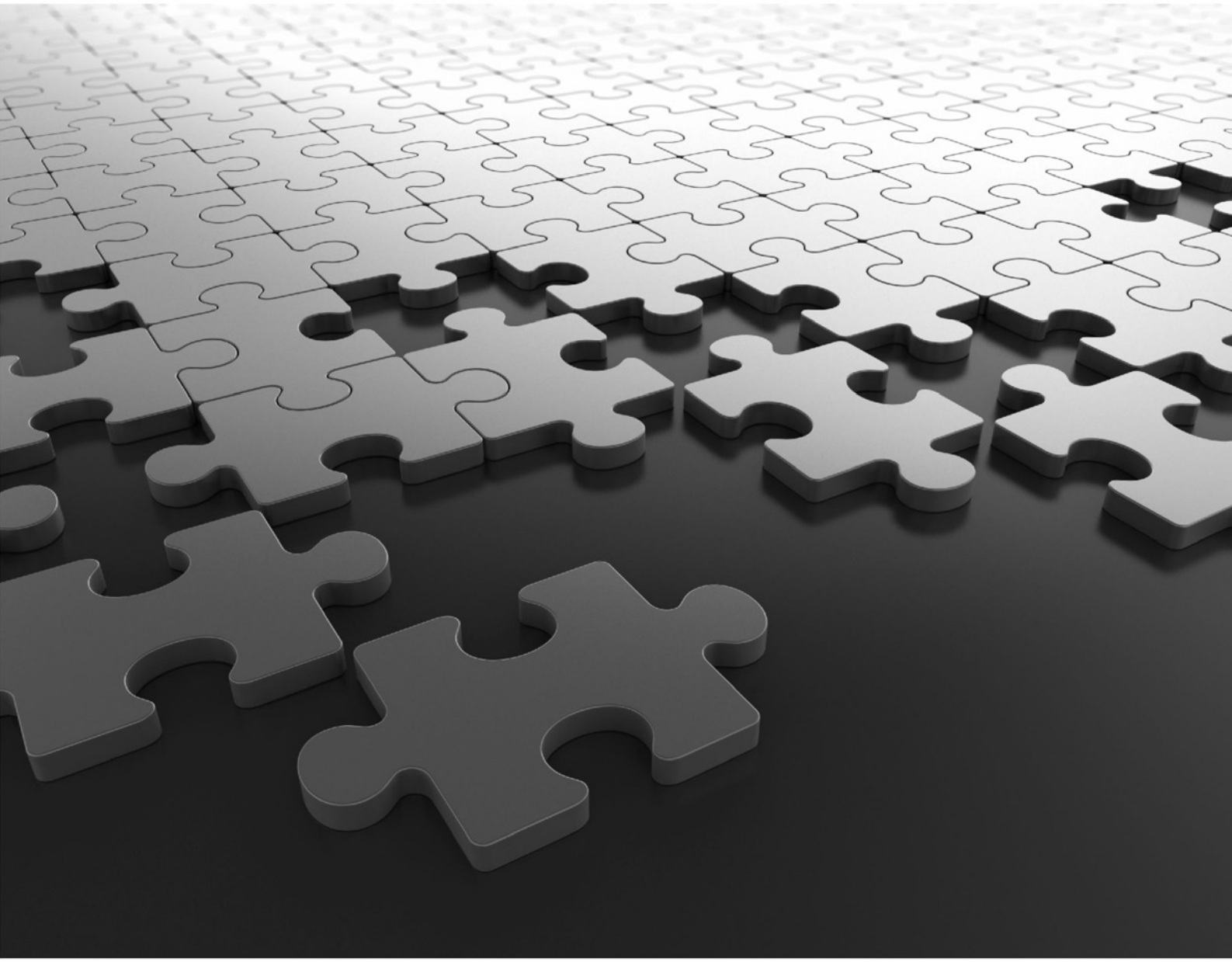
SPARSH VP710
User Guide



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SPARSH VP710

The Smart Video IP Deskphone

User Guide



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CHAPTER 1

Introduction

Welcome! Thank you for choosing the Matrix SPARSH VP710! You have now entered the exciting world of Internet Telephony. We hope you will make optimum use of this intelligent, feature-packed SIP-based IP phone. Please read this document carefully before installing and using the SPARSH VP710.

About this User Guide

This User Guide provides detailed information and instructions to install and operate Matrix SPARSH VP710.

Reference Documents

Refer to the Quick Start Guide in the package for:

- Basic setup of the phone
- Basic call features and phone customizations

The documentation can also be found at https://www.matrixtelesol.com/product-manuals.html

For product registration and warranty related details, please visit https://www.matrixcomsec.com/product-registration-form.html

Intended Audience

This document is aimed at,

- **System Engineers** and **System Administrators**, who will install, maintain and support SPARSH VP710. It is assumed that they have some experience in installing SIP phones, they are familiar with the VoIP technology and its functions, and they have basic knowledge of the various technical terms associated with it. They must be capable of configuring all the parameters of SPARSH VP710 and troubleshoot in-case it malfunctions.

- **End Users**, that is persons/organizations who will actually use SPARSH VP710. They include residential consumers, personnel of small and medium businesses, large enterprises, other commercial and public organizations/institutions.

  It is assumed that the End Users have some previous experience in operating key phones or mobile phones. End Users are not expected to configure the phone or program its features since they may not have the required technical understanding, but they must be capable to operate the phone. However, it is anticipated that some of them may have to or want to configure the elementary settings of the phone and/or program the features. Therefore, this document provides instructions on installation and configuration of the phone in a very simple and lucid manner.

Organization of this Document

This document contains following chapters:

- **Introduction** - This chapter provides an overview of this document, its purpose, intended audience, organization, terms and conventions used to present information and instructions.
- **Know your SPARSH VP710** - It describes the phone hardware components and software (User Interfaces) in details.
- **Information for your Safety and Comfort** - It describes some important measures for your personal safety as well as protection of the product.
- **Getting Started** - It provides step-by-step instructions on assembling the phone-body and handset, network and basic SIP configuration related information to connect the phone to the IP network.
- **Using the Touch Screen** - The IP phone supports three idle screens by default. You can swipe left or right to switch among different idle screens.
- **Customizing Your Phone** - It describes instructions regarding customizing the phone as per your requirement.
- **System Applications -** It provides basic operating instructions for using the system applications.The system applications like Camera, Browser, Clock etc are pre-installed on your IP phone and cannot be uninstalled,
- **Account Management -** It provides the information to configure the SIP account and Mobile Account of the IP phone in detail.
- **Accessing Basic Features** - It provides step-by-step instructions on using the basic features including making, receiving and managing calls once all necessary configurations are done.
- **Accessing Advanced Features** - It provides step-by-step instructions to access the advanced features of the phone. It is basically aimed for the technically sound users having primary knowledge of the SIP protocol and related features.
- **Configuring Audio Features** - It provides detailed information and instructions for configuring basic as well as advanced audio settings.
- **Configuring Video Features** - It provides detailed information and instructions for configuring basic as well as advanced video settings.
- **Voice Quality Monitoring** - Voice quality monitoring allows the IP phone to generate various quality metrics for listening and conversational quality.
- **Configuring Security Features** - It provides detailed information and instructions on how to secure your phone configuration and configure the security features.
- **Maintenance** - This chapter provides details regarding maintenance of your phone including firmware upgrade instructions, SNMP, TR-069 and advanced maintenance instructions.
- **Troubleshooting** - This chapter provides details regarding troubleshooting instructions of your phone.
- **Regulatory Notices** - It provides detailed information about regulatory notices.
- **Appendix** - It describes the physical features of the IP phone, Time Zones list, Glossary, Trusted Certificates, Configuration Parameters, Disclaimer.

How to Read this User Guide

This document is organized in a manner to help you get familiar with the phone, learn how to install it, connect it to the network and start making and receiving calls, use basic features, and move on to more advanced configuration and operation of the phone, all in the order of complexity. You are advised to read it in the sequence it is organized.

Instructions

The instructions in this document are written in bulleted lists, in a step-by-step format.

Cross references are provided in blue font, with hyperlinks to the headings. For example, you can refer to the relevant topic easily by clicking the text in blue font in this document.

Symbols Used

The following symbols have been used for notices to draw your attention to important items.



Important: *to indicate something that requires your special attention or to remind you of something you might need to do when you are using the phone.*



Caution: *to indicate an action or condition that is likely to result in malfunction or damage to the phone or your property.*



Warning: *to indicate a hazard or an action that will cause damage to the phone and/or cause bodily harm.*



Tip: *to indicate a helpful hint giving you an alternative way to operate the phone or carry out a procedure more efficiently.*

Terminology

The technical terms and Acronyms used in this document are standard terms, commonly used for the industries involved in dealing with telecommunication and/or data communication devices. Considering the broad group of intended users of this document, wherever possible, use of jargon is avoided.

Acronyms have been defined in the text and a complete list of Acronyms used in this document is provided in “Appendix E - Acronyms”.

The words IP phone, phone, VoIP phone or SIP phone are used interchangeably and synonymously to mean Matrix SPARSH VP710. Some of the terms specific to this document that you will encounter frequently are defined below:

- **ITSP:** Abbreviated form of Internet Telephony Service Provider. You can register the SIP account of SPARSH VP710 with the SIP servers supported by an ITSP.
- **OFF-Hook:** The phone goes in OFF-Hook state when you,
  - Lift the Handset from the cradle.

    Or

    - Press the Speaker key to activate the Speaker mode.

      Or

    - Press the Headset key to activate the Headset mode.

- **ON-Hook:** The phone goes in ON-Hook state when you,

    - Replace the Handset on the cradle.

      Or

    - Press the Speaker key to deactivate the Speaker mode.

      Or

    - Press the Headset key to deactivate the Headset mode.

- **Calling party/Caller:** The person who makes the call.
- **Called party/Callee:** The person to whom the call is made.
- **User:** The person who is in possession of the SPARSH VP710 IP phone and uses it.
- **Remote user/Remote party/Remote end/Far end:** The person with whom the user interacts.
- **Phone/IP Phone:** Both of these terms have been used synonymously in this document and refer to Matrix SPARSH VP710.

Using this document, we hope, you will be able to install, operate and make optimum use of SPARSH VP710. If you encounter any technical problems, contact your dealer/reseller or the Matrix Customer Care.

CHAPTER 2

Know Your SPARSH VP710

SPARSH VP710 is a Smart Video IP Deskphone engineered to deliver Video Conferencing Solution and Business Media Applications with a touch of Android experience. It is the next step towards collaboration offering flexibility and convenience providing lifelike unified communication. Deskphone is the perfect client to extract the capability of server and with a smarter deskphone it is optimizing the return over investment by utilizing the UC features to the extent.

Matrix SPARSH VP710 offers innovative features and sophisticated looks combining the best of both worlds targeted towards business-class professionals. The Smart Deskphone empower customers with latest technologies such as Wi-Fi, HDMI port, Bluetooth and USB connectivity bringing the world to their desk. With smart features as Internet Connectivity, Touch-screen functionality and Multimedia Apps, SPARSH VP710 is a new generation deskphone in line with smartphones, tablets and laptops.

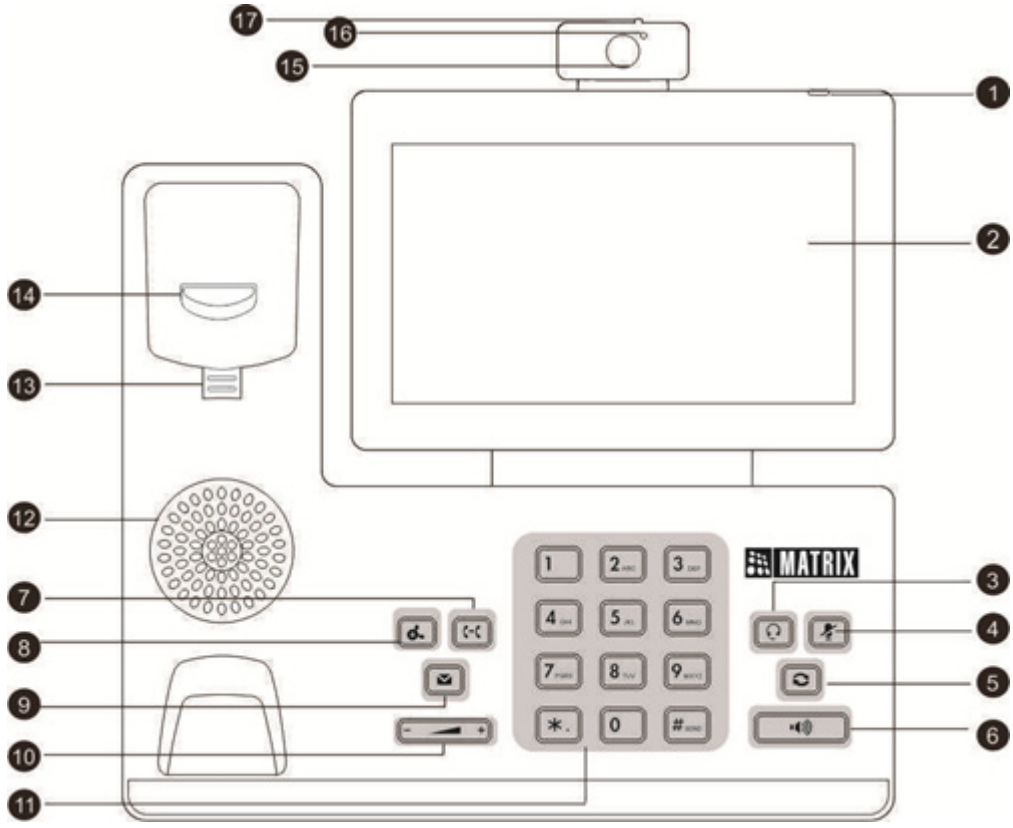


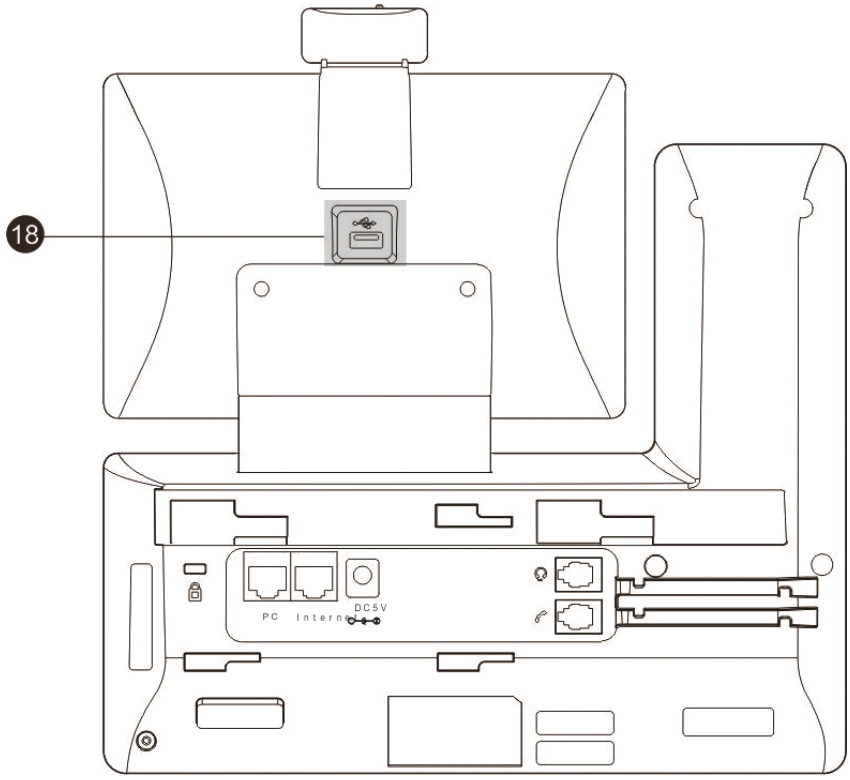
For a list of advanced technical information about the IP phone, refer "Appendix".

This chapter provides the overview of the SPARSH VP710 IP phone. If you require additional information or assistance with your new phone, contact your reseller/vendor.

Hardware Component Instructions

The main hardware components of the SPARSH VP710 IP phone are the touch screen, the keypad and the USB camera.





Hardware component instructions of the IP phone are:

Key Label	Item	Description
1	Power Indicator LED	Indicates call status, message status and phone's system status • Receives an incoming call-Fast-flashing • Receives a voice mail-Slow-flashing
2	Touch Screen	7 inch (1024 x 600) capacitive (5 point) touch screen. Tap to select and highlight screen items. The IP phone supports three idle screens by default. Shows information about calls, messages, time, date and other relevant data: • Default account • Call information—caller ID, call duration • Icons (for example,) • Missed call list or second incoming caller information • Prompt text (for example, "Save successfully!") • Time and date
3	HEADSET Key	Toggles and indicates the headset mode. The key LED illuminates solid green when headset mode is activated.
4	MUTE Key	Toggles and indicates mute feature. The key LED illuminates solid red when the call is muted.
5	REDIAL Key	Redials a previously dialed number.
6	Speakerphone Key	Toggles and indicates the hands-free (speakerphone) mode. The key LED illuminates solid green when the hands-free (speakerphone) mode is activated.
7	TRANSFER Key	Transfers a call to another party.
8	HOLD Key	Places a call on hold or resumes a held call.
9	MESSAGE Key	• Accesses voice mails. • Captures screenshots during a video call.
10	Volume Key	Adjusts the volume of the handset, headset, speaker, ringer or media. ringer or media.
11	Keypad	Provides the digits and special characters in context-sensitive applications.
12	Speaker	Provides hands-free (speakerphone) audio output.
13	Hookswitch Tab	Secures the handset in the handset cradle when the IP phone is mounted vertically. For more information on how to adjust the hookswitch tab, refer to the Quick Start Guide.

14	Hookswitch	• Picking up the handset from the handset cradle, the hookswitch bounces and the phone connects to the line. • Laying the handset down on the handset cradle, the phone disconnects from the line.
15	Camera Lens	2 Mega-pixel camera. Provides near-site video. The better distance between camera and images you want to capture should be in the range of 0.35 meters (1 foot) to 2 meters (6 feet).
16	Camera Indicator LED	Indicates video call status and camera status. • Receives a video call—Flashing green • The camera is inserted and detected successfully on the phone—Solid green
17	Shutter Switch	Covers and uncovers the camera. When the camera is switched off, the video image is black.
18	USB2.0 Port	Allows you to connect the USB flash drive/USB headset to your phone.

Icon Instructions

Feature Status Icons

Icons appearing on the status bar are described in the following table:

Icon	Description
	Wired network is unavailable
	Hands-free (speakerphone) mode
	Handset mode
	Headset mode
	Video/Voice Mail
	Auto Answer
	Do Not Disturb
	Call Forward
	Silent mode
	Camera is not detected

	Keep Mute
	Phone Lock
	Missed Calls
	Bluetooth mode is on
	Bluetooth headset is both paired and connected
	Bluetooth-Enabled mobile phone is both paired and connected
	Wi-Fi mode is on
	Wi-Fi is connected successfully and the signal strength is weak to strong
	The USB flash drive is detected
	Screenshot captured
	Downloading file
	Uploading file
	Upcoming alarm
	Unread email
	Warning

Line Key Icons

Icon indicator (associated with line key features):

Icon	Description
	Hold/Public Hold Private Hold
	DND
	Voice Mail
	Direct Pickup

	Group Pickup
	DTMF Prefix
	LDAP Local Group XML Group
	Conference
	Forward
	Transfer
	ReCall
	Record URL Record
	Recording starts successfully (Record/URL Record)
	Multicast Paging Group Listening Paging List
	Hot Desking
	Zero Touch
	URL
	Phone Lock
	Directory
	Speed Dial

Icon	Description
	The private line registers successfully
	Register failed
 Flashing	Registering

	The shared line registers successfully
	DND is enabled on this line
	Call forward is enabled on this line
	Mobile Account (Bluetooth-Enabled mobile phone is both paired and connected)
	Mobile Account (Bluetooth-Enabled mobile phone fails to connect)
 Flashing	Mobile Account (Bluetooth-Enabled mobile phone is connecting)

Icon	Description
	BLF/BLF list idle state
 (Flashing)	BLF/BLF list ringing state
 (Flashing)	BLF/BLF list callout state
	BLF/BLF list talking state
	BLF hold state
	BLF list call park state
	BLF/BLF list failed state

Icon	Description
	Intercom idle state
	Intercom ringing state
 Callout	Intercom callout state

 Talking	Intercom talking state
	Intercom failed state

Icon	Description
	The shared line is idle.
	The shared line is seized.
	The shared line receives an incoming call.
	The shared line is dialing.
	The shared line is in conversation.
	The shared line conversation is placed on public hold.
 (for local SCA phone)  (for monitoring SCA phone)	The shared line conversation is placed on private hold.
	The shared line conversation is barged in by the other shared line party.
	For a multi-party call, all the shared line parties participating in this call place the shared line conversation on hold.

Other Icons

Icon indicator (associated with an active call)

Icon	Description
	Call Hold (video)
	Call Hold (audio-only)
	Call Mute

	Call is encrypted

Icon indicator (associated with call recording)

Icon	Description
	Recording box is full (Using a record key)
	A call cannot be recorded (Using a record key)
	Recording starts successfully (Using a record key)
	Recording cannot be started (Using a record key)
	Recording cannot be stopped (Using a record key)

Icon indicator (associated with call history)

Icon	Description
	Received Calls
	Placed Calls
	Missed Calls
	Forwarded Calls

Icon indicator (associated with contact)

Icon	Description
	The default local caller photo and local contact icon
	The default local caller photo and local contact icon

LED Instructions

Power Indicator LED

LED Status	Description
Solid red	The phone is initializing.
Fast flashing red	The phone is ringing.
Slow flashing red	The phone receives video/ voice mail.
Off	The phone is powered off. The phone is busy. The phone is idle. The call is placed on hold or is held. The call is mute.

Camera Indicator LED

LED Status	Description
Solid green	The phone is powered on and the camera is properly connected to the phone. The camera is idle. The phone receives an audio-only call.
Flashing green	The phone receives a video call.
Solid red	There is an active video call. The video call is muted. The video call is held.
Slow-flashing red	The shutter switch is open, but the near-site video is stopped transmitting during a video call. The video call is placed on hold.
Off	The phone is powered off. The camera is not properly connected to the phone. The shutter switch is closed.



The above table displays the default LED status. The status of the power indicator LED is configurable via web user interface. Refer "Power Indicator LED" for details.

CHAPTER 3

Information for Your Safety and Comfort

Save these instructions. Read these safety instructions before use!

The following basic safety precautions should always be followed to reduce risk of fire, electrical shock, and other personal injury.

General Requirements

- Before you install and use the device, read the safety instructions carefully and observe the situation during operation.
- During the process of storage, transportation, and operation, please always keep the device dry and clean.
- During the process of storage, transportation, and operation, please avoid collision and crash of the device.
- Please attempt not to dismantle the device by yourself. In case of any discrepancy, please contact the appointed maintenance center for repair.
- Without prior written consent, no organization or individual is permitted to make any change to the structure or the safety design of the device. Matrix is under no circumstance liable to consequences or legal issues caused by such changes.
- Please refer to the relevant laws and statutes while using the device. Legal rights of others should be respected as well.

Environmental Requirements

- Place the device at a well-ventilated place. Do not expose the device under direct sunlight.

- Keep the device dry and free of dusts.
- Place the device on a stable and level platform.
- Please place no heavy objects on the device in case of damage and deformation caused by the heavy load.
- Keep at least 10 centimeters distance between the device and the closest object for heat dissipation.
- Do not place the device on or near any inflammable or fire-vulnerable object, such as rubber-made materials.
- Keep the device away from any heat source or bare fire, such as a candle or an electric heater.
- Keep the device away from any household appliance with strong magnetic field or electromagnetic field, such as a microwave oven or a refrigerator.

Operating Requirements

- Do not let a child operate the device without guidance.
- Do not let a child play with the device or any accessory in case of accidental swallowing.
- Please use the accessories provided or authorized by the manufacturer only.
- The power supply of the device shall meet the requirements of the input voltage of the device. Please use the provided surge protection power socket only.
- Before plugging or unplugging any cable, make sure that your hands are completely dry.
- Do not spill liquid of any kind on the product or use the equipment near water, for example, near a bathtub, washbowl, kitchen sink, wet basement or near a swimming pool.
- Do not tread on, pull, or over-bend any cable in case of malfunction of the device.
- During a thunderstorm, stop using the device and disconnect it from the power supply. Unplug the power plug and the Asymmetric Digital Subscriber Line (ADSL) twisted pair (the radio frequency cable) to avoid lightning strike.

- If the device is left unused for a rather long time, disconnect it from the power supply and unplug the power plug.

- When there is smoke emitted from the device, or some abnormal noise or smell, disconnect the device from the power supply, and unplug the power plug immediately. Contact the specified maintenance center for repair.

- Do not insert any object into equipment slots that is not part of the product or auxiliary product.

- Before connecting a cable, connect the grounding cable of the device first. Do not disconnect the grounding cable until you disconnect all other cables.

Cleaning Requirements

- Before cleaning the device, stop using it and disconnect it from the power supply.

- Use a piece of soft, dry and anti-static cloth to clean the device.

- Keep the power plug clean and dry. Using a dirty or wet power plug may lead to electric shock or other perils.

CHAPTER 4

Getting Started

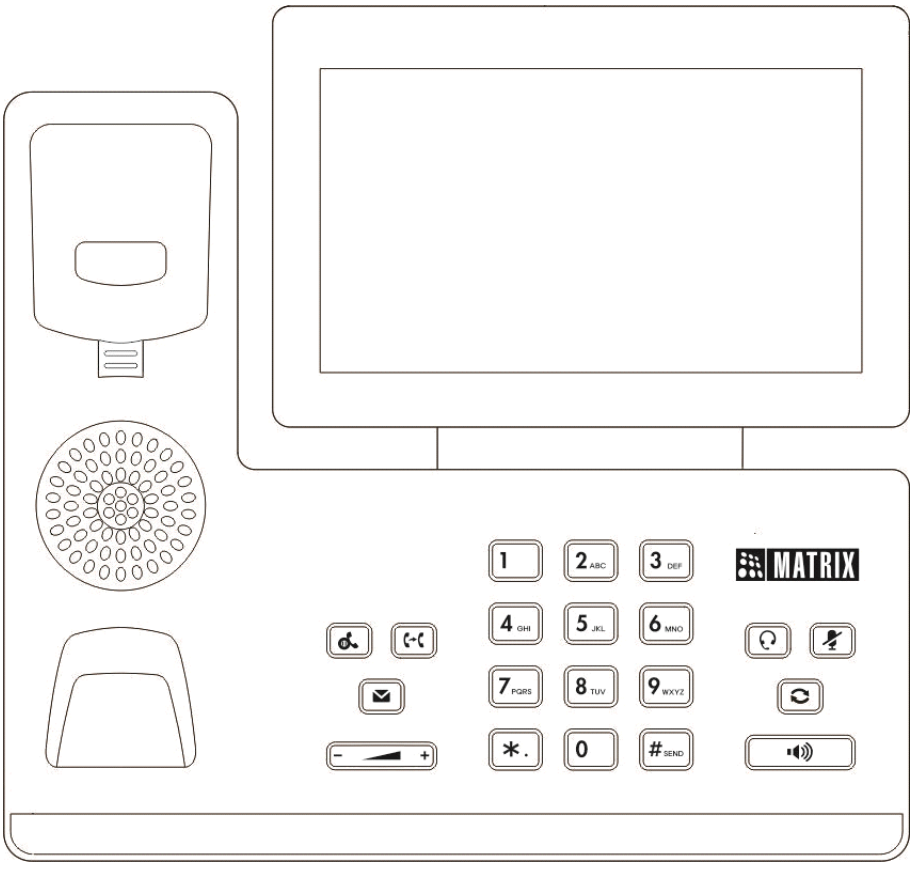
This chapter provides basic installation instructions and information for obtaining the best performance with the IP phone.

If you require additional information or assistance with your new phone, contact your re-seller/vendor.

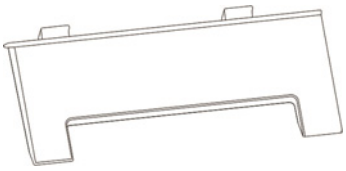
Packaging Contents

The following components are included in your IP phone package:

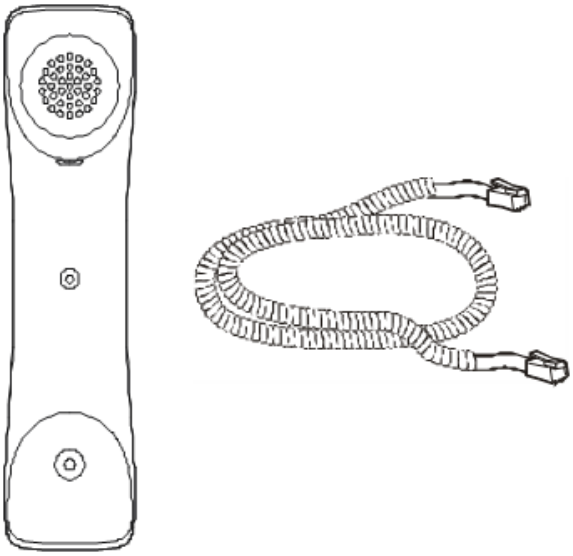
Matrix SPARSH VP710 IP Phone



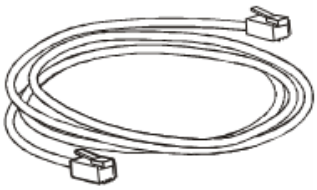
Phone Stand



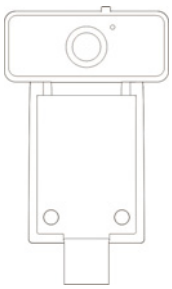
Handset & Handset Cord



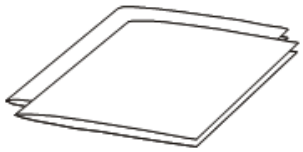
Ethernet Cable



Camera

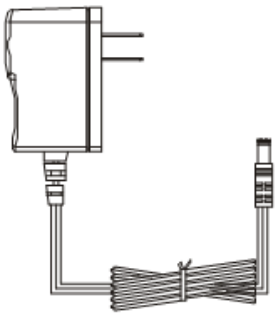


Quick Start Guide

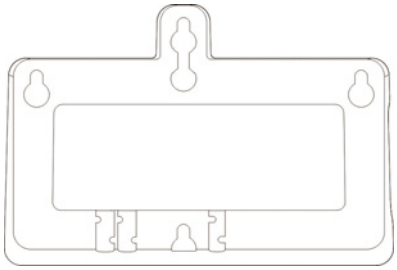


Optional Accessories

Power Adapter



Wall Mount Bracket



Check the list before installation. If you find anything missing, contact your reseller/vendor.

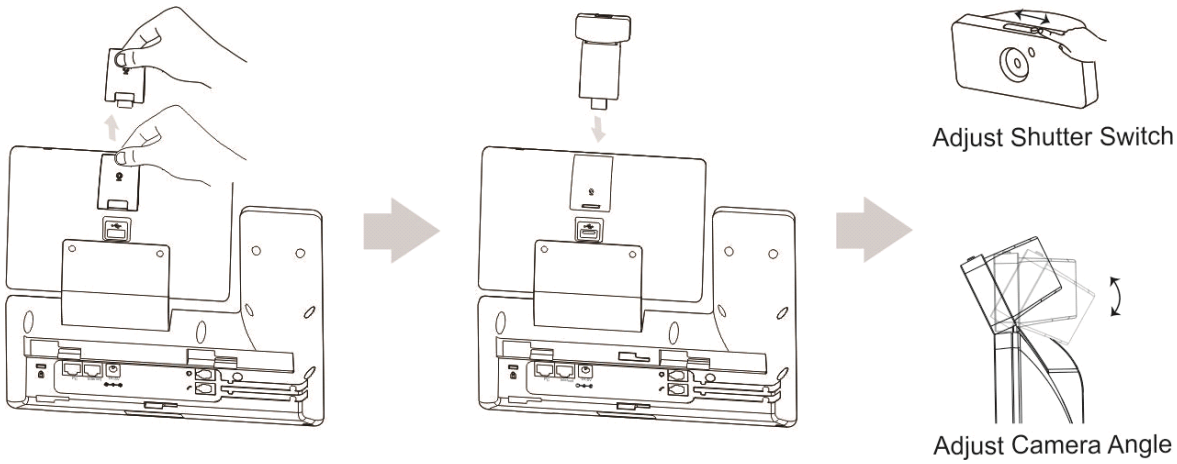
Phone Installation

If your phone is already installed, proceed to “Phone Initialization”.

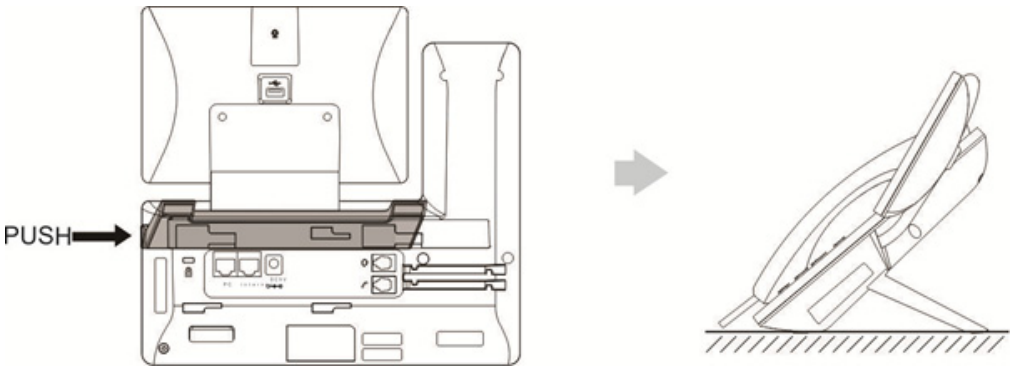
This section introduces how to install the IP phones with components in packaging contents.

1. Inserting the camera
2. Attaching the stand and the optional wall mount bracket
3. Adjusting the angle of touch screen
4. Attaching the optional wall mounting bracket
5. Connecting the handset and optional headset
6. Connecting the power and network
7. Connecting the optional USB flash drive

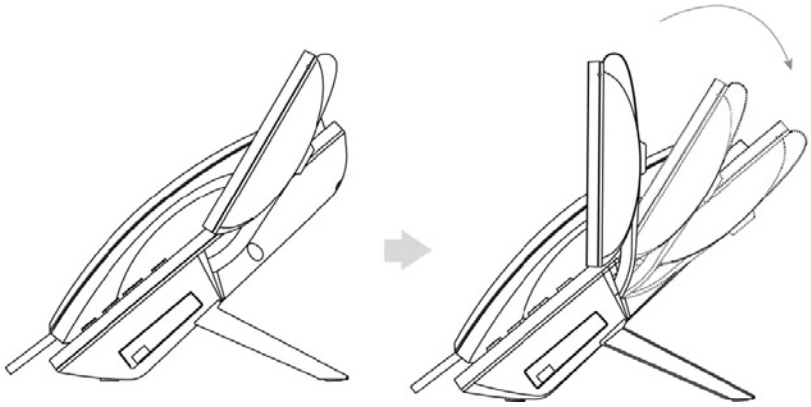
1. **Inserting the camera**



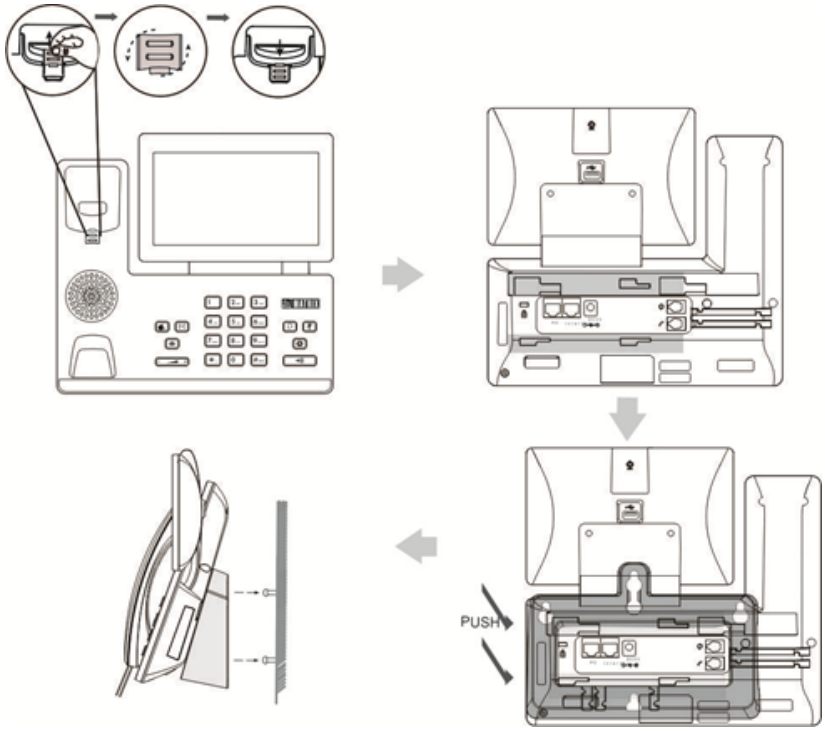
2. **Attaching the stand**



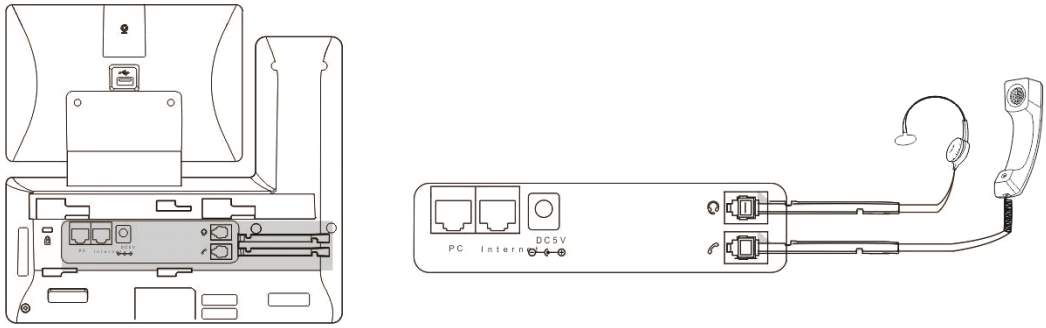
3. **Adjusting the angle of the touch screen**.



4. **Attaching the optional wall mounting bracket**

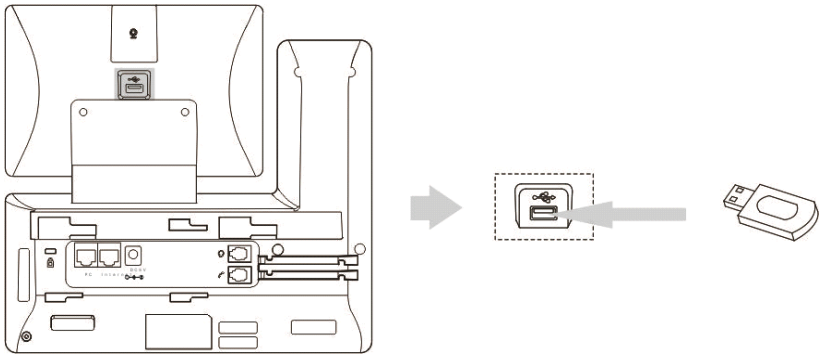


5. **Connect the handset and optional headset.**



A headset is not included in the packaging contents. Contact your re-seller/vendor for more information.

6. **Connect the optional USB Flash drive.**



7. **Connect the network and power.**

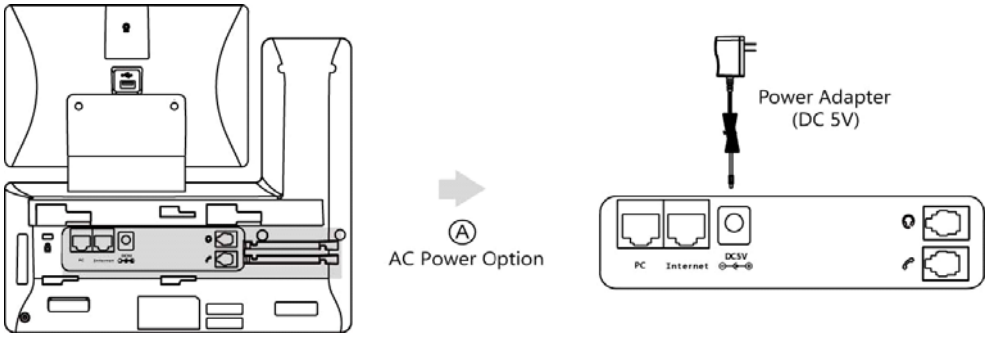
You have two options for power and network connections. Your network administrator will advise you for which one to use.

- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power:

- Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.

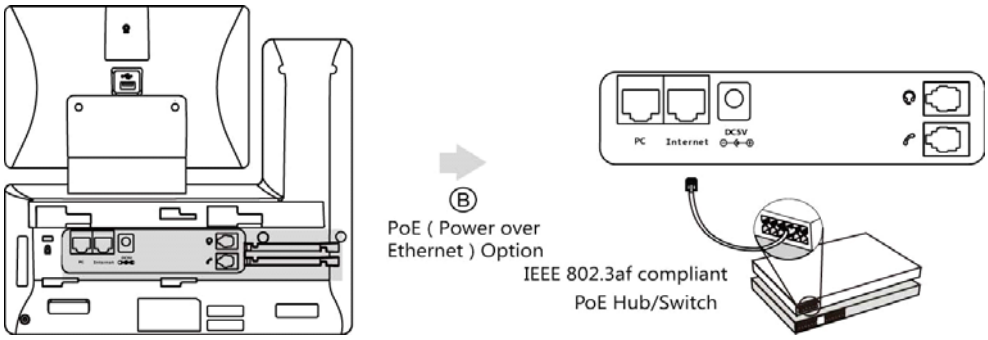


Power over Ethernet (PoE)

With the included or a regular Ethernet cable, the IP phone can be powered from a PoE-compliant switch or hub.

To connect the PoE:

- Connect the Ethernet cable between the Internet port on the phone and an available port on the in-line power switch/hub.



If in-line power switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

Do not unplug or remove power while the phone is updating firmware and configurations.

Network Connection

You have two options for network connection. You can select one of them according to the current office environment.

- Connecting to the wired network
- Connecting to the wireless network

Connecting to the Wired Network

You can connect your phone to a wired network.

Before connecting your phone to the wired network, it is important to note that the Wi-Fi feature should be disabled. For more information, refer to "Connecting to the Wireless Network".

Connecting to the Wireless Network

The IP phone supports Wi-Fi feature. You can connect your phone to a wireless network. The wireless network is more convenient and cost-effective than wired network. The PC port is unavailable when using wireless network. For more information on how to connect to a wireless network, refer to "Wireless Network Settings".

Phone Initialization

The initialization process of the IP phone is responsible for network connectivity and operation of the IP phone in your local network.

Once you connect your IP phone to the network and to an electrical supply, the IP phone begins its initialization process.

During the initialization process, the following events take place:

Automatic Phone Initialization

The phone finishes the initialization by loading the saved configuration. The touch screen displays “Welcome Initializing…please wait” during this process.

In the background, the IP phone executes the following steps:

Loading the ROM file

The ROM file resides in the flash memory of the IP phone. The IP phone comes from the factory with a ROM file pre-loaded. During initialization, the IP phone runs a bootstrap loader that loads and executes the ROM file.

Configuring the VLAN

If the IP phone is connected to a switch, the switch notifies the IP phone of the VLAN information defined on the switch (if using LLDP or CDP). The IP phone can then proceed with the DHCP request for its network settings (if using DHCP).

DHCP (Dynamic Host Configuration Protocol)

The IP phone is capable of querying a DHCP server. DHCP is enabled on the IP phone by default. The phone attempts to contact a DHCP server in your network to obtain valid IPv4 and IPv6 network settings (for example, IP address, subnet mask, default gateway address and DNS address) by default.



- *Basic network pre-configuration may be required to use the IP phone using DHCP. For more details, refer “DHCP”.*
- *You need to configure network parameters of the IP phone manually if any of them is not supplied by the DHCP server. For more information on configuring network parameters manually, refer “Configuring Network Parameters Manually”.*

Contacting the provisioning server

If the IP phone is configured to obtain configurations from the provisioning server, it will connect to the provisioning server and download the configuration file(s) during startup. The IP phone will be able to resolve and update configurations written in the configuration file(s). If the IP phone does not obtain configurations from the provisioning server, the IP phone will use configurations stored in the flash memory.

Updating firmware

If the access URL of firmware is defined in the configuration file, the IP phone will download firmware from the provisioning server. If the MD5 value of the downloaded firmware file differs from that of the image stored in the flash memory, the IP phone will perform a firmware update.

Downloading the resource files

In addition to configuration file(s), the IP phone may require resource files before it can deliver service. These resource files are optional, but if some particular features are being deployed, these files are required.

The followings show examples of resource files:

- Ring tones
- Contact files

Verifying Start-up

After connected to the power and network, the IP phone begins the initializing process by cycling through the following steps:

1. The power indicator LED illuminates solid red.
2. The message “Welcome Initializing…Please wait” appears on the touch screen when the IP phone starts up.
3. The main touch screen displays the following:
   - Time and date
   - Android Keys
4. Tap **Settings->Status** to check the IP phone status, the touch screen displays the valid IP address, MAC address, firmware version, etc.

If the IP phone has successfully passed through these steps, it starts up properly and is ready for use.

Reading Icons

Icons associated with different features may appear on the touch screen. Refer “Icon Instructions” for details.

Configuration Methods

IP phones can be configured automatically through configuration files stored on a central provisioning server, manually via the phone user interface or the web user interface, or by a combination of the automatic and manual methods.

The recommended method for configuring IP phones is automatically through a central provisioning server. If a central provisioning server is not available, the manual method will allow you to change most of the features.

The following section describes how to configure SPARSH VP710 using each of the mentioned methods:

- Phone User Interface
- Web User Interface
- Auto-Provisioning

Phone User Interface

An administrator or user can configure and use IP phones via phone user interface. You can swipe down from the top of the screen or swipe left/right to go to the second idle screen and then tap Settings to customize your phone. The Advanced option is only accessible to the administrator, and the default administrator password is “admin” (case-sensitive). All features are not accessible from the phone user interface.

For more information on customizing your phone with the available options from the phone user interface, refer “Customizing Your Phone”.

Web User Interface

In addition to the phone user interface, you can also customize your phone via web user interface. In order to access the web user interface, you need to know the IP address of your phone.

To obtain the IP address, swipe down from the top of the screen or swipe left/right to go to the second idle screen and then tap **Settings->Status** on the phone.

You can configure the IP phone via web user interface by logging into either of the following two modes. Each mode offers a different set of configuration privileges:

- Admin Mode
- User Mode

Browser Compatibility

We have tested the phone web interface with the following browsers and versions:

- Mozilla Firefox - Version 39.0
- Google Chrome - Version 46.0.2490.80 m

Admin Mode

This mode allows you to configure all the parameters applicable to the IP phone after logging into with your administrator privileges.

To obtain the IP address, swipe down from the top of the screen or swipe left/right to go to the second idle screen and then tap **Settings->Status** on the phone. Enter the IP address (e.g., http://192.168.0.10 or 192.168.0.10 for IPv4; http://[2005:1:1:1:215:65ff:fe64:6e0a] or [2005:1:1:1:215:65ff:fe64:6e0a] for IPv6) in the address bar of a web browser on your PC.

The default administrator user name and password are both "**admin**" (case-sensitive). The IP phones support both HTTP and HTTPS protocols for accessing the web user interface. For more information, refer "Web Server Type".

The options you can use to customize the IP phone via phone user interface and/or via web user interface are listed in the following table:

Options	Phone User Interface	Web User Interface
Status	√	√
--IP Address		
--MAC		
--Firmware		
--Network		
--Phone		
--Account		
Basic Phone Settings		
--Contrast	√	√
--Language	√	√
--Time & Date	√	√
--Administrator Password	√	√
--Key as Send	√	√
--Keypad Lock	√	√
-- Volume	√	
--Ring Tones	√	√
--Contact Management		
--Directory		√
--Local Directory	√	√
--Blacklist	√	√
--Remote Phone Book		√
--Call History Management	√	√
--Logo Customization		√
--Headset Prior		√
--Programmable Keys		√
--Account Registration	√	√
--Dial Plan		√
--Emergency Number		√
--Live Dialpad		√
--Hotline	√	√

Basic Call Features		√
--Recent Call In Dialing		
--Auto Answer	√	
--Auto Redial	√	
--Call Completion	√	
--Call Return		
--DND	√	
--Call Forward	√	
--Call Transfer	√	
--Call Waiting	√	
--Conference		
--Call Pickup	√	
--Anonymous Call	√	
--Anonymous Call Rejection	√	
Advanced Phone Features		√
--Hot Desking		
--Intercom	√	
--Multicast Paging		
--Music on Hold		
--Messages	√	
SIP Account		√
--User Options		
--Register Status	√	
--Activation	√	
--Label	√	
--Display Name	√	
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server1/2	√	
--Server Option		
--Server Port		
--Outbound Status	√	
--Outbound Proxy	√	
--NAT Traversal		
--STUN Status	√	
--STUN Server	√	

- *The table above lists most of the feature options. Please refer to the relevant sections for more information.*
- *In this document, configurations made either using the Phone User Interface or using the Web User Interface are termed as "Local" configurations since such methods will configure individual IP Phone parameters.*

  For mass configuration of IP Phones, the best method is using the "Auto-Provisioning" as described below.
- *To change the default administrator password, refer "Administrator Password".*

User Mode

This mode allows you to configure a limited set of features of the IP phone by logging into with your user privileges. This mode should be used by the end-users of the IP phone to configure user-specific features.

To obtain the IP address, swipe down from the top of the screen or swipe left/right to go to the second idle screen and then tap **Settings->Status** on the phone. Enter the IP address (e.g., http://192.168.0.10 or 192.168.0.10 for IPv4; http://[2005:1:1:1:215:65ff:fe64:6e0a] or [2005:1:1:1:215:65ff:fe64:6e0a] for IPv6) in the address bar of a web browser on your PC.

The default username is "**user**" (case-sensitive) and password is "**1111**". The IP phones support both HTTP and HTTPS protocols for accessing the web user interface. For more information, refer "Web Server Type".



To change the default user password, refer "User Password".

Auto-Provisioning

SPARSH VP710 IP Phones can be centrally provisioned from a provisioning server using the configuration file (<MAC>.cfg). SPARSH VP710 IP phones support FTP, TFTP, HTTP, and HTTPS protocols for auto- provisioning and are configured by default to use the TFTP protocol. Auto provisioning Methods enable SPARSH VP710 IP phones to update themselves automatically via downloading MAC-Oriented CFG files.

A MAC-Oriented CFG (or <MAC>.cfg) file contains parameters unique to a particular phone and will be effective for a specific IP Phone. The MAC-Oriented CFG file is named after the MAC address of the IP Phone. For, example, if the MAC address of the IP Phone is 00:1B:09:02:3D:01, the name of the MAC-Oriented CFG file must be **001B09023D01.cfg**.

Editing Configuration Files

Administrator needs to edit and customize the configuration file for each user as per requirement.

When modifying parameters, learn the following:

- Parameters in configuration files override those stored in the IP Phone's flash memory by default.
- The ***.cfg** extension of configuration files must be in lower case.
- Each line in a configuration file must use the following format and adhere to the following rules:
  - variable-name = value
  - Associate only one value with one variable.
  - Separate each variable name and value with an equal sign.
  - Set only one variable per line.
  - Put the variable and value on the same line, and do not break the line.
  - Comment the variable on a separate line. Use the pound (#) delimiter to distinguish the comments.

Encrypting Configuration Files

To protect against unauthorized access and tampering of sensitive information (e.g., login password, registration information), you can encrypt configuration files using Configuration Encryption Tool. AES keys must be 16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z. For more details, refer "Encrypting Configuration Files".

Location for Configuration Files

The MAC-Oriented Configuration files needs to be kept in the FTP/TFTP/HTTP/HTTPS Servers' root directory in the network.

Obtaining the Address of the Provisioning Server

SPARSH VP710 IP phones support obtaining the Provisioning Server Address in the following ways:

- Zero Touch
- Plug and Play (PnP) Server
- DHCP Options
- Static

The priority of obtaining the provisioning server address is as follows:

Zero Touch-->**PnP Server**-->**DHCP Options (Custom option**--> **Option 66 or Option 43)** --> **Phone Flash.**

If using any of the above mentioned methods, the Auto Provisioning Server URL needs to be send to the phone in below format depending on the provisioning protocol :

- In case of FTP: ftp://URL (IP Address or Domain Name)
- In case of TFTP : Direct URL (IP Address or Domain Name)
- In case of HTTP : http://URL (IP Address or Domain Name)
- In case of HTTPS : https://URL (IP Address or Domain Name)

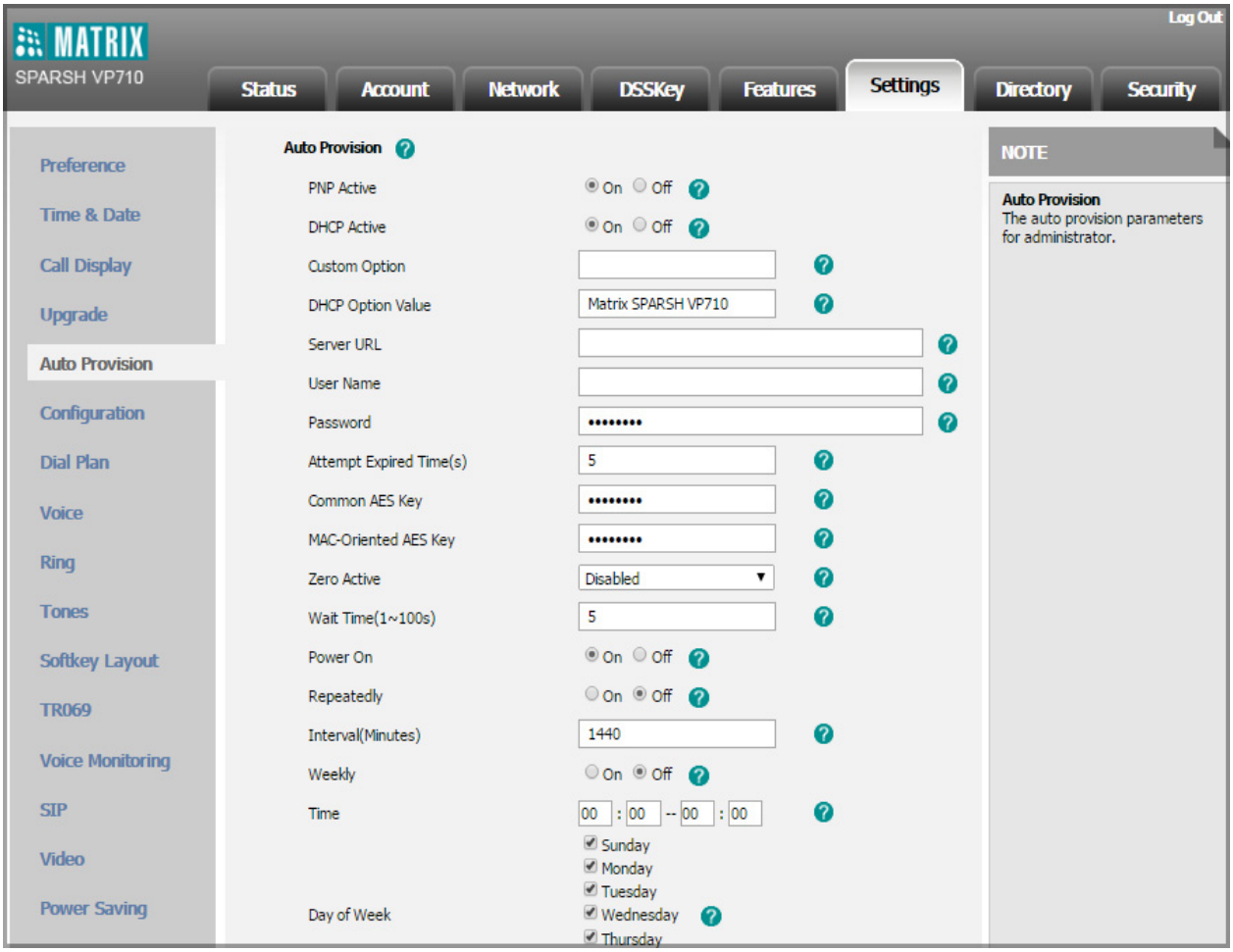
Zero Touch

Zero Touch allows you to configure the network parameters and provisioning server address via phone user interface during startup. This feature is helpful when there is a system failure on the phone. To use Zero Touch, make sure this feature is enabled.

To configure the Zero Touch via web user interface:

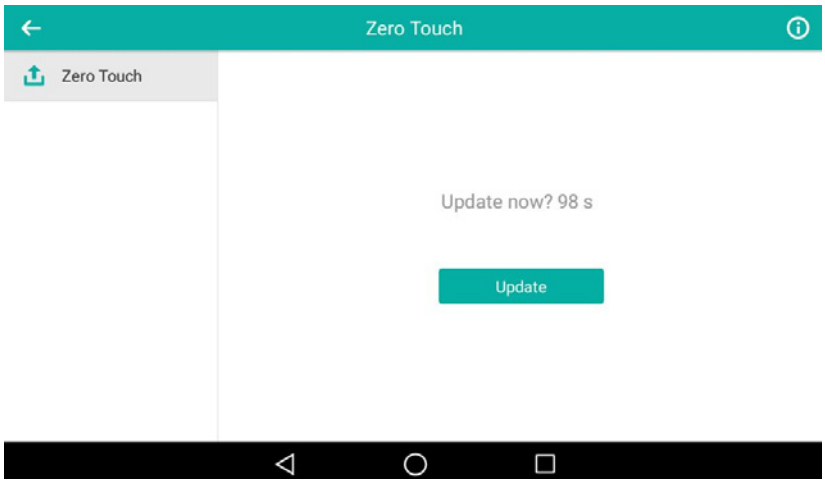
- Click on **Settings**->**Auto Provision**.
- Select **Enabled** from the **Zero Active** list.

- Select the desire value as the **Wait Time (1~100s)**.



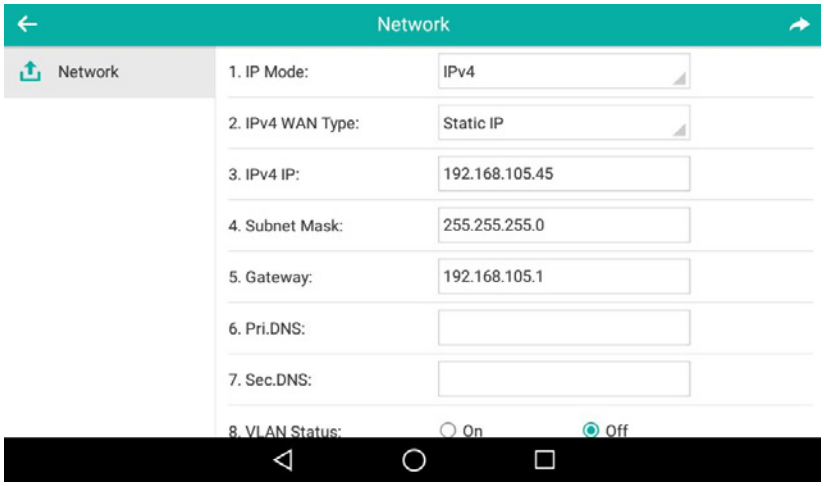
- Click **Confirm** to save the change.

When Zero Touch is enabled, configuration wizard as shown below will appear on the touch screen during start up:

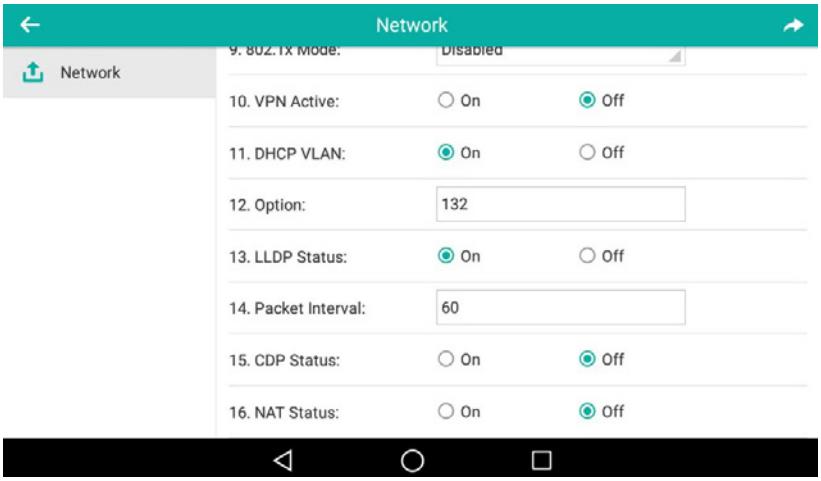


- Tap **Update** to configure the network settings.

- Enter the network parameters in the following fields.



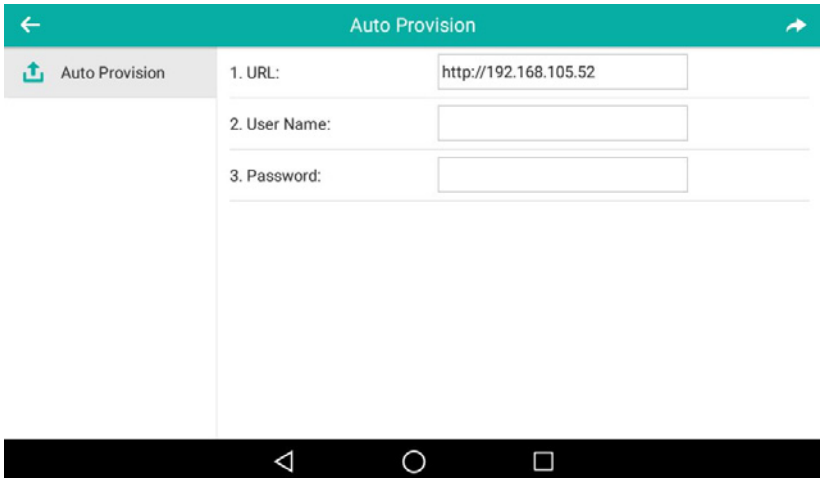
Scroll down the touch screen to enter all the parameters.



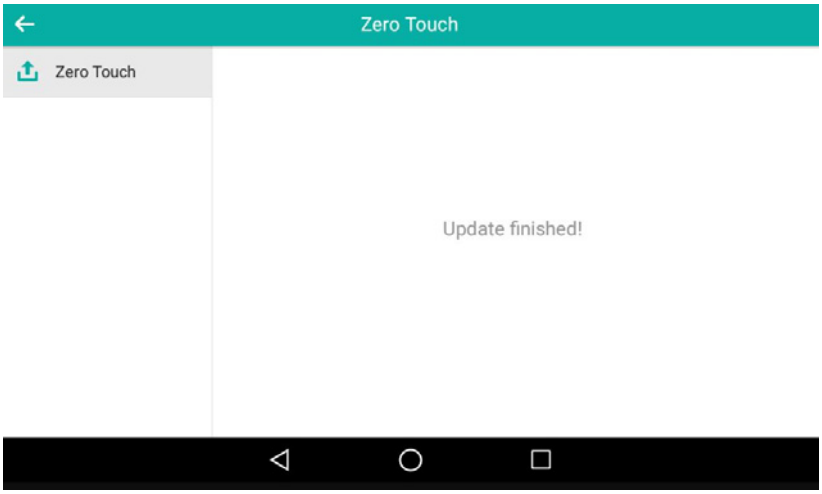
- Tap  soft key after configuring the network settings.
- In the Auto Provision screen, configure the Provisioning Server Address in the **URL** field.
- Enter the user name in the **User Name** field (optional).
- Enter the password in the **Password** field (optional).

,

- Tap  soft key to save the changes.



The IP Phone displays the following screen:

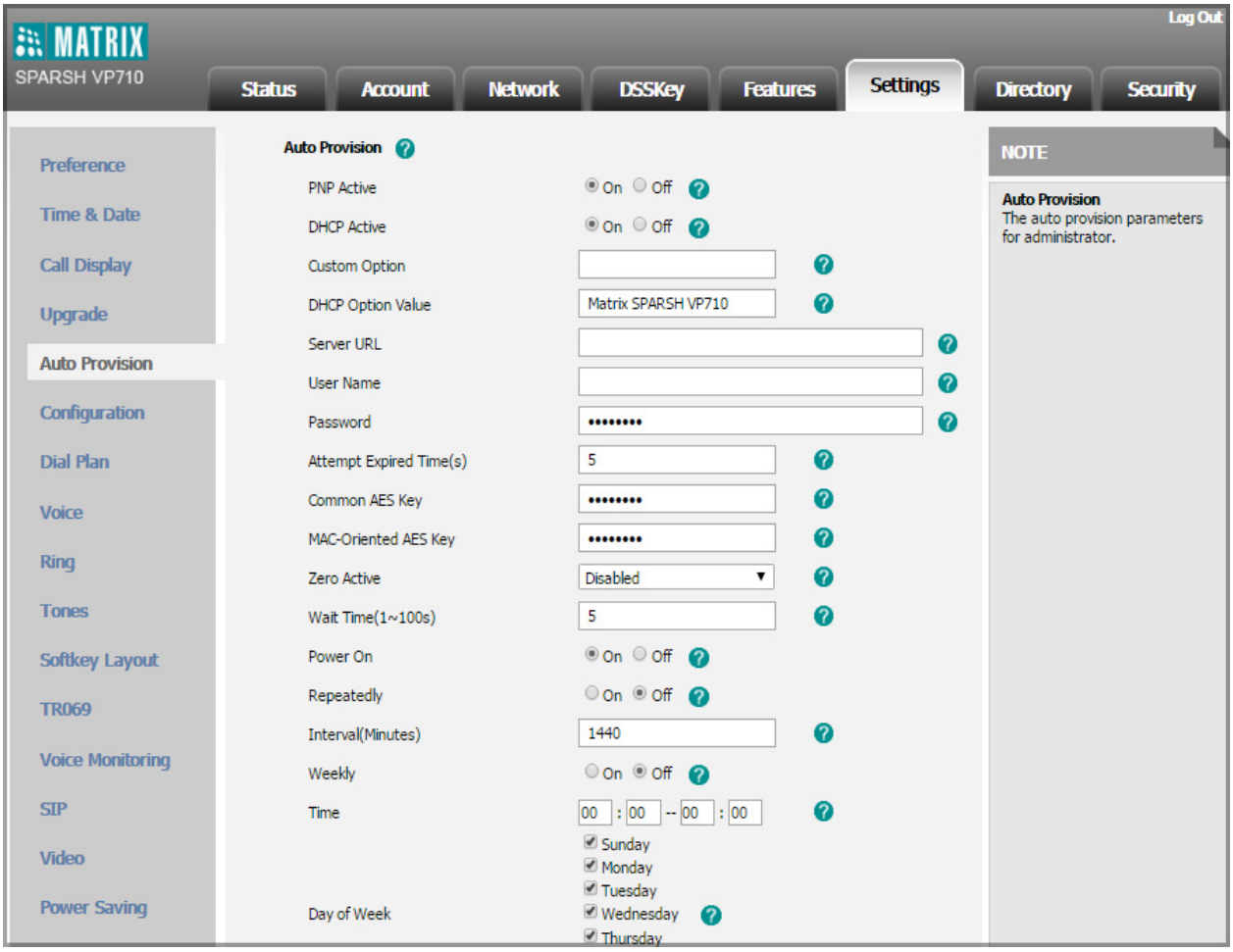


Plug and Play (PnP) Server

SPARSH VP710 IP phones support obtaining the Provisioning Server details from the PnP server. The phone broadcasts the PnP SUBSCRIBE message to obtain the Provisioning Server details during startup. To use Plug and Play, make sure this feature is enabled.

To enable PnP via web user interface:

- Click on **Settings**->**Auto Provision**.
- Select the **On** check box in the **PNP Active** field.



- Click **Confirm** to save the change.

Any PnP server activated in the network responses with a SIP NOTIFY message, and the Address of the Provisioning server as well as the Authentication Username and Password if configured, is contained in the message body. Then the phone can connect to the Provisioning Server and perform the auto provisioning process.

DHCP Options

SPARSH VP710 IP phones support obtaining the Provisioning Server details from DHCP options. You can configure the phone to obtain the Provisioning Server Address from a custom DHCP option, or the phone will automatically detect the Option 66 and Option 43. The Option 66 is used to identify the TFTP server. To obtain the Provisioning Server Address by a custom DHCP option, make sure the DHCP option is set properly.

SPARSH VP710 identifies the protocol of Provisioning Server as per the details configured in the DHCP option.

For example in DHCP Option 66,

- If only URL is configured then SPARSH VP710 will generate TFTP query
- If ftp://URL is configured then SPARSH VP710 will generate FTP query
- If http://URL is configured then SPARSH VP710 will generate HTTP query
- If https://URL is configured then SPARSH VP710 will generate HTTPS query

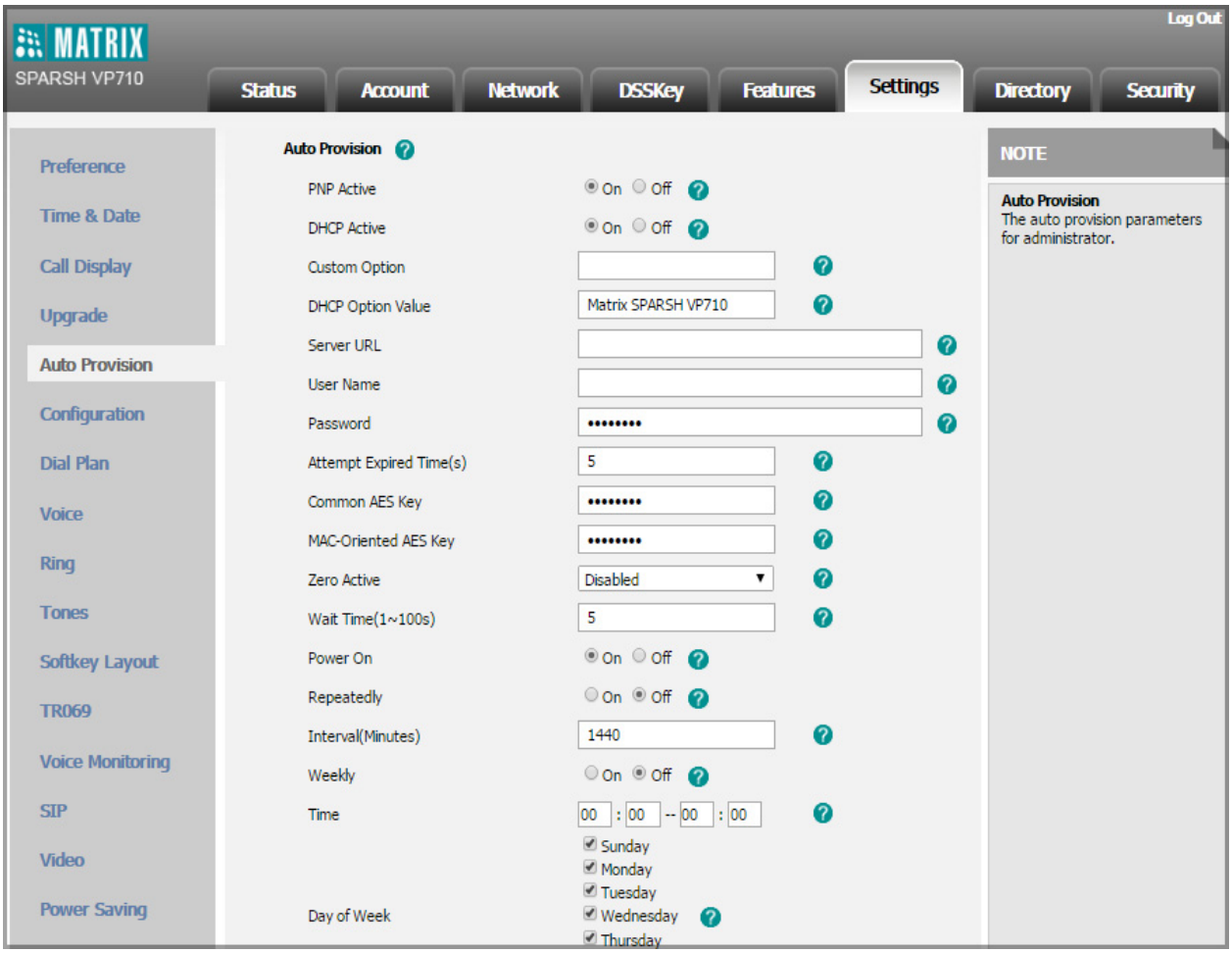
In any of the above methods, Authentication User name & Password can be send in below format:

http://username:password@URL
[Here we have taken http as an example]

The custom DHCP option must be in accordance with the one defined in the DHCP server.

To configure the DHCP option via web user interface:

- Click on **Settings** -> **Auto Provision**.
- Select the **On** check box in the **DHCP Active** field.
- Enter the desired value in the **Custom Option (128~254)** field.
- Enter the desired value in the **DHCP Option Value** field. The default value is **Matrix SPARSH VP710.**



- Click **Confirm** to save the change.

Downloading the Configuration Files

Once obtaining a provisioning server address in one of the ways introduced above, the phone will connect to the provisioning server and download configuration files. During the auto provisioning process, the phone will try to download the MAC-Oriented CFG file related to it from the provisioning server. If resource files need to be updated and the access URLs have been specified in configuration files, the phone will then try to download and update the resource files.

Resolving and Updating Configurations

After downloading, the phone resolves the configuration files, downloads the resource files requested in the configuration files, and then updates the configurations and resource files to the phone flash. Generally, updated configurations will automatically take effect after the auto provisioning process is completed. For update of some specific configurations which require a reboot before taking effect, for example, network configurations, the phone will reboot to make the configurations effective after the auto provisioning process is completed.

The phone calculates the MD5 values of the downloaded files. If the MD5 values of the Common and MAC-Oriented configuration files are the same as those of the last downloaded configuration files, this means these two configuration files on the provisioning server are not changed. The phone will complete the auto provisioning without repeated update. This is used to avoid unnecessary restart and impact of phone usage. If configuration files have been AES-encrypted, the phone will decrypt them after downloading the configuration files.

Update Mode

The update mode is used to set the desired time for the phone to perform the auto provisioning process.

Let us understand the following update modes in detail:

- Power On
- Repeatedly
- Weekly
- Auto Provision Now
- Multi-mode Mixed
- SIP NOTIFY Message

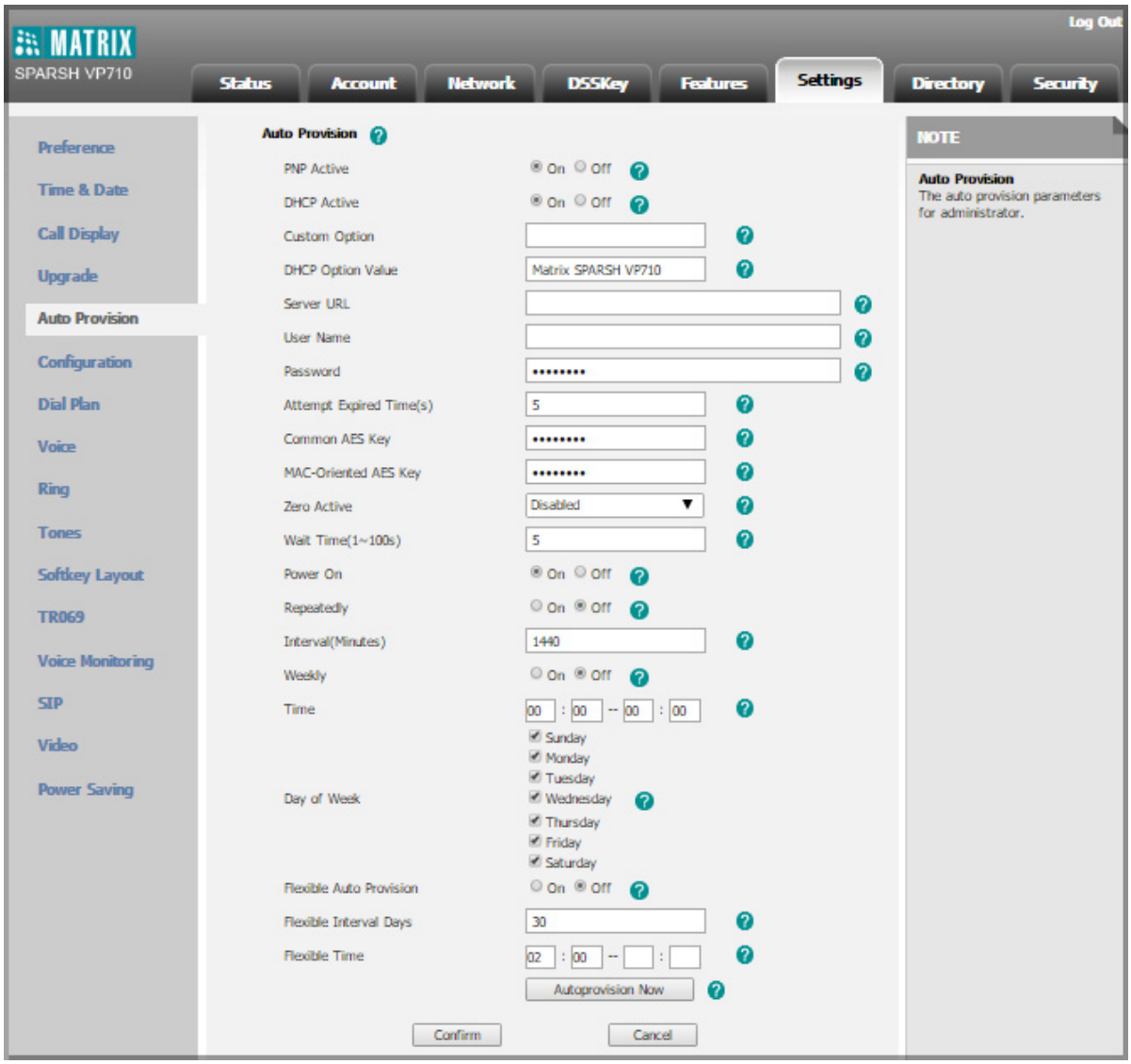
Power On

The phone performs the auto provisioning process when the phone is powered on.

To activate the Power On mode via a web user interface:

- Click on **Settings**->**Auto Provision**.

- Select the **On** check box in the **Power On** field.



- Click **Confirm** to save the change.

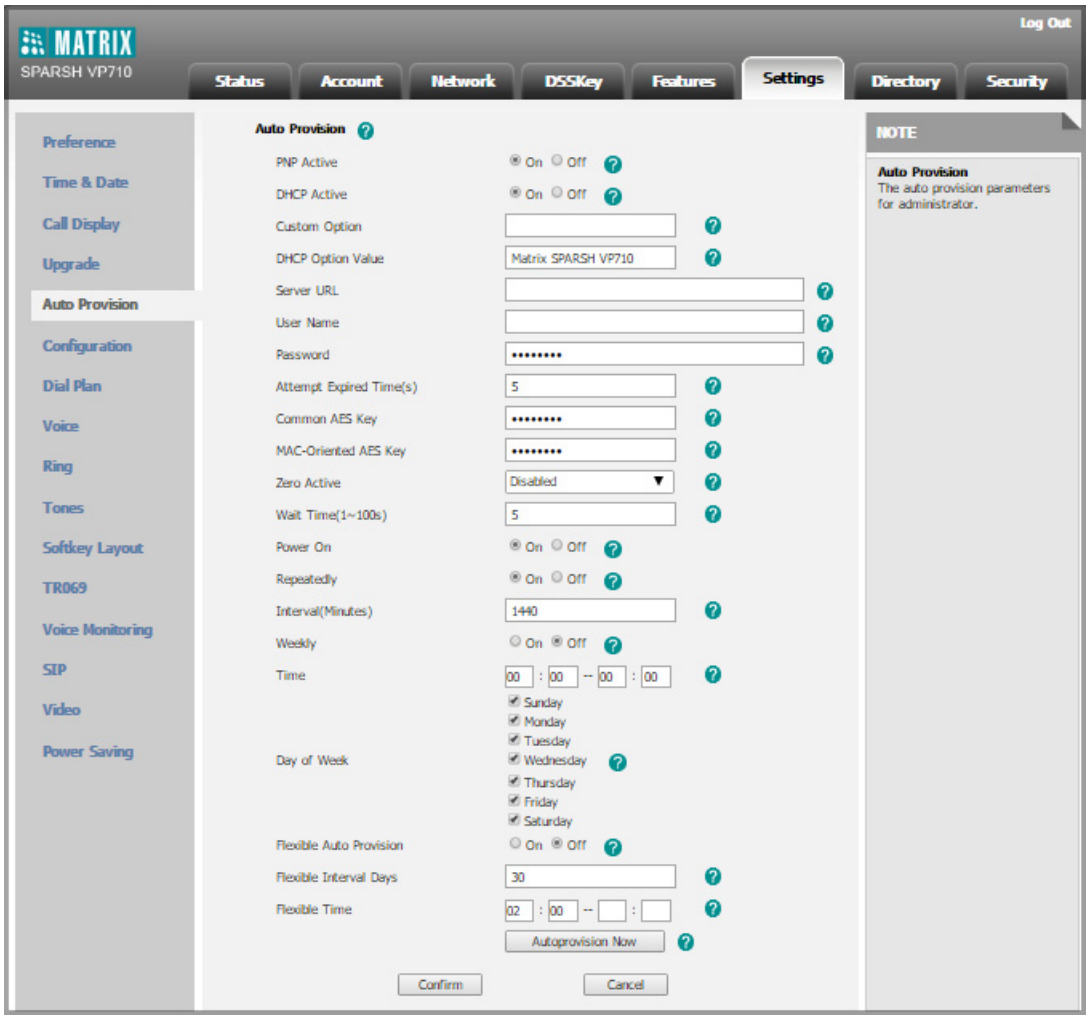
Repeatedly

The phone performs the auto provisioning process at regular intervals. You can configure the interval for the Repeatedly mode. The default interval is 1440 minutes.

To activate the Repeatedly mode via web user interface:

- Click on **Settings**->**Auto Provision**.
- Select the **On** check box in the **Repeatedly** field.

- Enter the interval time (in minutes) in the **Interval (Minutes)** field.



- Click **Confirm** to save the change.

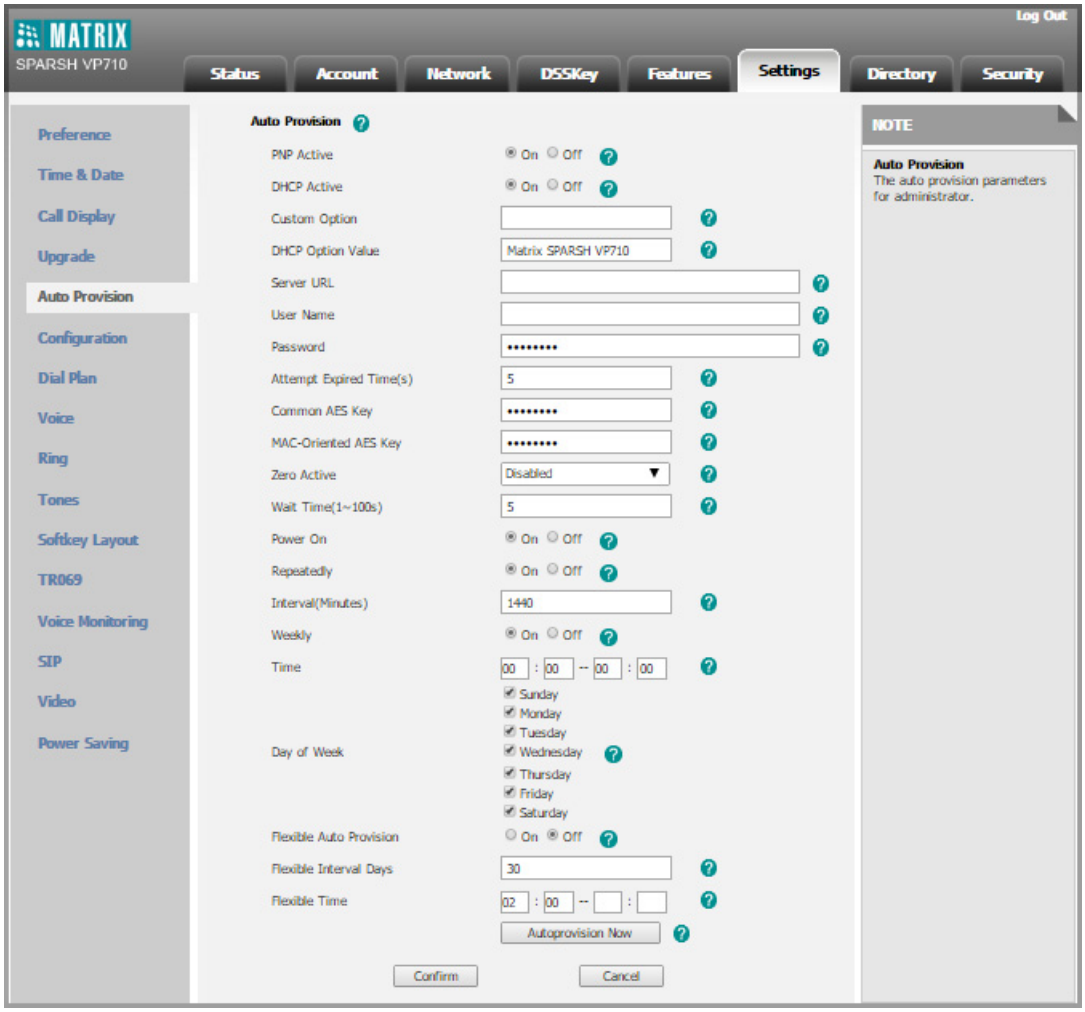
Weekly

The phone performs the auto provisioning process at the fixed time every week. You can configure what time of the day and which day of the week to trigger the phone to perform the auto provisioning process. For example, you can configure the phone to check and update new configuration between 2 to 3 o'clock every Friday and Sunday.

To activate the Weekly mode via web user interface:

- Click on **Settings**->**Auto Provision**.
- Select the **On** check box in the **Weekly** field.
- Enter the desired time in the **Time** field.

- Select one or more check boxes in the **Day of Week** field.



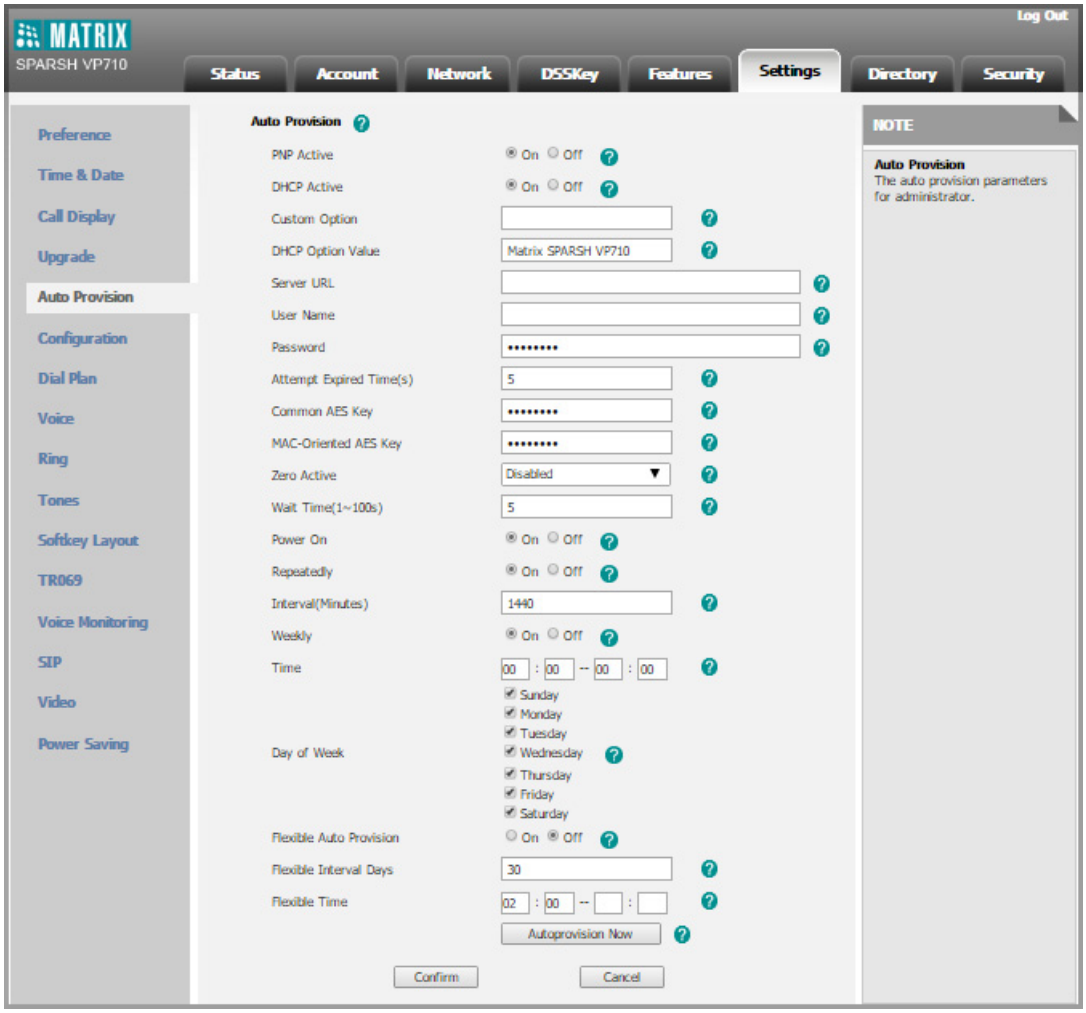
- Click **Confirm** to save the change.

Auto Provision Now

You can use Auto Provision Now mode to manually trigger the phone to perform the auto provisioning process immediately.

To use the Auto Provision Now mode via web user interface:

- Click on **Settings**->**Auto Provision**.



- Click **Autoprovision Now**.

The phone will perform the auto provisioning process immediately.

Multi-mode Mixed

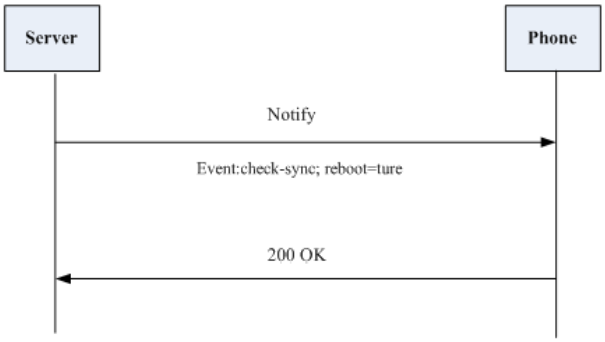
You can activate more than one update mode for auto provisioning. For example, you can activate the **Power On** and **Repeatedly** modes simultaneously. The phone will perform the auto provisioning process when it is powered on and at a specified interval.

SIP NOTIFY Message

The phone will perform the auto provisioning process when receiving a SIP NOTIFY message which contains the header “Event: check-sync”. If the header of the SIP NOTIFY message contains an additional string “reboot=true”,

the phone will reboot immediately and then perform the auto provisioning process. This update mode requires server support.

The following figure shows the message flow:



Viewing the Phone Status

You can view phone status via phone user interface or web user interface.

Available information of phone status includes:

- General information (IP address, MAC address and firmware version)
- Network status (IPv4 status or IPv6 status, IP mode and MAC address)
  - IPv4 uses a 32-bit address.
  - IPv6 is an updated version of the current Internet Protocol to meet the increased demands for unique IP addresses, using a 128-bit address.
- Phone status (product name, hardware version, firmware version, product ID, MAC address, device certificate status and Android version)
- Account status (register status of SIP accounts)
- Storage information (internal storage, internal SD and USB storage

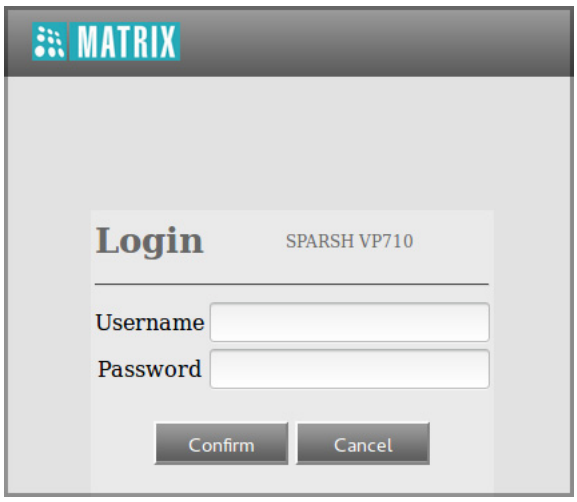
The storage information is sorted and displayed with different categories. The following table lists usage of each category

Category	Usage
Apps	Display the memory cost of the application data and media content.
Cached data	Tap to clear cached data for all apps.
Pictures, videos	Tap to view the pictures/videos by **Gallery** application.
Audio	Tap to open the file system of the storage. You can browse the file system to view all audios in the internal SD card or USB flash drive.
Downloads	Display the memory cost of the downloaded files.
Misc.	Tap to display all folders on the IP phone. You can then select the folders to be cleaned up.

You can view the device certificate status and storage information via phone user interface only.

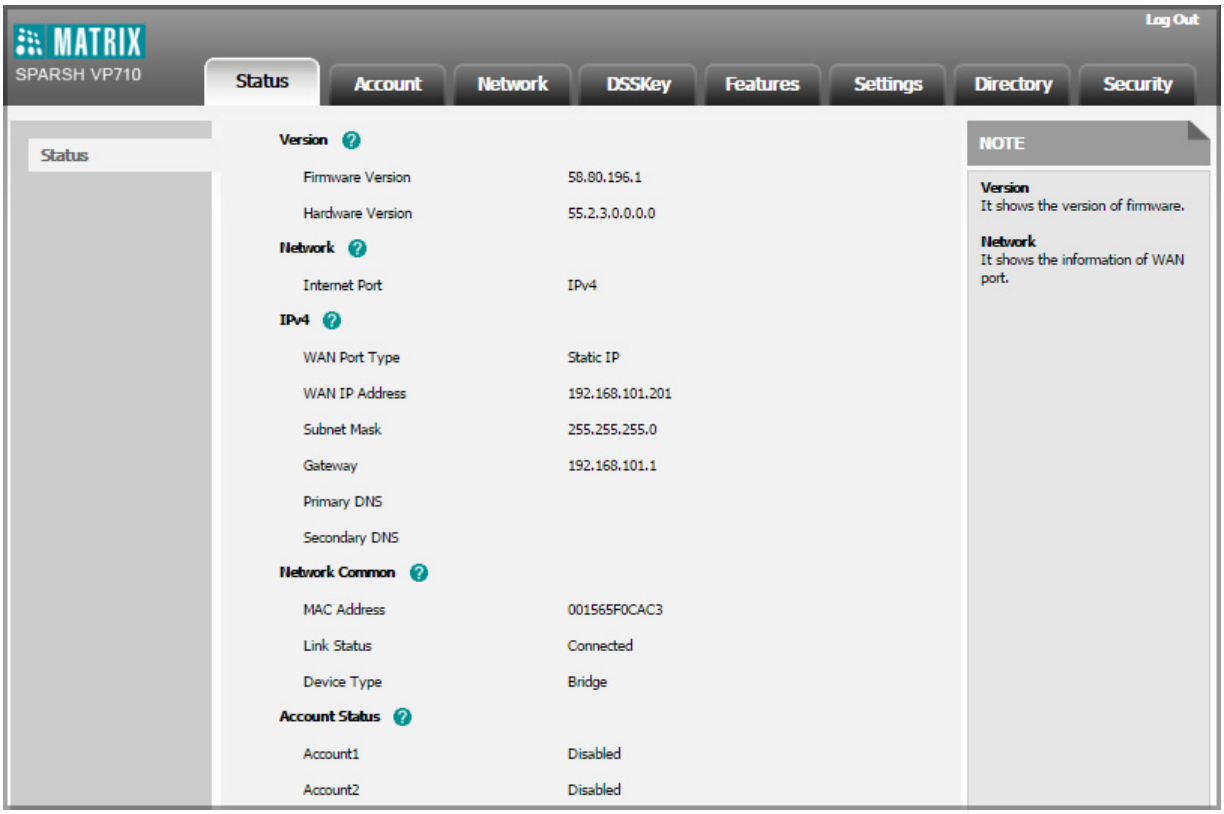
To view the phone status via web user interface:

- Open the web browser of your computer.
- Enter the IP address in the browser's address bar, and then press **Enter**.
- Enter the **Username** (**admin**) and **Password** (default: **admin**) in the login page.



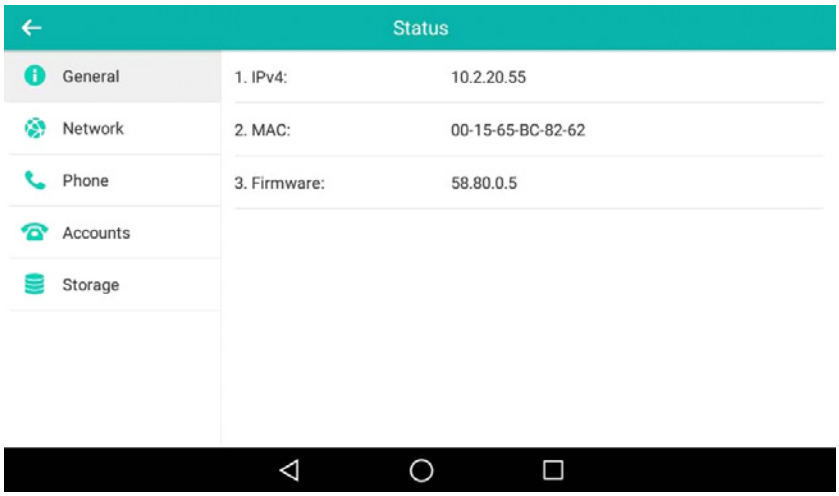
- Click **Confirm** to login.

The phone status is displayed on the first page of the web user interface.



To view the phone status via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Status**.



- Tap the desired item to view the desired information

Registration

Generally, your phone will be deployed with multiple other phones. In this case, the phone parameters will be configured before hand, so that after phone start up, the phone will be registered and ready for use. If your phone is not registered, you may have to register it by configuring it manually. For more information on how to configure and register your phone, refer “Account Management”.

CHAPTER 5

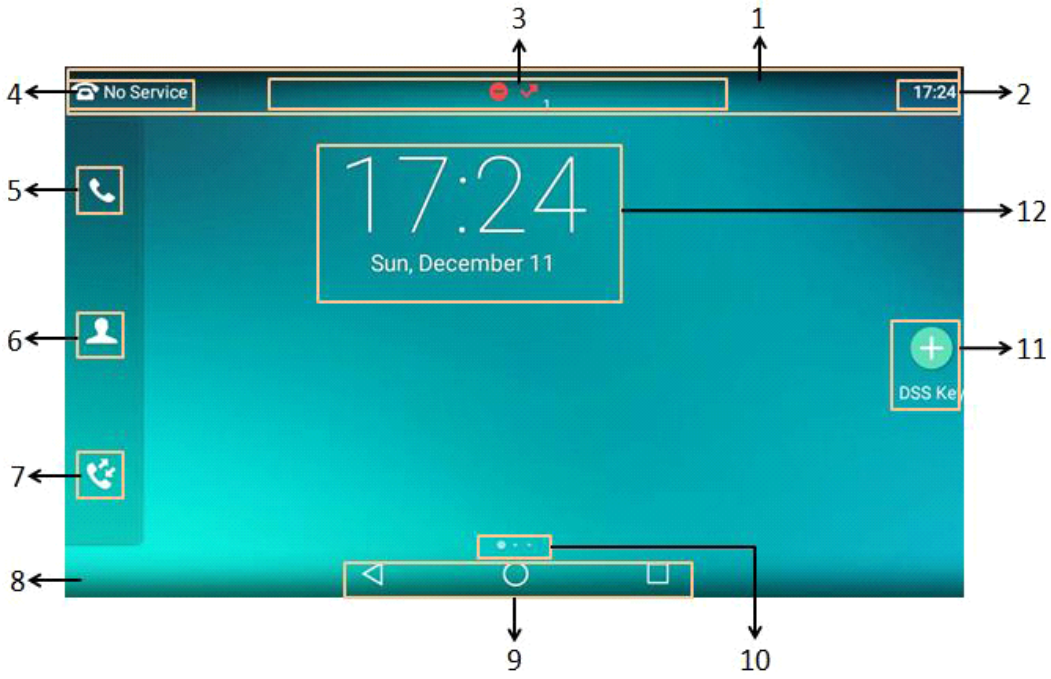
Using the Touch Screen

Idle Screen

The IP phone supports multiple idle screens. You can swipe left or right to switch among different idle screens. The IP phone supports three idle screens by default. You can add a new idle screen by moving an application or widget to the right side of the third idle screen. The idle screen will be deleted automatically if there is no application or widget on the screen. For more information, refer to "Managing Applications and Widgets".

The Home Screen

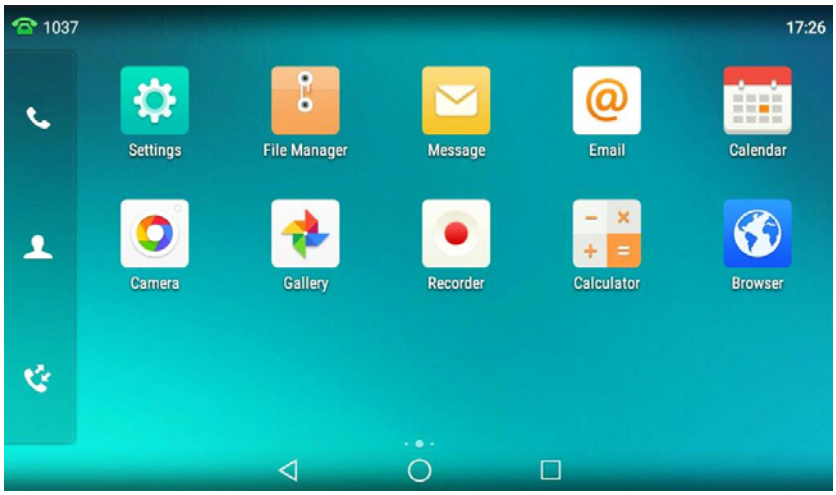
The home screen displays the following:



No.	Name	Description
1	Status Bar	Displays the phone's default account, icons and time.
2	Time	The phone's time is displayed on the right of the status bar. For more information, refer to "Time & Date".
3	Icons	Icons are displayed in the middle of the status bar. For more information, refer to "Icon Instructions".
4	Default account	The label of the default account is displayed on the left of the status bar. For more information, refer to "Default Account" If there is no account registered on the IP phone, this field will display **No Service**.
5	Phone Dialer	Tap to enter the dialing screen. For more information, refer to "Dialing Screen Display".
6	Directory	Tap to enter the Directory screen and view contacts. For more information, refer to "Contact Management".
7	History	Tap to enter the History screen and view call history. For more information, refer to "Call History Management".
8	Wallpaper	Shows the specified wallpaper, which can be customized. For more information, refer to "Wallpaper"
9	Android Keys	 : tap to go back to the previous screen.  : tap to return to the idle screen.  : tap to view and manage the list of recently used applications.
10	Screen Indicator	Indicates which idle screen is displayed.
11	DSS Key	Tap to add a line key, and then line key list will be displayed on the right of the home screen. For more information, refer to "DSS Keys".
12	Digital Clock Widget	Displays the phone's time and date. For more information, refer to "Clock".

The Second Idle Screen

The second idle screen displays phone's system applications by default.

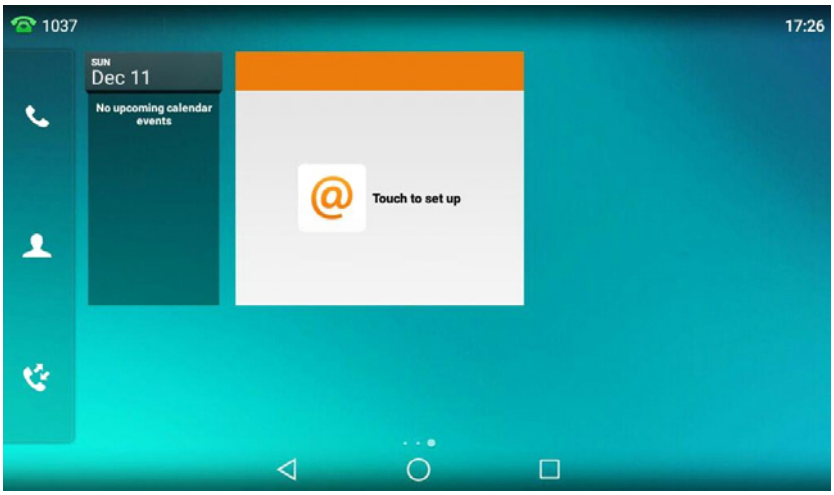


No.	Application	Description
1	Settings	You can use this application to access phone settings and features. You can also swipe down from the top of the screen and then tap **Settings** to access phone settings and features.
2	FileManager	You can use this application to manage files in internal SD card as well as USB flash drive. For more information, refer to "FileManager".
3	Message	You can use this application to access the video/voice mails or leave video/voice mails for someone. For more information, refer to "Message".
4	Email	You can use this application to send and receive emails. For more information, refer to "Email".
5	Calendar	You can use this application as a reminder or to manage and arrange your events. For more information, refer to "Calendar".
6	Camera	You can use this application to take pictures or record videos. For more information, refer to "Camera".

7	Gallery	You can use this application to view photos and videos. For more information, refer to "The pictures are transferred as mail attachment. The attachment files size must be within 5MB. You can also share pictures with other devices via Bluetooth or email when previewing the pictures. And the operating instructions are the same with sharing pictures by Gallery application. For more information, refer to "Sharing Pictures/Videos".".
8	Recorder	You can use this application to record audio. For more information, refer to "Recorder".
9	Calculator	You can use this application to calculate. For more information, refer to "Calculator".
10	Browser	You can use this application to present and explore content on the World Wide Web. For more information, refer to "Browser".

The Third Idle Screen

The third idle screen displays the calendar and email widgets by default.

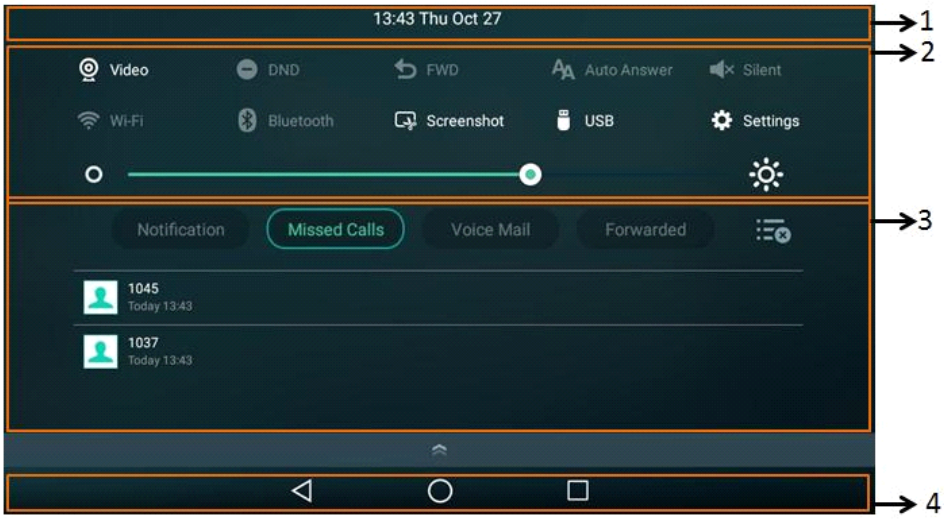


No.	Application	Description
1	Calendar Widget	Displays the data and upcoming events. For more information, refer to "Calendar".
2	Email Widget	Displays email list of the specified folder. For more information, refer to "Email".

Control Center/Notification Center

Control center or notification center allows you to access common features or view important notifications quickly.

Swipe down from the top of the screen to enter the control center and the notification center:

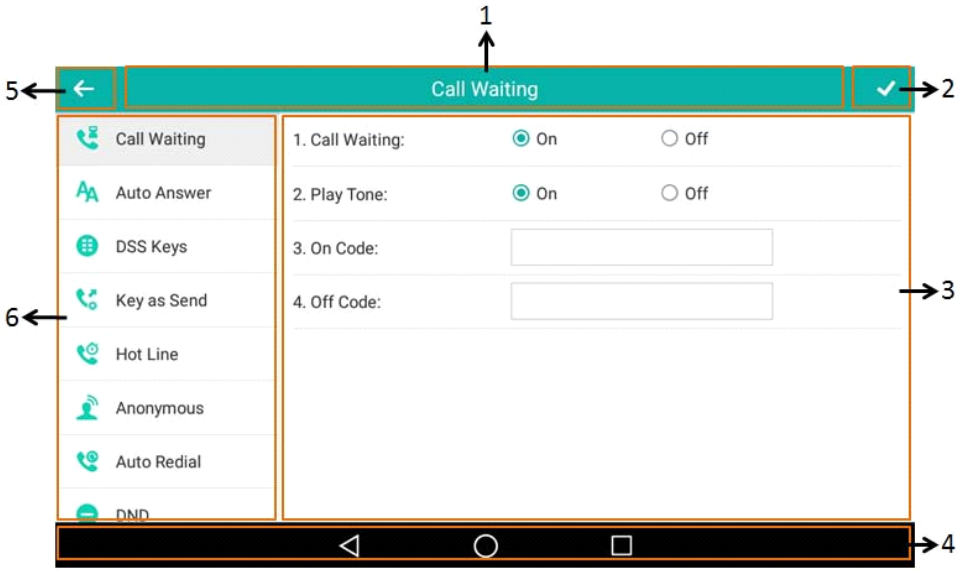


No.	Name	Description	
1	Time and date	Display the phone's time and date. For more information, refer to "Time & Date".	
2	Control Center	Video	• Tap to access **Camera** application when the IP phone is idle. For more information, refer to "Camera". • Tap to stop transmitting video during a video call. For more information, refer to "Stopping Transmitting Video during a Call".
		DND	Tap to turn on/off DND feature quickly. For more information, refer to "Do Not Disturb (DND)".
		FWD	Tap to enter Call Forward setting screen. For more information, refer to "Call Forward".
		Auto Answer	Tap to enter Auto Answer setting screen. For more information, refer to "Auto Answer".

2	Control Center	Silent	Tap to turn on/off silent feature quickly.
		Wi-Fi	• Tap to turn on/off Wi-Fi feature quickly. For more information, refer to “Wireless Network Settings”. • Long tap to enter Wi-Fi setting screen.
		Bluetooth	• Tap to turn on/off Bluetooth feature quickly. For more information, refer to “Bluetooth Headset”. • Long tap to enter Bluetooth setting screen.
		Screenshot	Tap to capture a screenshot. For more information, refer to “Screenshot”.
		USB	Tap to access the **FileManager** application to manage the files in the USB flash drive. For more information, refer to “FileManager”.
		Settings	Tap to enter the Settings screen.
		Backlight Slider	Drag the slider to adjust the screen brightness quickly. You can also enter the Backlight setting screen to adjust the screen brightness. For more information, refer to “Backlight”.
3	Notification Center	• Tap **Notification**/**Missed Calls**/**Voice Mail**/**Forwarded** to view the corresponding notification list. • Tap the desired notification message to view the details. Swipe left or right to delete a specific notification. • Tap  to delete all notifications.	
4	Android Keys	 : tap to go back to the previous screen.  : tap to return to the idle screen.  : tap to view and manage the list of recently used applications.	

You can swipe up from the bottom of the screen to hide this screen.

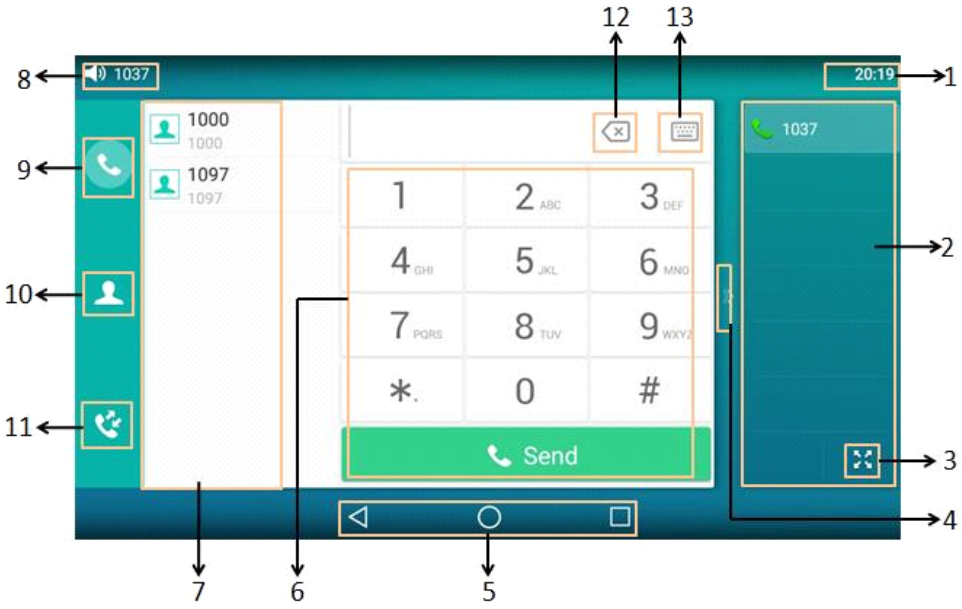
Configuration Screen Display



No.	Name	Description
1	Title Area	Displays the title of the current menu screen.
2		Tap to accept the settings.
3	Configuration Area	Displays configuration items of the highlighted menu item in the menu tree area.
4	Android Keys	 : tap to go back to the previous screen.  : tap to return to the idle screen.  : tap to view and manage the list of recently used applications.
5		Tap to go back to previous screen.

6	Menu Tree Area	• Displays menu items. • Tap the desired menu item to enter the corresponding configuration screen.

Dialing Screen Display



No.	Name	Description
1	Time	The phone's time is displayed on the right of the status bar. For more information, refer to "Time & Date".
2	Line Key List	The line key list displays 6 line keys by default. You can drag up and down to scroll through the line keys. Various features can be assigned to line keys. Each line key can display the label and indicate the status of the assigned feature. **Note**: The line key list appears only if you have added at least one line key.
3		Tap to unfold line key list and display all line keys. For more information, refer to "DSS Keys".
4		Tap it or swipe right to hide the line key list.

5	Android Keys	: tap to go back to the previous screen. : tap to return to the idle screen. : tap to view and manage the list of recently used applications.
6	Dial Pad	Tap the desired character(s) and then tap Send to dial out the number.
7	Placed Calls List	You should enable the recent call in dialing feature to view the placed calls list. For more information, refer to "Recent Call In Dialing"
8	Talking Mode and Default Account	Shows which account and talking mode you are using to make a call. The talking mode includes handset, speakerphone or headset; you can alternate among the three modes when making a call. • Tap to change the default account. For more information, refer to "Default Account"
9		Tap to go back to the idle screen.
10	Directory	Tap to enter the Directory screen and view contacts.
11	History	Tap to enter the History screen and view call history.
12		Tap to delete the entered characters one by one. • Long tap to delete two or more characters.
13		Tap to display/hide the onscreen keyboard.

Video Call Screen Display

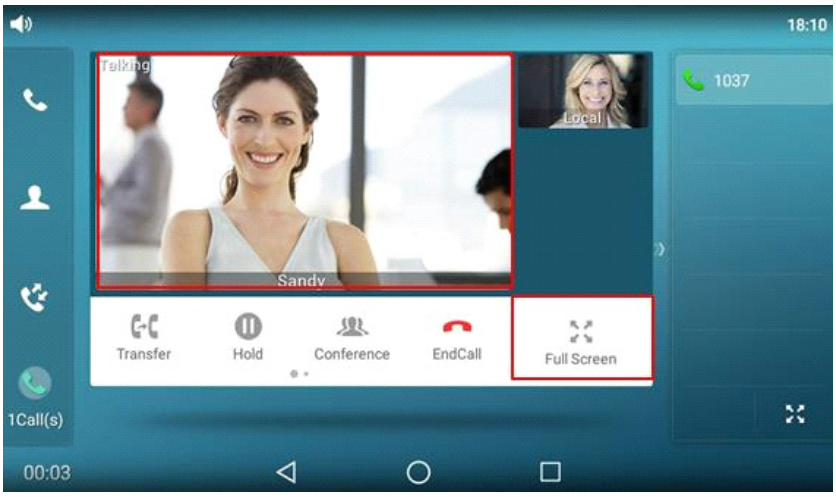


No.	Name	Description
1	Time	The phone's time is displayed on the right of the status bar. For more information, refer to "Time & Date".
2	Line Key List	The line key list displays 6 line keys by default. You can drag up and down to scroll through the line keys. Various features can be assigned to line keys. Each line key can display the label and indicate the status of the assigned feature. **Note**: The line key list appears only if you have added at least one line key.
3		Tap to unfold line key list and display all line keys. For more information, refer to "DSS Keys".
4		Tap it or swipe right to hide the line key list.
5	Full Screen	Tap to show the video on the phone in full screen. Only the near site and far site are shown. For more information on full screen, refer to "Video Call Full Screen Display".

6	Soft Keys	• Label automatically to identify their context-sensitive features. • Swipe left to see other soft keys.
7	Far-Site Information	Shows the information (name or number) of the far site.
8	Active Call Window	Shows the video of far site and near site. By default, a small window (near site) is located on the top-right of the touch screen, and a larger window (far site) is centered in the middle of the touch screen. **Note:** If you do not connect the CAM50 to your phone or turn off the near-site video during the call, the IP phone will stop transmitting the near-site video. For more information on turn off the near-site video, refer to"Stopping Transmitting Video during a Call".
9	Call Duration Timer	Shows the duration time of current call.
10	Talking Mode	Shows that you are talking using the handset, speakerphone or headset. You can alternate among the three modes during calls.
11	Phone Dialer	Tap to enter the dialing screen. For more information, refer to "Dialing Screen Display".
12	Directory	Tap to enter the Directory screen and view contacts.
13	History	Tap to enter the History screen and view call history.
14		• A number under the icon indicates the number of call(s) on the phone. For example, 1 indicates that there is a call on the phone. • Tap to exit or return to the talking screen.
15	Android Keys	: tap to go back to the previous screen. : tap to return to the idle screen. : tap to view and manage the list of recently used applications.

Video Call Full Screen Display

From the video call screen, do one of the following to view full-screen video:

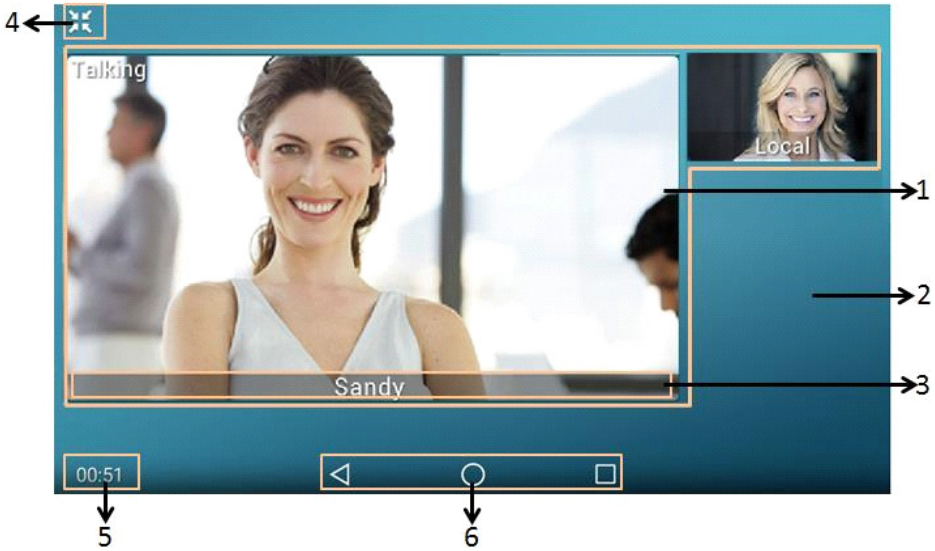


- Tap the big size image

  OR

- Tap **Full Screen**.

  The phone will enter the full screen automatically within 5 seconds during a call.



No.	Name	Description
1	Video Display Area	Shows video in different layout. For more information, refer to "Changing the Video Layout during Calls". **Note:** If you do not connect the CAM50 to your phone or turn off the near-site video during the call, the IP phone will stop transmitting the near-site video. For more information on turn off the near-site video, refer to"Stopping Transmitting Video during a Call".
2	Blank Area	Tap to exit the full-screen display.
3	Far-Site Information	Shows the information (name or number) of the far site by default.
4		Tap to exit the full-screen display.
5	Call Duration Timer	Shows the duration time of current call.
6	Android Keys	 : tap to go back to the previous screen.  : tap to return to the idle screen.  : tap to view and manage the list of recently used applications.

Basic Network Settings

In order to get your IP phones running, you must do the basic network configuration, such as IP address and subnet mask, gateway. This section describes how to configure the basic network parameters for IP phones. The IP phone can support either or both IPv4 and IPv6 addresses.

DHCP

DHCP (Dynamic Host Configuration Protocol) is a network protocol used to dynamically allocate network parameters to network hosts. The automatic allocation of network parameters to hosts eases the administrative burden of maintaining an IP network. IP phones comply with the DHCP specifications documented in RFC 2131. If using DHCP, IP phones connected to the network become operational without having to be manually assigned IP addresses and additional network parameters. Static DNS address (es) can be configured and used when DHCP is enabled.

DHCP Option

DHCP provides a framework for passing information to TCP/IP network devices. Network and other control information are carried in tagged data items that are stored in the options field of the DHCP message. The data items themselves are also called options.

DHCP can be initiated by simply connecting the IP phone with the network. IP phones broadcast DISCOVER messages to request the network information carried in DHCP options, and the DHCP server responds with specific values in corresponding options.

The following table lists common DHCP options supported by IP phones.

Parameter	DHCP Option	Description
Subnet Mask	1	Specify the client's subnet mask.
Time Offset	2	Specify the offset of the client's subnet in seconds from Coordinated Universal Time (UTC).
Router	3	Specify a list of IP addresses for routers on the client's subnet.
Time Server	4	Specify a list of time servers available to the client.
Domain Name Server	6	Specify a list of domain name servers available to the client.
Log Server	7	Specify a list of MIT-LCS UDP servers available to the client.

Host Name	12	Specify the name of the client.
Domain Server	15	Specify the domain name that client should use when resolving host names via DNS.
Broadcast Address	28	Specify the broadcast address in use on the client's subnet.
Network Time Protocol Servers	42	Specify a list of NTP servers available to the client by IP address.
Vendor-Specific Information	43	Identify the vendor-specific information.
Vendor Class Identifier	60	Identify the vendor type.
TFTP Server Name	66	Identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options.
Boot file Name	67	Identify a boot file when the 'file' field in the DHCP header has been used for DHCP options.

For more information on DHCP options, refer to, http://www.ietf.org/rfc/rfc2131.txt?number=2131 or http://www.ietf.org/rfc/rfc2132.txt?number=2132.

If you do not have the ability to configure the DHCP options for discovering the provisioning server on the DHCP server, an alternate method of automatically discovering the provisioning server address is required. Connecting to the secondary DHCP server that responds to DHCP INFORM queries with a requested provisioning server address is one possibility. For more information, refer to http://www.ietf.org/rfc/rfc3925.txt?number=3925.

Procedure

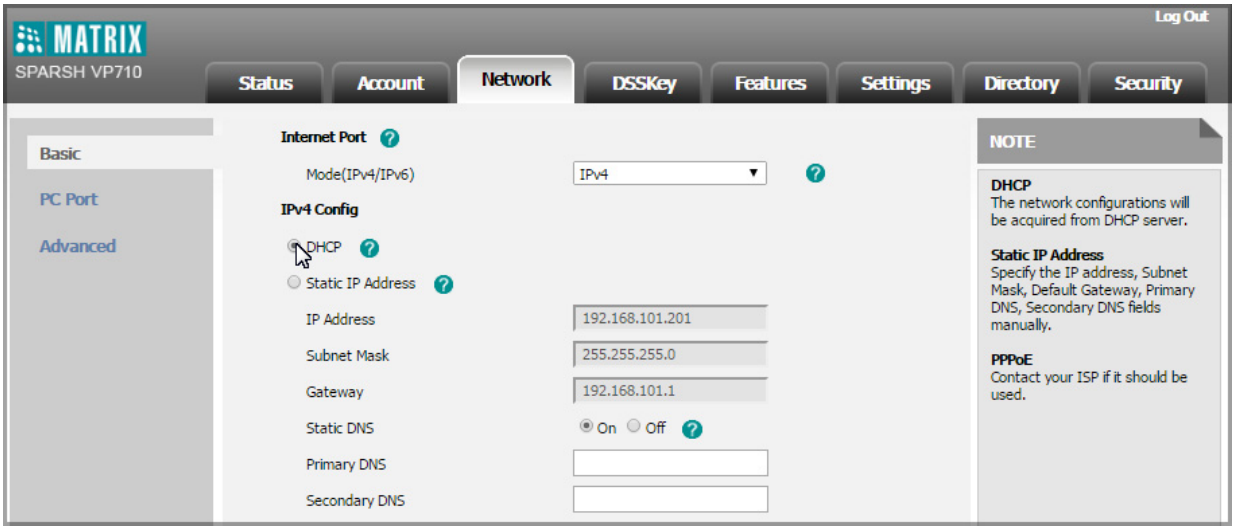
DHCP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure DHCP on the IP phone. Configure static DNS address For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure DHCP on the IP phone. Configure static DNS address when DHCP is used. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network&q=load
	Phone User Interface	Configure DHCP on the IP phone.

To configure DHCP via web user interface:

- Click on **Network->Basic**.
- In the **IPv4 Config** block, click **DHCP**.



- Click **Confirm** to save the change.

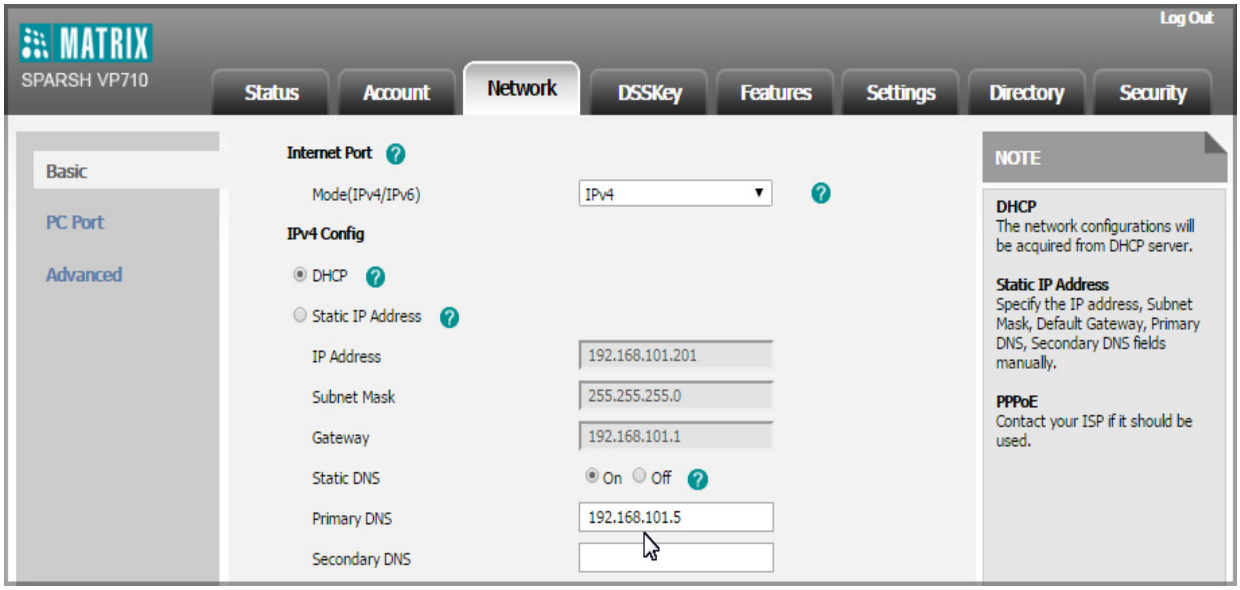
  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To configure static DNS address when DHCP is used via web user interface:

- Click on **Network->Basic**.
- In the **IPv4 Config** block, click **DHCP**.
- Click **Static DNS**.

- Enter the desired values in the **Primary DNS** and **Secondary DNS** fields.



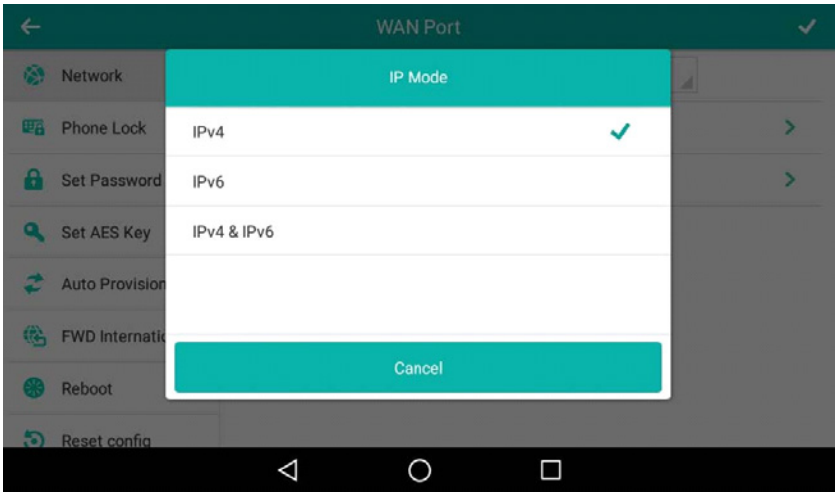
- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

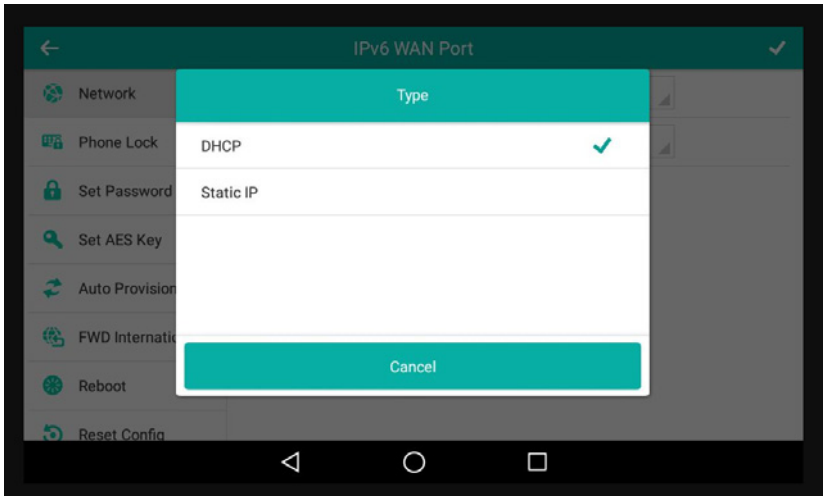
To configure DHCP via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv4.**
- Tap the **Type** field.
- Tap **DHCP** in the pop-up dialog box.



- Tap ✓ to accept the change or ◁ to cancel.
- Similarly, to configure DHCP for IPv6, tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv6.**

- Tap the **Type** field.
- Tap **DHCP** in the pop-up dialog box.



- Tap ✓ to accept the change or ◁ to cancel.

*You can configure a static IPv4 address for the IP phone. Before configuring it, make sure that the **IP mode** is configured as **IPv4** or **IPv4 & IPv6**.*

Configuring Network Parameters Manually

If DHCP is disabled or IP phones cannot obtain network parameters from the DHCP server, you need to configure them manually. The following parameters should be configured for the IP phones to establish network connectivity:

- IP Address
- Subnet Mask
- Default Gateway
- Primary DNS
- Secondary DNS

Procedure

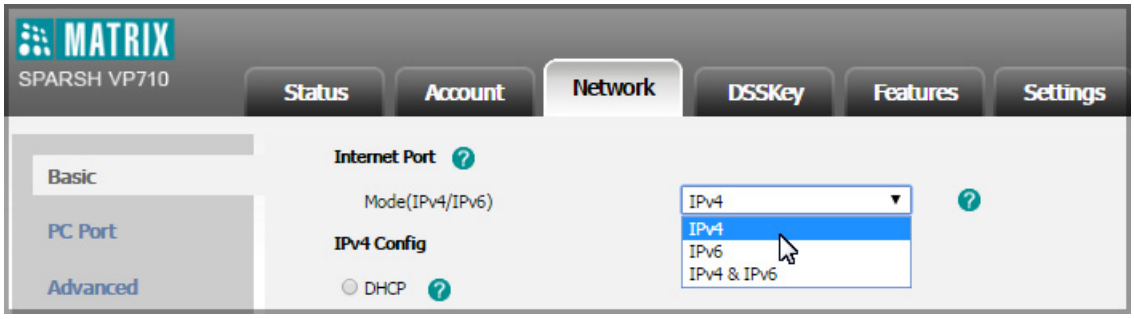
Network parameters can be configured manually using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure network parameters of the IP phone manually. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure network parameters of the IP phone manually. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network&q=load
	Phone User Interface	Configure network parameters of the IP phone manually.

To configure the IP address mode via web user interface:

- Click on **Network->Basic**.
- Select desired value from the **Mode (IPv4/IPv6)** list.



- Click **Confirm** to save the change.

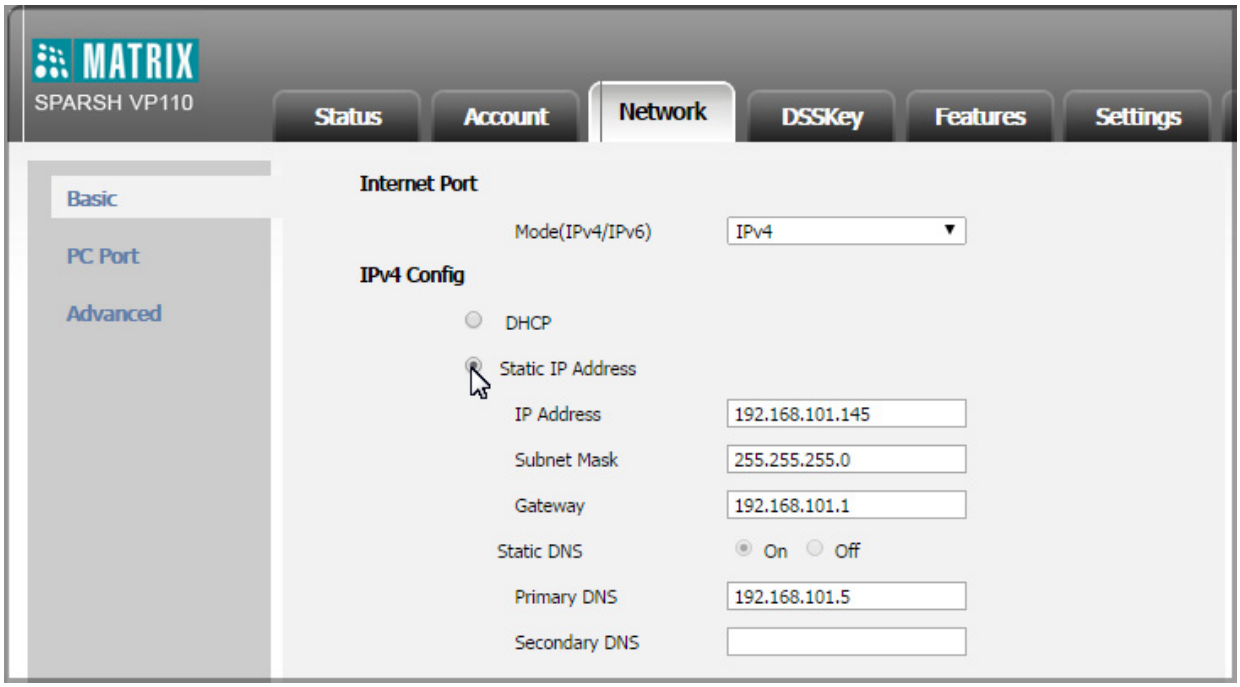
  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To configure a static IPv4 address via web user interface:

- Click on **Network->Basic**.
- In the **IPv4 Config** block, click **Static IP Address**.

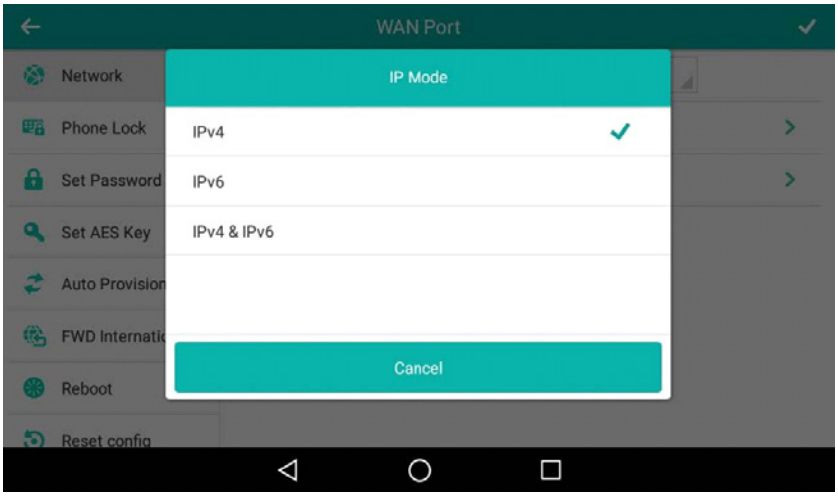
- Enter the desired values in the IP Address, Subnet Mask, Gateway, Primary DNS and Secondary DNS fields.



- Click **Confirm** to save the change.
  A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the IP phone.

To configure the IP address mode via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port.**
- Tap the **IP Mode** field.
- Tap the desired value (**IPv4, IPv6 or IPv4 & IPv6**) in the pop-up dialog box.

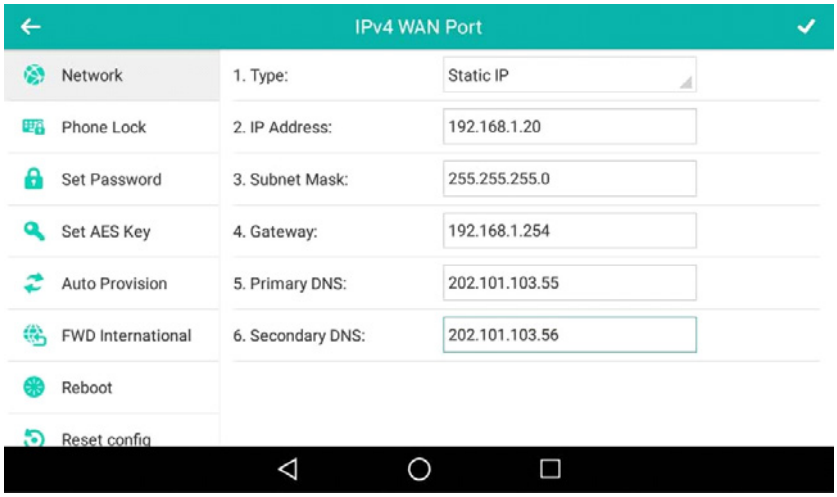


- Tap ✓ to accept the change or ◁ to cancel.

*You can configure a static IPv4 address for the IP phone. Before configuring it, make sure that the **IP mode** is configured as **IPv4** or **IPv4 & IPv6**.*

To configure a static IPv4 address via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv4.**
- Tap the **Type** field.
- Tap **Static IP** in the pop-up dialog box.
- Enter the desired value in the **IP Address, Subnet Mask, Gateway, Primary DNS** and **Secondary DNS** field respectively.



- Tap ✓ to save the change or ◁ to cancel.

*You can configure a static IPv4 address for the IP phone. Before configuring it, make sure that the **IP mode** is configured as **IPv4** or **IPv4 & IPv6**.*

To configure a static IPv6 address via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv6.**
- Tap the **Type** field.
- Tap **Static IP** in the pop-up dialog box.

- Enter the desired value in the **IP Address, IPv6 IP Prefix, Gateway, Primary DNS** and **Secondary DNS** field respectively.



- Tap ✓ to accept the change or ◁ to cancel.

If you are using an xDSL modem for IPv4 network connection, you can connect your phone to the Internet via PPPoE mode. Set the WAN port as a PPPoE port. The PPPoE port will perform a PPP negotiation to obtain the IP address. Contact your service provider for the PPPoE username and password.

PPPoE

PPPoE (Point-to-Point Protocol over Ethernet) is a network protocol used by Internet Service Providers (ISPs) to provide Digital Subscriber Line (DSL) high speed Internet services. PPPoE allows an office or building-full of users to share a common DSL connection to the Internet. PPPoE connection is supported by the IP phone Internet port. Contact your ISP for the PPPoE user name and password.

Procedure

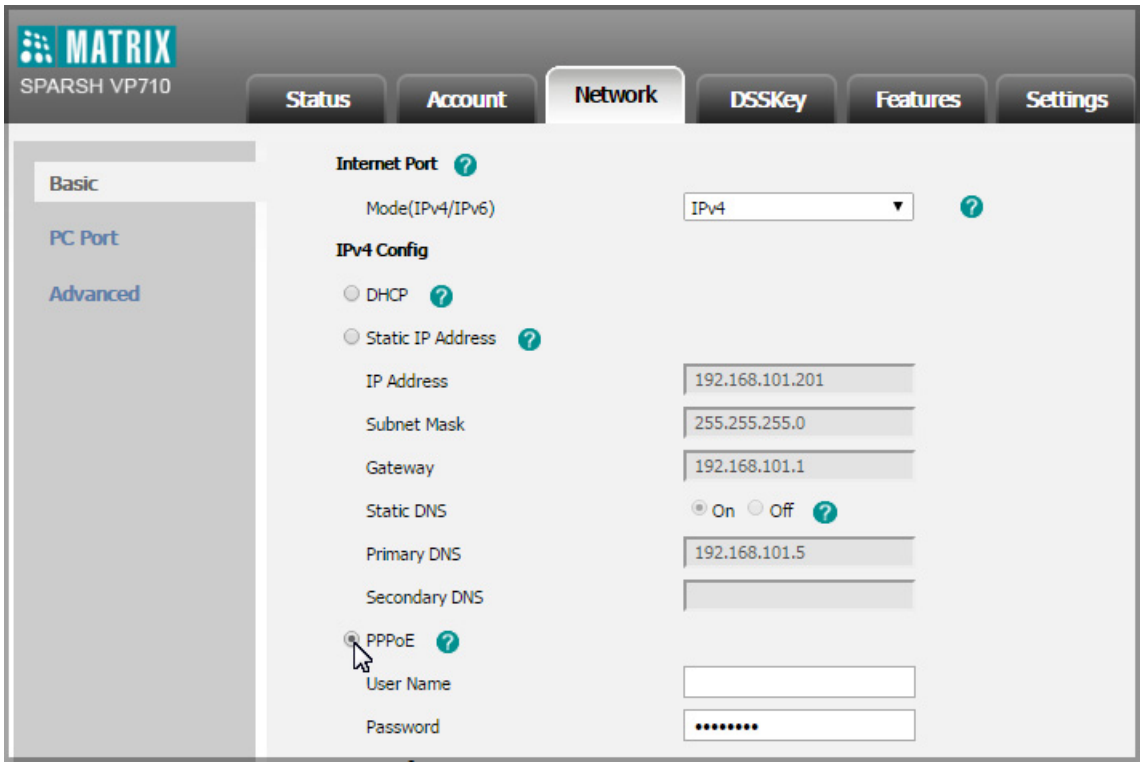
PPPoE can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure PPPoE on the IP phone. Configure the user name and password for PPPoE on the IP phone. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure PPPoE on the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network&q=load
	Phone User Interface	Configure PPPoE on the IP phone.

To configure PPPoE via web user interface:

- Click on **Network->Basic**.
- In the **IPv4 Config** block, click **PPPoE**.
- Enter the **User Name** and **Password** in corresponding fields.



- Click **Confirm** to save the change.

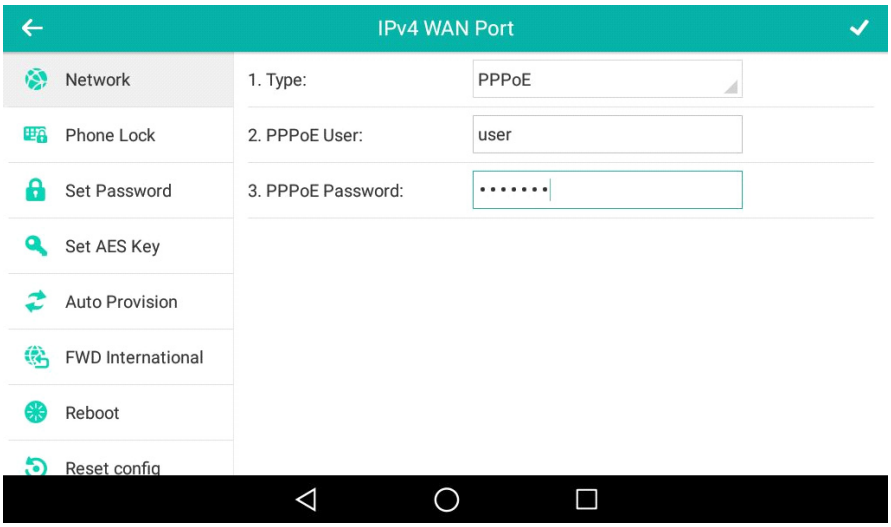
  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To configure PPPoE via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv4**.
- Tap the **Type** field.

- Tap **PPPoE** in the pop-up dialog box.
- Enter the PPPoE user name and password in the corresponding fields.



- Tap ✓ to accept the change or ◁ to cancel.

Using the wrong network parameters may result in inaccessibility of your phone and may also have an impact on your network performance. For more information on these parameters, contact your network administrator.

Wireless Network Settings

Wi-Fi feature enables users to connect their phones to the organization's wireless network. The wireless network is more convenient and cost-effective than wired network.

When the Wi-Fi feature is enabled, the IP phone will automatically scan the available wireless networks. All the available wireless networks will display in scanning list on the touch screen.



When you connect the Ethernet cable, you can enable the Wi-Fi feature. But you have to disable the Wi-Fi feature if you want to use the wired network.

Enabling Bluetooth may degrade your wireless network connection. For improved wireless network performance, you should disable Bluetooth when it is not in use.

You need to take a note of the following when using the IP phones in the wireless network:

- Check whether the wireless network is normal when the account registered fails or sometimes there is no sound during an active call.
- Ensure that the bandwidth of your wireless network is able to provide stable and real-time data transmission otherwise the quality of video calls may be affected. We recommend you to use the wired network for video calls.
- We recommend you to use a good quality router in your home/office environment.
- We recommend you to set the password for the wireless network so as to ensure the network resource will not be occupied by any unknown user.

Procedure

Wi-Fi parameters can be configured manually using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure Wi-Fi feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	NA
	Phone User Interface	Configure Wi-Fi feature.

Connecting to the Wireless Network

Three ways to the IP phone to the wireless network:

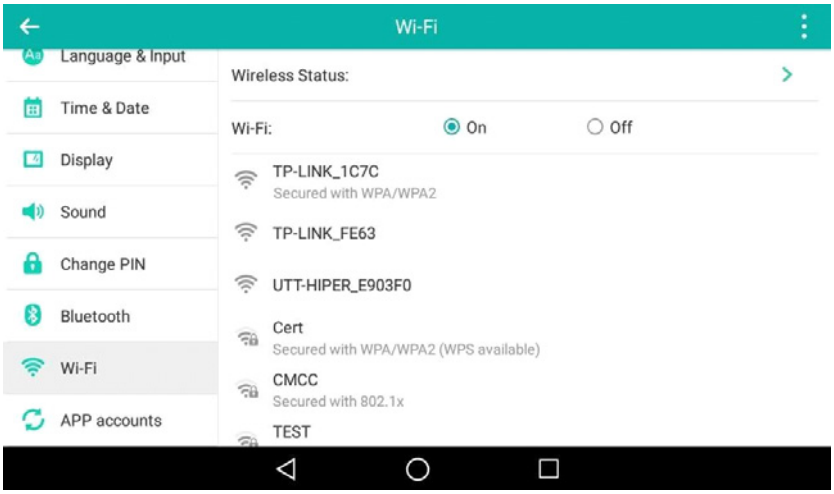
- Manually connect to an available wireless network
- Wi-Fi Protected Setup (WPS)
- Manually add a wireless network

Manually connect to an available wireless network

To manually connect to an available wireless network via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the **On** radio box in the **Wi-Fi** field.

  The phone will automatically search for available wireless networks in your area.

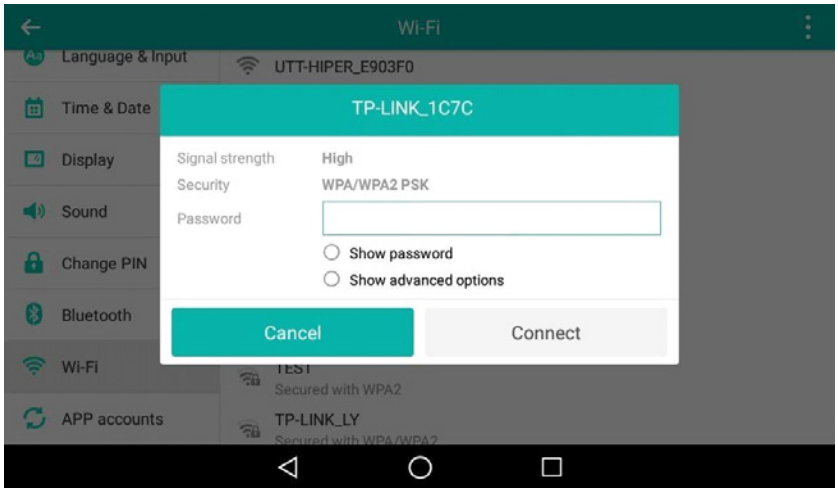


- To re-search the available network, tap and then tap **Scan** (Optional).
- Tap the desired wireless network (SSID) to connect to it.
- If the network is secure, enter its password in the **Password** field.

  Contact your network administrator for the Wi-Fi password.

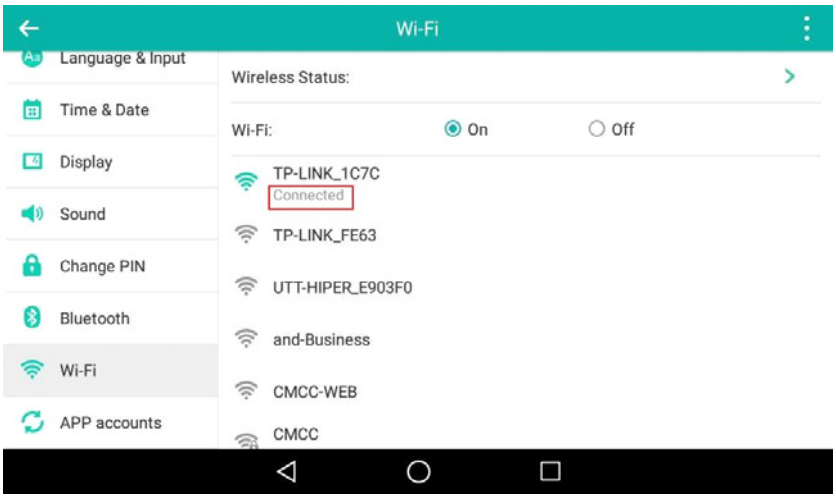
  You can do the following:

  - Tap the **Show password** radio box to make the password visible.
  - Tap the **Show advanced options** radio box to configure the HTTP proxy for **Browser** application.



- Tap **Connect** to connect to the wireless network.

  Once the connection has completed successfully, the prompt “Connected” appears under the corresponding SSID.



  You can also swipe down from the top of the screen to enter the control center, and toggle **Wi-Fi** on. The phone will automatically search for wireless network in your area, and try to connect the wireless network that you have connected last time. If there are no available wireless networks or the phone cannot connect the wireless network that you have connected last time, you will get a notification “Wi-Fi networks available”. Swipe down from the top of the screen to enter the notification center, tap the notification.

  Repeat the above steps to connect to the desired wireless network.

Wi-Fi Protected Setup (WPS)

Wi-Fi Protected Setup (WPS) provides simplified mechanisms to configure secure wireless networks. WPS can automatically configure a wireless network with a network name (SSID) and strong WPA data encryption and authentication.

Two methods supported by the IP phone in the Wi-Fi protected setup:

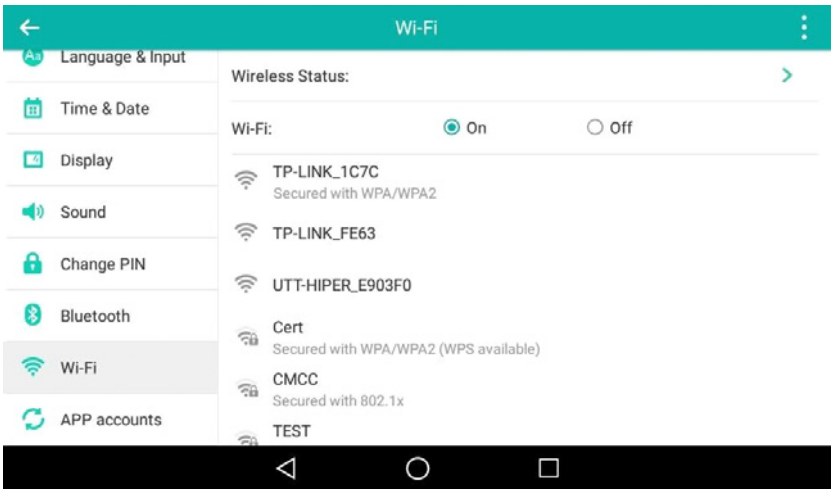
- **Push Button Configuration (PBC):** The user simply has to push the WPS key on both the IP phone and gateway/router to connect.
- **Personal Identification Number (PIN):** The user has to enter a WPS PIN generated randomly by the IP phone on the gateway/router to connect.

Push Button Configuration (PBC)

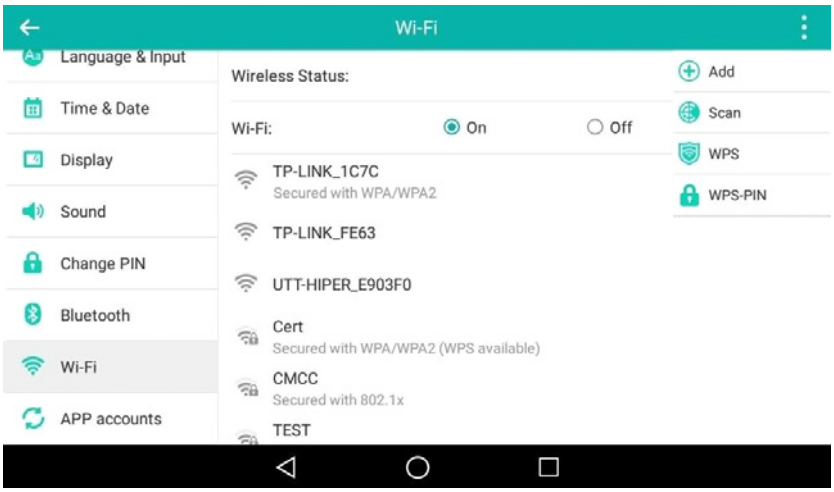
To connect to the wireless network using PBC method via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the **On** radio box in the **Wi-Fi** field.

  The phone will automatically search for available wireless networks in your area.



- Tap and then tap **WPS**.



The touch screen prompts “Press the Wi-Fi Protected Setup button on your router. It may be called “WPS” or contain this symbol:”.

- Long press the **WPS** key on your gateway/router.

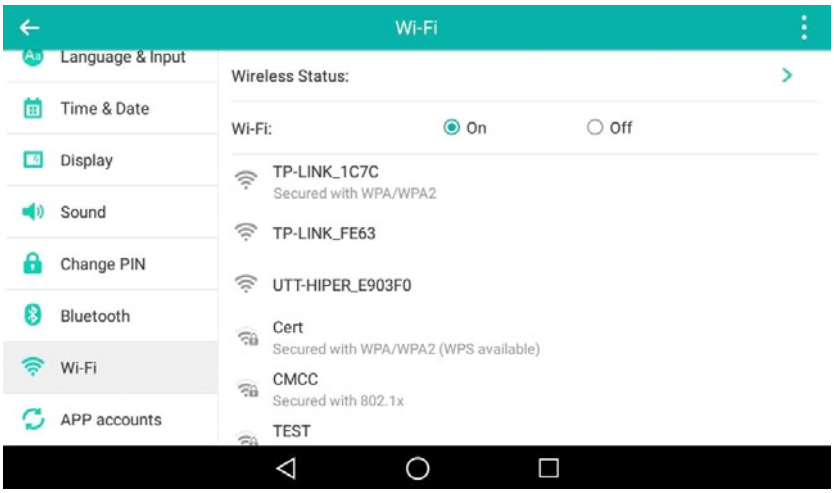
  Once WPS setup has completed successfully, the touch screen will prompt “Connected”.

Personal Identification Number (PIN)

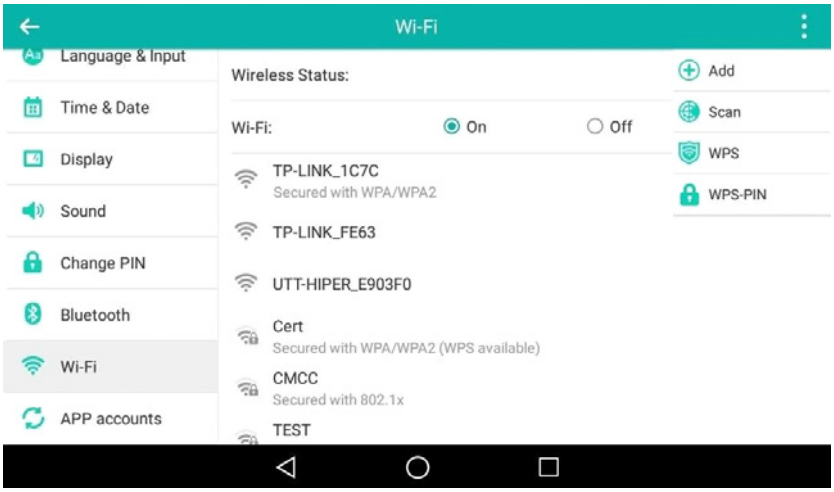
To connect to the wireless network using PIN method via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the **On** radio box in the Wi-Fi field.

The phone will automatically search for available wireless networks in your area.



- Tap and then tap **WPS-PIN**.



The touch screen prompts “Enter pin XXX on your Wi-Fi router. The setup can take up to two minutes to complete.”.

- Note the PIN code.
- Enter the gateway/router address in the address bar of a web browser on your PC connected to the gateway/router.
- Enter the user name and password in the login page.
- Locate the **WPS Wizard** configuration menu.
- Enter the PIN code in the field that allows you to setup wireless devices by using a PIN.
- Click the corresponding button in the gateway/router’s web interface to search WPS clients.

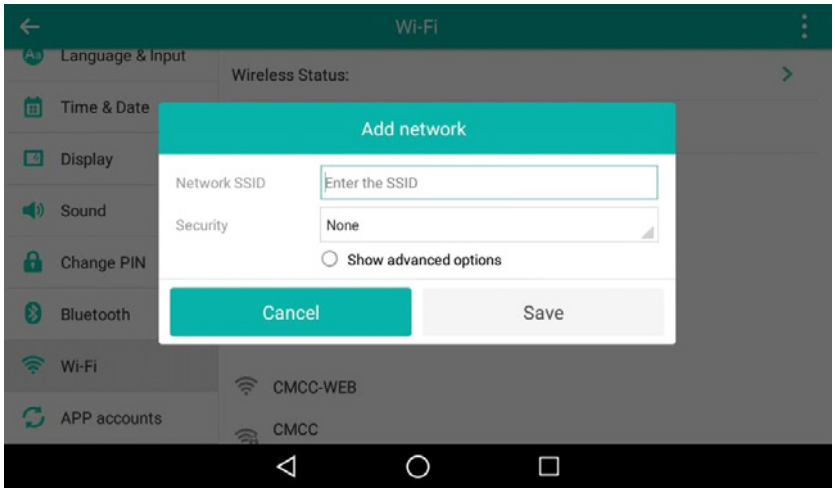
  Once WPS setup has completed successfully, the touch screen will prompt “Connected”.

Manually Add a Wireless Network

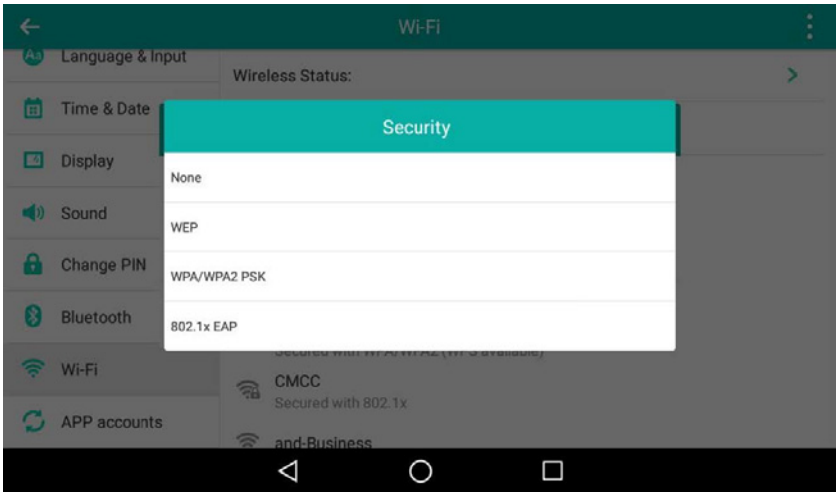
If your gateway/router has SSID broadcast disabled, it might not appear in the scanning results. In that case, you must manually add a wireless network.

To add a wireless network via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the **On** radio box in the Wi-Fi field.
- Tap and then tap **Add**.



- Enter the desired value in the **Network SSID** field.
- Tap the **Security** field.



- Tap the desired value.
- If you select **WEP** or **WPA/WPA2 PSK**:
  - Enter the password in the **Password** field.
- If you select **802.1x EAP**:

- Tap the **EAP method** field.
- Tap the desired EAP method in the pop-up dialog box.

- If you select **PEAP/TTLS**:
  - Tap the **Phase-2 authentication** field.
  - Tap the desired Phase-2 authentication method in the pop-up dialog box.
  - Enter the identity (username) in the **Identity** field.
  - Enter the anonymous identity (username) in the **Anonymous identity** field (to be used as the unencrypted identity).
  - Enter the password in the **Password** field.

- If you select **TLS**:
  - Enter the username in the **Identity** field

- If you select **PWD**:
  - Enter the username in the **Identity** field.
  - Enter the password in the **Password** field.

- You can do the following:
  - Tap the **Show password** radio box to make the password visible.
  - Tap the **Show advanced** options radio box to configure the HTTP proxy for Browser application.

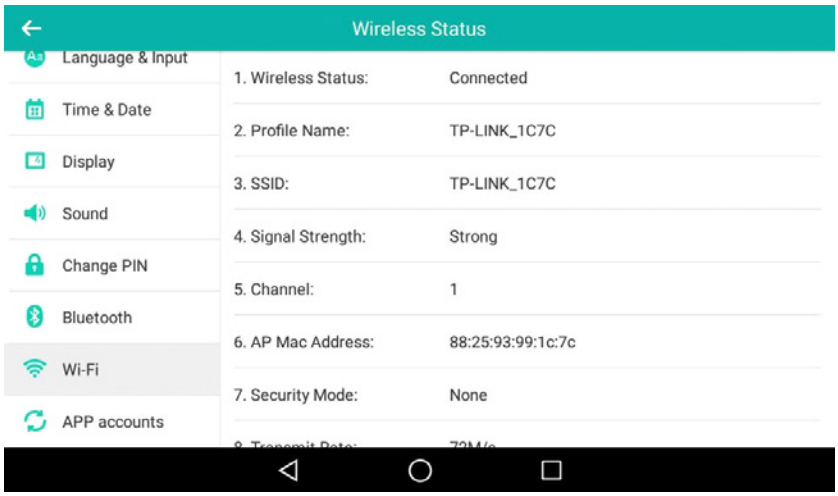
- Tap **Save** to accept the change.

Viewing the Connected Wireless Network Information

When the phone is connected to a wireless network, the Wi-Fi icon will display on the status bar. The Wi-Fi icon indicates the signal strength. The more green arcs you see, the stronger the signal strength is.

To view the connected wireless network information via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap **Wireless Status** to view the detailed wireless network information (e.g., Profile Name, SSID or Signal Strength).



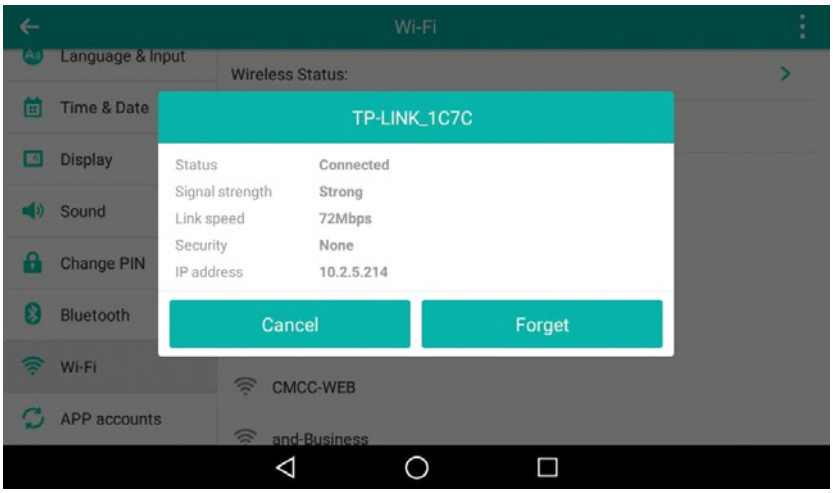
- Drag up and down to scroll through the list of wireless network information (Optional).

Disconnecting the Wireless Network Connection

You can disconnect the wireless network connection from your phone.

To disconnect the wireless network via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the connected SSID (the top one).

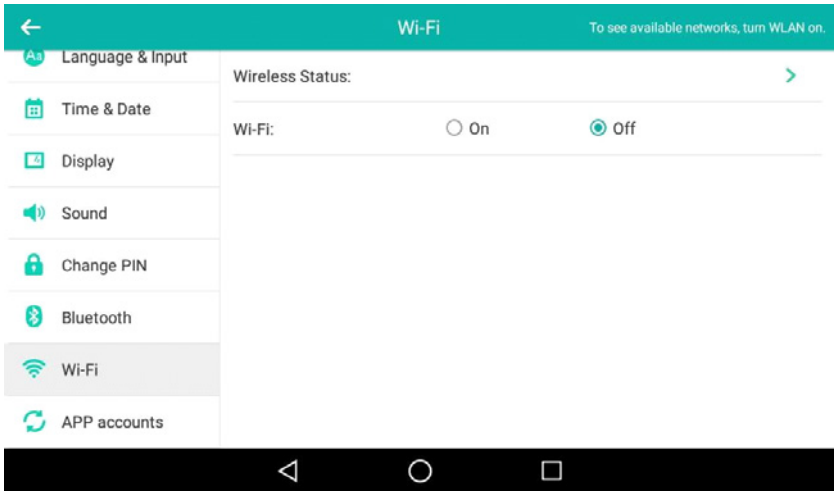


- Tap **Forget**.

Disabling Wi-Fi Feature

To disable Wi-Fi feature via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Wi-Fi.**
- Tap the Off radio box in the Wi-Fi field.



- Swipe down from the top of the screen to enter the control center, and then tap 
- The  icon disappears from the status bar, and the Wi-Fi label resumes to Wi-Fi.

Configuring Transmission Methods of the Internet Port and the PC Port

Two Ethernet ports on the back of the IP phone: Internet port and PC port. Three optional methods of transmission configuration for IP phone Internet or PC Ethernet ports:

- Auto-negotiation
- Half-duplex
- Full-duplex

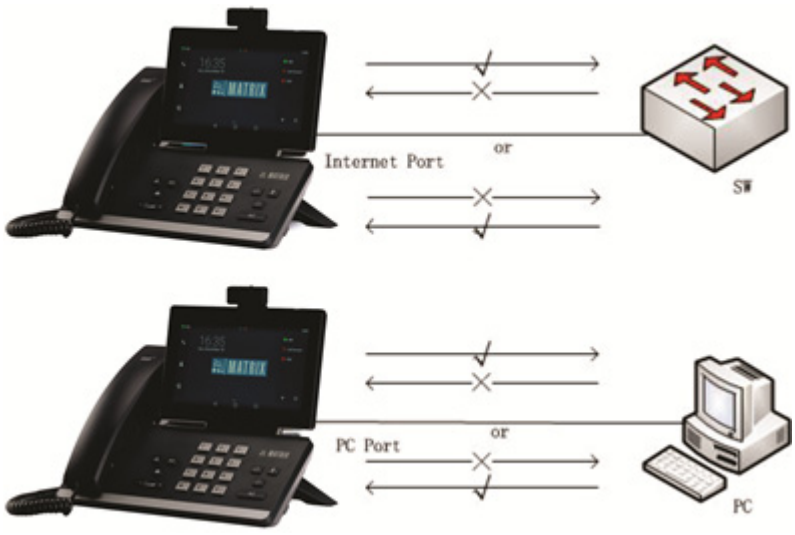
Auto-negotiation is configured for both Internet and PC ports on the IP phone by default.

Auto-negotiation

Auto-negotiation means that two connected devices choose common transmission parameters (e.g., speed and duplex mode) to transmit voice or data over Ethernet. This process entails devices first sharing transmission capabilities and then selecting the highest performance transmission mode supported by both. You can configure the Internet port and PC port on the IP phone to automatically negotiate during the transmission.

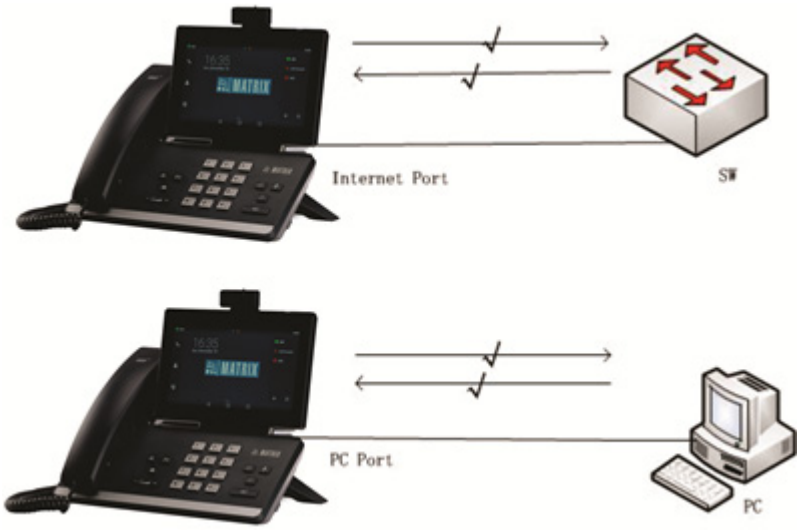
Half-duplex

Half-duplex transmission refers to transmitting voice or data in both directions, but in one direction at a time; this means one device can send data on the line, but not receive data simultaneously. You can configure the half-duplex transmission on both Internet port and PC port for the IP phone to transmit in 10Mbps or 100Mbps.



Full-duplex

Full-duplex transmission refers to transmitting voice or data in both directions at the same time; this means one device can send data on the line while receiving data. You can configure the full-duplex transmission on both Internet port and PC port for the IP phone to transmit in 10Mbps or 100Mbps.



Procedure

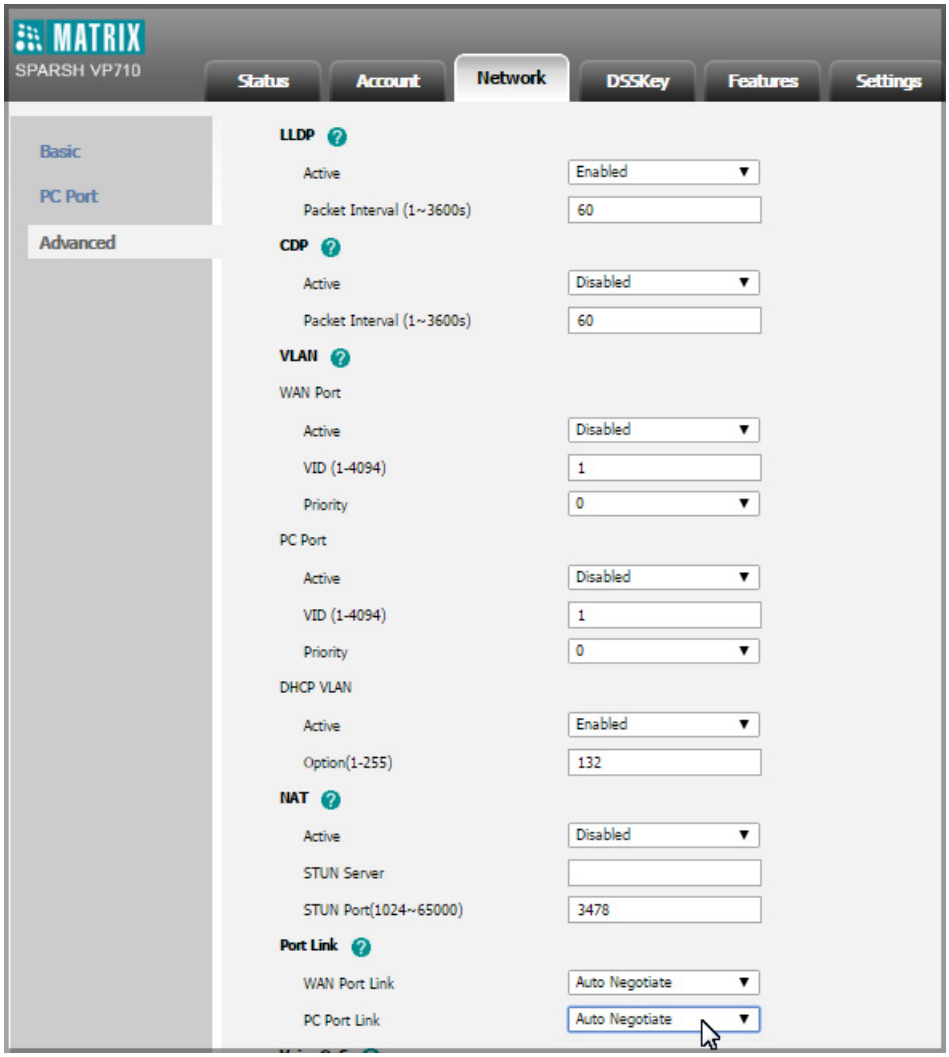
The transmission methods of Ethernet ports can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transmission methods of Ethernet ports. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the transmission methods of Ethernet ports. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_ data&p=network-adv&q=load

To configure the transmission methods of Ethernet ports via web user interface:

- Click on **Network->Advanced**.
- Scroll to **Port Link** and select the desired value from the **WAN Port Link** list.

- Select the desired value from the **PC Port Link** list.



- Click **Confirm** to save the change.
  A dialog box pops up to prompt that settings will take effect after a reboot.
- Click OK to reboot the phone.

Configuring PC Port Mode

The PC port on the back of the IP phone is used to connect a PC, which can be configured in the following mode:

- **Bridge:** The IP phone functions as a bridge, and the connected PC appears on the network as a stand-alone device with its own IP address.

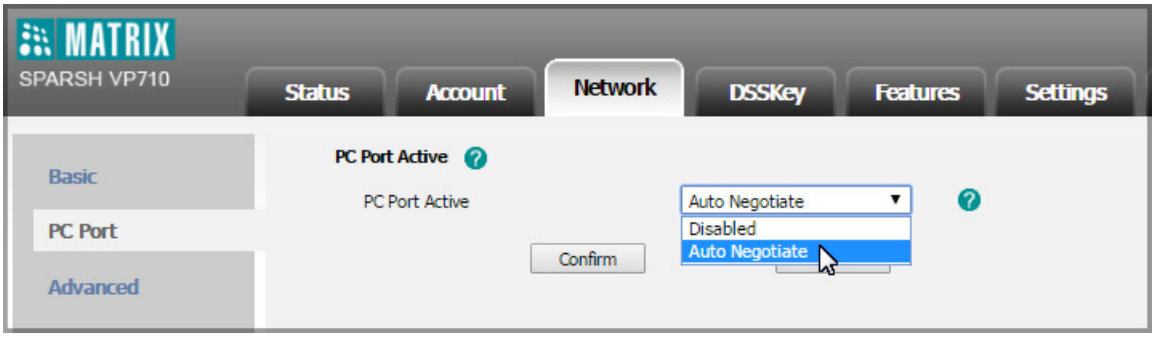
Procedure

PC port mode can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the PC port mode. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the PC port mode. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-pcport&q=load
	Phone User Interface	None

To configure the PC port mode via web user interface:

- Click on **Network->PC Port**.
- Select the desired value from the **PC Port Active** list.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To disable the PC port via web user interface:

- Click on **Network->PC Port**.
- Select **Disabled** from the **PC Port Active** list.
- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

CHAPTER 6

Customizing Your Phone

You can customize your IP phone personally by configuring certain settings, for example, contrast, language and time & date. You can add contacts to the phone's local directory manually or from call history. You can also personalize different ring tones for different callers.

This chapter provides basic operating instructions for customizing your phone. Topics include:

- General Settings
- Audio Settings
- Contact Management
- Call History Management
- System Customizations

General Settings

Wallpaper

Procedure

Wallpaper can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the wallpaper displayed on the IP phone. Specify the access URL of the custom wallpaper. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure the wallpaper displayed on the IP phone. http://<phoneIPAddress>/servlet?m=mod_data&p=settings-preference&q=load
	Phone User Interface	Configure the wallpaper displayed on the IP phone.

You can customize the wallpaper of the IP phone. You can also change the wallpaper image via phone user interface or web user interface. The custom wallpaper can be a built-in image, a picture stored in internal SD card/ USB flash drive (if connected) or a picture uploaded via web user interface. The IP phone supports the wallpaper images in *.jpg/*.png/*.bmp/*.jpeg format.

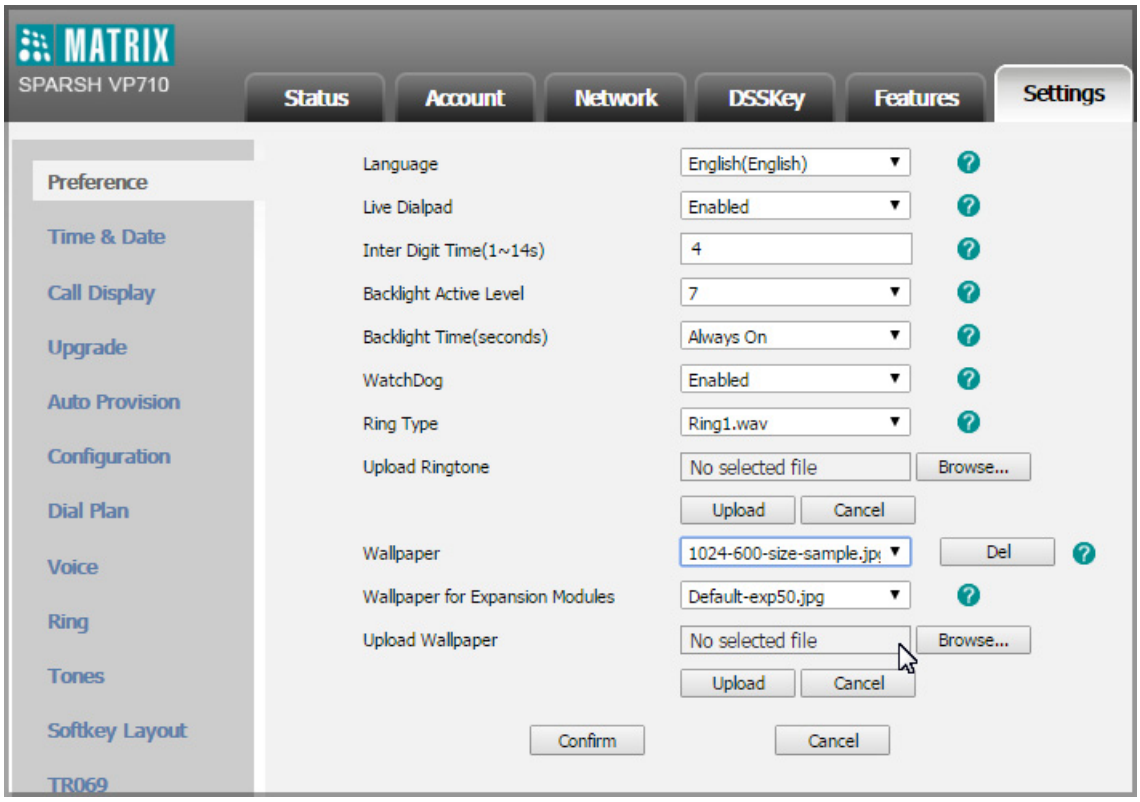
You can select a wallpaper from different categories:

- **Wallpapers:** Set a recently used picture as the wallpaper, or tap 'Pick image' to display the available pictures from the Recent/Images/Downloads/Gallery list and then select a desired picture as the wallpaper.
- **Gallery:** Launch Gallery application, tap a Gallery albums and then select a desired picture as the wallpaper.

To upload custom wallpaper via web user interface:

- Click on **Settings->Preference.**
- In the **Upload Wallpaper** field, click **Browse** to locate and upload the picture from your local system.

- Click **Upload** to upload the file.



The uploaded custom picture will apply to the IP phone and appear in the **Wallpaper** list. You can delete the custom wallpaper by clicking **Del** when selecting the desired custom wallpaper in the **Wallpaper**.

To change the wallpaper via web user interface:

- Click on **Settings->Preference**.
- Select the desired wallpaper from the **Wallpaper** list.
- Click **Confirm** to save the change.

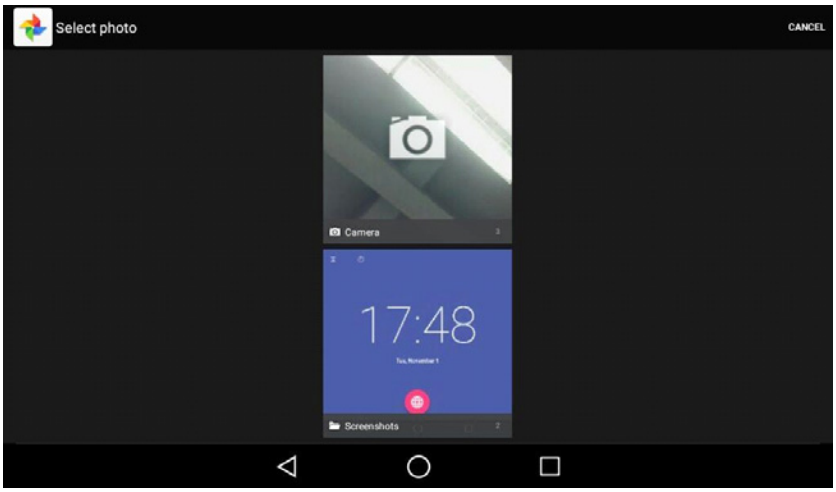
To change the wallpaper via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Display**.
- Tap **Wallpaper**.

Do one of the following:

- Tap **Gallery**.

The touch screen displays all Gallery albums.

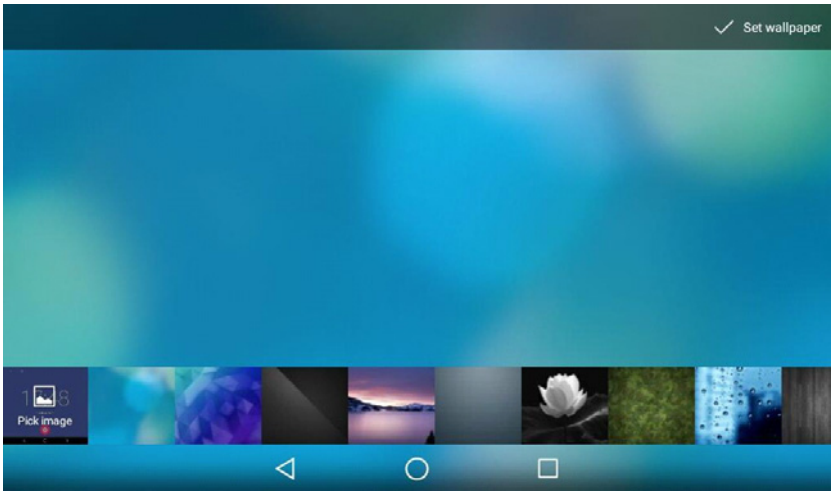


- Select a desired Gallery album.
- Tap a desired picture to preview, and then tap **Set wallpaper.**



OR

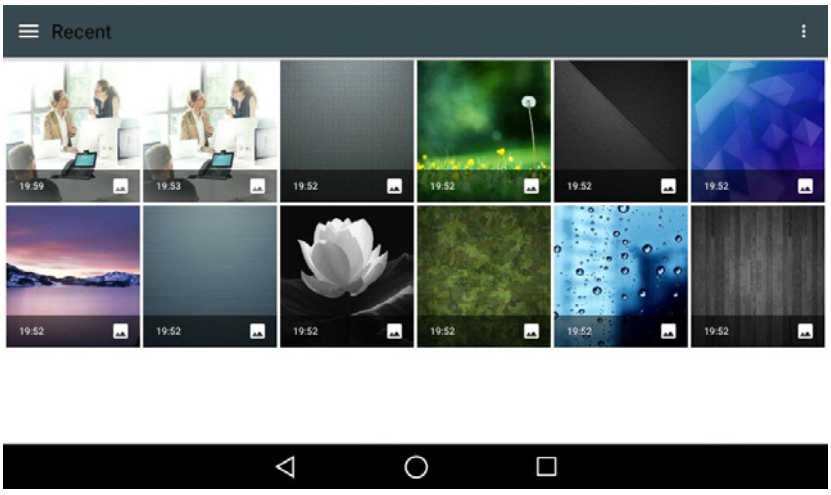
- Tap **Wallpapers**.
  - The touch screen displays the current used wallpaper in the background, built-in wallpapers and the recently used wallpapers list on the bottom of the touch screen.



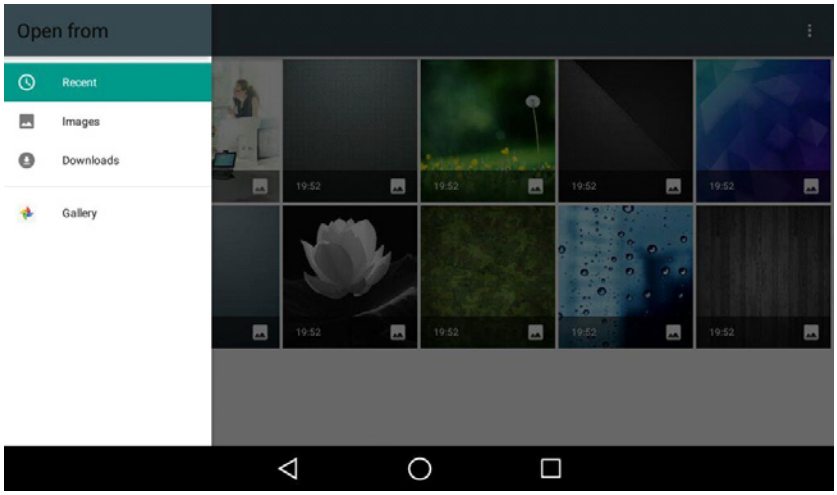
Do one of the following:

- Select a desired wallpaper from the build-in wallpapers and recently used wallpapers list, and then tap **Set wallpaper**.

  OR
- Tap **Pick** image. The touch screen displays the recently saved pictures.



- Tap  on the top-left of the touch screen.



Do one of the following:

- Tap **Images** to see all pictures stored in internal SD card or USB flash drive (if connected).
  OR
- Tap **Downloads** to see all pictures you have downloaded.
  OR
- Tap **Gallery** to see all pictures by Gallery application.

  Select a desired picture to preview.

  The picture will appear in the recently used wallpapers list.



- Tap **Set wallpaper.**

You can also long tap the empty spot on the idle screen and then tap **WALLPAPERS**, the effect is equal to tapping **Settings->Basic->Display->Wallpaper->Wallpapers**. Then you can change the phone wallpaper as per the instructions mentioned above.

You can also change the phone wallpaper when viewing pictures by FileManager or Gallery application. For more information, refer to “FileManager” and “The pictures are transferred as mail attachment. The attachment files size must be within 5MB. You can also share pictures with other devices via Bluetooth or email when previewing the pictures. And the operating instructions are the same with sharing pictures by Gallery application. For more information, refer to “Sharing Pictures/Videos”.”.

The phone screen size is 7 inches (Resolution: 1024x600). If the size of the custom picture doesn’t meet the IP phone screen size, the wallpaper will be stretched or/and zoomed out according to the screen size.

Screen Saver

The screen saver will automatically start each time your phone is idle for a certain amount of time. The screen saver is used to blank the screen or fill it with moving images or patterns.

The screen saver will be stopped when:

- Press any key.
- Tap the touch screen.
- Pick up or hang up the handset.
- There is an incoming call.
- A new prompt (e.g., missed call, new voice mail or forwarded call).
- The status of BLF key changes

If your phone is idle again for a specified period of time, the screen saver will start again.

The IP phone supports four screen saver types: Clock, Colors, Photo Frame and Photo Table.

Screen saver is configurable via phone user interface only.

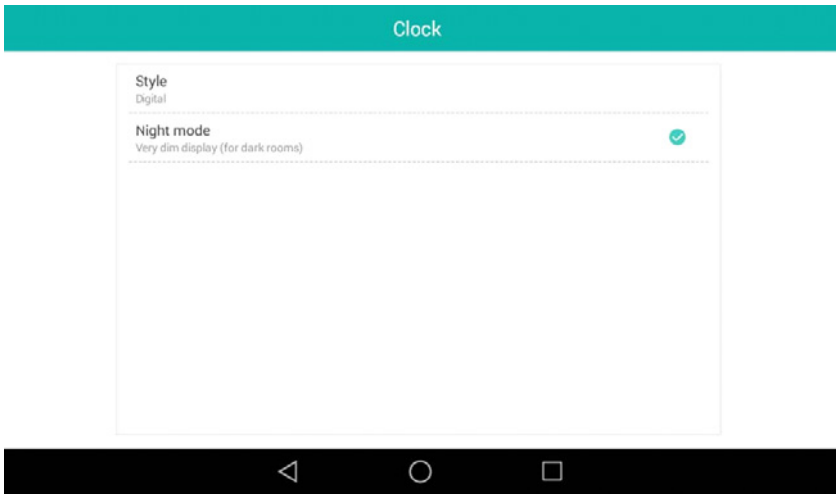
To configure the screen saver via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Display->Screen Saver.**
- Tap the **Screen Saver Wait Time** field.
- Tap the desired time in the pop-up dialog box.

  Do one of the following:

  - Tap the **Clock** radio box.

Tap next to the radio box (Optional).



- Tap the **Style** field to set the clock type to **Analog** or **Digital**.
- Tap the **Night mode** radio box to display the screen saver dimly for dark rooms.
- Tap to return to the Screen Saver setting screen.

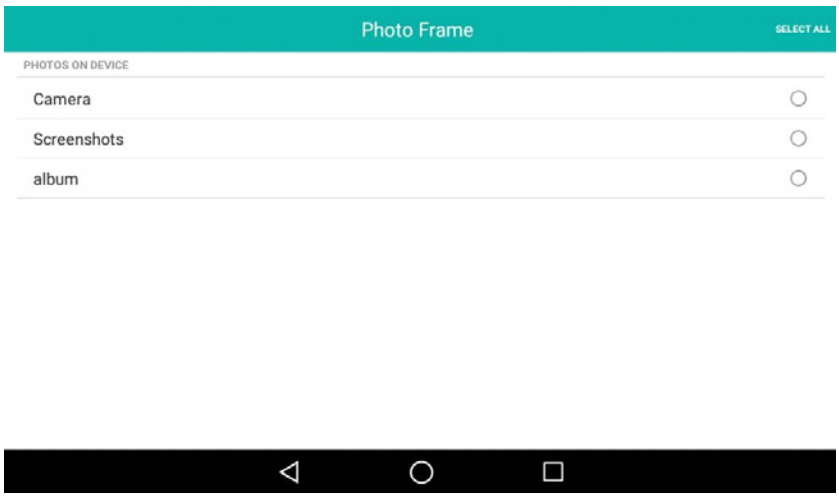
OR

- Tap the **Colors** radio box.

OR

- Tap the **Photo Frame** radio box.

- Tap next to the radio box to select the desired Gallery album(s).
- Tap the desired checkbox or **SELECT ALL** on the top-right of the touch screen.

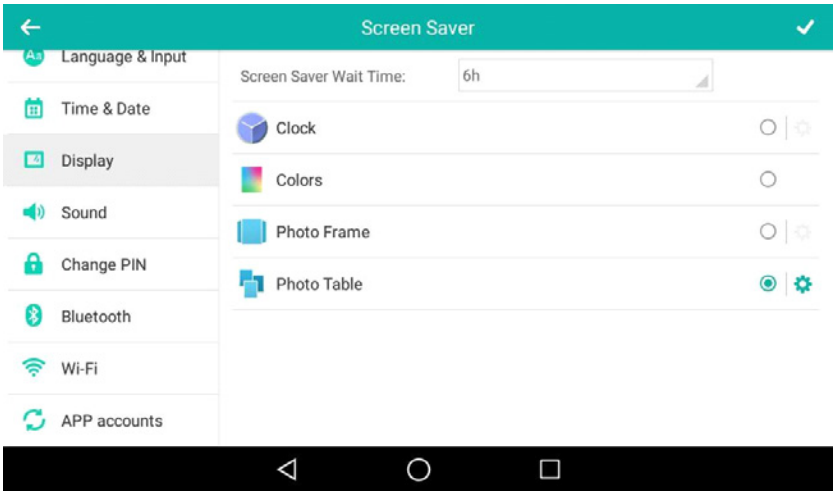


- Tap to return to the Screen Saver setting screen.

OR

- Tap the **Photo Table** radio box.

- Tap next to the radio box to select the desired Gallery album(s).
- Tap the desired checkbox or **SELECT ALL** on the top-right of the touch screen.
- Tap to return to the Screen Saver setting screen.



- Tap to accept the change or to cancel.

Power Saving

The power saving feature is used to turn off the backlight and screen to conserve energy. The IP phone enters power-saving mode after it has been idle for a certain period of time.

The IP phone will exit power-saving mode when:

- Press any key.
- Tap the touch screen.
- Pick up or hang up the handset.
- There is an incoming call.
- A new prompt (e.g., missed call, new voice mail or forwarded call).
- The status of BLF key changes.

Power saving is configurable via web user interface only.

If you disable the power saving feature, the IP phone will automatically enter power-saving mode to protect the screen when the phone is inactive for 72 hours.

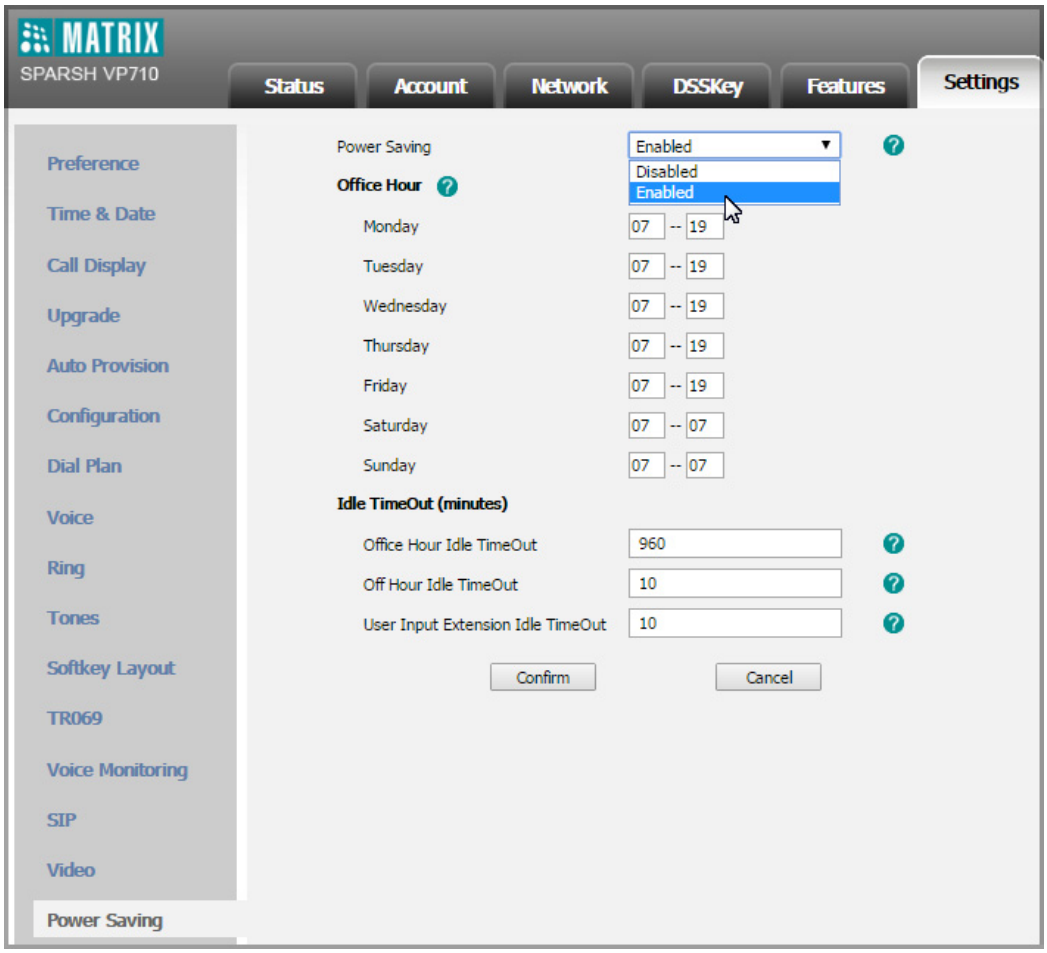
Procedure

Power Saving can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the power saving feature. Configure the office hour. Configure the idle timeout. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the power saving feature. Configure the office hour. Configure the idle timeout. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-powersaving&q=load

To enable the power saving feature via web user interface:

- Click on **Settings->Power Saving.**
- Select **Enabled** from the **Power Saving** list.



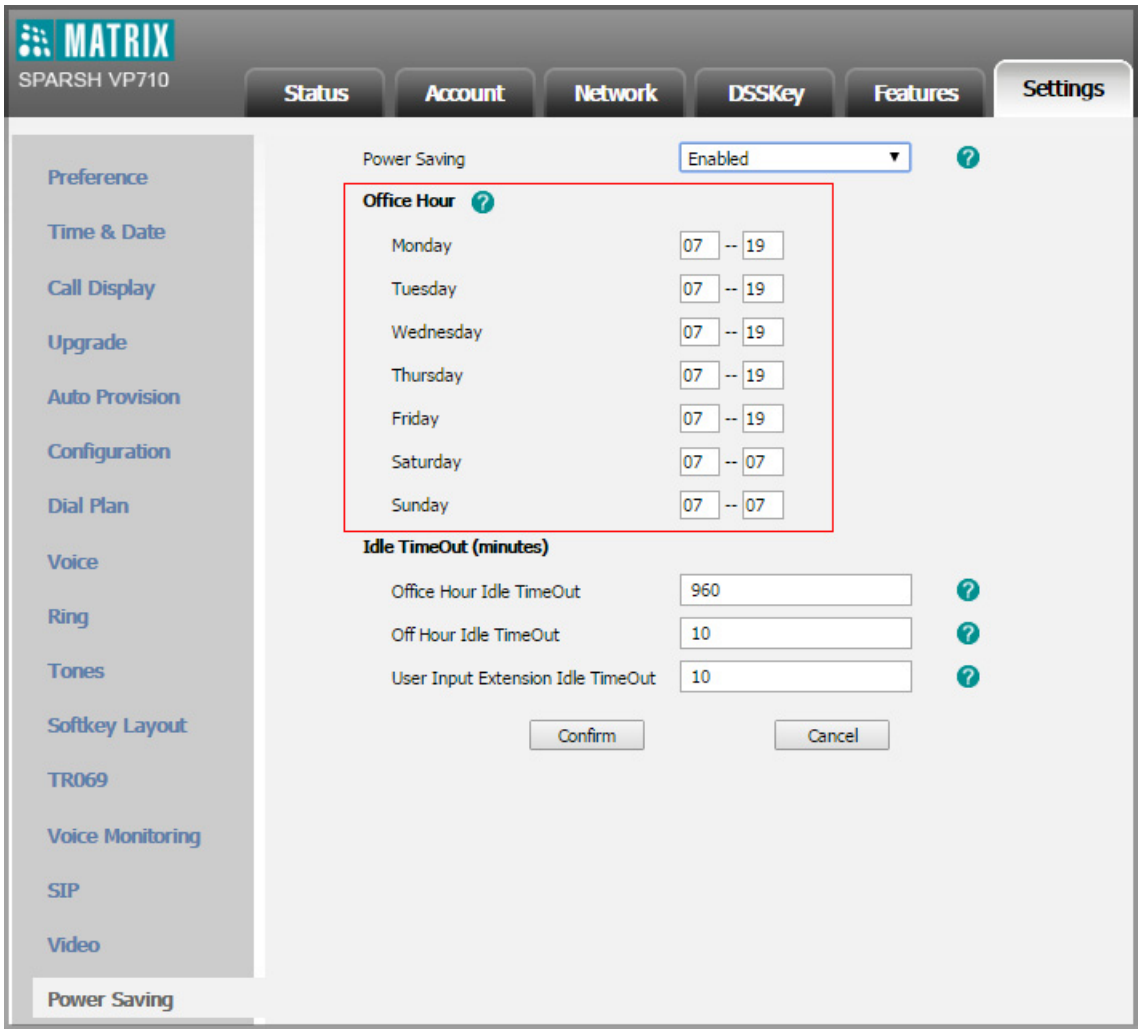
- Click **Confirm** to save the change.

To configure the office hour via web user interface:

Office Hour specifies the starting time and ending time in the office each day.

- Click on **Settings->Power Saving.**
- Select a desired day of the week.

  Enter the starting time and ending time respectively in the desired day field.

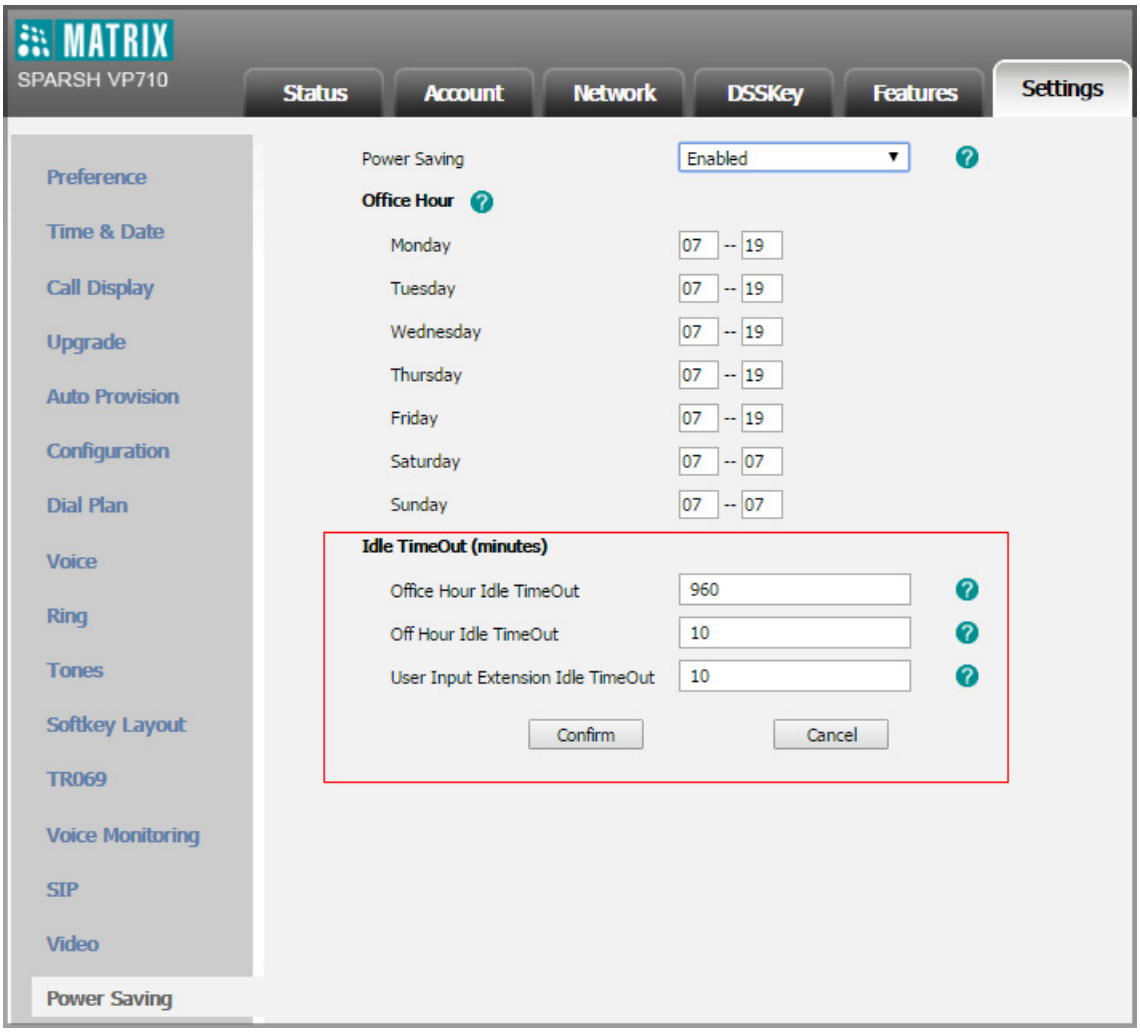


- Click **Confirm** to save the change.

To configure the idle timeout via web user interface:

- Click on **Settings->Power Saving.**
- Enter the desired value in the **Office Hour Idle TimeOut** field.
- The default value is 960, you can set to 1-960.
- Enter the desired value in the **Off Hour Idle TimeOut** field.
- The default value is 10, you can set to 1-10.
- Enter the desired value in the **User Input Extension Idle TimeOut** field.

- The default value is 10, you can set to 1-30.



- Click **Confirm** to save the change.

Backlight

Backlight determines the brightness of the touch screen display, allowing users to read easily in dark environments. Backlight time specifies the delay time to turn off the backlight when the IP phone is inactive. Backlight turns off quickly if a short backlight time is configured, this may not give users enough time to read messages.

Backlight has two options:

- **Backlight Active Level:** The intensity of the touch screen when the phone is active. You can change the intensity of the touch screen by dragging the backlight active level slider.
- **Backlight Time:** The delay time to change the intensity of the touch screen when the phone is inactive. You can select a desired time to change the intensity or turn on the backlight permanently.
  - **Always On:** Backlight is on permanently.
  - **15s, 30s, 60s, 120s, 300s, 600s or 1800s:** Backlight is changed when the phone is inactive after the designated time (in seconds).

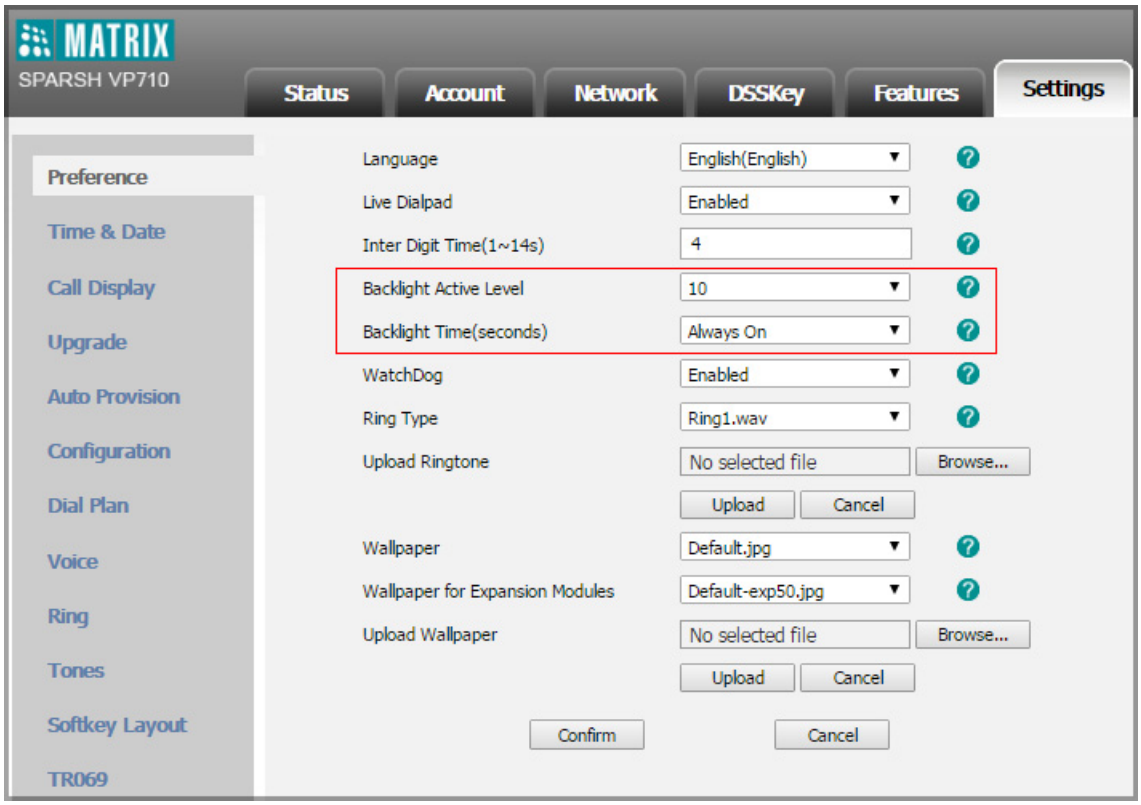
Procedure

Backlight can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the backlight of the touch screen. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the backlight of the touch screen. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_ data&p=settings-preference&q=load
	Phone User Interface	Configure the backlight of the touch screen.

To configure the backlight via web user interface:

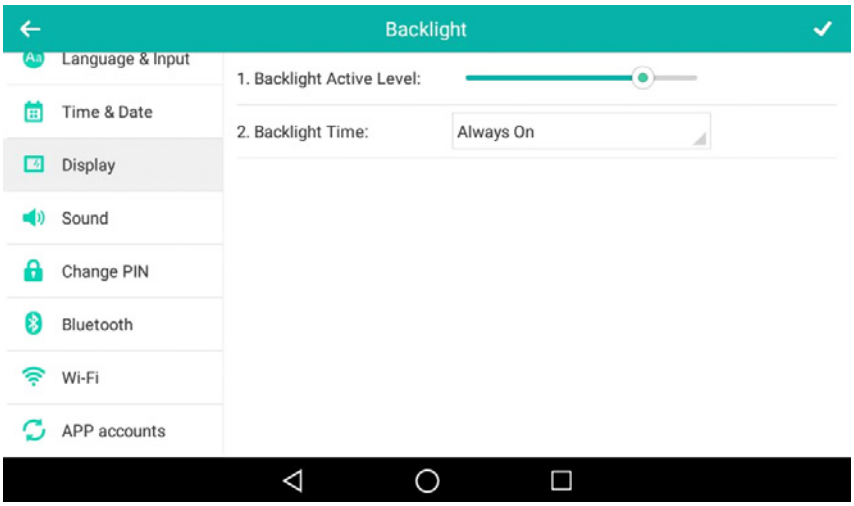
- Click on **Settings->Preference.**
- Select the desired value from the **Backlight Active Level** list.
- Select the desired value from the **Backlight Time(seconds)** list.



- Click **Confirm** to save the change.

To configure the backlight via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Display->Backlight.**
- Drag the **Backlight Active Level** slider to change the intensity of the touch screen.
- Tap the **Backlight Time** field.
- Tap the desired time in the pop-up dialog box.



- Tap ✓ to accept the change or ◁ to cancel.

You can also drag the backlight slider on the control center to change the intensity of the touch screen.

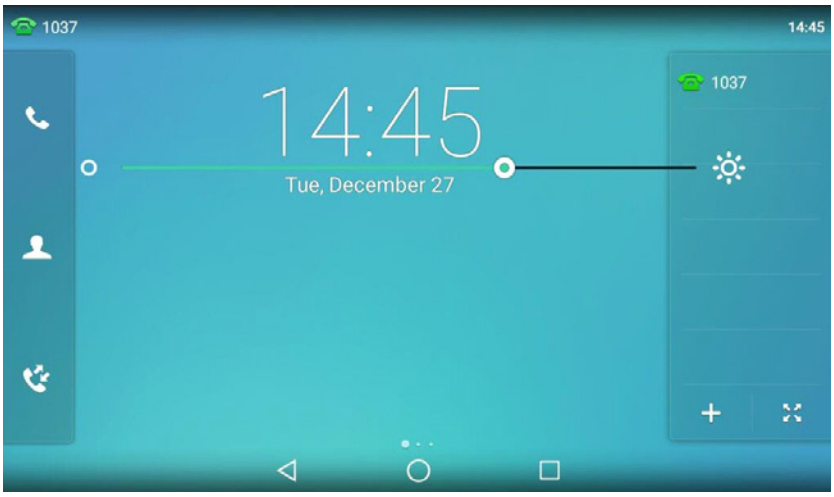
To configure the backlight active level on the control center:

- Swipe down from the top of the screen to enter the control center.

  You can do one of the following:

  - Drag the backlight slider.

When dragging the slider, the control center (except the backlight slider) and notification center will be hidden. You can view the intensity changes of the touch screen in real-time



If you lift your finger, the control center and notification center will be shown again.

OR

- Tap . The intensity of the touch screen is changed to the highest.

OR

- Tap . The intensity of the touch screen is changed to the lowest.

Bluetooth

Bluetooth enables low-bandwidth wireless connections within a range of 10 meters (32 feet). The best performance is in the 1 to 2 meters (3 to 6 feet) range. You can activate/deactivate the Bluetooth mode and then pair and connect the Bluetooth device with your phone. You can pair upto countless Bluetooth devices with your phone; however, only one Bluetooth device can be connected at a time. You can pair and connect a Bluetooth headset/ speakerphone/microphone or a Bluetooth-enabled mobile phone to your IP phone.

Users can transfer files via Bluetooth, sharing images/videos with other Bluetooth devices. You can personalize the Bluetooth device name for the IP phone. The pre-configured Bluetooth device name will display in scanning list of other devices. It is helpful for the other Bluetooth devices to identify and pair with your IP phone.

For more information on Bluetooth-enabled mobile phone, refer to Bluetooth-Enabled Mobile Phone.

Procedure

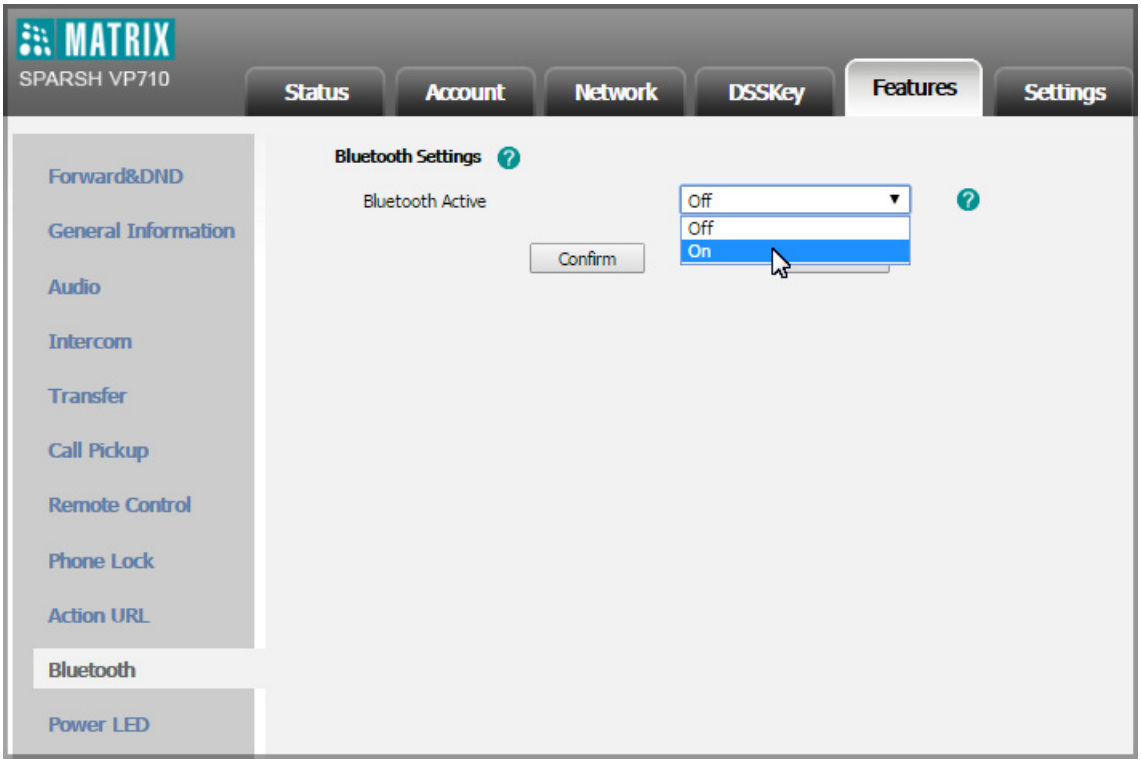
Bluetooth can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure Bluetooth mode. Configure the Bluetooth device name. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure Bluetooth mode. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-bluetooth&q=load
	Phone User Interface	Configure Bluetooth mode. Configure the Bluetooth device name.

Activating/Deactivating the Bluetooth Mode

To active the Bluetooth mode via web user interface:

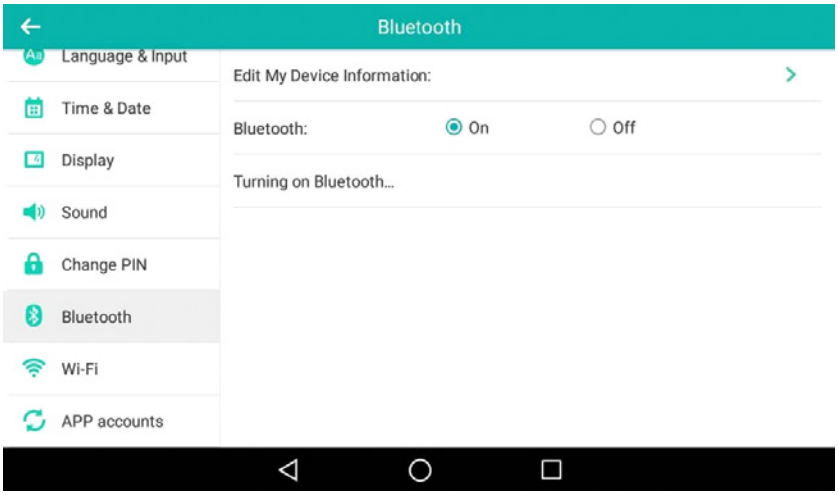
- Click on **Features->Bluetooth.**
- Select the desired value from the **Bluetooth Active** list.



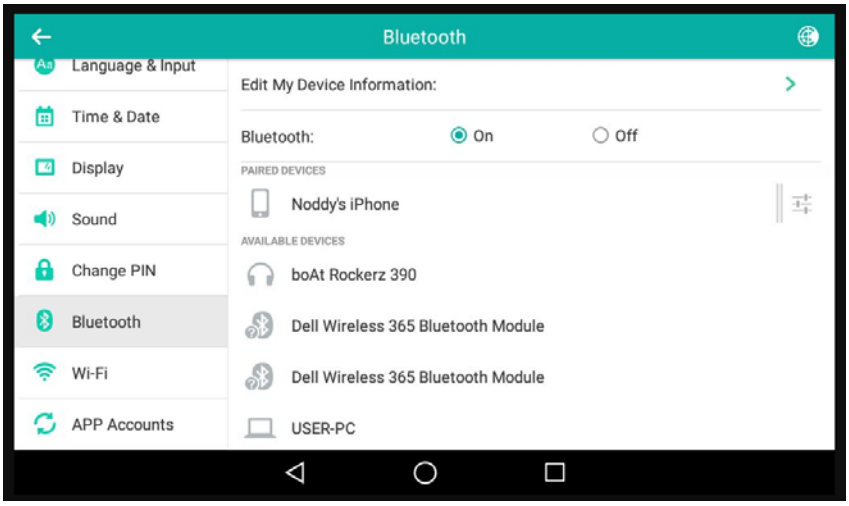
- Click **Confirm** to save the change.

To activate the Bluetooth mode via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the **On** radio box in the **Bluetooth** field.



The IP phone scans the available Bluetooth device automatically. After scanning, the touch screen displays the Bluetooth device information (device type and device name).

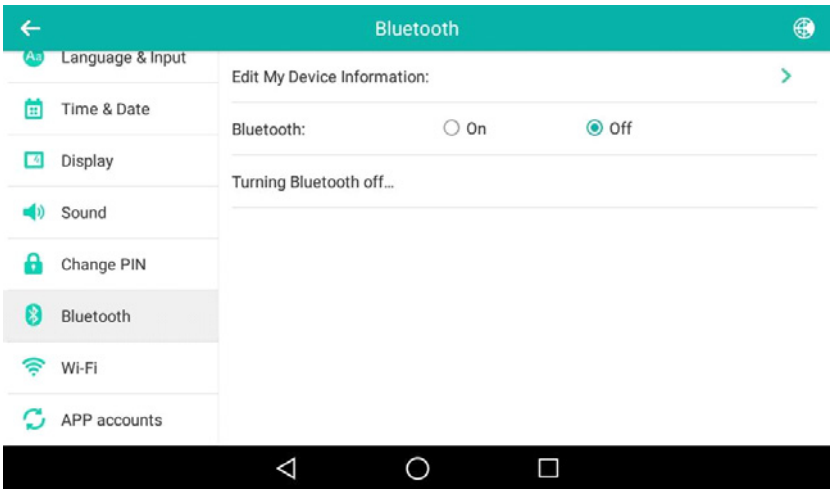


- The touch screen displays the Bluetooth icon.

  You can also swipe down from the top of the screen to enter the control center, and toggle Bluetooth on.

To deactivate the Bluetooth mode via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the **Off** radio box in the **Bluetooth** field



The Bluetooth icon disappears from the status bar.

Bluetooth Headset

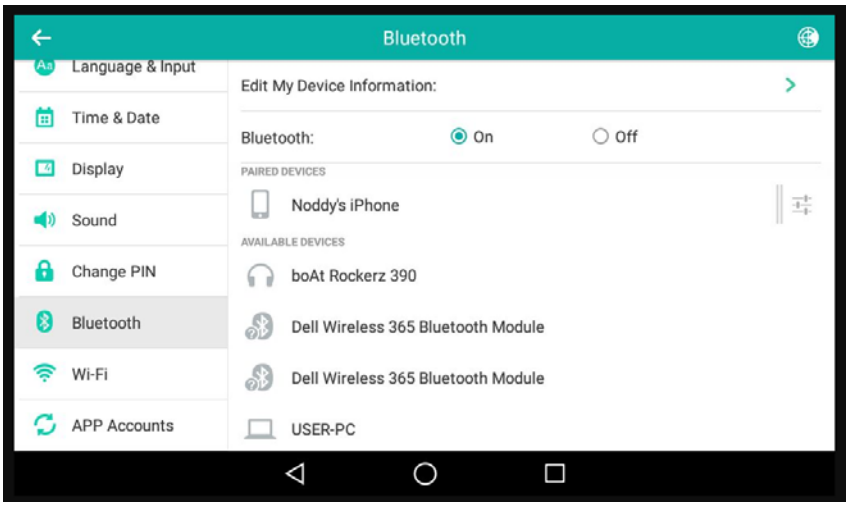
Pairing and Connecting the Bluetooth Headset

Before you pair a Bluetooth headset, check the indicator on the Bluetooth headset to make sure that the Bluetooth headset is discoverable. For more information, refer to the documentation from the Bluetooth headset manufacturer.

To pair and connect your Bluetooth headset to your phone:

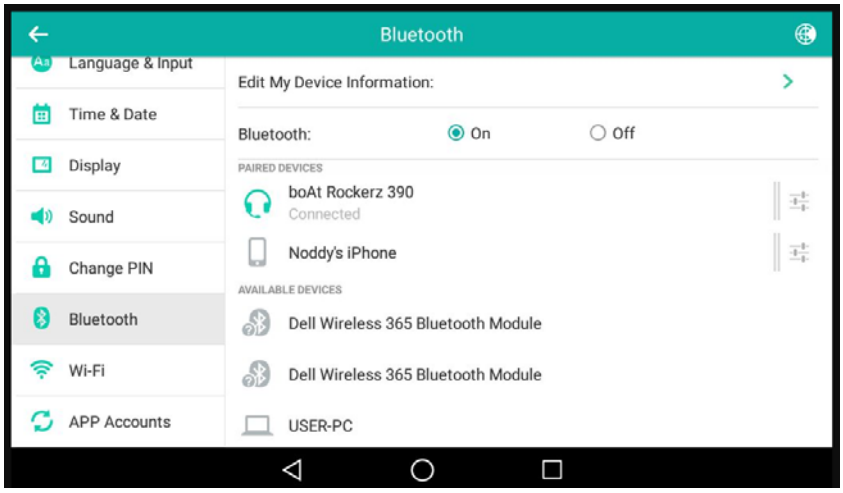
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the **On** radio box in the **Bluetooth** field.

  If there is no Bluetooth device found on the touch screen, tap  to search for the Bluetooth device.

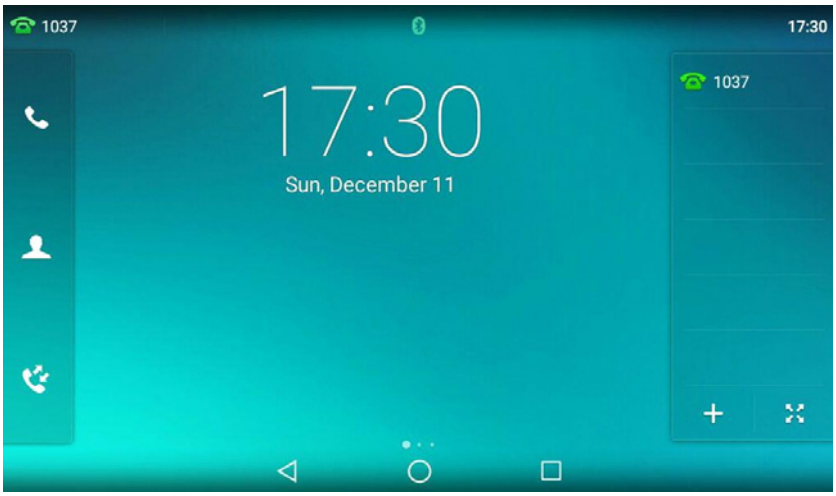


- Tap the desired Bluetooth headset to connect to your phone.

Once the connection has completed successfully, the prompt “Connected” appears under the Bluetooth headset name.



- The Bluetooth icon appears on the status bar.



With the Bluetooth headset paired and connected, you can use the Bluetooth headset to place and answer calls. For more information, refer to the documentation from the Bluetooth headset manufacturer.

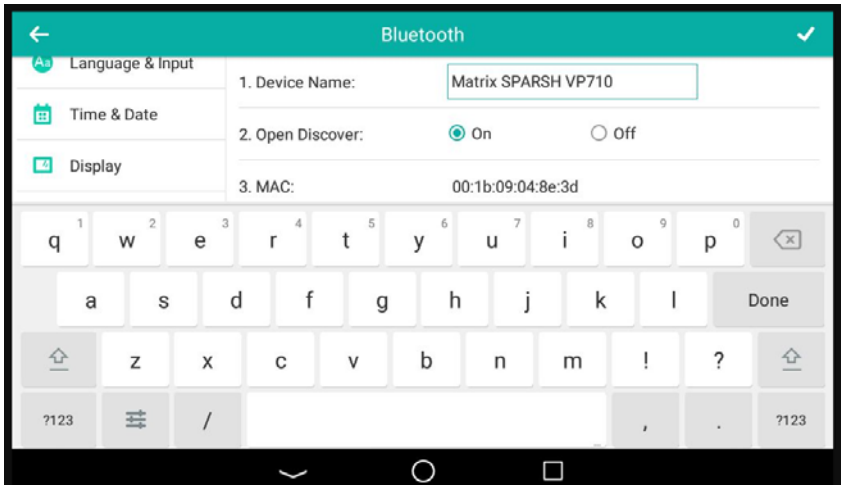
Making the Phone Discoverable

If you make your IP phone visible to other Bluetooth devices, the other Bluetooth devices can scan and find your IP phone. But if your IP phone is not visible to other Bluetooth devices, your IP phone’ device name will not display in scanning list of other device.

To make the phone discoverable:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the **On** radio box in the **Bluetooth** field.
- Tap **Edit My Device Information**.

- Tap the **On** radio box in the **Open Discover** field.

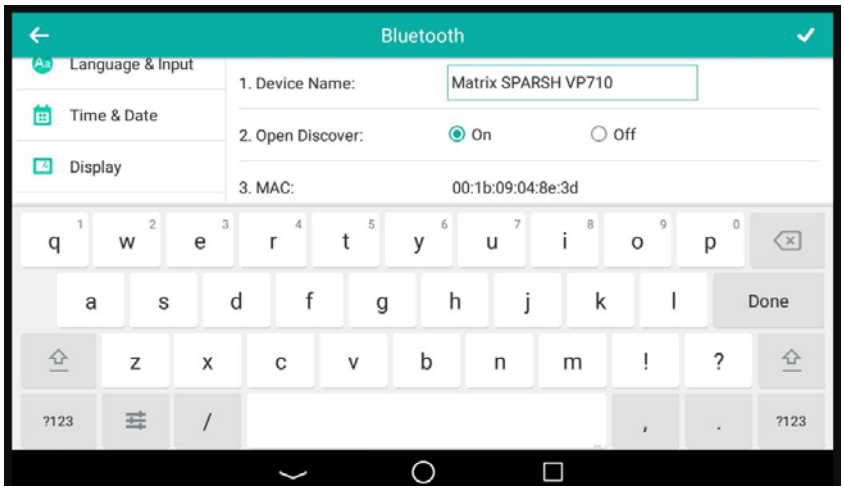


- The IP phone will be visible to all Bluetooth devices nearby.

Editing Device Information

To edit device information via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth->Edit My Device Information.**
- Enter the desired name in the **Device Name** field.
- The default device name is “Matrix SPARSH VP710”



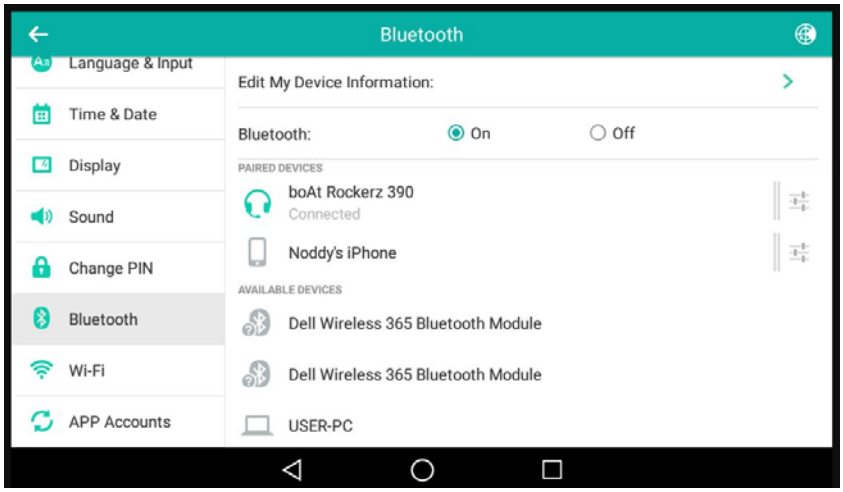
- Tap ✓ to accept the change or ◁ to cancel.
- You should make your phone discoverable first, and then the pre-configured Bluetooth device name will display in scanning list of other device.

Renaming the Paired Bluetooth Device

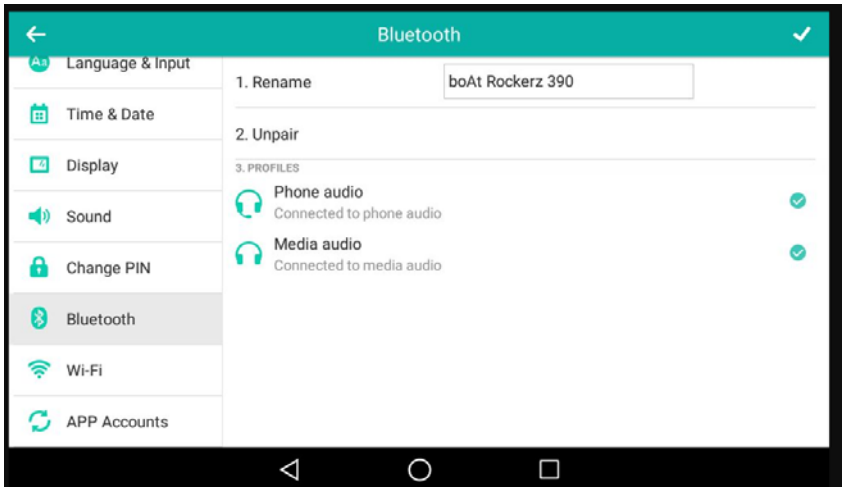
To rename the paired Bluetooth device:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**

  The touch screen displays the paired and connected Bluetooth headset.



- Tap  after the connected Bluetooth headset name.
- Enter the desired headset name in the **Rename** field.



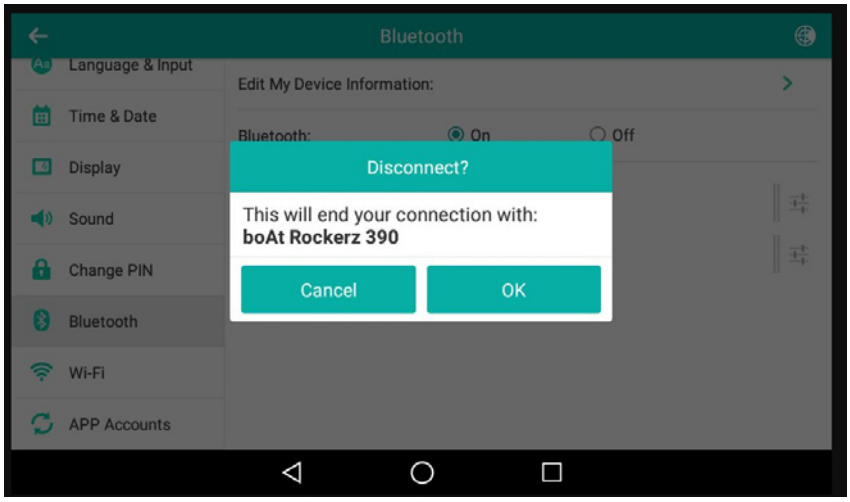
- Tap  to accept the change or  to cancel.

  The configured Bluetooth headset name will display in the **PARED DEVICES** list on your phone. The Bluetooth headset name displayed in scanning list of other devices will not be changed.

Disconnecting the Bluetooth Headset

To disconnect your Bluetooth headset from your phone:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the connected Bluetooth headset.



- Tap **OK** to accept the change or **Cancel** to cancel.
- The Bluetooth headset is disconnected. To connect it to your phone again, tap the Bluetooth headset again.

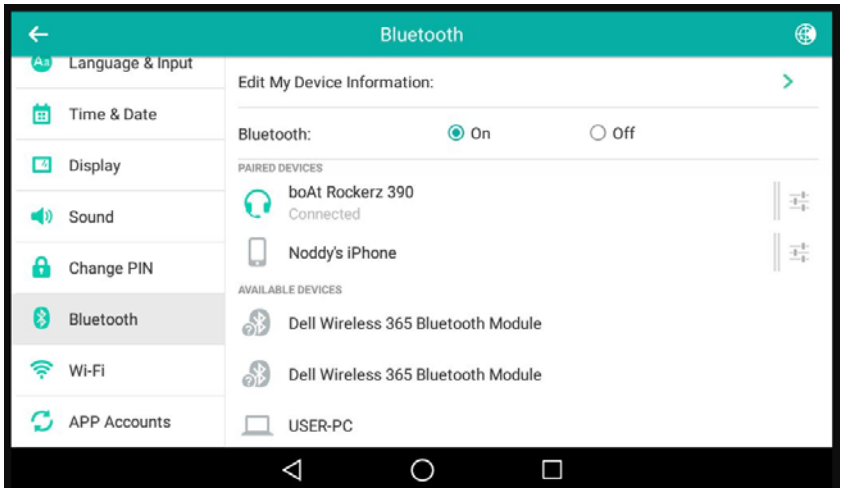
Unpairing the Bluetooth Headset

You can unpair your Bluetooth headset and your phone. When you unpair the Bluetooth headset, it disappears from the **PAIRED DEVICES** list. To connect it to your phone again, you just need to tap the desired Bluetooth headset in the **AVAILABLE DEVICES** list.

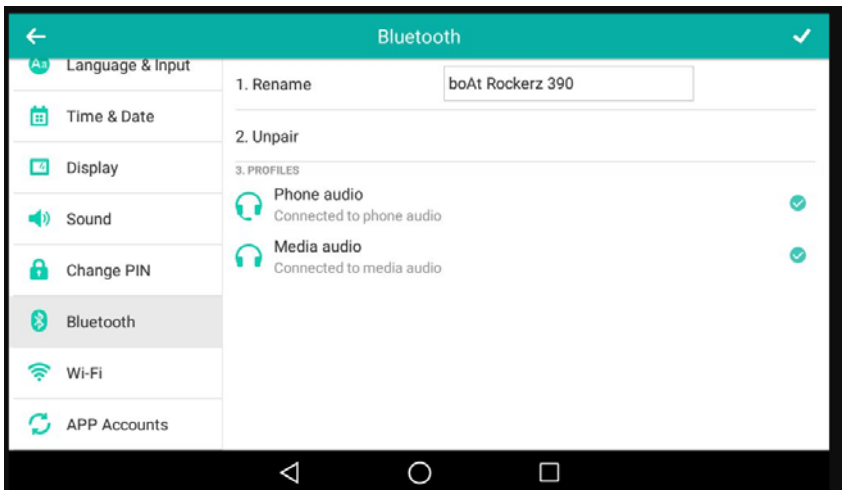
To unpair Bluetooth headset via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**

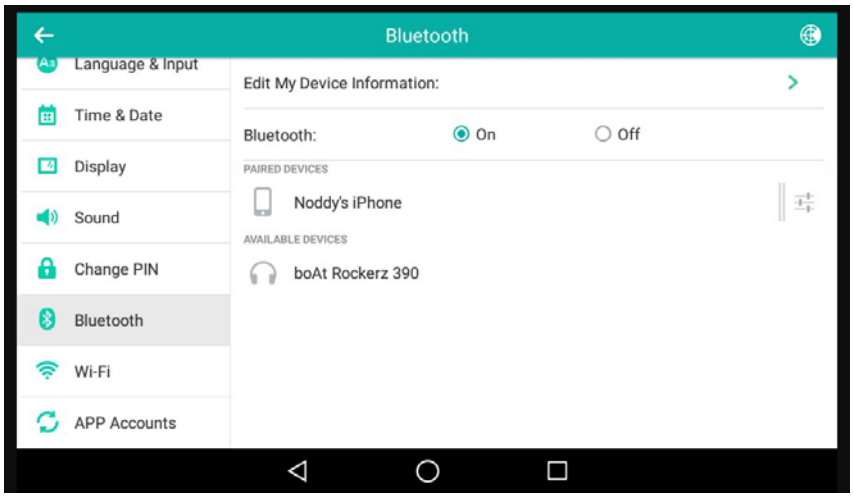
- The touch screen displays the paired and connected Bluetooth headset.



- Tap  after the connected Bluetooth headset name.



- Tap **Unpair**.



- The Bluetooth headset disappears from the **PAIRED DEVICES** list

Bluetooth-Enabled Mobile Phone

The IP phone supports the use of your IP phone in conjunction with Bluetooth-enabled mobile phone. You can do the following:

- Pair and connect your IP phone with a Bluetooth-enabled mobile phone to make and receive mobile calls. Your mobile phone is automatically assigned a line key on your IP phone.
- Import your mobile contacts into the IP phone.
- Use your IP phone as a hands free device for your mobile phone.

For operating instructions on activating/deactivating the Bluetooth mode, making the phone discoverable, editing device information, renaming the paired Bluetooth device and disconnecting/unpairing the paired Bluetooth-enabled mobile phone, refer to the operating instructions of "Bluetooth Headset".

Pairing and Connecting the Bluetooth-Enabled Mobile Phone

You can only connect one Bluetooth device at a time. After connection, the IP phone will automatically find an available line key and assign the line key for Mobile Account. The Mobile Account key's default label is "My Mobile". If there is no available line key, you may assign it manually.

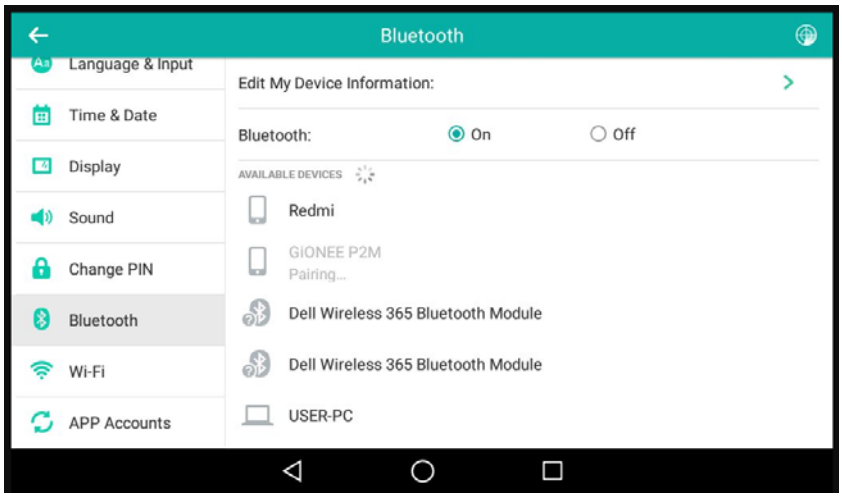
The mobile phone will disconnect from your phone in the following scenarios:

- Deactivate the Bluetooth mode from the IP phone.
- Connect a Bluetooth headset to the IP phone.
- Deactivate the Bluetooth mode from mobile phone.
- The distance between mobile phone and IP phone is out of the range of 10 meters.

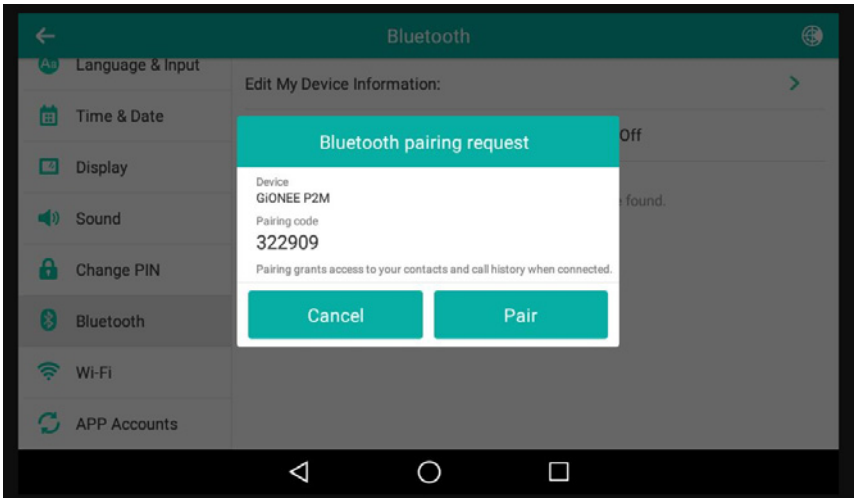
Before you pair a Bluetooth-Enabled mobile phone, check the Bluetooth mode on the Bluetooth-Enabled mobile phone to make sure that the Bluetooth-Enabled mobile phone is discoverable.

To pair and connect the Bluetooth-Enabled Mobile Phone with your phone:

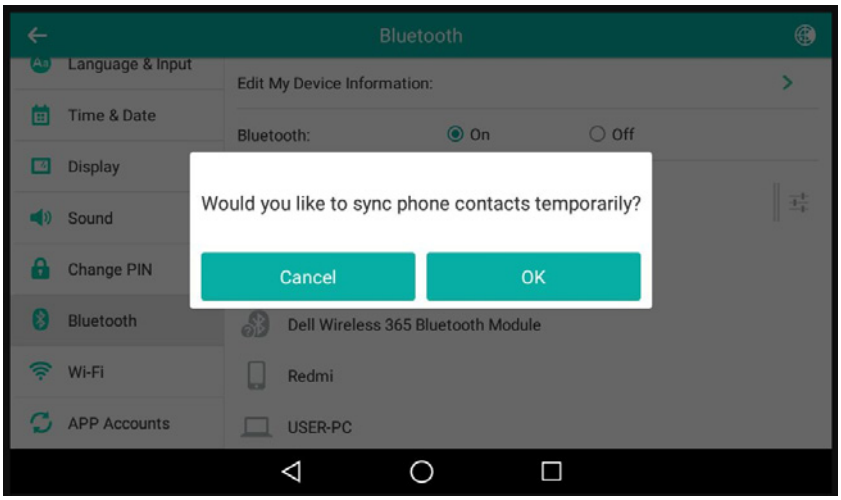
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap the **On** radio box in the **Bluetooth** field.
- If there is no Bluetooth-Enabled mobile phone found on the touch screen, tap the **Scan** soft key to search for the Bluetooth-Enabled mobile phone.
- Tap the desired Bluetooth-Enabled mobile phone in the Bluetooth devices list to connect the mobile phone to your phone.



- Tap **Pair** on both mobile phone and IP phone.

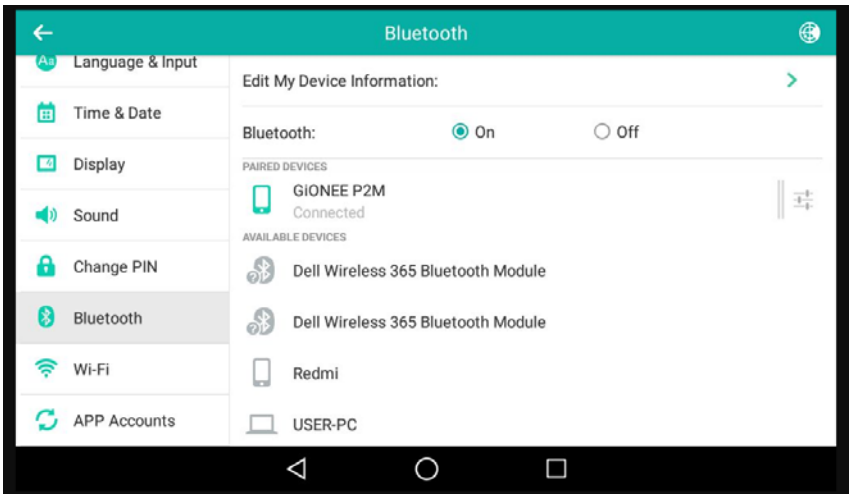


- Then the IP phone will prompt “Would you like to sync phone contacts temporarily?”

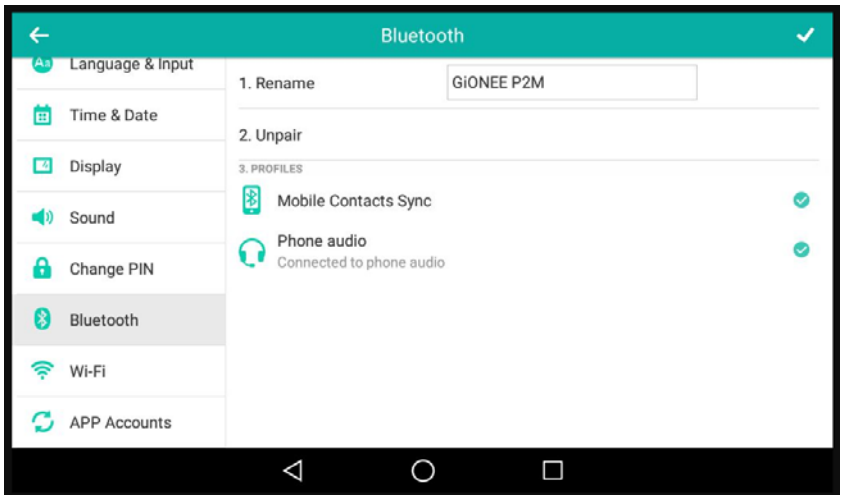


- Tap **OK** to enable the mobile contacts sync feature and the IP phone will sync the mobile contacts temporarily, or tap **Cancel** to disable mobile contacts sync feature and the IP phone will not sync the mobile contacts. If you tap **Cancel**, you can also manually enable the mobile contacts sync feature on the IP phone. For more information, refer to "Syncing the Mobile Contacts to the IP Phone"

- If you enable mobile contacts sync feature, you also need to authorize the IP phone to sync the contacts temporarily on the mobile phone. Once the connection has completed successfully, the prompt "Connected" appears under the Bluetooth-Enabled mobile phone name.



- The Bluetooth icon will display on the status bar. And the IP phone automatically assigns a Mobile Account key.



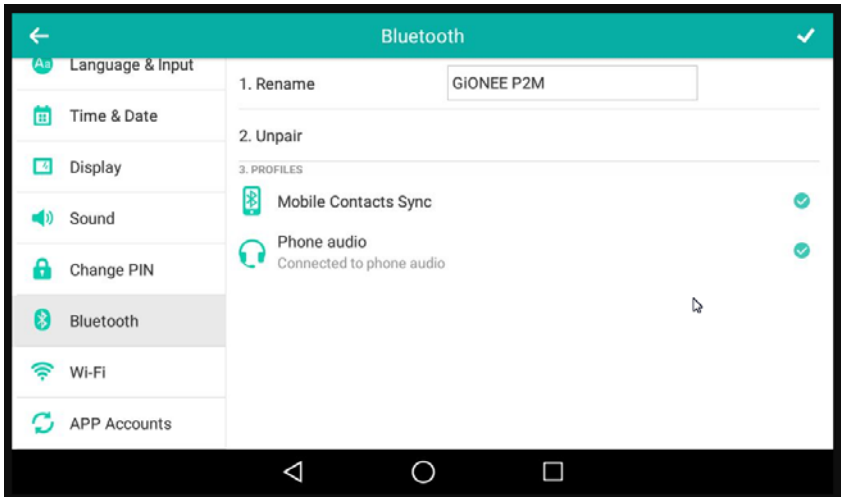
For more information on how to configure the Mobile Account key, refer to "Mobile Account". Before you configure the Mobile Account key, make sure the Bluetooth-Enabled mobile phone is connected. Otherwise the IP phone will prompt "Mobile Account is not connected!". If the distance between mobile phone and IP phone is more than 10 meters or the Bluetooth mode on the mobile phone is deactivated, you can tap the Mobile Account key to reconnect the last paired Bluetooth-Enabled mobile phone.

Syncing the Mobile Contacts to the IP Phone

If you disable the mobile contacts sync feature when you pair and connect the Bluetooth-Enabled mobile phone to the IP phone (refer to "Pairing and Connecting the Bluetooth-Enabled Mobile Phone"), you need to enable the mobile contacts sync feature manually when you want to sync the mobile contacts the IP phone.

To enable the mobile contacts sync feature manually:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Bluetooth.**
- Tap  after the desired Bluetooth-Enabled mobile phone name.
- Tap the **On** radio box in the **Mobile Contacts Sync** field.

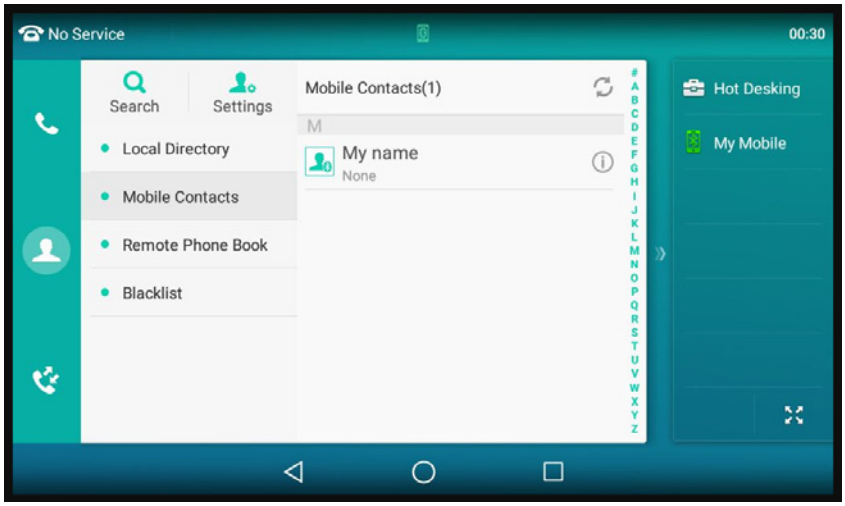


- You need to authorize the IP phone to sync the contacts temporarily on the mobile phone. And then the mobile contacts directory will appear in the IP phone directory list.

To view your mobile contacts on the IP phone:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.

- Tap  **->Mobile Contacts.**



- The contacts in the mobile phone will display in the Mobile Contacts list.
- For operating instructions on adding contact to local/blacklist directory from mobile contacts directory, update the mobile contacts directory or placing calls to and searching for contacts in the mobile contacts directory, refer to the operating instructions of "Mobile Contacts".

Making a Mobile Phone Call by Using the IP Phone

When you make a mobile phone call by using your IP phone, the IP phone acts as a hands free device for your mobile phone. The call is made through your mobile phone, but the audio is present on and the call control is done by using the IP phone. You can have both mobile phone call and IP phone call on your IP phone, and two types of calls will not affect each other.

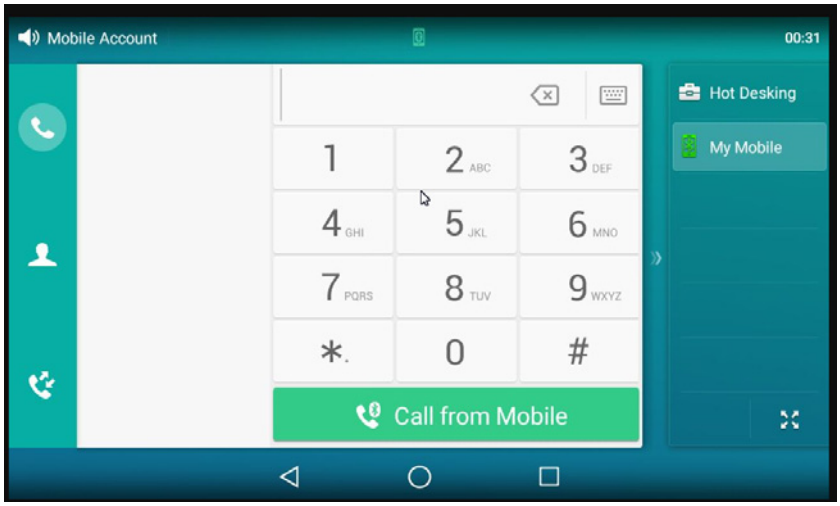
To make a call through a mobile phone:

- Tap the **My Mobile** line key.



- Enter the desired number.

- Tap **Call from Mobile.**



The outgoing call information will appear on both your IP phone and mobile phone screen. After the call is picked up, the audio goes through the IP phone, although the mobile phone screen also shows a connected call. You can hold, mute or end the call on the IP phone, and after you hold the call you can resume the call or place a new call.

You can control the call's audio to go through the mobile phone or IP phone on your mobile phone. For example, if you choose Bluetooth, the audio will go through the IP phone; if you choose Handset earpiece or Speaker, the audio will go through the mobile phone. The phone will first match the contacts in the mobile contacts directory to present the caller/callee identity when receiving/placing a mobile phone call.

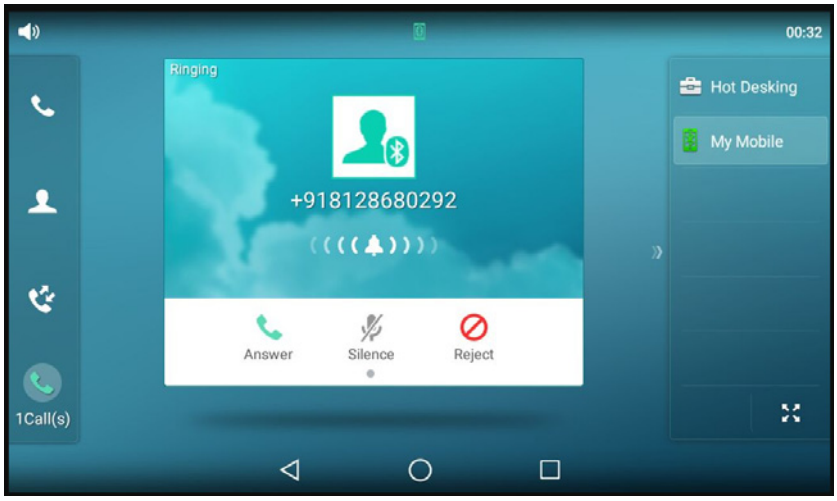
Receiving Mobile Phone Calls by Using the IP Phone

If your mobile phone is paired and connected with the IP phone, an incoming call to your mobile phone is also shown on the IP phone.

To handle a mobile phone call by using the IP phone, do one of the following:

- Press , tap the **Answer** soft key or pick up the handset to connect to the call.

- Tap the **Reject** soft key to reject the call.



Sending Files via Bluetooth

The **Gallery** and **FileManager** applications support sending files via Bluetooth. You can send pictures, videos or audios to a Bluetooth-enabled mobile phone or other Bluetooth devices displayed in the **AVAILABLE DEVICES** list. For more information, refer to “” and “Sharing Pictures/Videos”.

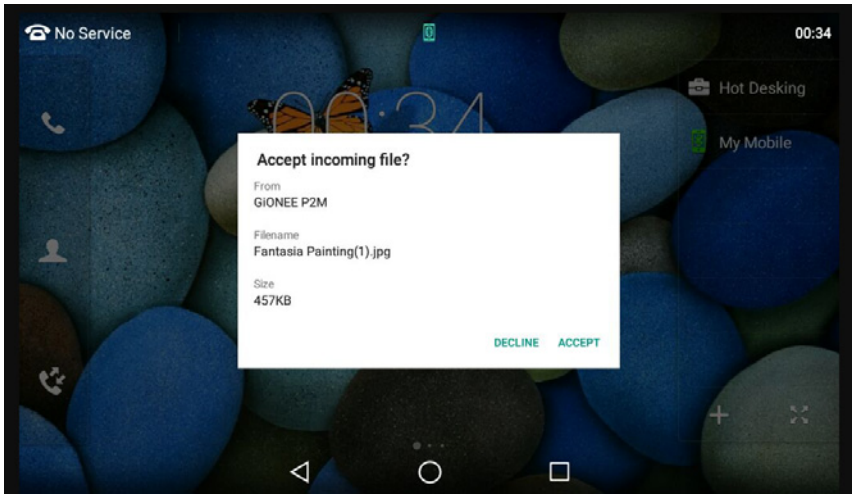
Receiving Files via Bluetooth

The Bluetooth-enabled mobile phone or other Bluetooth devices can share files with your IP phone. The file type is not restricted, you can receive pictures, audios, videos, documents, and so on. You can use the corresponding system applications to view the files. You can only receive files from the Bluetooth devices displayed in the in the **AVAILABLE DEVICES** list.

To receive file via Bluetooth:

- Share a file with the IP phone on your Bluetooth device.

  For more information, refer to the documentation from the Bluetooth device manufacturer The IP phone will play a notification sound, and the touch screen prompts “Accept incoming file?”.

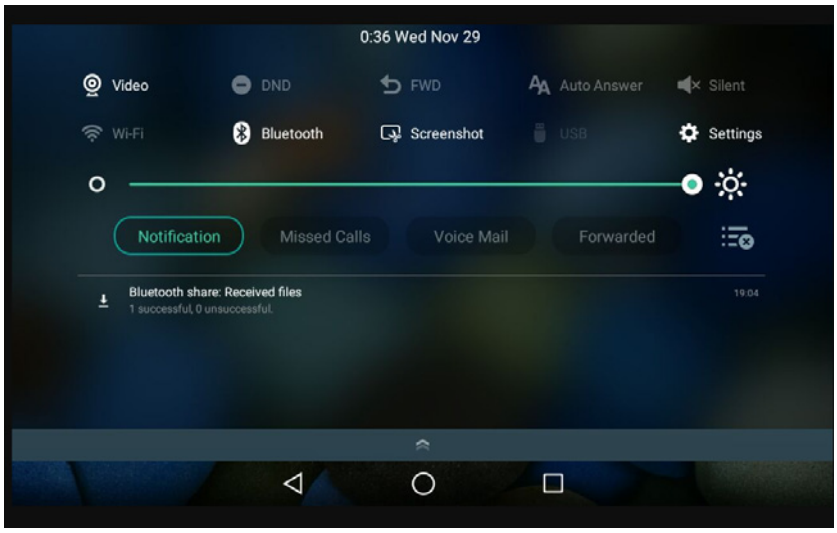


- Tap **Accept** to start receiving the file.

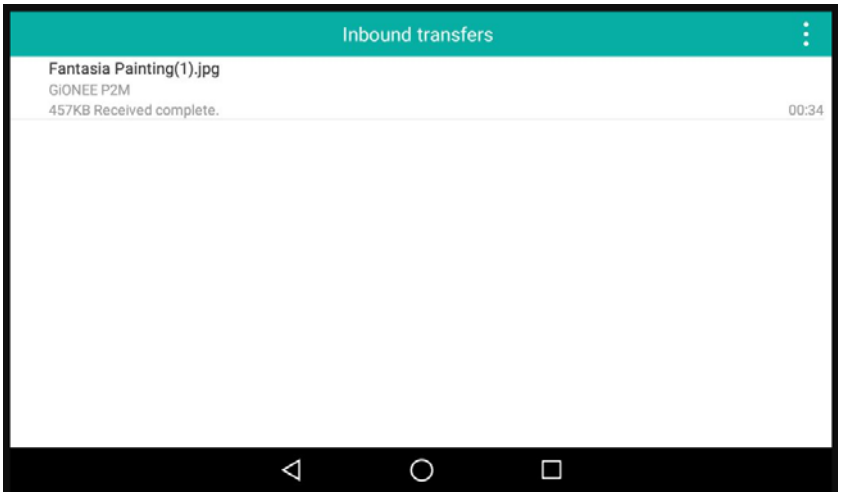
  The touch screen prompts “The file will be received. Check progress in the Notification panel.”. The icon will display on the status bar.



- Swipe down from the top of the screen to enter the notification center.



- Tap the notification “Bluetooth share: Received file”.



- You can use the corresponding system applications to view the file.

Language

The default language of the phone user interface is English. If the language of your web browser is not supported by the phone, the web user interface will use English by default. You can change the language for the phone user interface and the web user interface respectively.

The following table lists languages supported by the phone user interface and the web user interface respectively.

Phone /Web User Interface
English
Chinese Simplified
Chinese Traditional
French
German
Polish
Italian
Portuguese
Spanish
Turkish
Russian

Specifying the Language to Use

The default language used on the phone user interface is English. The default language used on the web user interface depends on the language preferences in the browser (if the language is not supported by the IP phone, the web user interface uses English). You can specify the languages for the phone user interface and the web user interface respectively.

Procedure

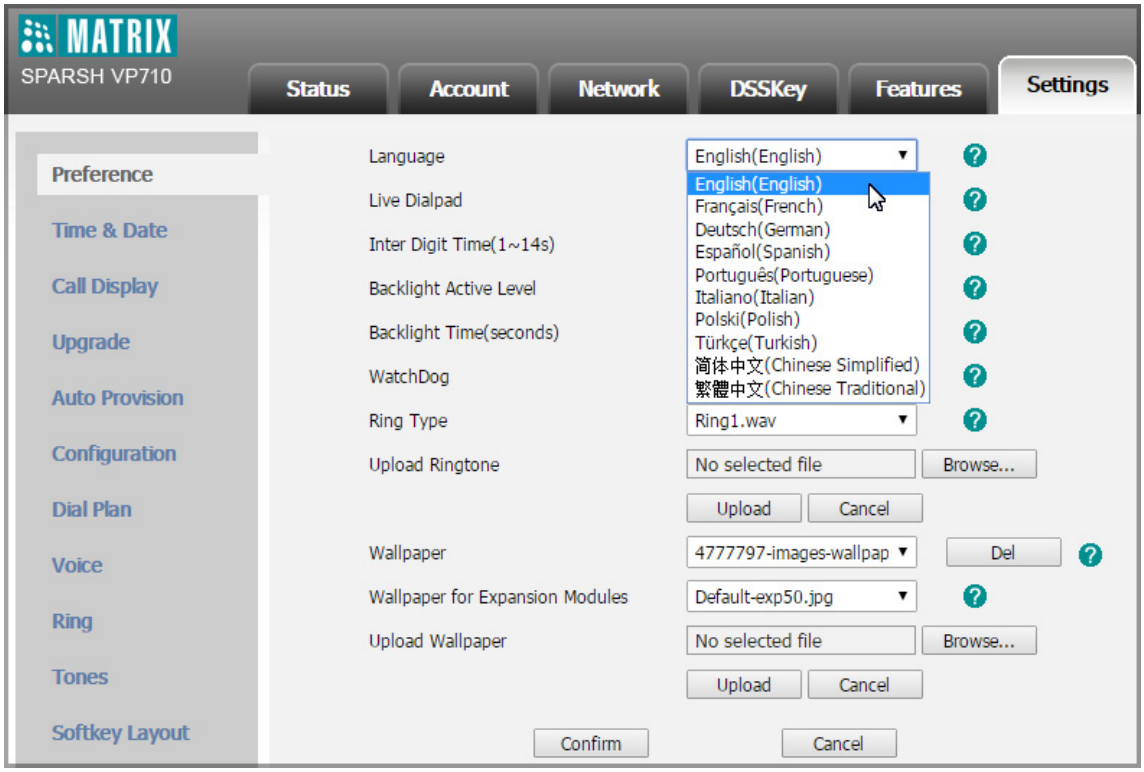
Specify the language for the phone user interface or the web user interface using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify the languages for the phone user interface and the web user interface. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify the language for the web user interface. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-preference&q=load
	Phone User Interface	Specify the language for the phone user interface.

To specify the language for the web user interface via web user interface:

- Click on **Settings->Preference**.

- Select the desired language from the **Language** list.



- Click **Confirm** to save the change.

  Text displayed on the web user interface will change to the selected language.

To change the language for the phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Language & Input->Language.**
- Drag up and down to scroll through the list of available languages.
- Tap the desired language.



- Tap ✓ to accept the change or ◁ to cancel.

  Text displayed on the phone user interface will change to the selected language.

Time & Date

The IP phones maintain a local clock. By default, a digit clock widget will be displayed on the home screen when the IP phone starts up. You can check the current time and date on the home screen. You can also check the current time and date on the control center. In addition, phone's time will be also displayed on the right of the status bar. You can check the current time on the three idle screens.

Time and date are synced automatically from the NTP server by default. The NTP server can be obtained by DHCP or configured manually. If IP phones cannot obtain the time and date from the NTP server, you need to manually configure them.



*In this case, if the **Manual Time** option is also disabled (see the web interface) and the IP phone receives date and time in 200 OK SIP response from the SIP server, then the IP phone will automatically update the time as obtained from the SIP server. To update the time from the NTP server next time (provided the NTP server is functional and the IP phone can query for the updated time from the NTP server), you must restart the IP phone.*

The time and date display can use one of several different formats.

Time Zone

A time zone is a region on Earth that has a uniform standard time. It is convenient for areas in close commercial or other communication to keep the same time. When configuring the IP phone to obtain the time and date from the NTP server, you must set the time zone.

Daylight Saving Time

Daylight Saving Time (DST) is the practice of temporary advancing clocks during the summertime so that evenings have more daylight and mornings have less. Typically, clocks are adjusted forward one hour at the start of spring and backward in autumn. Many countries have used the DST at various times, details vary by location. The DST can be adjusted automatically from the time zone configuration. Typically, there is no need to change this setting.

The following table lists available configuration methods for time and date.

Option	Configuration Methods
NTP time server	Configuration Files Web User Interface Phone User Interface
Time Zone	Configuration Files Web User Interface Phone User Interface

Time	Web User Interface Phone User Interface
Time Format	Configuration Files Web User Interface Phone User Interface
Date	Web User Interface Phone User Interface
Date Format	Configuration Files Web User Interface Phone User Interface
Daylight Saving Time	Configuration Files Web User Interface

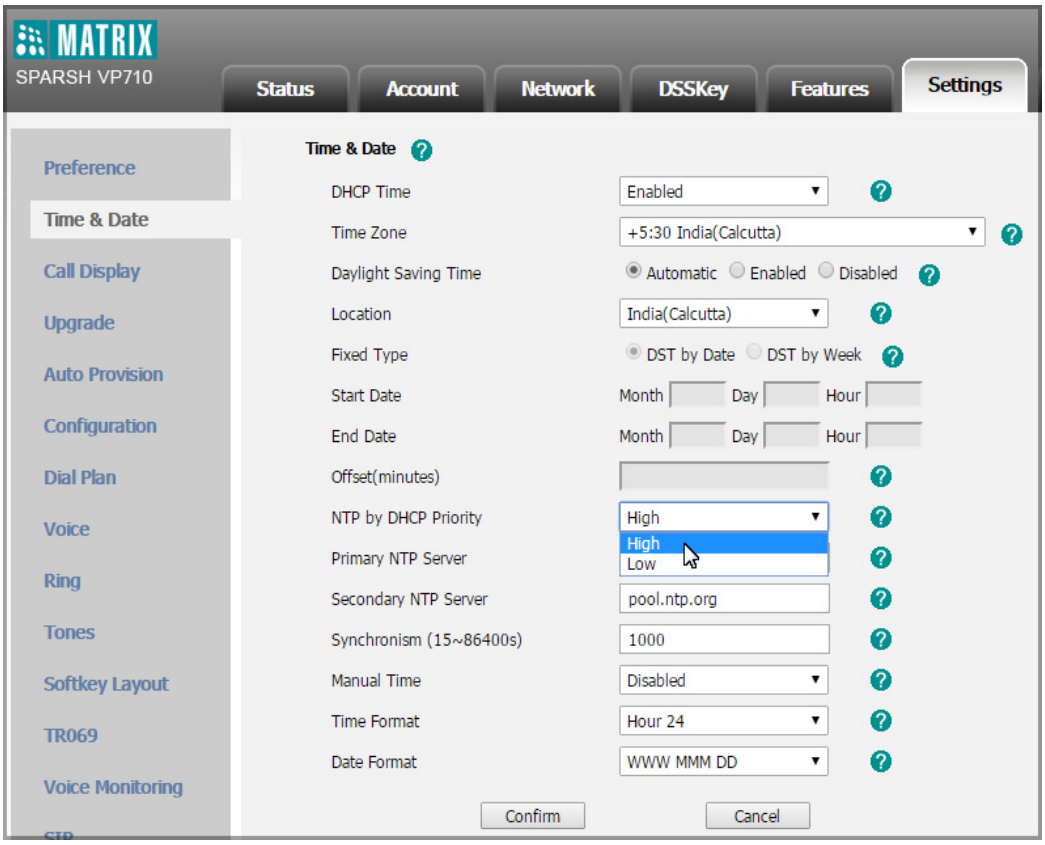
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure NTP by DHCP priority feature and DHCP time feature. Configure the NTP server, time zone and DST. Configure the time and date manually. Configure the time and date formats. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure NTP by DHCP priority feature. Configure the NTP server, time zone and DST. Configure the time and date manually. Configure the time and date formats. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-datetime&q=load
	Phone User Interface	Configure the NTP server and time zone. Configure the time and date manually. Configure the time and date formats.

To configure NTP by DHCP priority feature via web user interface:

- Click on **Settings->Time & Date**.
- Select the desired value from the **NTP By DHCP Priority** list.



- Click **Confirm** to save the change.

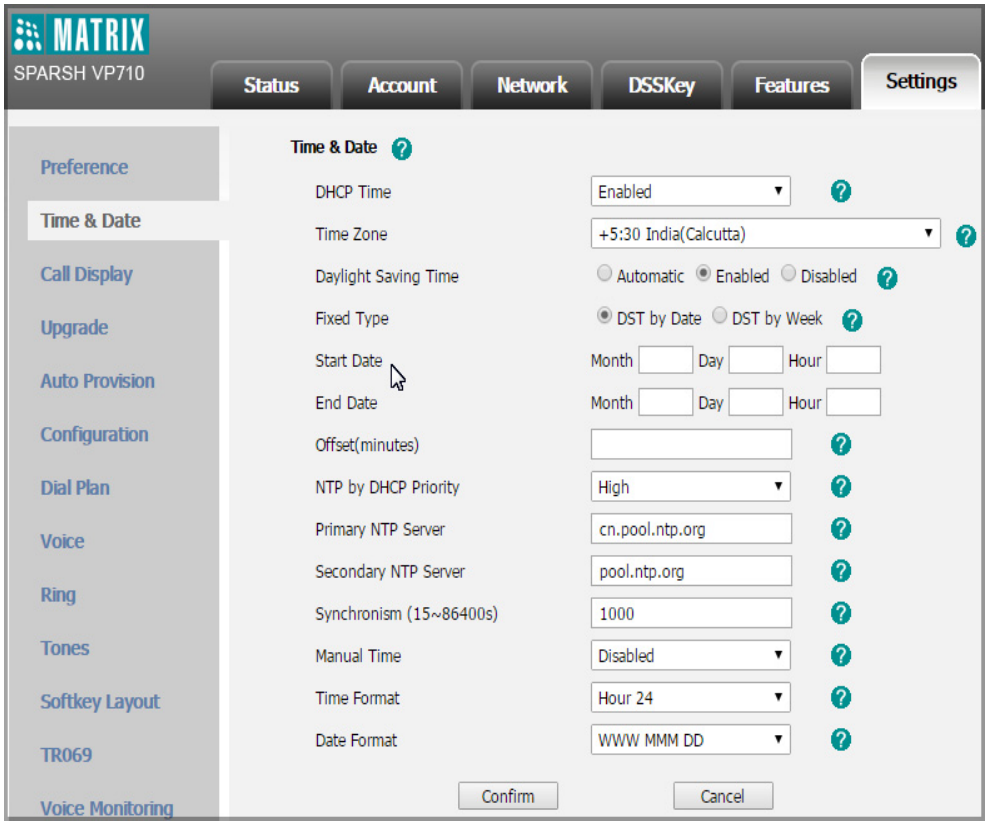
To configure the NTP server, time zone and DST via web user interface:

- Click on **Settings->Time & Date**.
- Select **Disabled** from the **Manual Time** list.
- Select the desired time zone from the **Time Zone** list.
- Enter the domain names or IP addresses in the **Primary NTP Server** and **Secondary NTP Server** fields respectively.
- Enter the desired time interval in the **Synchronism (15~86400s)** field.
- Select the desired value from the **Daylight Saving Time** list.

  If you select **Enabled**, do one of the following:

  Click **DST By Date** in **Fixed Type**.
  Enter the start time in the **Start Date** field.

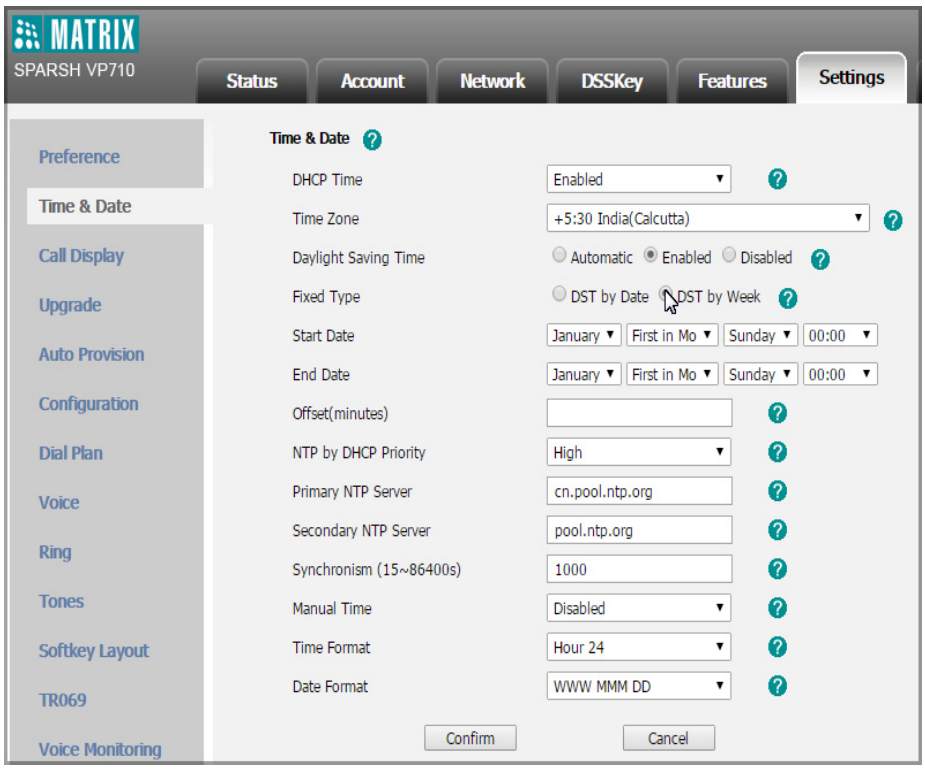
Enter the end time in the **End Date** field.



OR

Click **DST By Week** in **Fixed Type**.

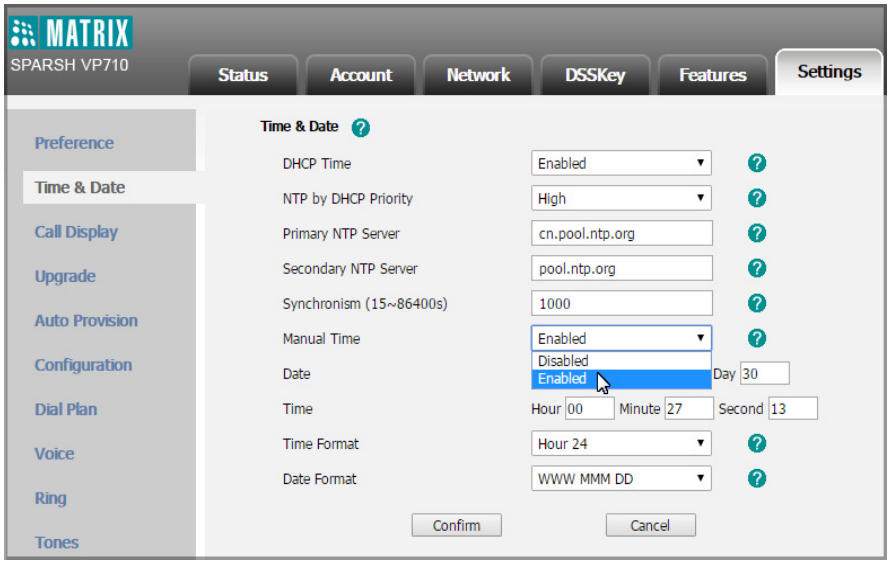
Select the desired values in the following lists: DST Start Month, DST Start Week of Month, DST Start Day of Week, Start Hour of Day; DST Stop Month, DST Stop Week of Month, DST Stop Day of Week and End Hour of Day.



- Enter the desired offset time in the **Offset** (minutes) field.
- Click **Confirm** to save the change.

To configure the time and date manually via web user interface:

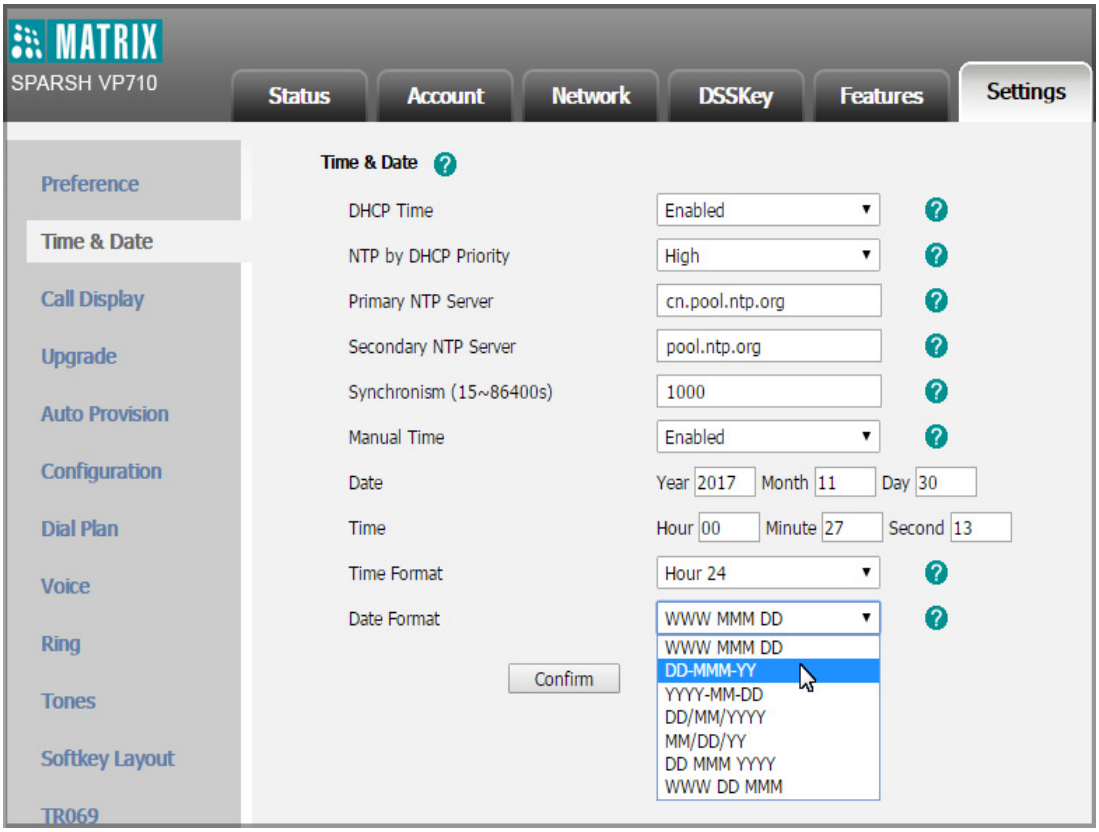
- Click on **Settings->Time & Date**.
- Select **Enabled** from the **Manual Time** list.



- Enter the time and date in the corresponding fields.
- Click **Confirm** to save the change.

To configure the time and date format via web user interface:

- Click on **Settings->Time & Date**.
- Select the desired value from the **Time Format** list.
- Select the desired value from the **Date Format** list.



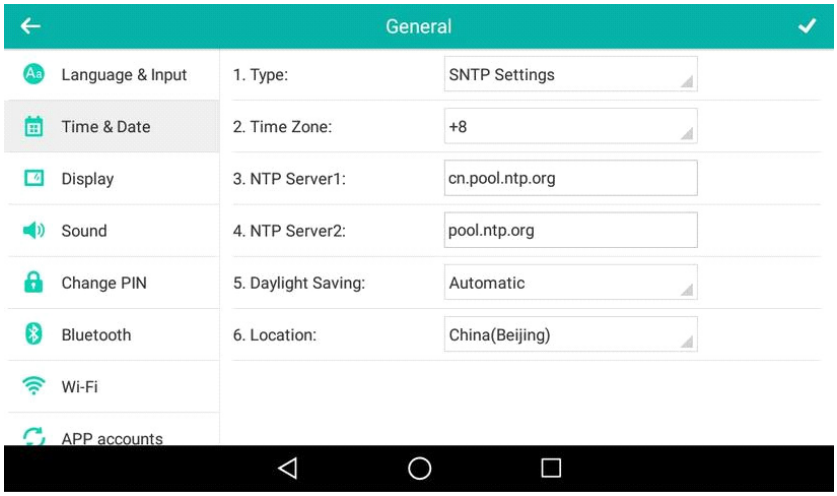
- Click **Confirm** to save the change.

To configure the SNTP setting via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Time & Date->General.**
- Tap the **Type** field.
- Tap **SNTP Settings** in the pop-up dialog box.
- Tap the **Time Zone** field.
- Tap the time zone that applies to your area in the pop-up dialog box. The default time zone is “+8”.
- Enter the domain name or IP address of SNTP server in the **NTP Server1** and **NTP Server2** field respectively.
- Tap the **Daylight Saving** field.
- Tap the desired value in the pop-up dialog box.
- Tap the **Location** field.

  This field appears only if Daylight Saving field is selected to Automatic.

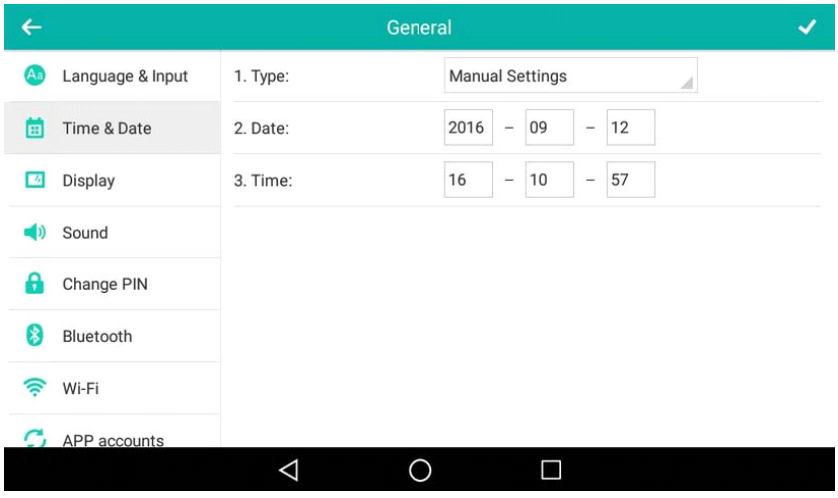
- Tap the desired time zone name in the pop-up dialogbox.



- Tap ✓ to accept the change or ◁ to cancel.

To configure the time and date manually via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Time & Date->General.**
- Tap the **Type** field.
- Tap **Manual Settings** in the pop-up dialog box.
- Enter the specific date and time in the corresponding fields.

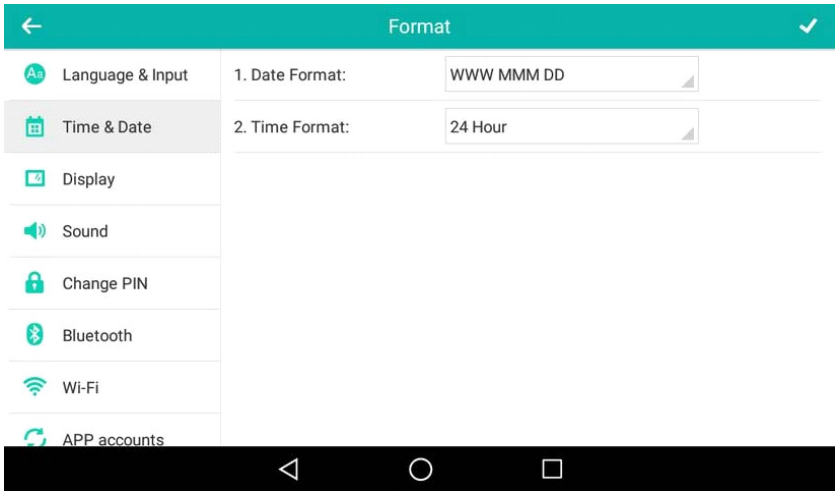


- Tap ✓ to accept the change or ◁ to cancel.

Please refer to "Appendix A - Time Zone" for the list of available time zones on the IP phone.

To configure the date and time format via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Time & Date->Time & Date Format.**
- Tap the **Date Format** field.
- Tap the desired date format in the pop-up dialog box.
- Tap the **Time Format** field.
- Tap the desired time format (12 Hour or 24 Hour) in the pop-up dialog box.



- Tap ✓ to accept the change or ◁ to cancel.

Power Indicator LED

Power indicator LED indicates power status and phone status. There are six configuration options for power indicator LED:

Common Power Light On

Common Power Light On allows the power indicator LED to be turned on.

Ringing Power Light Flash

Ring Power Light Flash allows the power indicator LED to flash when the IP phone receives an incoming call. If this option is disabled, the status of the power indicator LED is determined by the option “Common Power Light On”.

Voice/Text Mail Power Light Flash

Voice/Text Mail Power Light Flash allows the power indicator LED to flash when the IP phone receives a voice mail or a text message. If this option is disabled, the status of the power indicator LED is determined by the option "Common Power Light On".

Mute Power Light Flash

Mute Power Light Flash allows the power indicator LED to flash when a call is mute. If this option is disabled, the status of the power indicator LED is determined by the option "Common Power Light On".

Hold/Held Power Light Flash

Hold/Held Power Light Flash allows the power indicator LED to flash when a call is placed on hold or is held. If this option is disabled, the status of the power indicator LED is determined by the option "Common Power Light On".

Talk/Dial Power Light On

Talk/Dial Power Light On allows the power indicator LED to be turned on when the IP phone is busy. If this option is disabled, the status of the power indicator LED is determined by the option "Common Power Light On".

Procedure

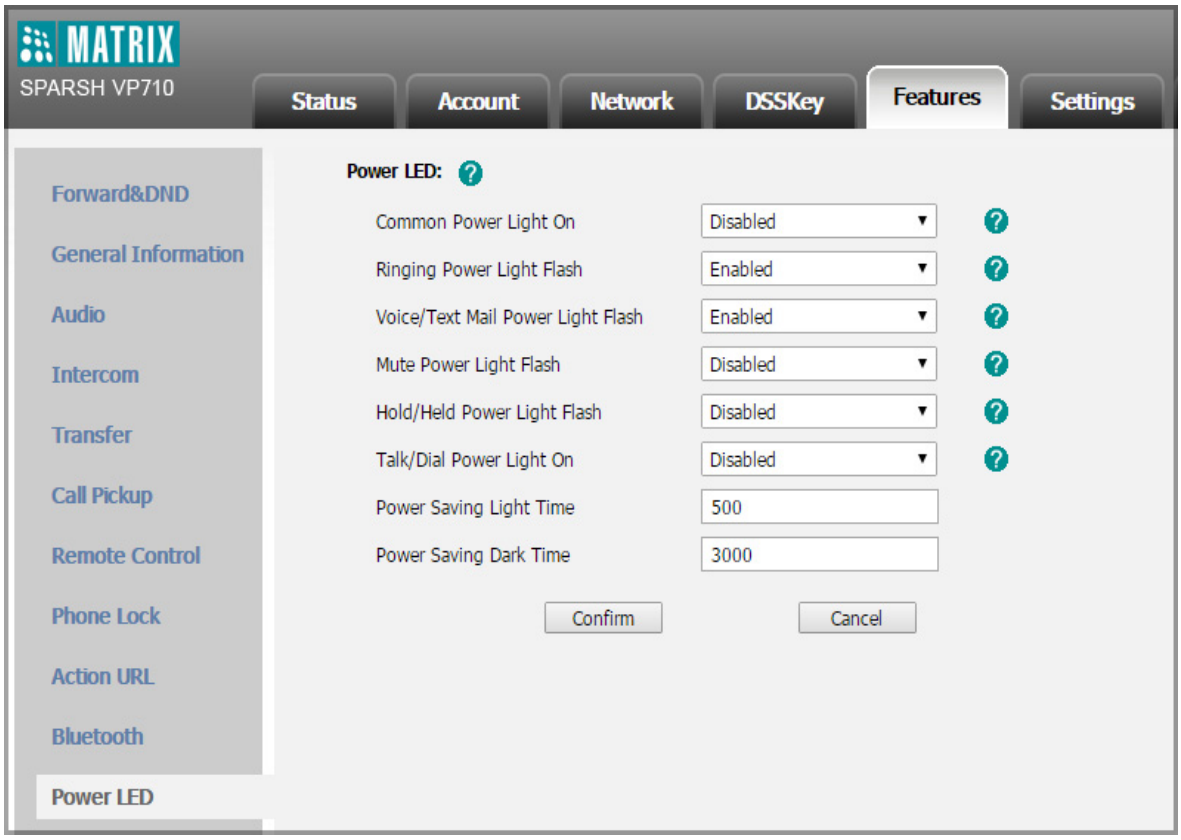
Power indicator LED can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the power indicator LED. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the power indicator LED. **Navigate to**: http://<phoneIPAddress>/servlet?p=features-powerled&q=load

To configure the power Indicator LED via web user interface:

- Click on **Features->Power LED**.
- Select the desired value from the **Common Power Light On** list.
- Select the desired value from the **Ringing Power Light Flash** list.
- Select the desired value from the **Voice/Text Mail Power Light Flash** list.
- Select the desired value from the **Mute Power Light Flash** list.
- Select the desired value from the **Hold/Held Power Light Flash** list.

- Select the desired value from the **Talk/Dial Power Light On** list.



- Click **Confirm** to save the change.

Administrator Password

The Advanced option is only accessible to the administrator. The default administrator password is “admin”. For security reasons, you should change the default administrator password as soon as possible.

Procedure

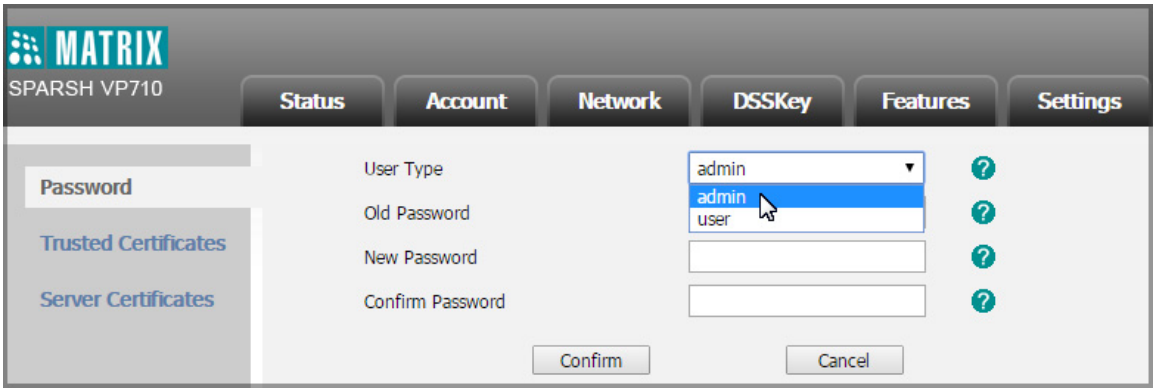
Administrator password can be changed using the configuration files or locally.

Configuration File	<MAC>.cfg	Change the administrator password. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Change the administrator password. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=security&q=load
	Phone User Interface	Change the administrator password.

To change the administrator password via web user interface:

- Click on **Security->Password**.
- Select **admin** from the **User Type** list.
- Enter the current administrator password in the **Old Password** field.
- Enter new password in the **New Password** and **Confirm Password** fields.

  Valid characters are ASCII characters 32-126(0x20-0x7E) except 58(3A).

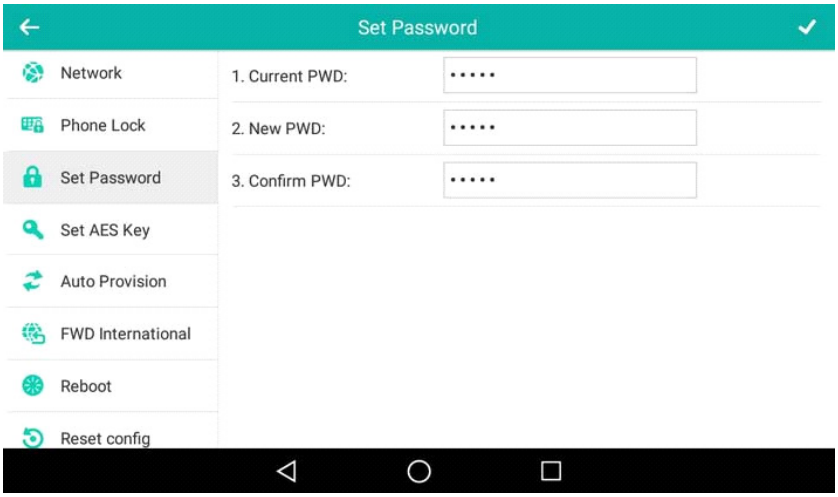


- Click **Confirm** to save the change.

*If logging into the web user interface of the phone with the user credential, you need to enter the old user password in the **Old Password** field.*

To change the administrator password via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced (default password: admin) ->Set Password.**
- Enter the current password in the **Current PWD** field.
- Enter the new password in the **New PWD** field.
- Re-enter the new password in the **Confirm PWD** field.



- Tap ✓ to accept the change or ◁ to cancel.

User Password

Some menu options are protected by two privilege levels, user and administrator, each with its own password. When logging into the web user interface, you need to enter the user name and password to access various menu options.

A user or an administrator can change the user password. The default user password is “user”. For security reasons, the user or administrator should change the default user password as soon as possible.

Procedure

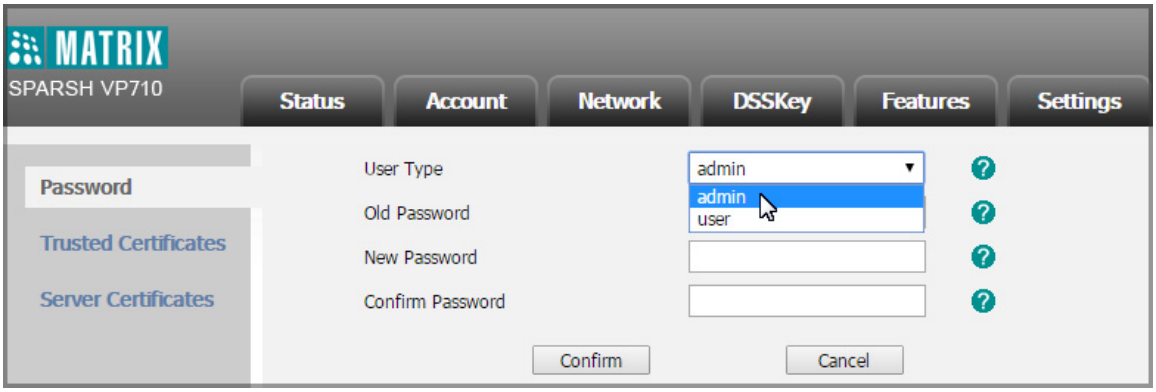
User password can be changed using the configuration files or locally.

Configuration File	<MAC>.cfg	Change the user password of the IP phone. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Change the user password of the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=security&q=load

To change the user password via web user interface:

- Click on **Security->Password**.
- Select **user** from the **User Type** list.
- Enter new password in the **New Password** and **Confirm Password** fields.

  Valid characters are ASCII characters 32-126(0x20-0x7E) except 58(3A).



- Click **Confirm** to save the change.

*If logging into the web user interface of the phone with the user credential, you need to enter the old user password in the **Old Password** field.*

Key as Send

Key as send allows assigning the pound key (“#”) or asterisk key (“*”) as a send key. Send sound allows the IP phone to play a key tone when a user presses the send key. Key tone allows the IP phone to play a key tone when a user presses any key. Send sound works only if Key tone is enabled.

Procedure

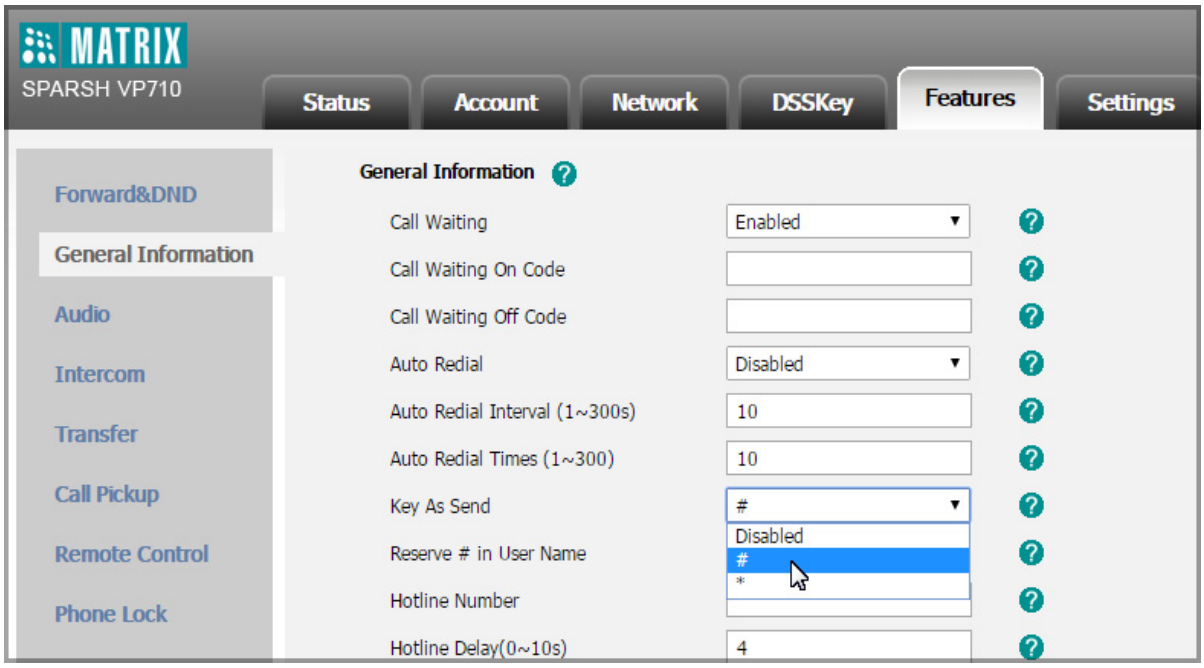
Key as send can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure a send key. Configure a send sound. Configure a key tone. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Configure a send key. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load Configure a send sound and key tone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-audio&q=load
	Phone User Interface	Configure a send key. Configure a key tone.

To configure a send key via web user interface:

- Click on **Features->General Information**.

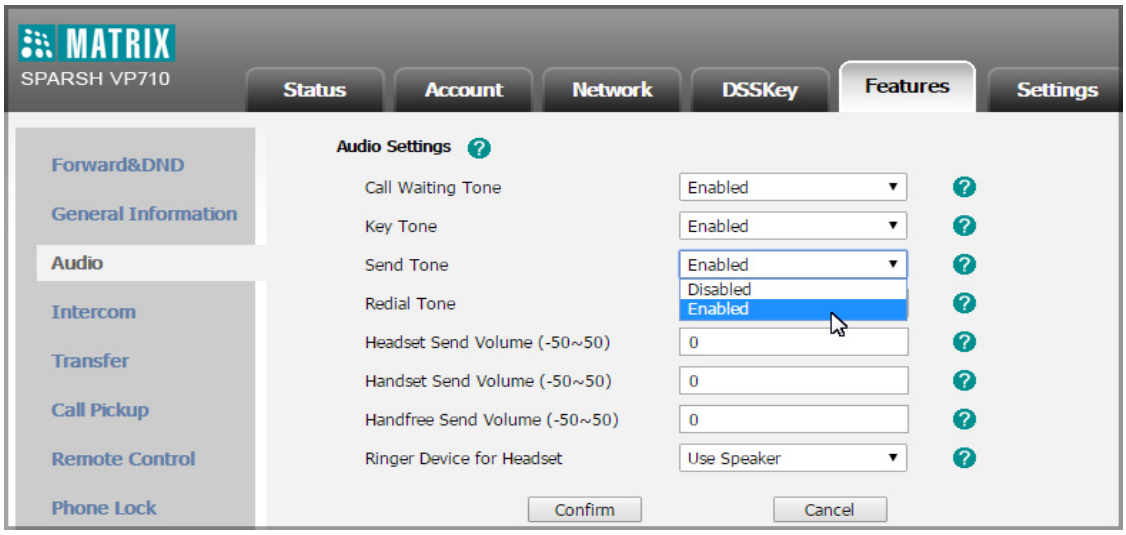
- Select the desired value from the **Key As Send** list.



- Click **Confirm** to save the change.

To configure a send tone and key tone via web user interface:

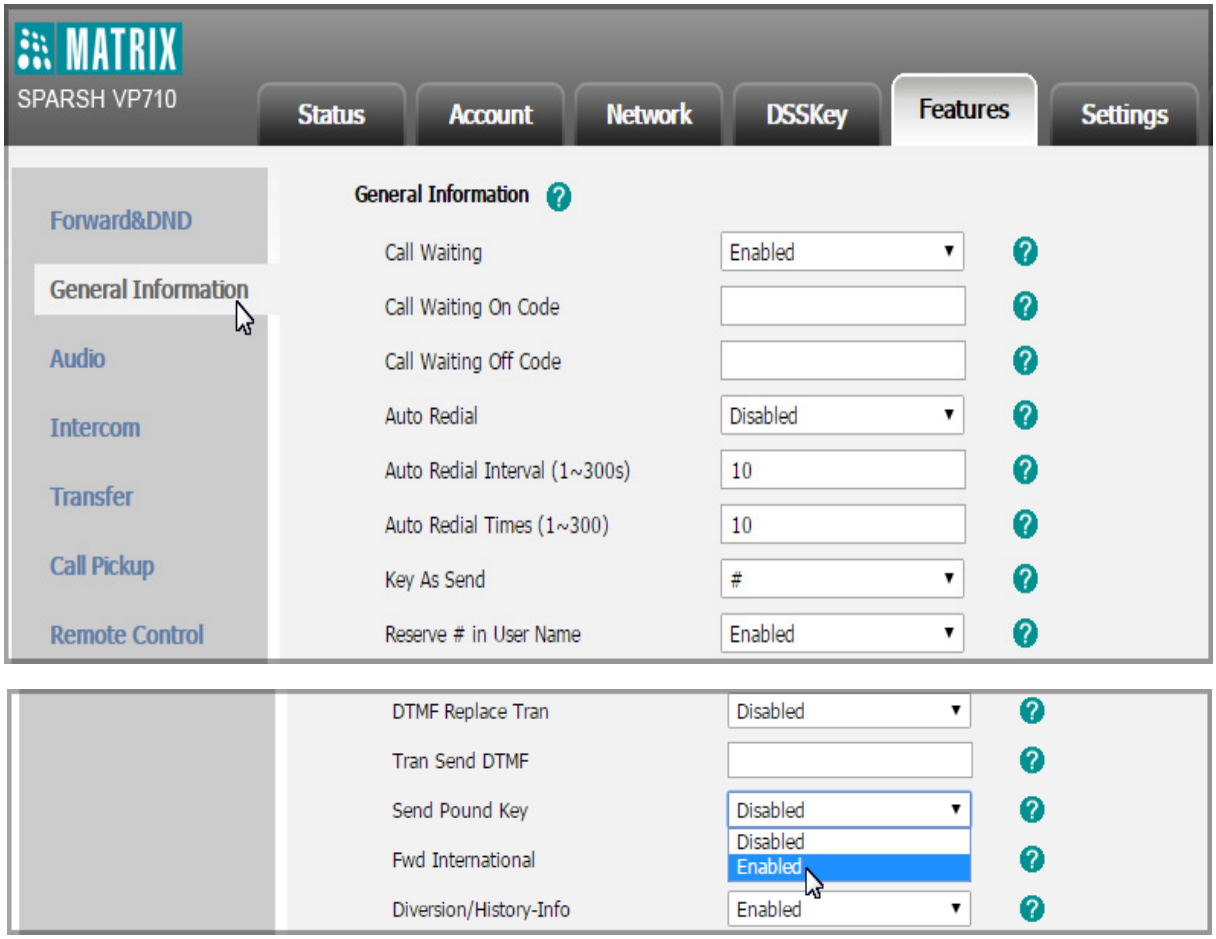
- Click on **Features->Audio**.
- Select the desired value from the **Key Tone** list.
- Select the desired value from the **Send Tone** list.



- Click **Confirm** to save the change.

To configure send pound key via web user interface:

- Click on **Features->General Information**.
- Select the desired value from the **Send Pound Key** list.

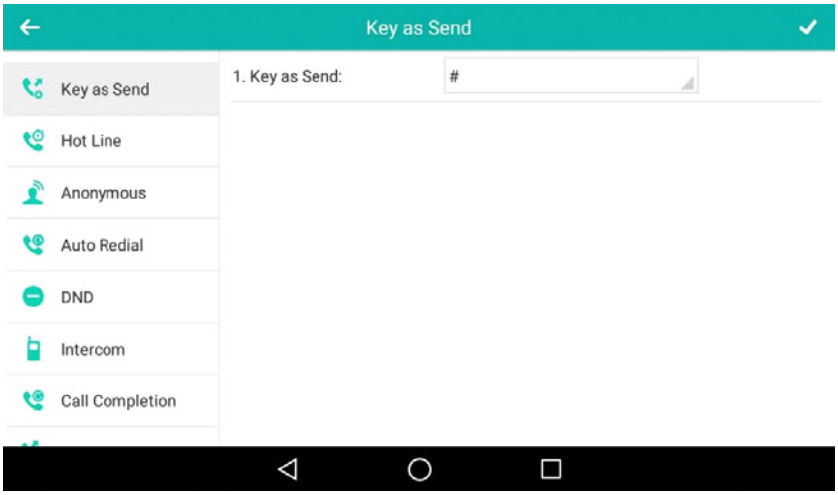


- Click **Confirm** to save the change.

To configure send key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Key as Send.**
- Tap the **Key as Send** field.

- Tap **#** or ***** in the pop-up dialog box, or tap **Disabled** to disable this feature.



- Tap ✓ to accept the change or ◁ to cancel.

Keypad Lock (Phone Lock)

Phone lock is used to lock the IP phone to prevent it from unauthorized use. Once the IP phone is locked, a user must enter the password to unlock it.

All keys and screen are locked except the HEADSET key, Volume key, digit keys, # key, * key and Speakerphone key. You are only allowed to dial emergency numbers, reject incoming calls by tapping the Reject soft key, answer incoming calls by lifting the handset, pressing the Speakerphone key, the HEADSET key or tapping the Answer soft key, and end the call by hanging up the handset, pressing the Speakerphone key or tapping the EndCall soft key.

The emergency number setting, if desired, must be made before lock activation.For more information, refer “”.

The IP phone will not be locked immediately after the phone lock type is configured. One of the following steps is also needed:

- Long press the pound key when the IP phone is idle.
- Press the keypad lock key (if configured) when the IP phone is idle.

In addition to the above steps, you can configure the IP phone to automatically lock the keypad after a period of time.

Procedure

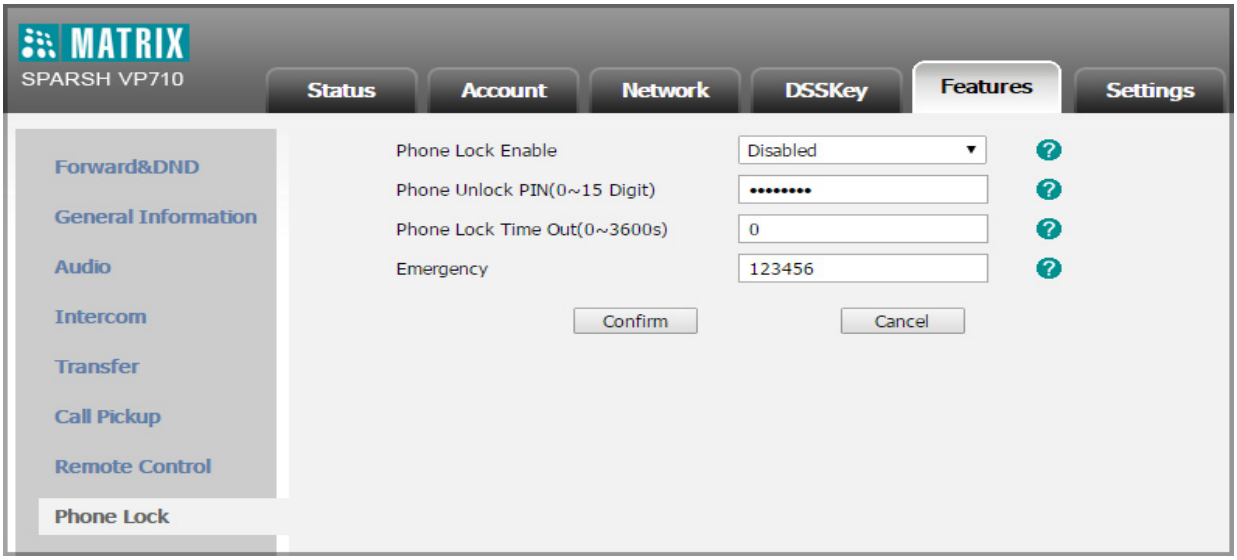
Phone lock can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure phone lock. Change the unlock PIN. Configure the IP phone to automatically lock the keypad after a time interval. Configure emergency numbers. Assign a phone lock key. Assign a keypad lock key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure phone lock. Configure the type of phone lock. Change the unlock PIN. Configure the IP phone to automatically lock the keypad after a time interval. Configure emergency numbers. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data a&p=features-phonelock&q=load Assign a keypad lock key. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data a&p=dsskey&model=2&q=load
	Phone User Interface	Configure the phone lock feature. Change the unlock PIN. Configure the IP phone to automatically lock the phone after a time interval. Assign a phone lock key.

To configure phone lock via web user interface:

- Click on **Features->Phone Lock**.
- Select **Enabled** from the **Phone Lock Enable** list.
- Enter the unlock PIN in the **Phone Unlock PIN** (0~15 Digit) field.

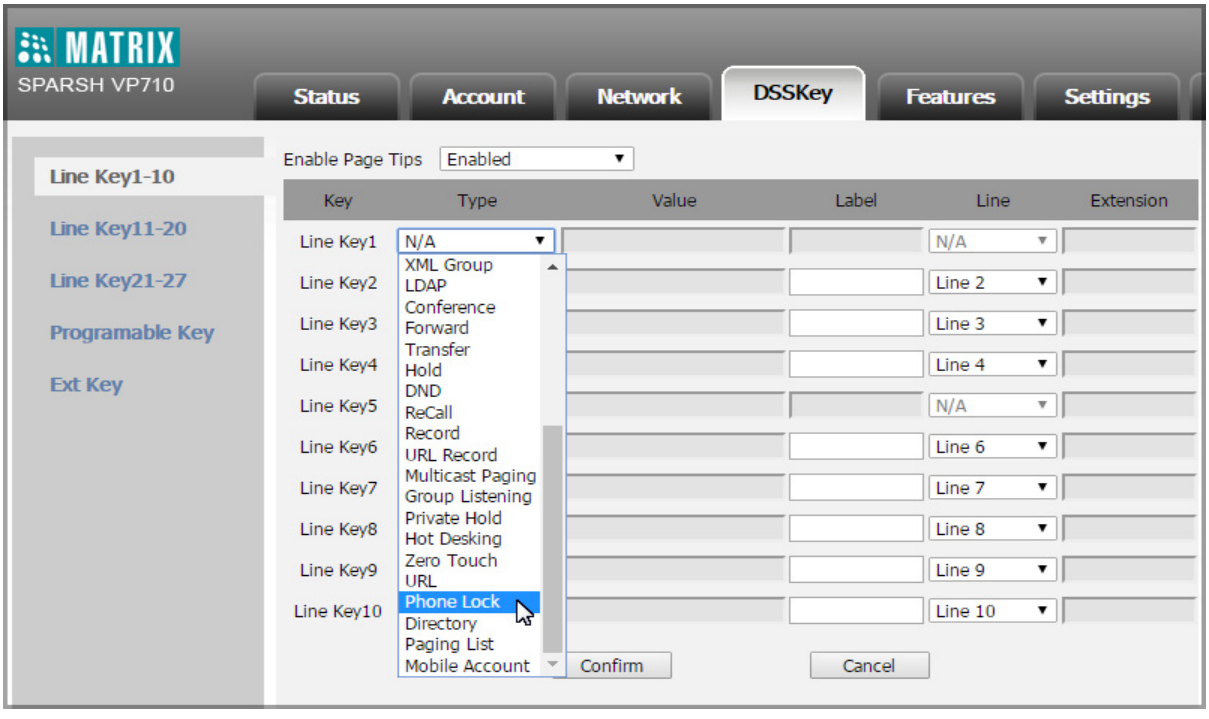
- Enter the desired time in the **Phone Lock Time Out** (0~3600s) field.



- Click **Confirm** to save the change.

To configure a keypad lock key via web user interface:

- Click on **DSS Key-> Line Key**.
- In the desired DSS key field, select **Phone Lock** from the **Type** list.



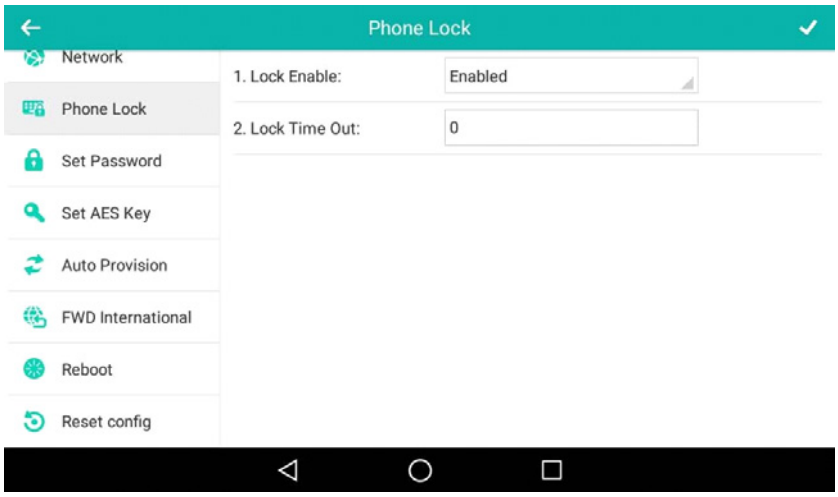
- Click **Confirm** to save the change.

To activate the phone lock via phone user interface:

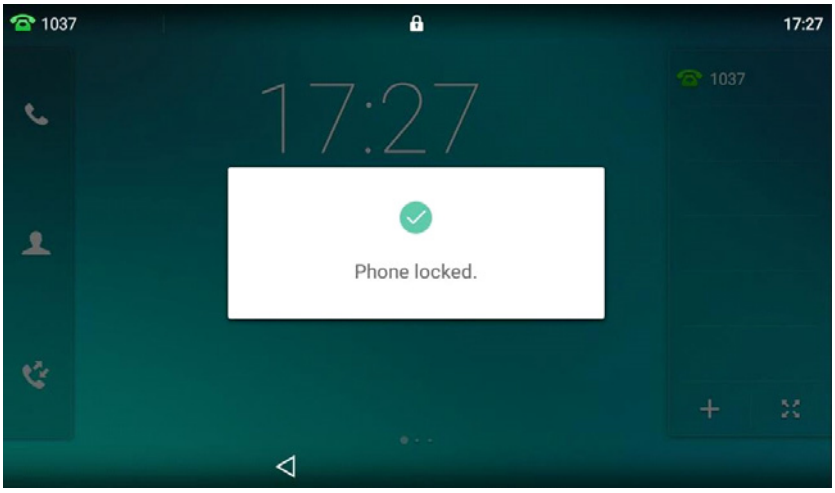
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced (default password: admin) ->Phone Lock.**
- Tap the **Lock Enable** field.
- Tap **Enabled** in the pop-up dialog box to enable this feature.
- Enter the desired interval of automatic phone lock in the **Lock Time Out** field.

  The default timeout is 0. It means the phone will not be automatically locked. You need to long press # to lock it immediately when the phone is idle.

  If set to other values except 0 (e.g., 5), the phone will be locked when the phone is inactive in idle screen for the designated time (in seconds.



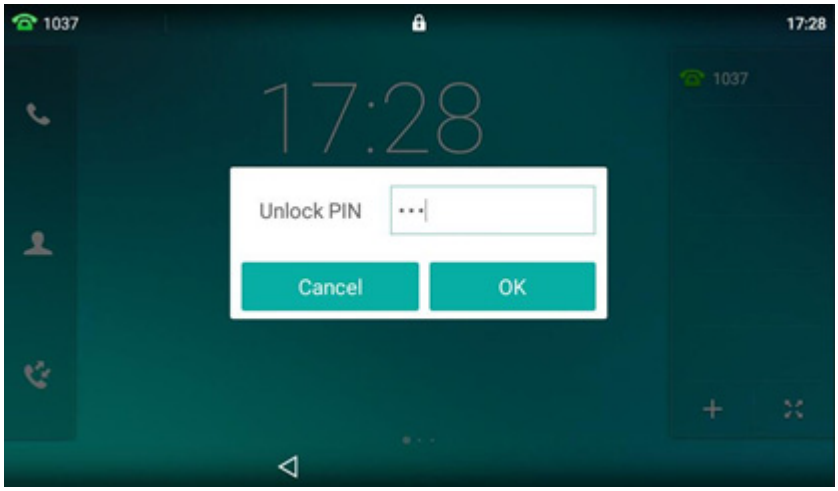
- Tap ✓ to save the change or ◁ to cancel.
- When the phone is locked, the touch screen prompts “Phone locked.” and displays the icon on the status bar.



- To unlock the phone, you must know the phone unlock PIN. The default phone unlock PIN is “123”.

To unlock the phone via phone user interface:

- Tap the screen or tap the locked key, the touch screen prompts “Unlock PIN”.
- Enter the PIN in the **Unlock PIN** field.



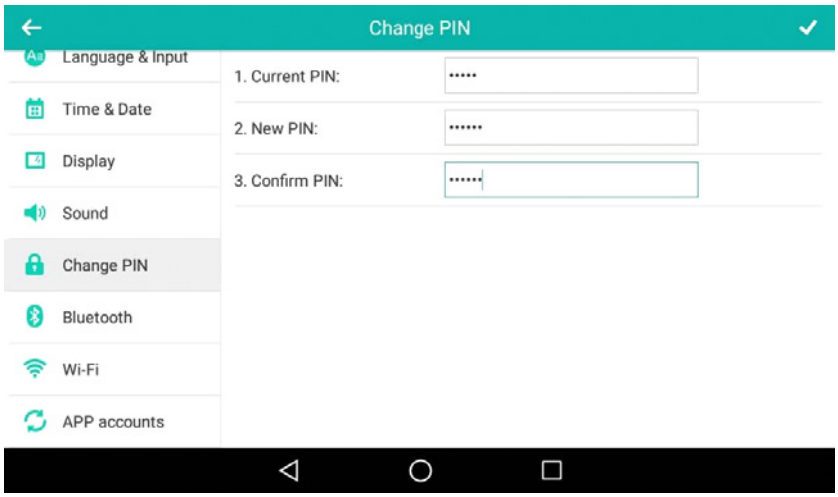
- Press **OK** to unlock the phone.
- The icon disappears from the status bar.
- You can long press # SEND or wait for a period of time (if configured) to lock the phone again.

You can also unlock the phone by administrator password. When you enter the administrator password to unlock the phone, the phone will turn to the Change PIN screen.

To change the phone unlock PIN via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Change PIN.**

- Enter the desired value in the **Current PIN**, **New PIN** and **Confirm PIN** field respectively.

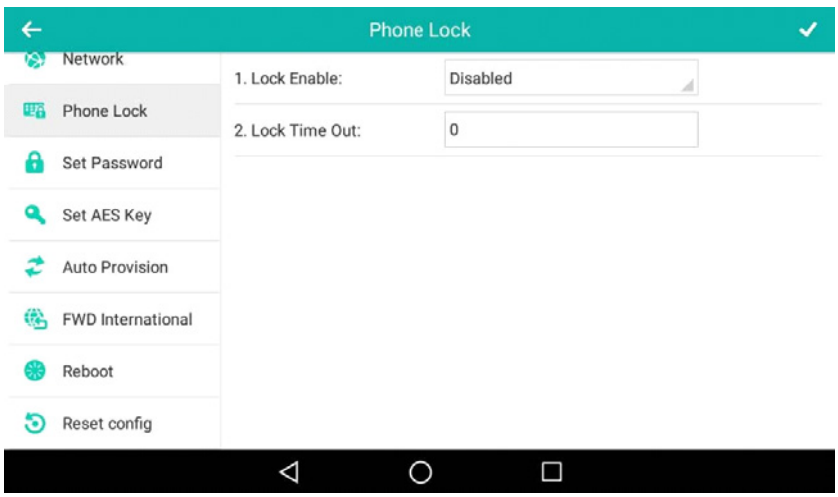


- Tap ✓ to accept the change or ◁ to cancel.

The unlock PIN length must be within 15 digits.

To deactivate the phone lock via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced (default password: admin) ->Phone Lock.**
- Tap the **Lock Enable** field.
- Tap **Disabled** in the pop-up dialog box to disable this feature.



- Tap ✓ to accept the change or ◁ to cancel.

Auto-Logout Time

Auto-logout time defines a specific period of time during which the IP phones will automatically log out if you have not performed any actions via web user interface. Once logging out, you must re-enter username and password for web access authentication.

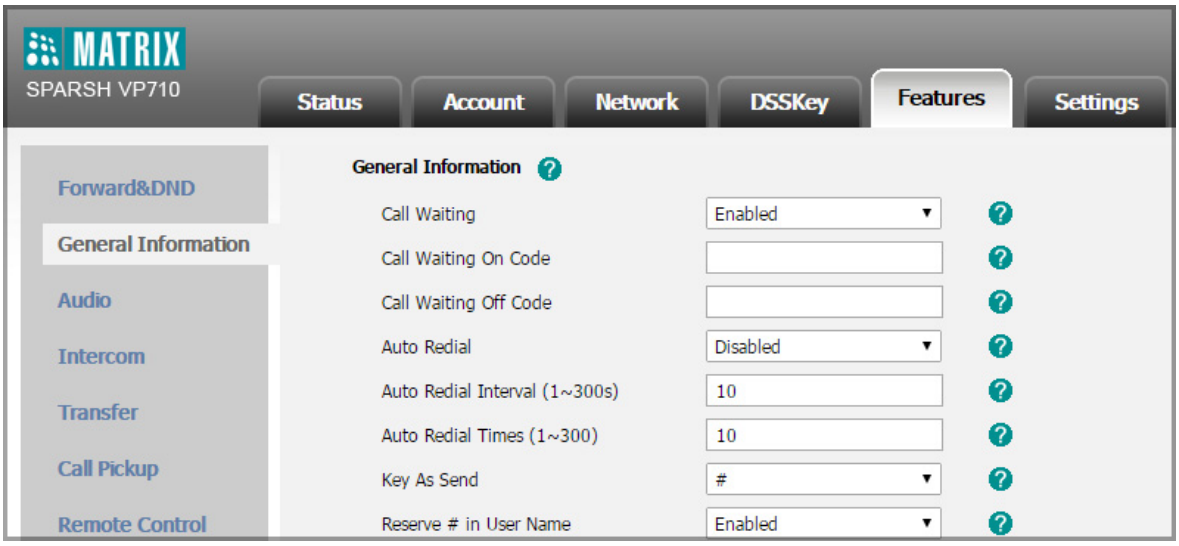
Procedure

Auto-Lougout Time can be configured using the configuration files or locally.

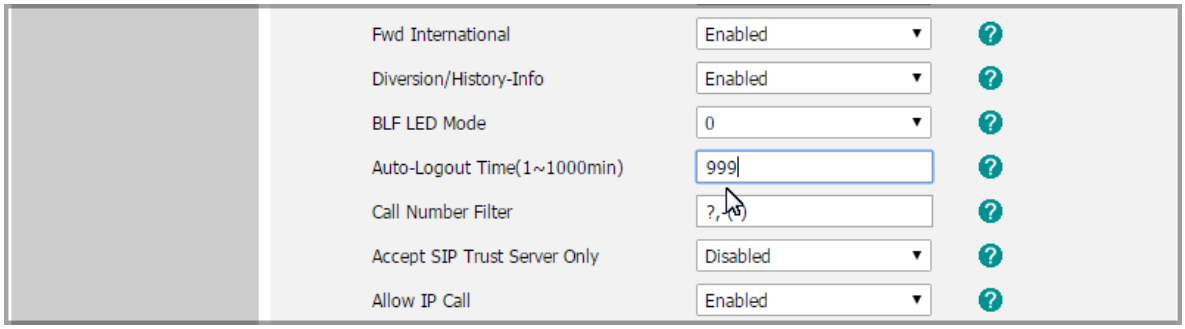
Configuration File	<MAC>.cfg	Configure auto-logout time. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure auto-logout time. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure the auto-logout time via web user interface:

- Click on **Features->General Information.**



- Scroll to **Auto-Logout Time(1~1000min)** and enter the desired auto-logout time.



- Click **Confirm** to save the change.

Call Display

Display called party information allows the IP phone to present the callee identity in addition to the presentation of caller identity when it receives an incoming call.

The following figure shows an example of screen display when Display Called Party Information feature is enabled on the phone (a call from Marry (phone number: 1008) to Tom.

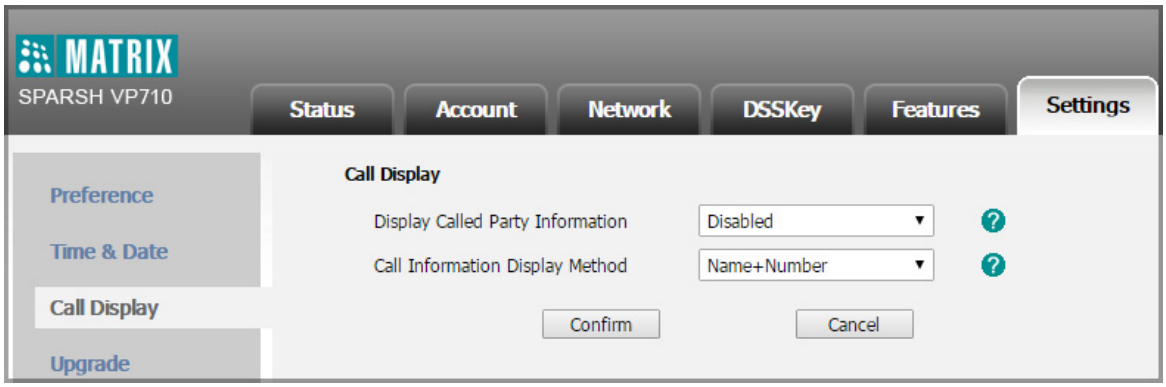


You can customize the call information to be displayed on the IP phone as required. IP phones support five call information display methods: Number+Name, Name, Name+Number, Number or Full Contact Info (display name<sip:xxx@domain.com>).

To configure call display features via web user interface:

- Click on **Settings->Call Display**.
- Select the desired value from the **Display Called Party Information** list.

- Select the desired value from the **Call Information Display Method** list.



- Click **Confirm** to save the change.

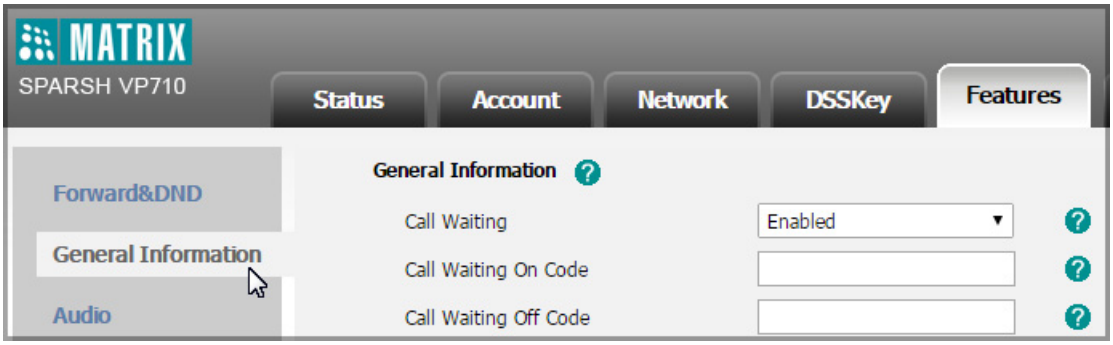
Display Method on Dialing

When the IP phone is on the pre-dialing or dialing screen, the account information will be displayed on the top-left corner of the touch screen.

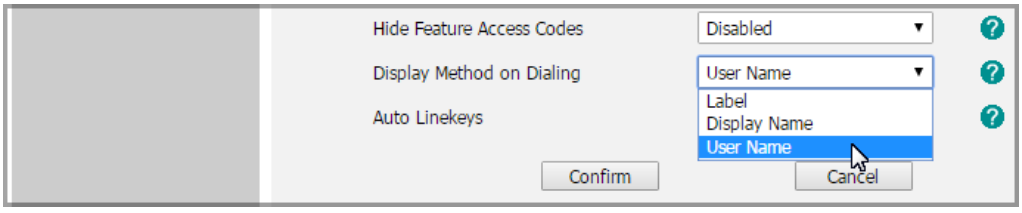
You can customize the account information to be displayed on the IP phone as required. IP phones support three account information display methods: Label, Display Name or User Name.

To configure display method on dialing via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Display Method on Dialing** list.



- Click **Confirm** to save the change.

Audio Settings

Audio Setting allows you to configure the following:

- Volume
- Ring Tones

Volume

You can press the Volume key to adjust the ringer volume when the phone is idle. You can also press the Volume key to adjust the receiver volume of currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.

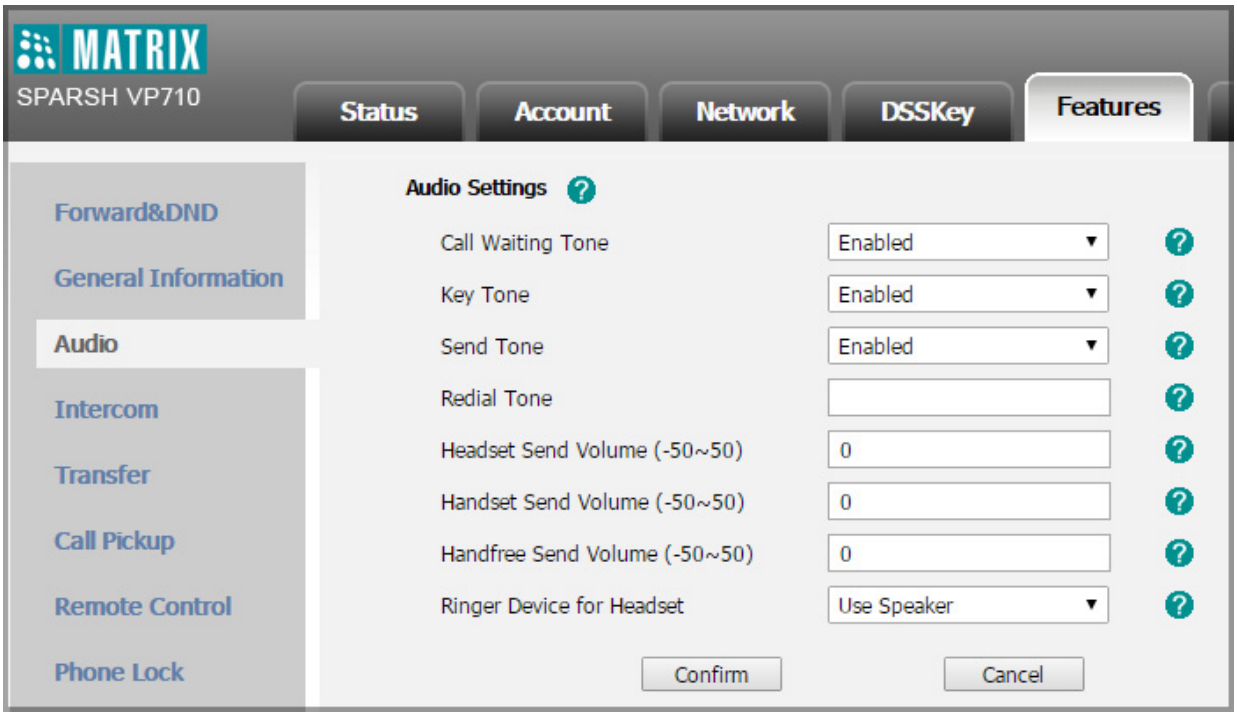
Procedure

Sending volume can be configured using the following methods.

Configuration File	<MAC>.cfg	Configure the sending volume of the speaker. Configure the sending volume of the handset. Configure the sending volume of the headset. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the sending volume of the speaker/handset/headset. http://<phoneIPAddress>/servlet?m=mod_data&p=features-audio&q=load
	Phone User Interface	Configure the Ringer volume, Talking volume, Key tone volume and Media volume.

To configure the sending volume of the speaker/handset/headset.:

- Click on **Features->Audio.**
- Enter the desired value in **Headset Send Volume (-50~50)**.
- Enter the desired value in **Handset Send Volume (-50~50).**
- Enter the desired value in **Handfree Send Volume (-50~50)**



- Click **Confirm** to save the change.
  A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

You can adjust the following volume level using the phone:

- **Ringer volume:** Press the Volume key to adjust the ringer volume when the phone is idle or ringing. When adjusting ringer volume, the touch sound and notification sound volume will change as well.
- **Talking volume:** Press the Volume key to adjust the receiver volume of currently engaged audio devices (handset, speakerphone or headset) when the phone is in use.
- **Key tone volume:** Press the Volume key to adjust the key tone volume when the phone is on the dialing screen.
- **Media volume:** Press the Volume key to adjust the media volume when playing audios/videos.

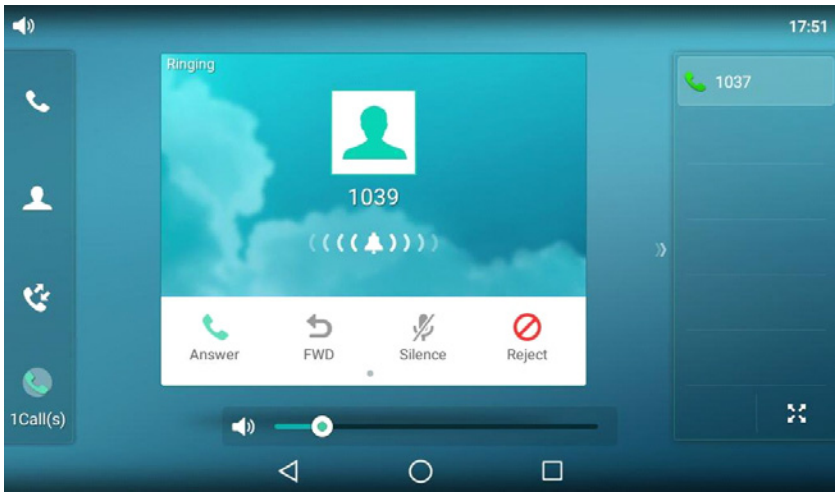
To adjust the ringer volume when the phone is idle:

- Press  to adjust the ringer volume.



To adjust the ringer volume when the phone is ringing:

- Press  to adjust the ringer volume.



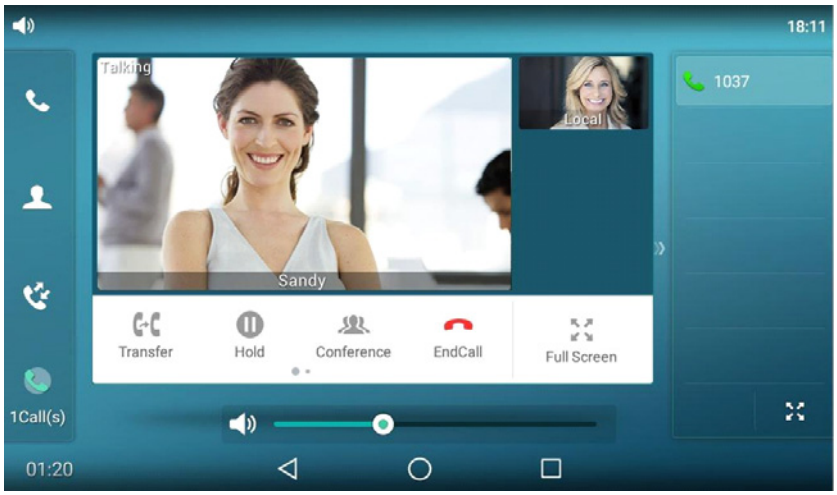
You can also press to adjust the ringer volume when selecting a ring tone.

If the ringer volume is adjusted to minimum, the  *icon will appear on the status bar.*

You can also drag the volume slider to adjust the ringer volume.

To adjust the talking volume during a call:

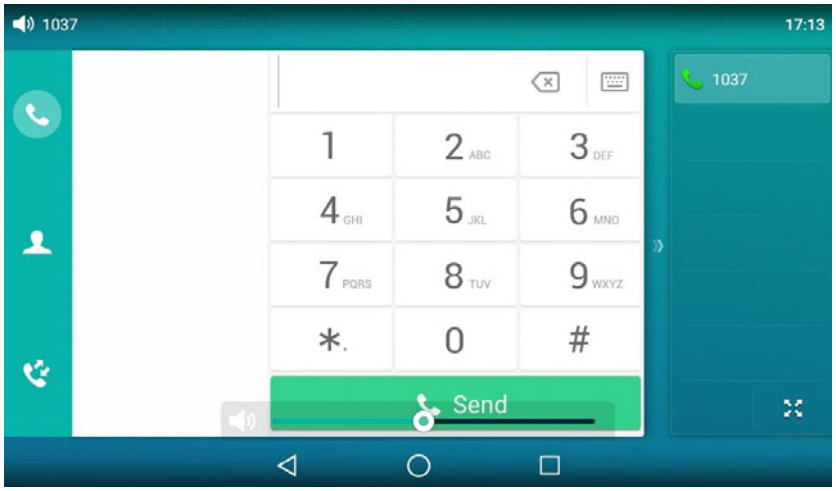
- Press  to adjust the volume of currently engaged audio device (handset,speakerphone or headset).



- You can also drag the volume slider to adjust the talking volume.

To adjust the key tone volume when the phone is on the dialing screen:

- Press  to adjust the key tone volume.

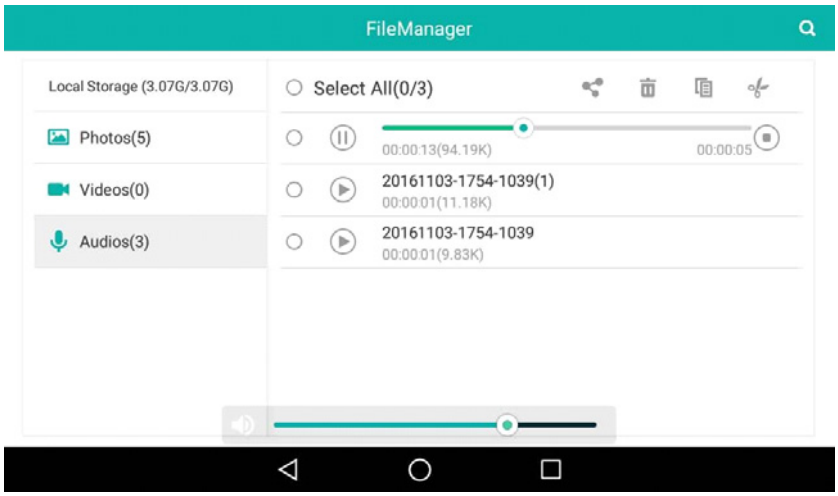


You can also drag the volume slider to adjust the key tone volume.

To adjust the media volume:

You can adjust the media volume in multiple scenarios. The following takes playing the audios by FileManager application as example:

- Press  to adjust the media volume.



You can also drag the volume slider to adjust the key tone volume.

Ring Tones

Ring tones are used to indicate incoming calls. You can select different ring tones to distinguish your phone from your neighbor's.

Procedure

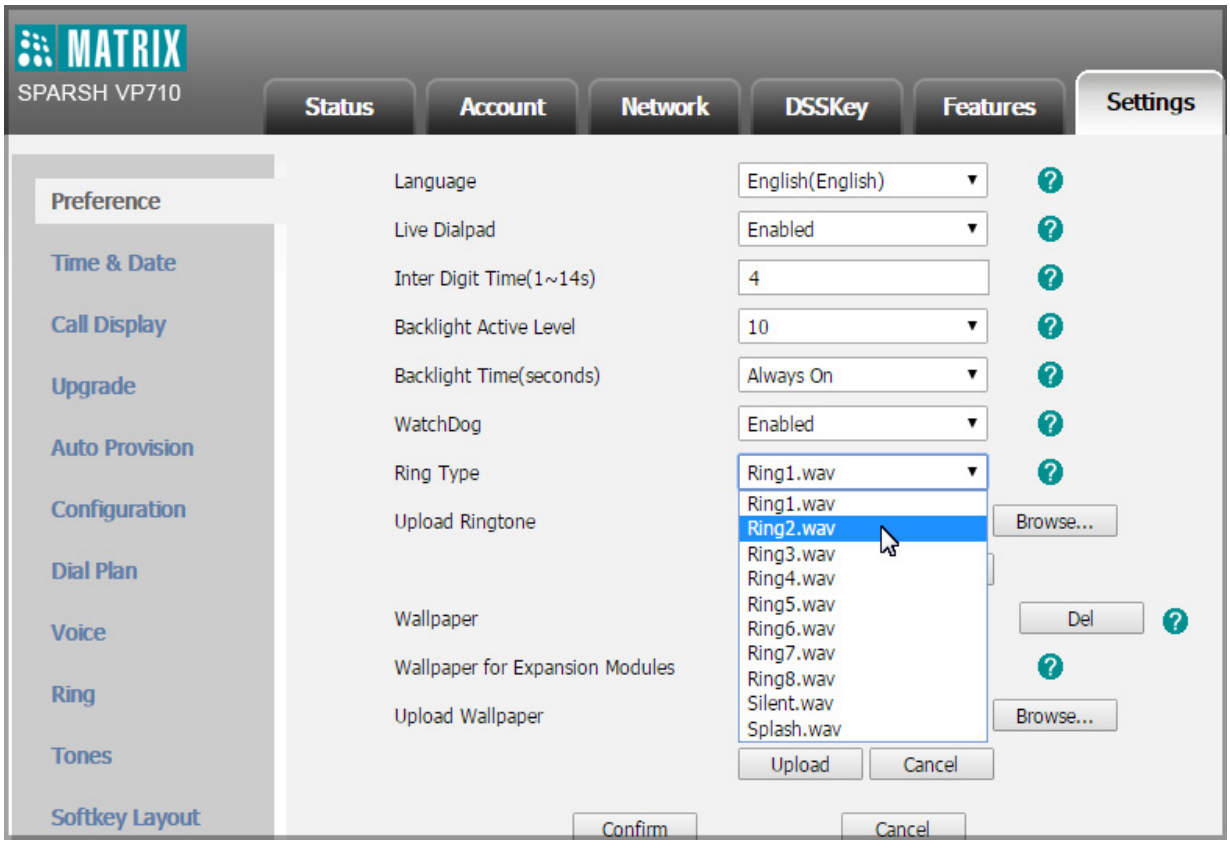
Ring tones can be configured using the following methods.

Configuration File	<MAC>.cfg	Configure a ring tone for the IP phone. Specify the access URL of the custom ring tone. Delete all custom ring tone files. Configure a ring tone on a per-line basis. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Upload the custom ring tones. Configure a ring tone for the IP phone. Configure a ring tone on a per-line basis. **Navigate to:** http://<phoneIPAddress>/ servlet?m=mod_ data&p=settings-preference&q=load http://<phoneIPAddress>/ servlet?m=mod_ data&p=account-basic&q=load&acc=0
	Phone User Interface	Configure a ring tone for the IP phone. Configure a ring tone for the account

To select a ring tone via web user interface:

- Click on **Settings->Preference**.
- Select the desired ring tone from the **Ring Type** list.

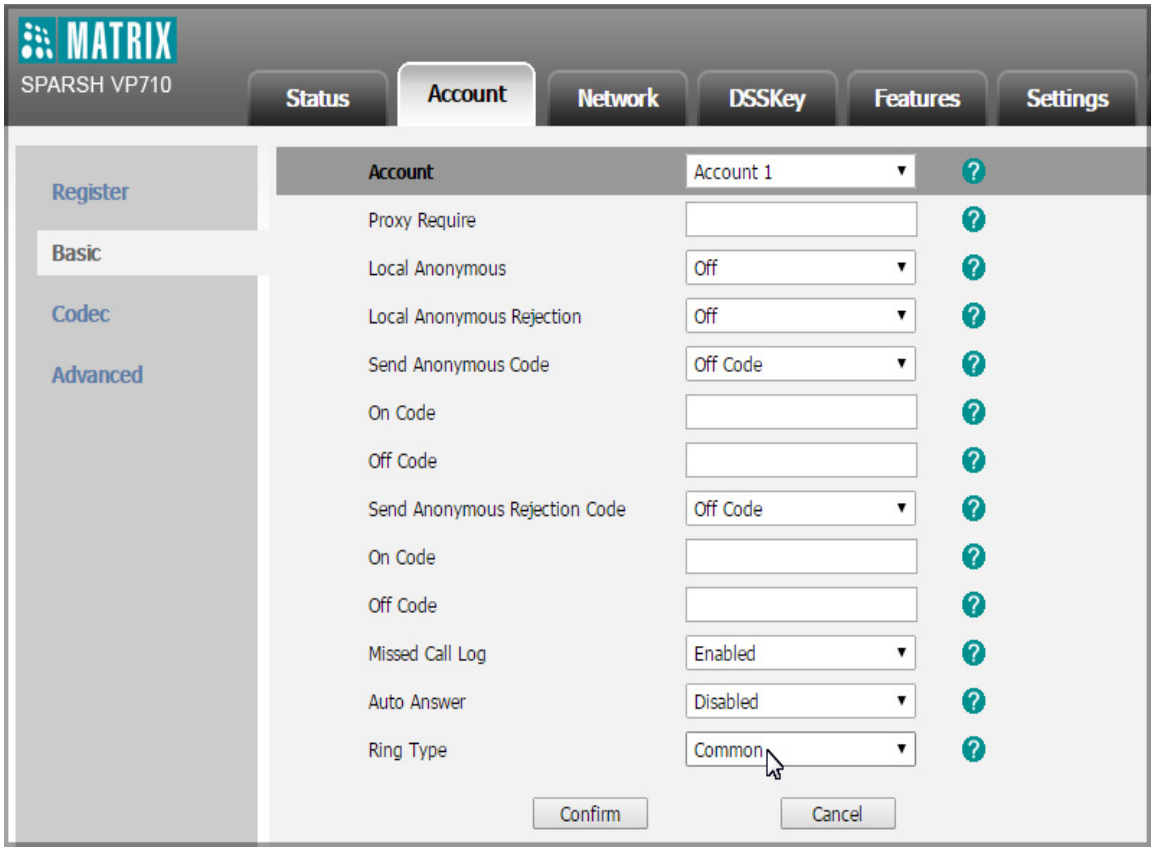


- Click **Confirm** to save the change.

A ring tone for the account is configurable via web user interface only.

To select a ring tone for the account via web user interface:

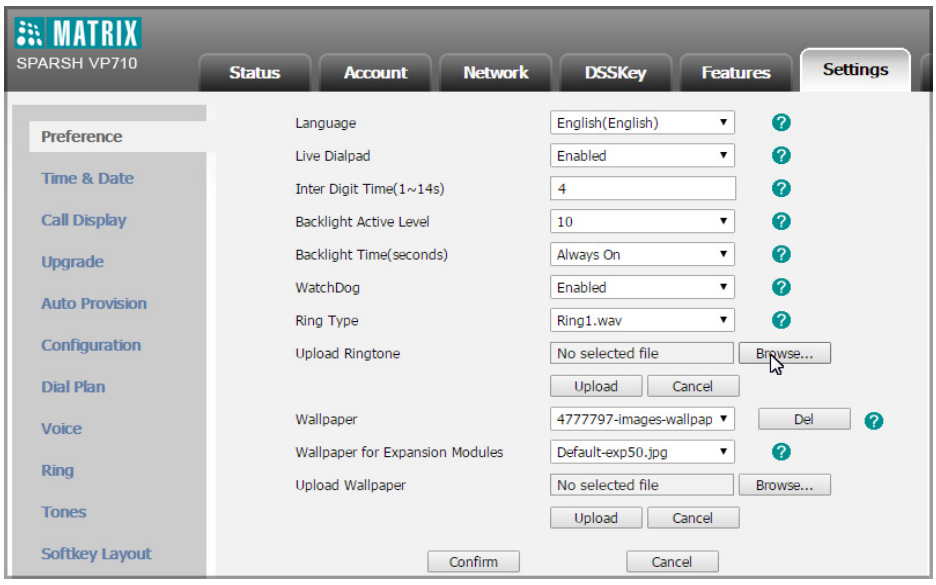
- Click on **Account->Basic**
- Select the desired ring tone from the **Ring Type** list.



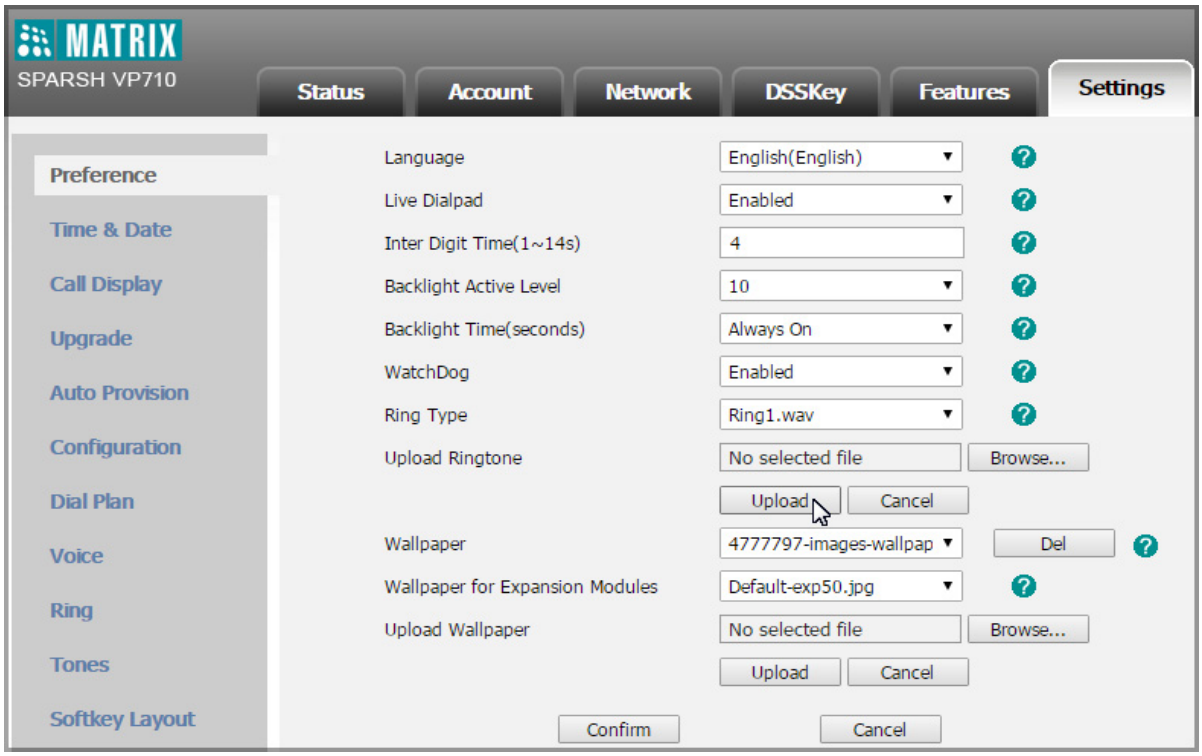
- If **Common** is selected, this account will use the ring tone selected for the phone
- Click **Confirm** to save the change.

To upload a custom ring tone via web user interface:

- Click on **Settings->Preference.**
- In the **Upload Ringtone** field, click **Browse** to locate a ring tone file (the file format must be *.wav) from your local system.



- Click **Upload** to upload the file.



The custom ring tone appears in the Ring Type list.

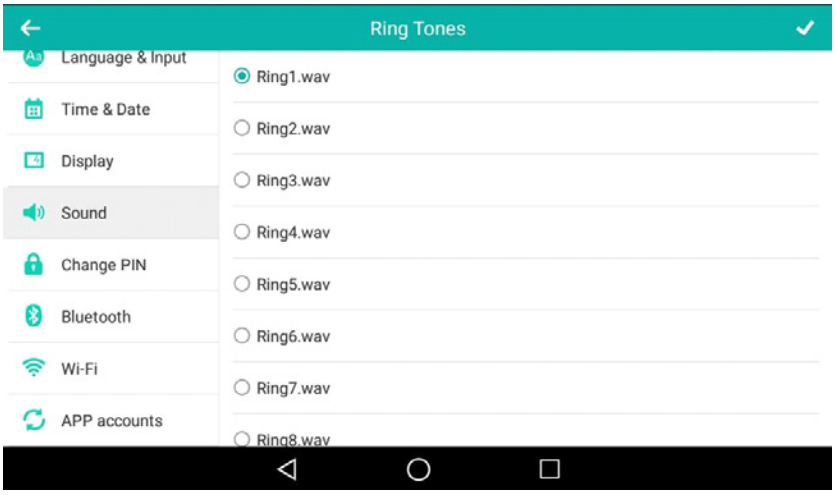
- You can only delete the custom ring tone by clicking **Del** in the **Ring Type** field.

The ring tone for an incoming call on the phone may be different. For example, when the phone receives an incoming call from a contact stored in the local directory, it will play the ring tone assigned to the contact in the contact directory (refer to ""). If no ring tone is assigned to the contact, the phone will play the ring tone assigned to the associated group (refer to "Adding Groups"). Otherwise, the phone will play the ring tone assigned to the account. If no ring tone is assigned to the account, the phone will play the ring tone assigned for the phone.

Single custom ring tone file must be within 8MB. Uploading custom ring tones for your phone is configurable via web user interface only.

To select a ring tone for the phone via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Sound->Ring Tones->Common.**
- Drag up and down to scroll through the list of available ring tones.
- Tap the desired ring tone.

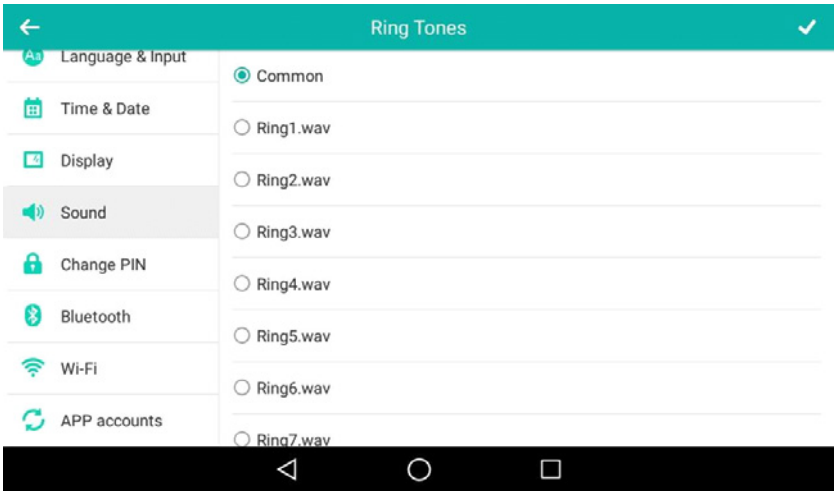


- Press  to adjust the ringer volume
- Tap  to accept the change or  to cancel.

To select a ring tone for the account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Sound->Ring Tones.**
- Tap the desired account.
- Drag up and down to scroll through the list of available ring tones.
- Tap the desired ring tone.

- If **Common** is selected, this account will use the ring tone selected for the phone.



- Press  to adjust the ringer volume
- Tap  to accept the change or  to cancel.

Key Tone

If you enable key tone, the phone will produce a sound when you press the keypad.

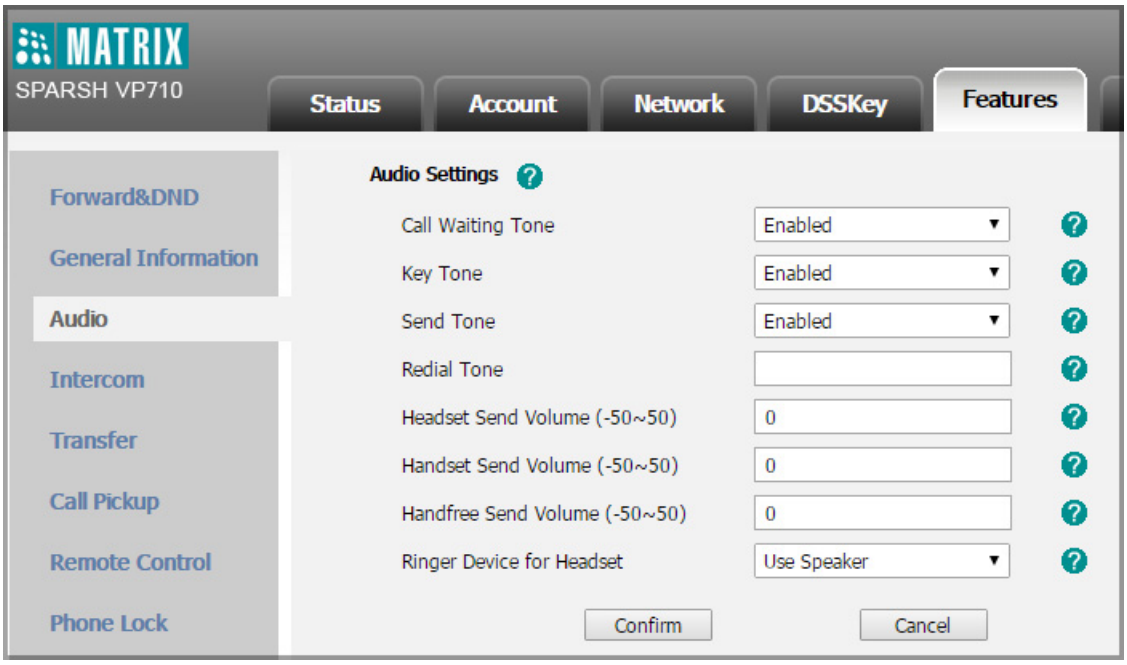
Procedure

Key Tone can be configured using the following methods.

Configuration File	<MAC>.cfg	Configure a key tone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure a send tone or key tone. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-audio&q=load
	Phone User Interface	Configure a key tone.

To configure key tone via web user interface:

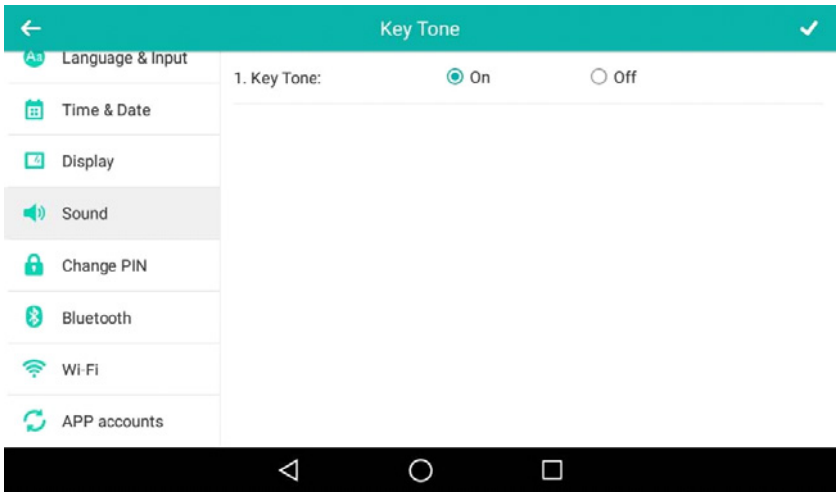
- Click on **Features->Audio.**
- Select the desired value from the **Key Tone** list.



- Click **Confirm** to save the change.

To enable key tone via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Sound->Key Tone.**
- Tap the On radio box in the **Key Tone** field.



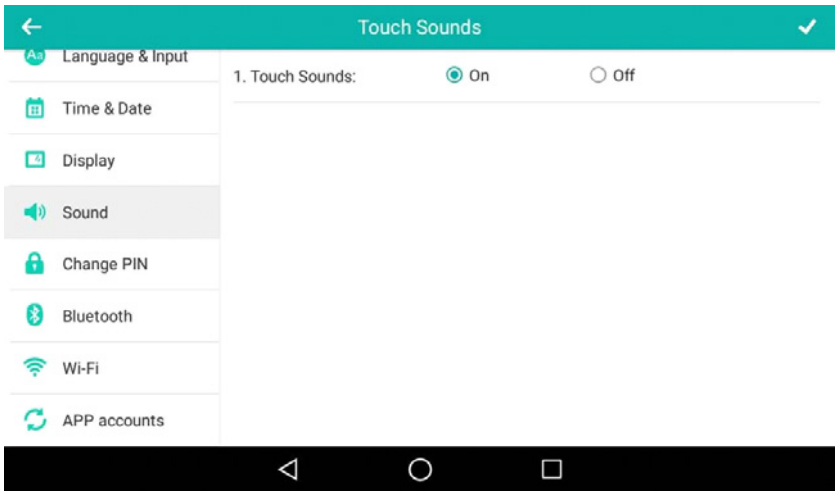
- Tap ✓ to accept the change or ◁ to cancel.

Touch Sounds

If you enable touch sounds, the phone will produce a sound when you tap an option on the touch screen.

To enable touch sounds via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Sound->Touch Sounds.**
- Tap the **On** radio box in the **Touch Sounds** field.



- Tap ✓ to accept the change or ◁ to cancel.

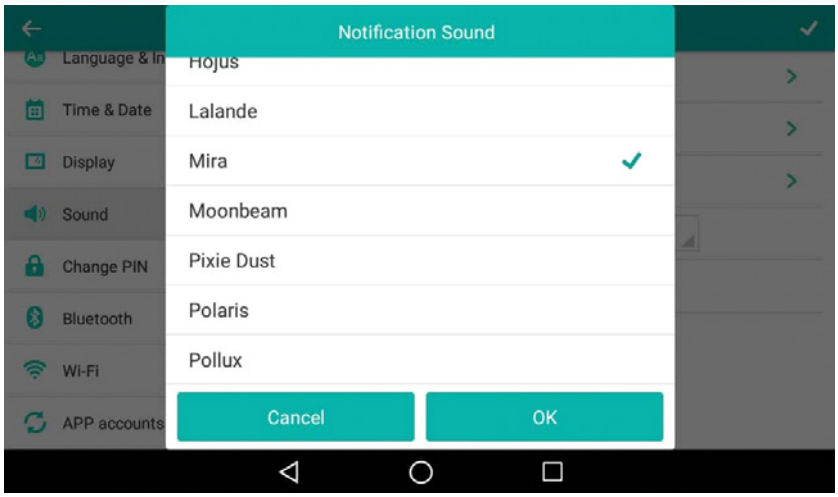
Touch sound is configurable via phone user interface only.

Notification Sound

The phone will produce a notification sound when receiving a notification (e.g., incoming email notification).

To select a notification sound for the phone via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->Sound.**
- Tap the **Notification Sound** field.
- Tap the desired notification sound in the pop-up dialog box



- Press  to adjust the notification sound.
- Tap **OK** to accept the change or **Cancel** to cancel.

Notification sound is configurable via phone user interface only.

Silent Mode

If you turn on the silent mode, the IP phone won't produce ring tone/key tone/touch sound/notification sound from phone's speaker.

You can use silent mode feature to block the incoming call/message from producing ring tone/notification tone from phone's speaker. It is helpful for users not to be disturbed by the tone.

Procedure:

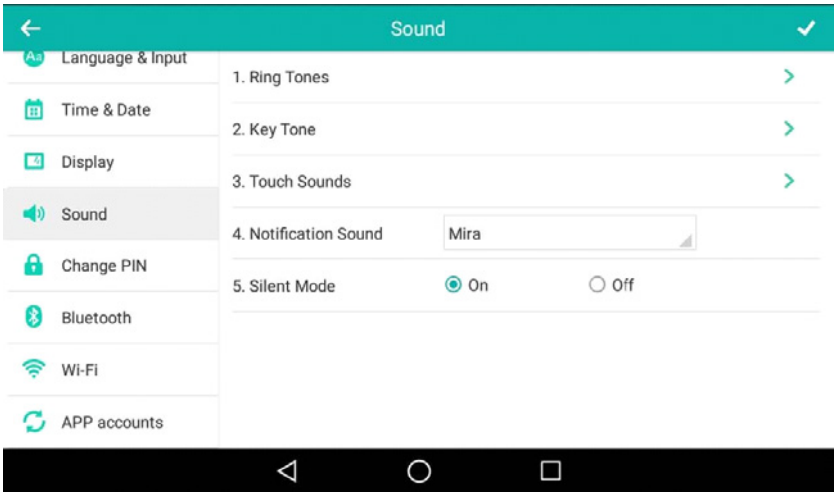
Silent Mode can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the using permission of silent mode feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Phone User Interface	Enable/Disable Silent Mode.

To enable the silent mode via phone user interface:

Do one of the following:

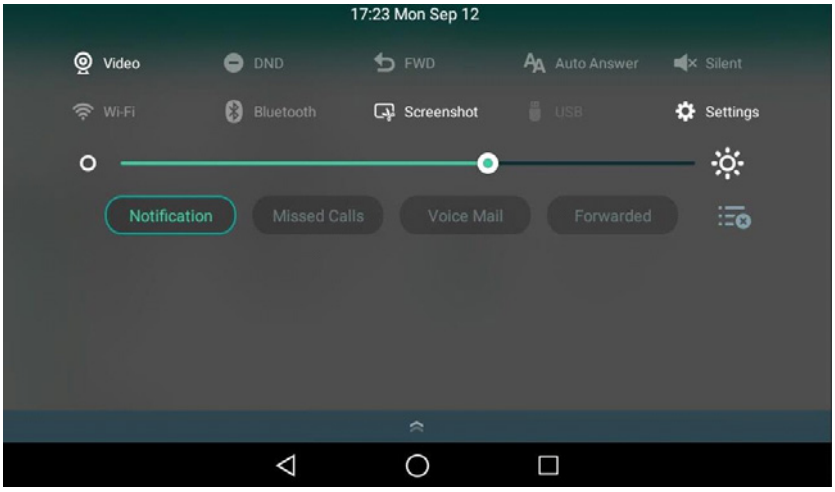
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
  - Tap **Settings->Basic->Sound.**
  - Tap the **On** radio box in the **Silent Mode** field.



  - Tap  to accept the change or  to cancel.

- Press  to adjust the adjust the ringer volume to minimum.

- Swipe down from the top of the screen to enter the control center and toggle Silent on.



Silent mode is configurable via phone user interface only.

You may not have the permission to turn on the silent mode. Contact your system administrator for more information.

Contact Management

This section provides the operating instructions for managing contacts. Topics include:

- Directory
- Local Directory
- Blacklist
- Remote Phone Book
- Mobile Contacts

Local Directory

The built-in phone directory can store contact names and numbers. You can store up to 1000 contacts and 48 groups (including the default groups: Company, Family and Friend) in your phone's local directory; add new contacts and groups; edit, delete or search for a contact; or simply dial a contact number from the local directory.

Contacts and groups can be added either one by one or in batch using a local contact file.

Procedure

Configuration changes can be performed using the configuration files or locally.

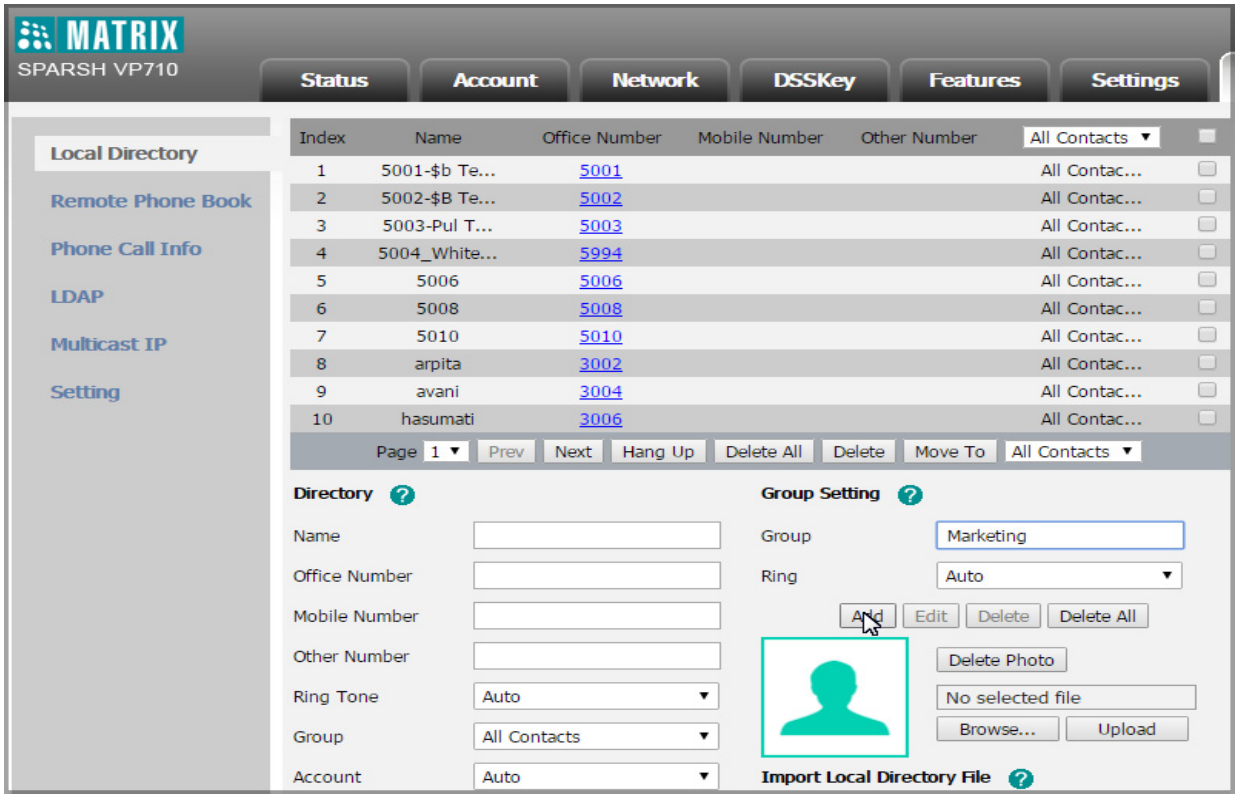
Configuration File	<MAC>.cfg	Specify the access URL of the local contact file. Specify the access URL of a contact avatar file. Specify the access URL of a contact icon file Specify the access URL of a TAR contact avatar file. Specify the access URL of the compressed TAR file consisting of the avatars TAR file and contact XML file. Specify the access URL of a TAR contact icon file. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Add a new group and a contact to the local directory. To import or export the local contact file. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=contactsbasic&q=load&group=0&page=1
	Phone User Interface	Add a group and a contact to the local directory.

Adding Groups

To add a group to the local directory via web user interface:

- Click on **Directory->Local Directory**.
- In the **Group Setting** block, enter the desired group name in the **Group** field.
- Select the desired ring tone from the **Ring** list.

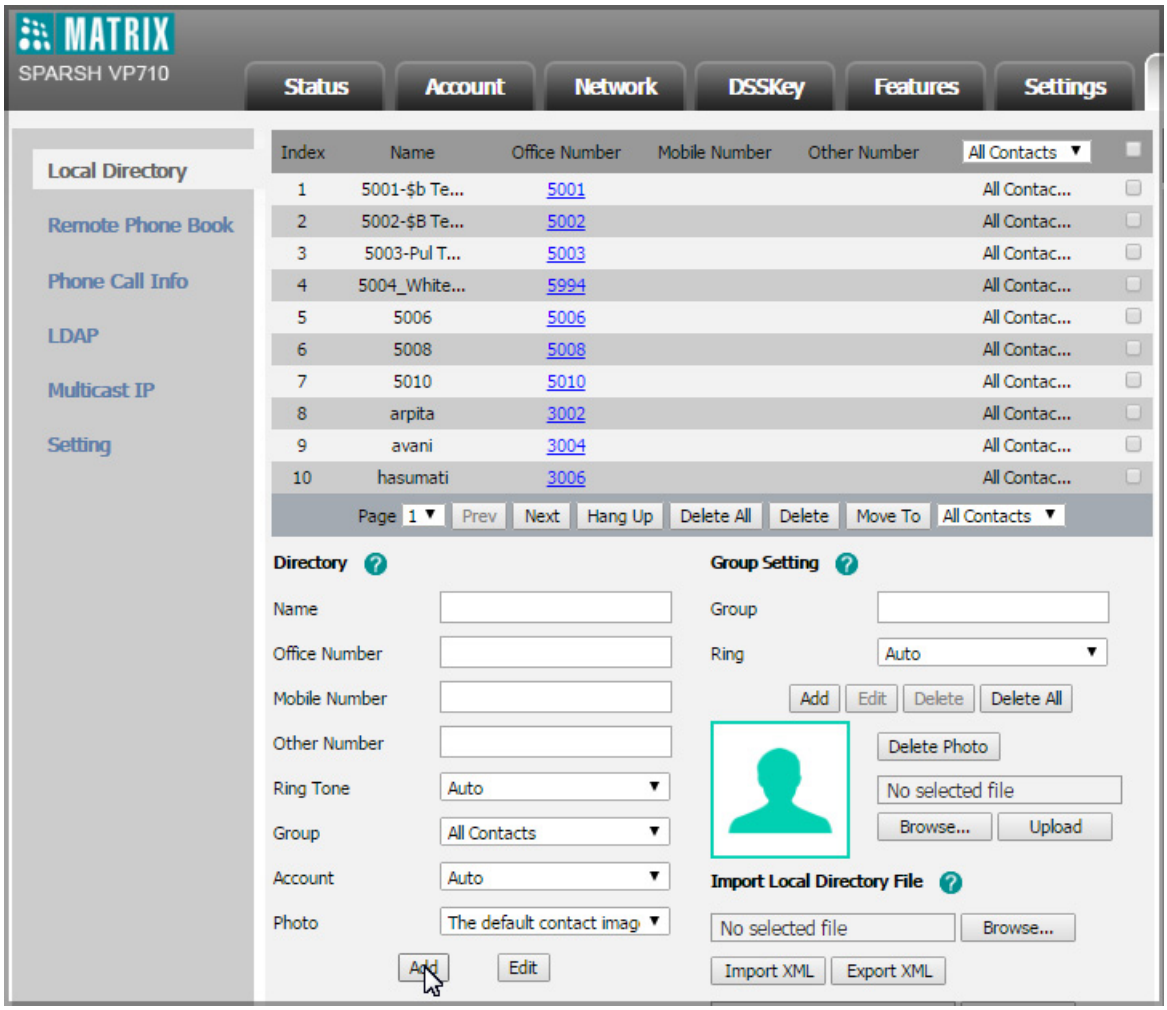


- Click **Add** to add the group.

To add a contact to the local directory via web user interface:

- Click on **Directory->Local Directory.**
- In the **Directory** block, enter the name and the office, mobile or other numbers in the corresponding fields.
- Select the desired ring tone from the **Ring Tone** list**.**
- Select the desired group from the **Group** list.
- Select the desired account from the **Account** list.

  If Auto is selected, the IP phone will use the default account when placing calls to the contact from the local directory.
- Select the desired account from the **Photo** list.

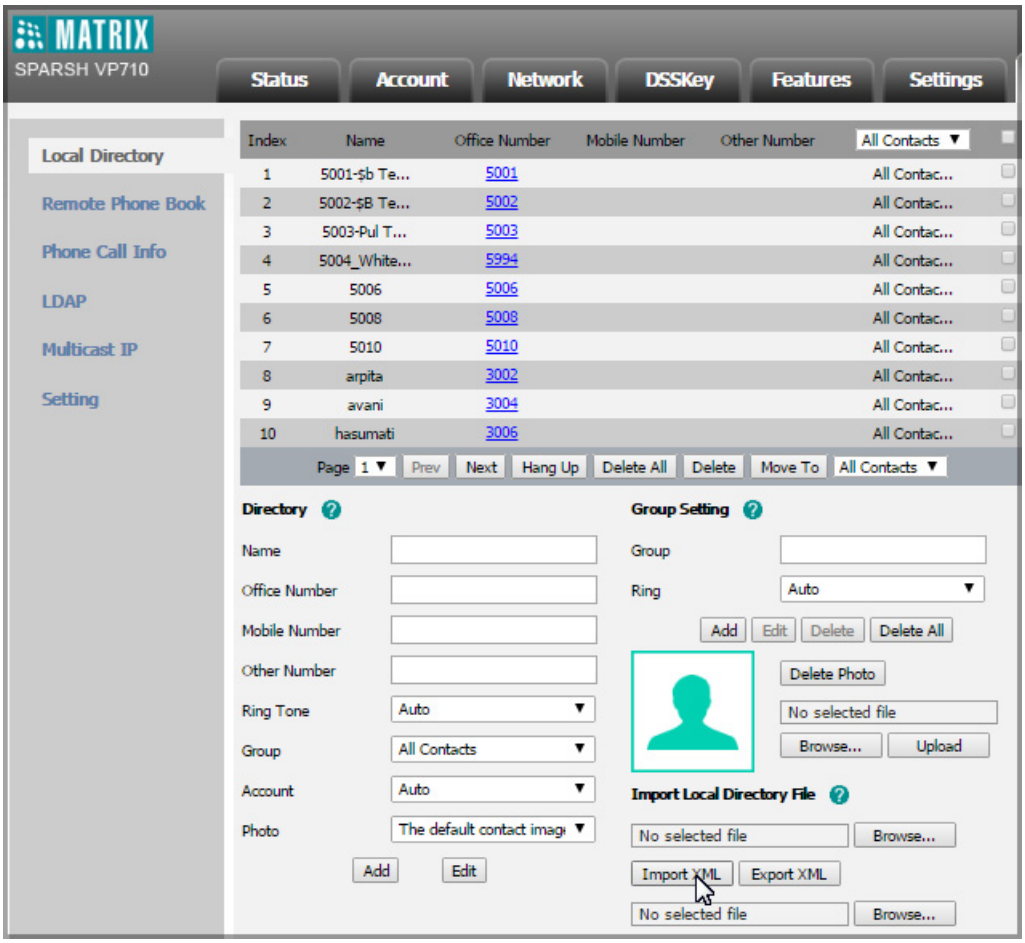


- Click **Add** to add the contact.

To import an XML contact list file via web user interface:

- Click on **Directory->Local Directory.**

- Click **Import XML** to locate and import a contact list file (the file format must be *.xml) from your local system



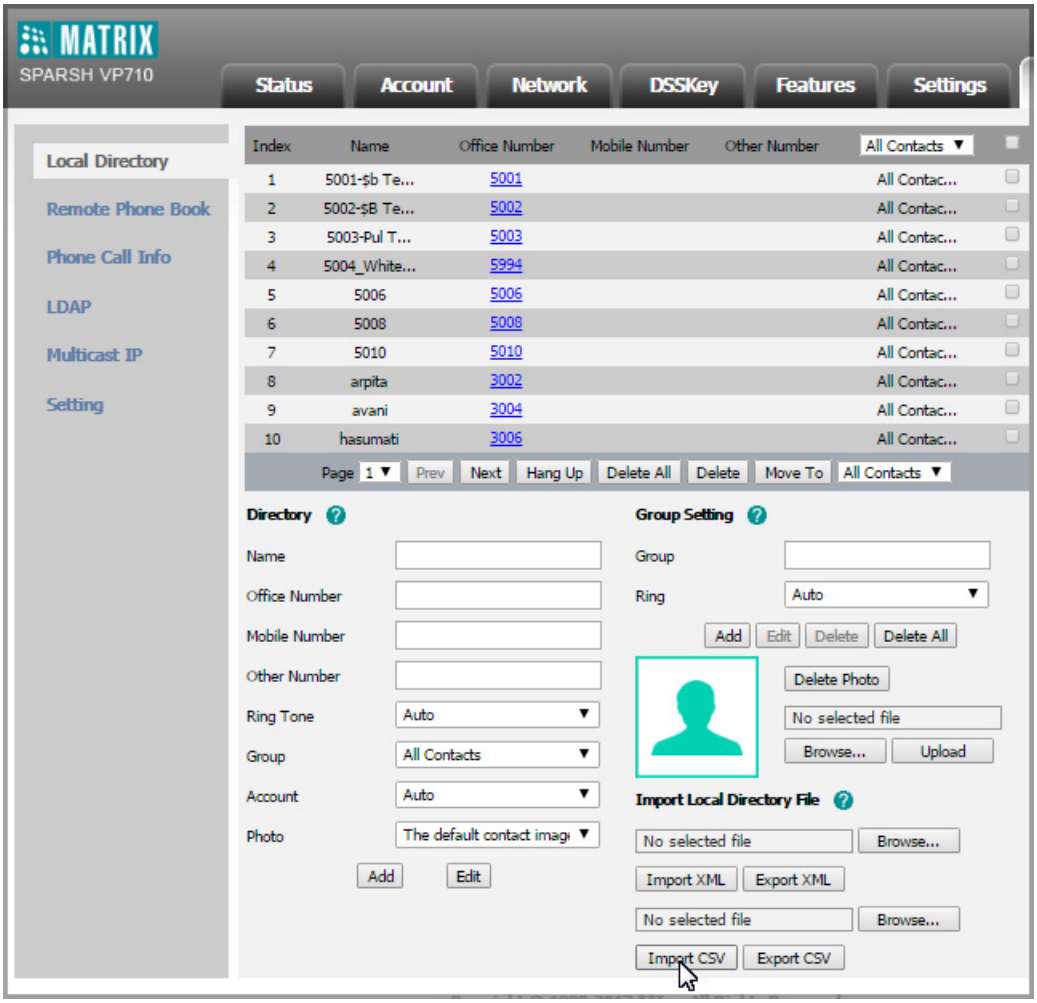
The web user interface prompts "The original contact will be covered, continue?".

- Click **OK** to complete importing the contact list.

To import a CSV contact list file via web user interface:

- Click on **Directory->Local Directory.**
- Click **Import CSV** to locate and import a contact list file (the file format must be *.csv) from your local system.
- Check the **Show Title** checkbox.
  It will prevent importing the title of the contact information which is located in the first line of the CSV file.
- Check the **Delete Old Contacts** checkbox.
  It will delete all existing contacts while importing the contact list.
- Select the contact information you want to import into the local directory from the list.

At least one item should be selected to be imported into the local directory.

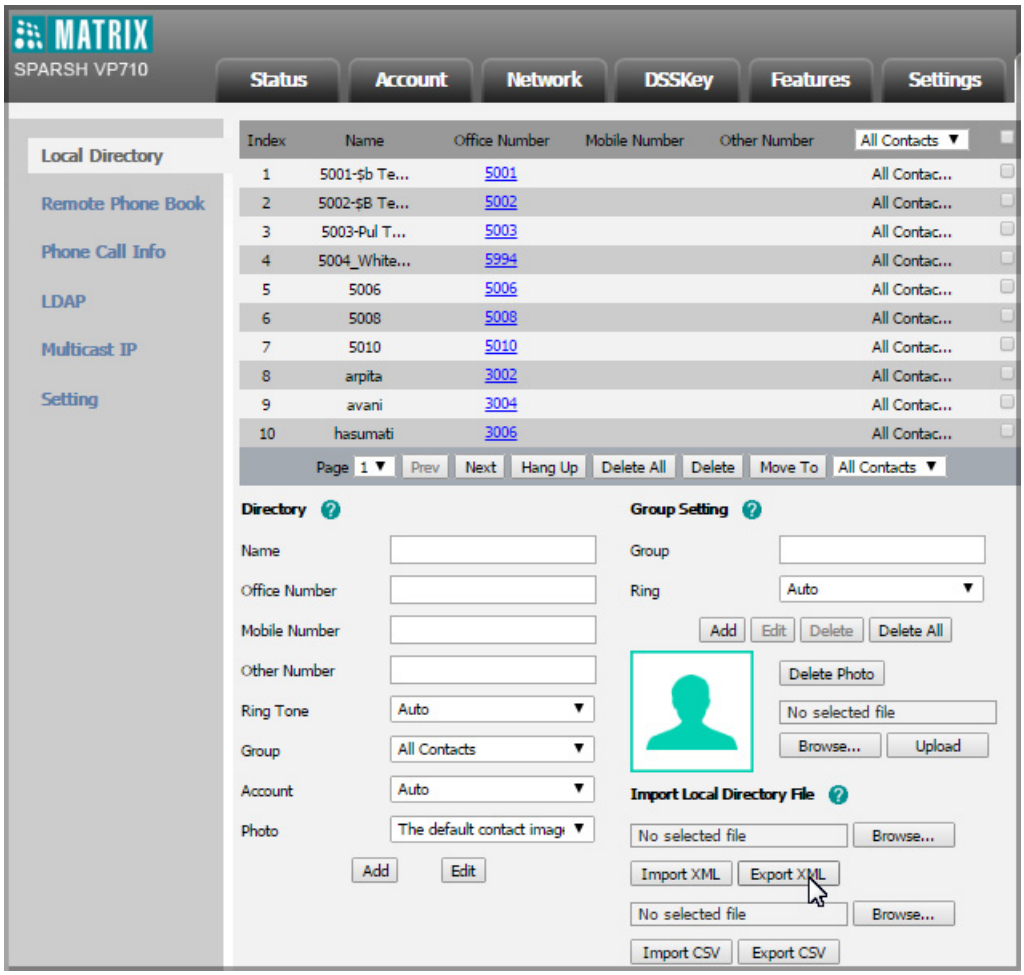


- Click **Import** to complete importing the contact list.

To export a contact list via web user interface:

- Click on **Directory->Local Directory.**

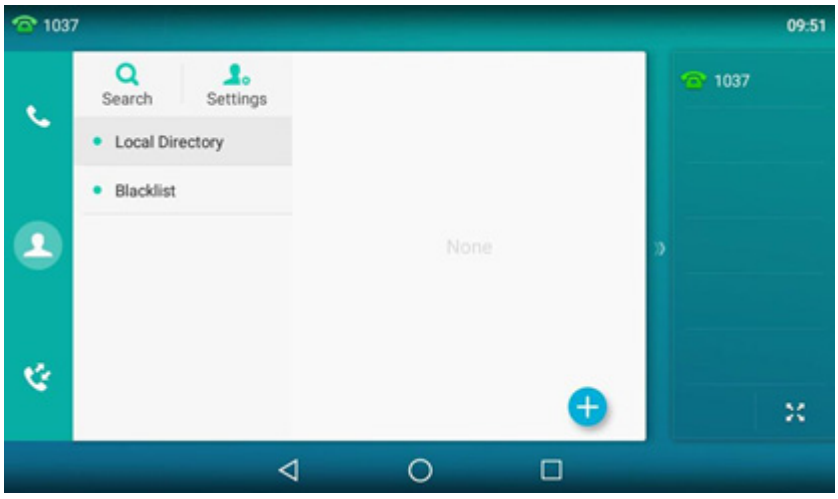
- Click **Export XML (or Export CSV)**.



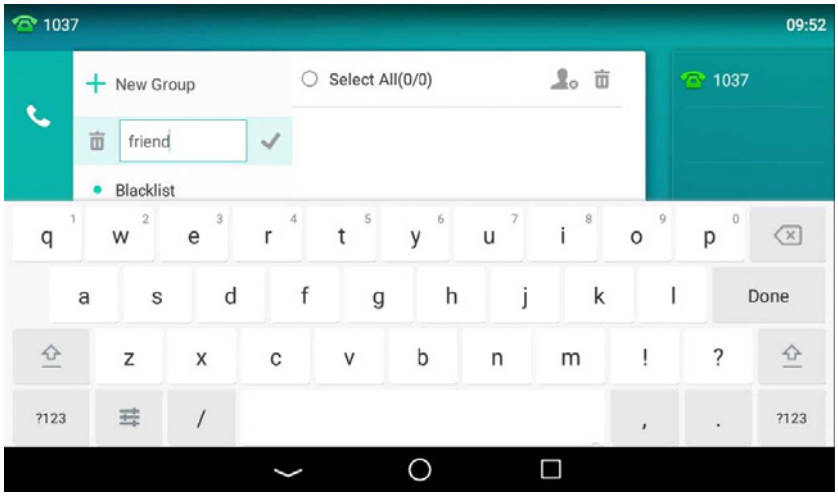
- Click **Save** to save the contact list to your local system.

To add a group to the local directory via phone user interface:

- Tap.  .

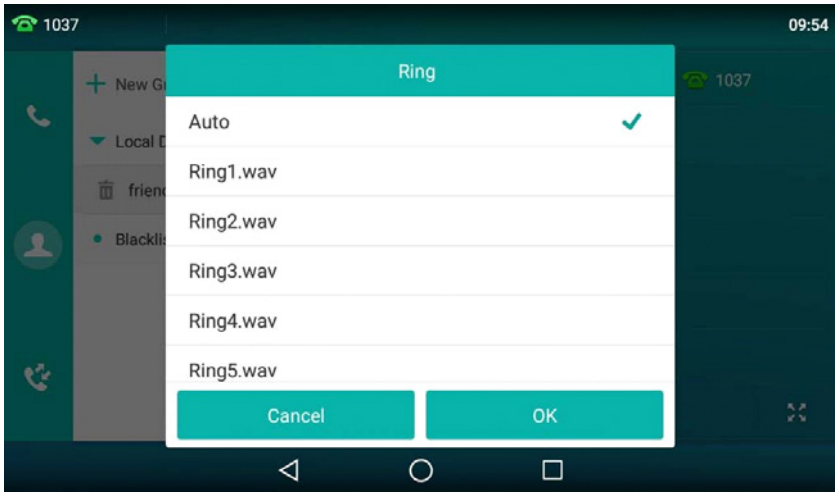


- Tap **Settings**.
- Tap **New Group**.



- Enter the desired group name in the highlighted field.
- Tap to accept the change.
- Tap after the group name to specify a ring tone for the group.

  Tap the desired ring tone in the pop-up dialog box.



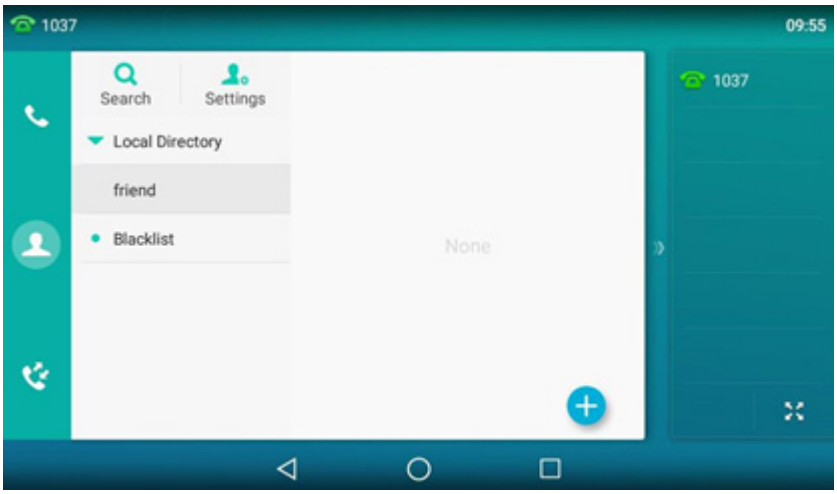
  If Auto is selected, this group will use the ring tone according to the priority.For more information on ringtone for an account, refer “Ring Tones”.

- Tap **OK** to accept the change.

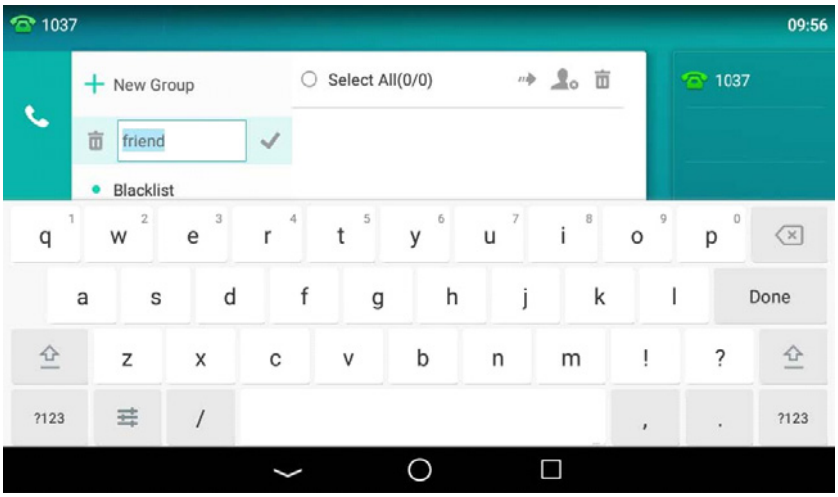
Editing a Group

To edit a group in the local directory:

- Tap. .



- Tap **Settings**.
- Tap the desired group.
- Tap after the group name.

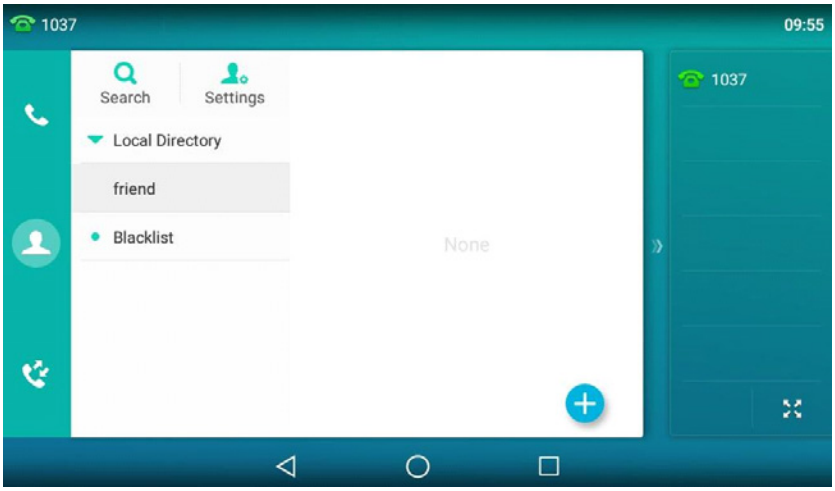


- Edit the group name in the highlighted field.
- Tap to accept the change

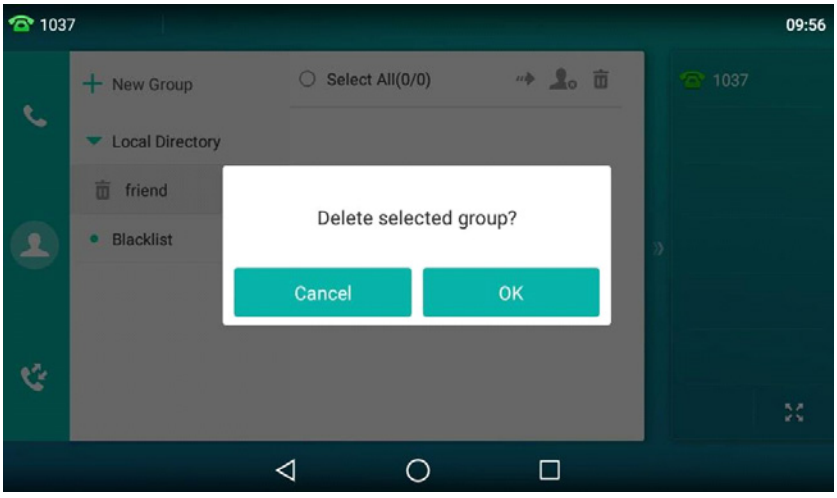
Deleting a Group

To delete a group from the local directory:

- Tap. .



- Tap **Settings**.
- Tap before the desired group name.



- The touch screen prompts the following warning:
- Tap **OK** to accept the change or **Cancel** to cancel.

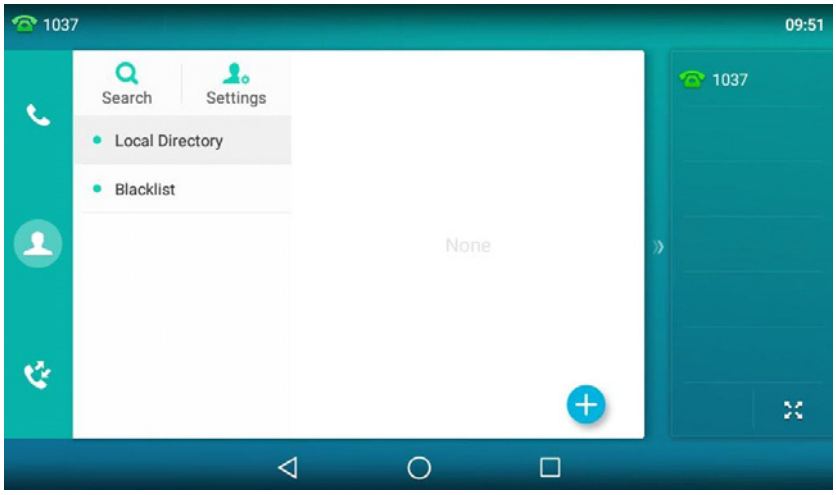
Adding a Contact

You can add contacts to the local directory in one of the following ways:

- Manually
- From call history. Refer “Call History Management” for more information.
- From a remote phone book. Refer “Remote Phone Book” for more information.

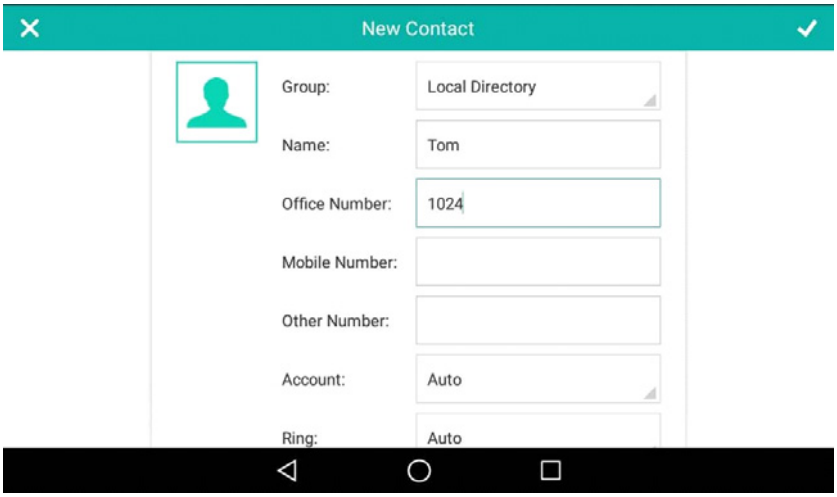
To add a contact to the local directory manually:

- Tap. .



- Tap

  If you want to add a contact to the specified contact group, you can tap the contact group first, and then tap .
- Enter the name and the office, mobile or other numbers in the corresponding fields.



- Tap the **Account** field.
- Tap the desired account in the pop-up dialog box.

  If **Auto** is selected, the phone will use the default account when placing calls to the contact from the local directory.

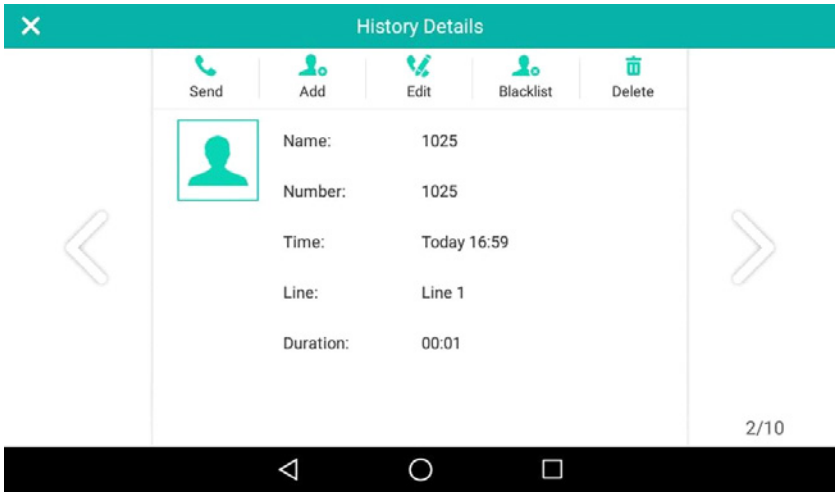
- Tap the **Ring** field.
- Tap the desired ring tone in the pop-up dialog box.

  If **Auto** is selected, this contact will use the ring tone according to the priority.

- Tap the **Photo** field.
- Tap the desired photo in the pop-up dialog box.
- Tap to accept the change or to cancel.

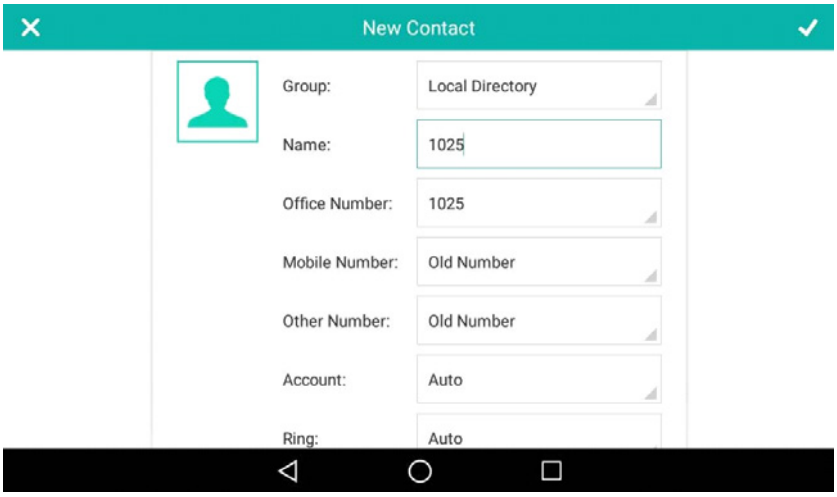
To add a contact to the local directory from call history:

- Tap .
- Drag up and down to view the list of call history.
- Tap after the desired entry.



- Tap **Add**.

- Edit the corresponding fields.

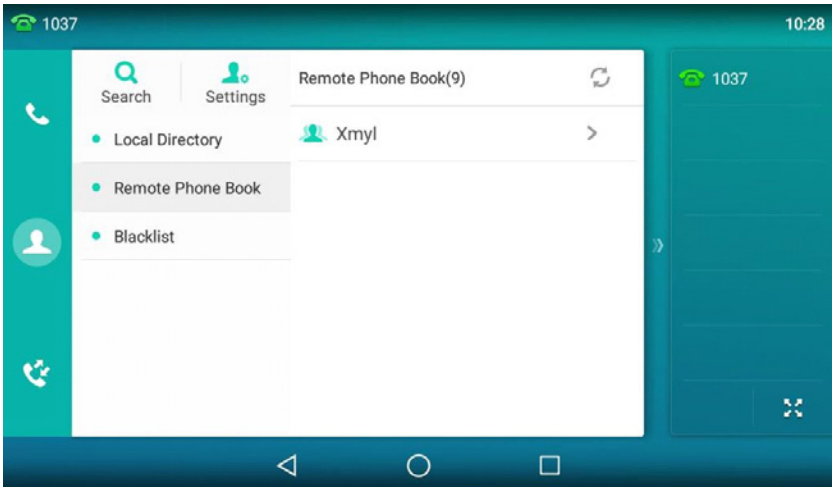


- Tap to accept the change.

  The entry is successfully saved to the local directory.

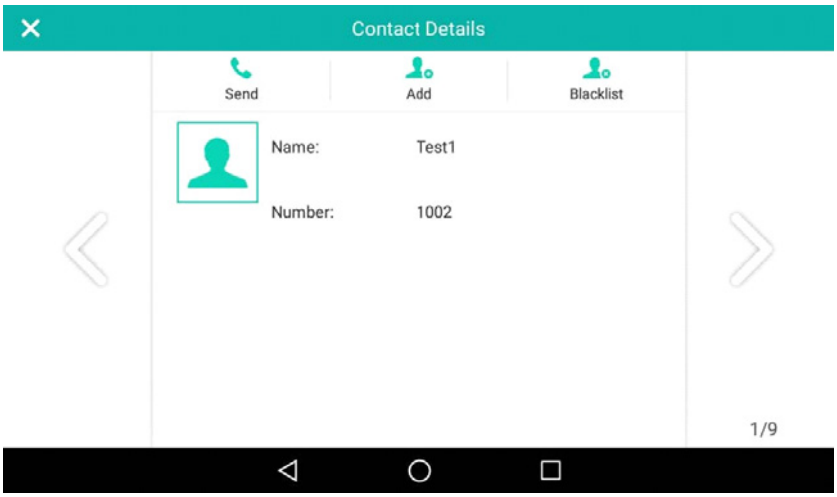
To add a contact from a remote phone book:

- Tap. Remote Phone Book.

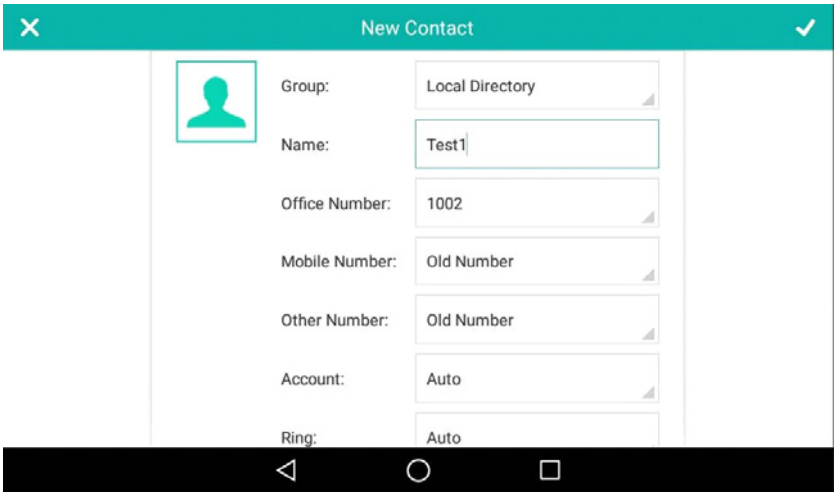


- Tap the desired remote phone book.
- The phone then connects to the remote phone book and proceeds to load it. The contacts in the remote phone book are displayed on the touch screen.

- Tap ⓘ after the desired contact in the remote phone book.



- Tap **Add**.
- Edit the corresponding fields.

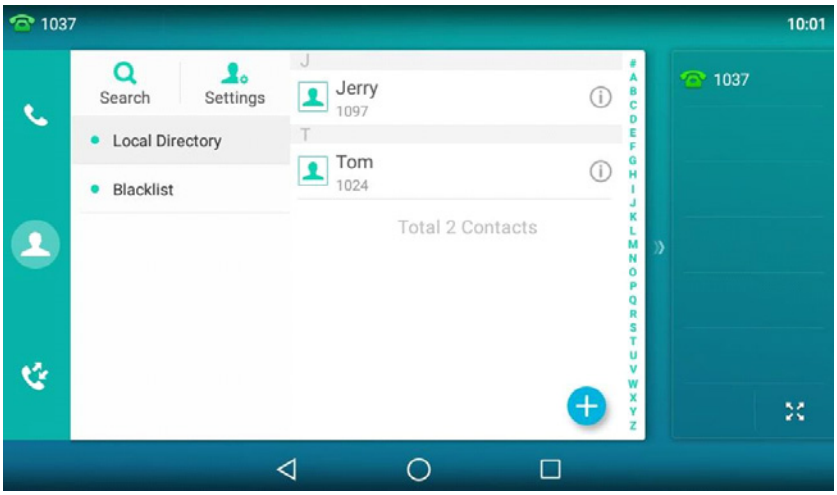


- Tap ✓ to accept the change or ◁ to cancel.

Editing a Contact

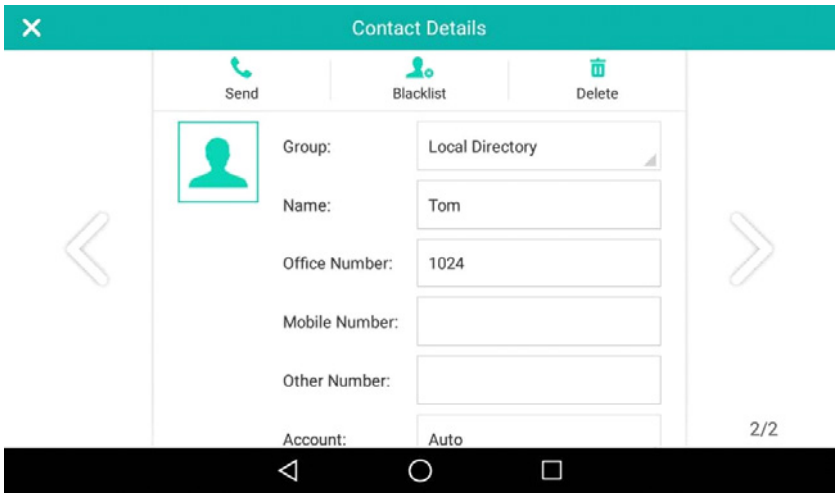
To edit a contact in the local directory:

- Tap. .



- Tap after the desired contact.

  If the contact was added to a specified contact group, you can tap the contact group first, and then tap after the desired contact.
- Tap the desired field to edit the contact information.

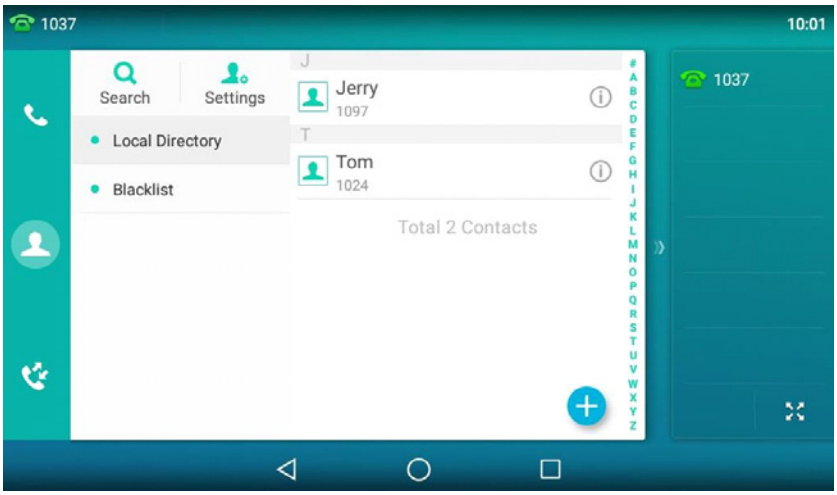


- You can tap or to change the contact you want to edit.
- Tap to accept the change or and then tap Discard to cancel.

Deleting a Contact

To delete a contact from the local directory:

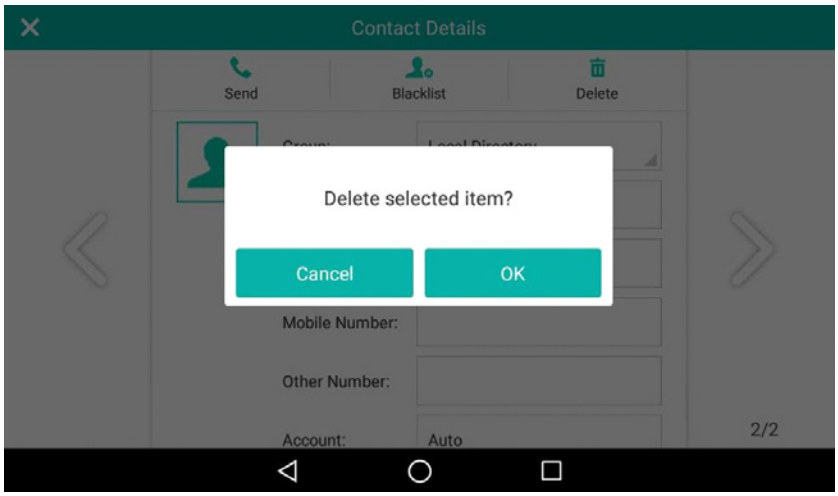
- Tap. .



- Tap after the desired contact.

  If the contact was added to a specified contact group, you can tap the contact group first, and then tap after the desired contact.
- Tap **Delete**.

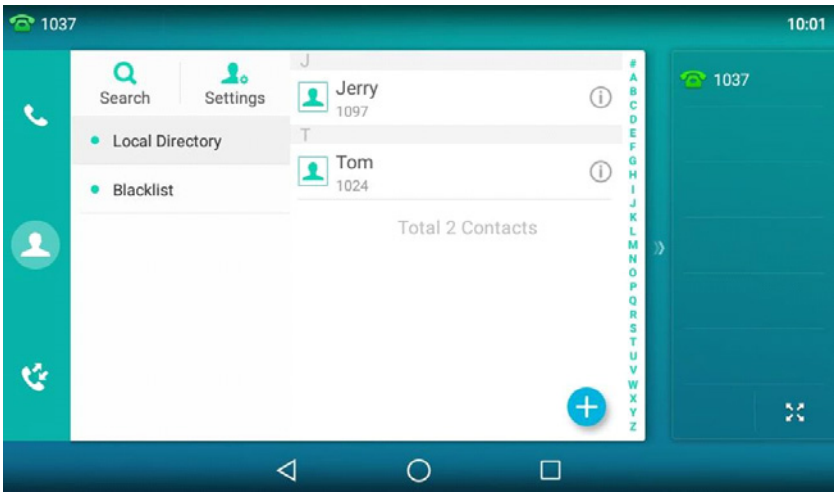
  The touch screen prompts the following warning:



- Tap **OK** to accept the change or **Cancel** to cancel.

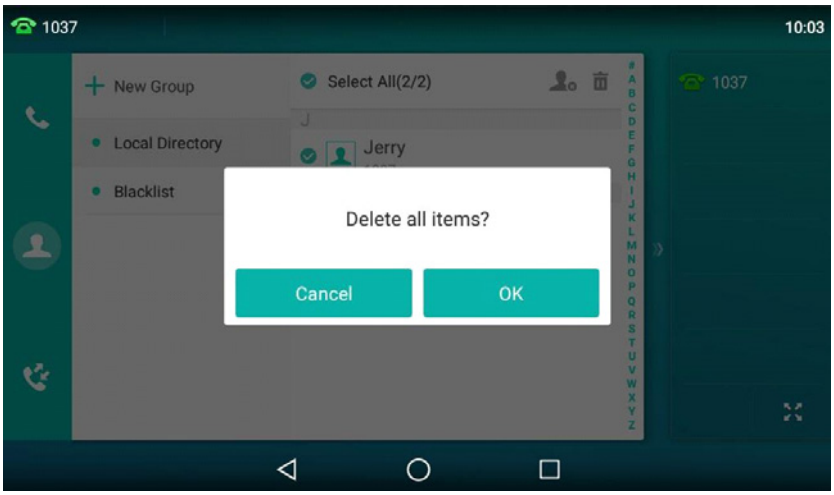
To delete all contacts:

- Tap .



- Tap **Settings**.
- Tap the **Select All** radio box.
- Tap .

  The touch screen prompts the following warning:



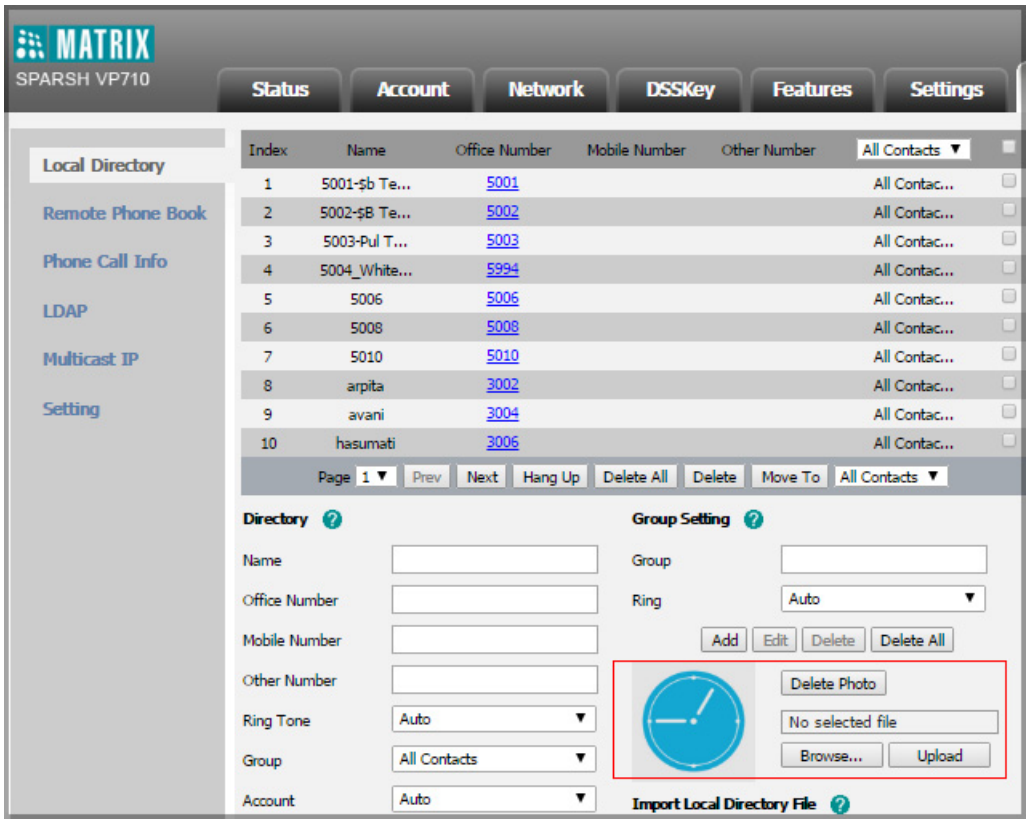
- Tap **OK** to accept the change or **Cancel** to cancel.

Uploading Contact Photos

To upload a new custom photo for the contact via web user interface:

- Click on **Directory->Local Directory.**
- Click **Upload Photo**, and then locate and upload a photo file from your local computer.

The web user interface prompts “Upload success!”.



You can click **Delete** Photo to delete the custom photo.

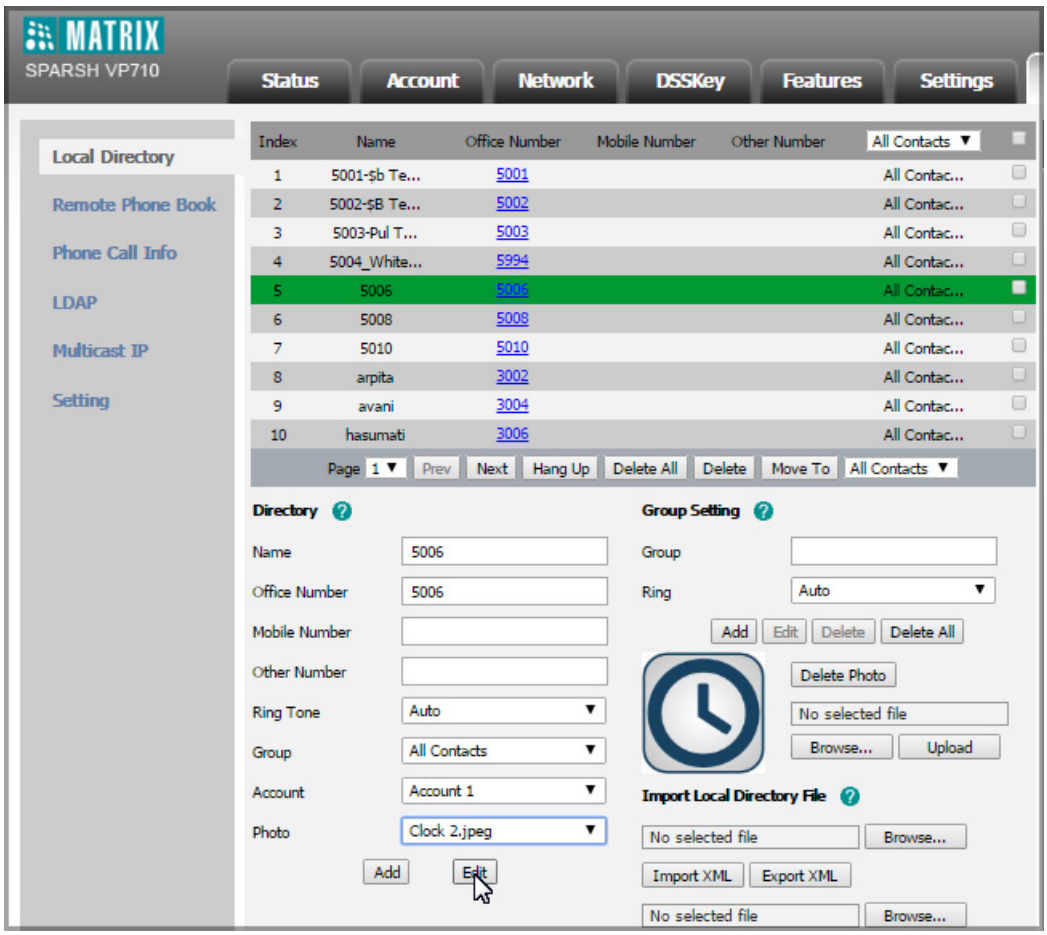
*The phone only supports *.png, *.jpg, *.jpeg and *.bmp format files. The file size must be less than 2M.*

You can only delete custom photos

To change the custom photo for the contact via web user interface:

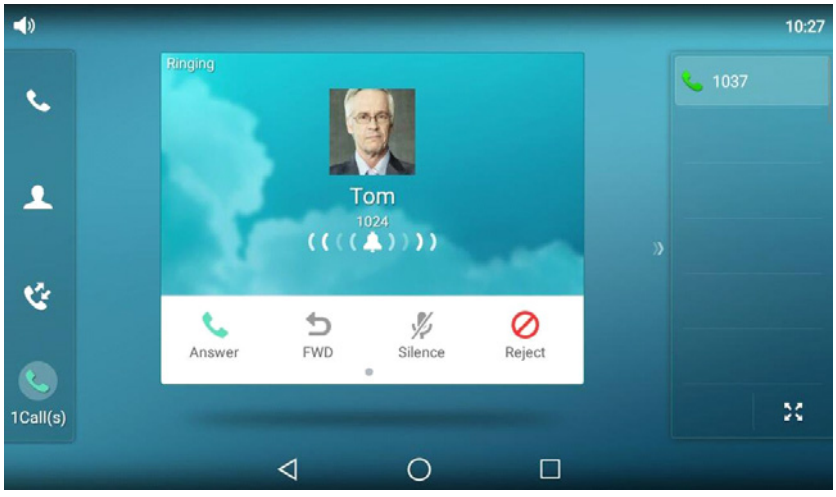
- Click on **Directory->Local Directory.**
- Click the desired contact you want to edit.

- Select the desired photo from the **Photo** list.



- Click **Edit** to accept the change.

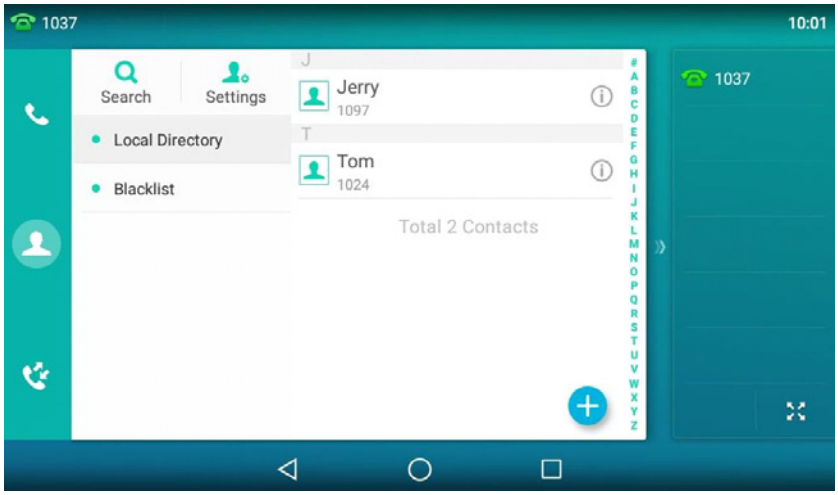
  When you place a call to the contact or receive a call from the contact, the touch screen will display the contact photo.



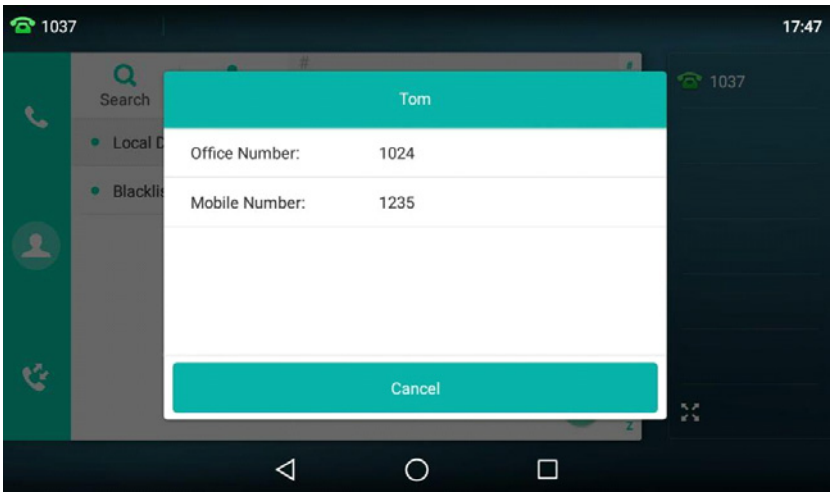
Placing Calls to Contacts

To place a call to a contact from the local directory:

- Tap .



- If the contact was added to a specified contact group, you can tap the contact group first, and then do one of the following:
  - Tap the desired contact.
    - If only one number for the contact is stored in the local directory, the IP phone will dial out the number.
    - If multiple numbers for the contact are stored in the local directory, tap the desired number to dial out.
- Tap after the desired contact.
  - If only one number for the contact is stored in the local directory, tap **Send** to dial out the number.
  - If multiple numbers for the contact are stored in the local directory, tap **Send** to display all numbers.

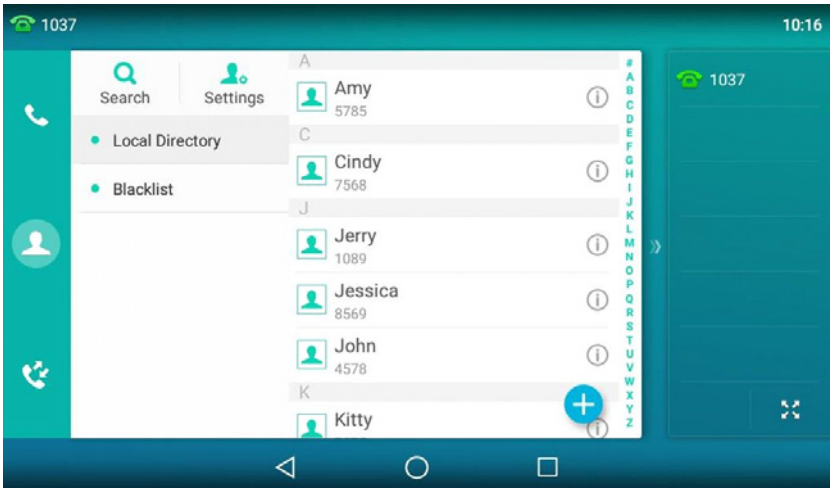


- Tap the desired number to dial out.

Searching for Contacts

To search for a contact in the local directory:

- Tap .

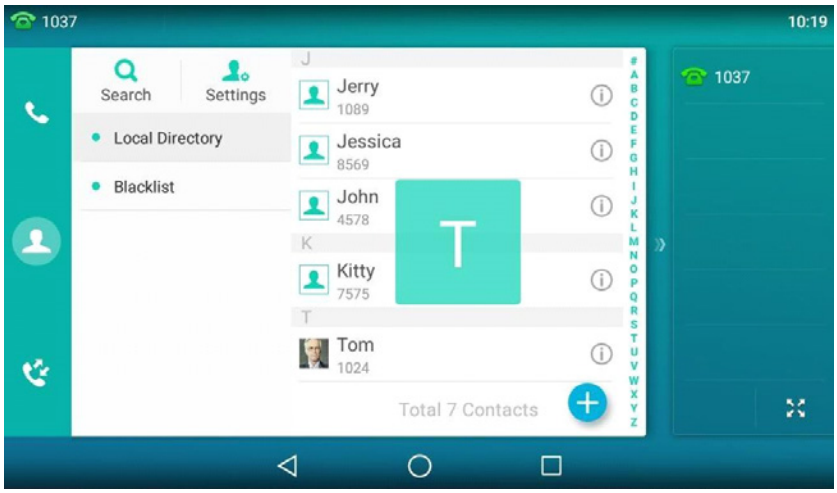


- Do one of the following:
  - Tap **Search**.

    Enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) using the onscreen keyboard.

    The contacts whose name or phone number matches the characters entered will appear on the touch screen. You can dial from the result list.

  - Tap the desired letter (e.g., T) along the right side to jump to contacts whose names start with T or t.



Drag up and down to select the desired contact to view or dial out.

Search Source List in Dialing

You can search for a contact in your desired lists when the phone is in the dialing screen. The lists can be Local Directory, History, Remote Phone Book and LDAP.

Procedure

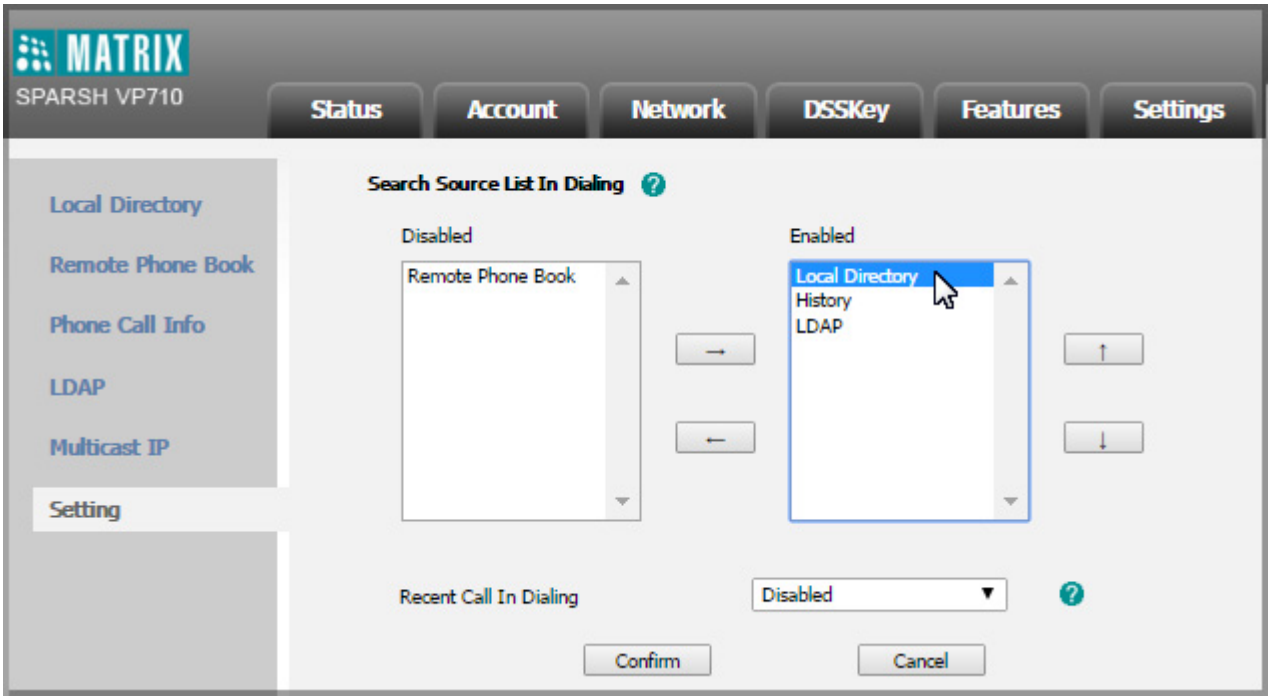
Search source list in dialing can be configured using the configuration files or locally.

Local	Web User Interface	Configure the search source list in dialing. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=contacts-favorite&q=load

To configure search source list in dialing via web user interface:

- Click on **Directory->Setting**.
- In the **Search Source List In Dialing** block, select the desired list from the **Disabled** column and then click  .
  The selected list appears in the **Enabled** column.
- Repeat above step to add more lists to the **Enabled** column.
- To remove a list from the **Enabled** column, select the desired list and then click  .
- To adjust the display order of search results, select the desired list and then click  or  .

•



- Click **Confirm** to save the change.

  The touch screen displays the search results in the adjusted order.

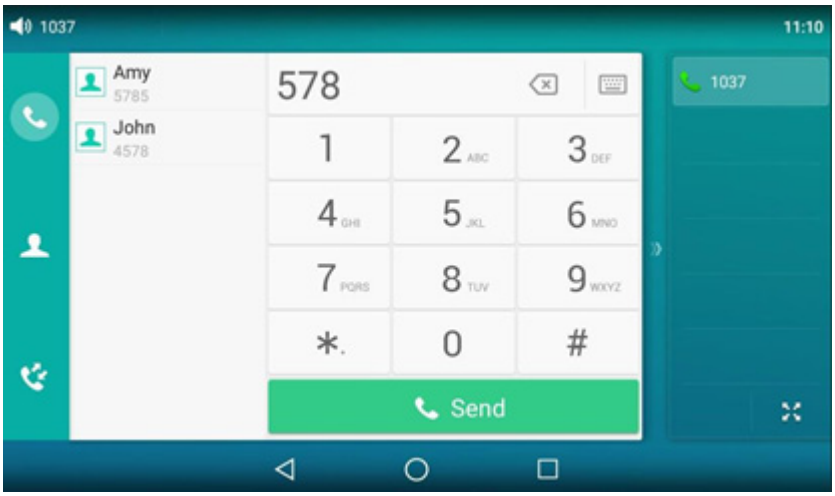
Search source list in dialing is configurable via web user interface only.

To search for a contact in the enabled search source lists:

- Pick up the handset, press the Speakerphone key or tap the line key.
- Enter a few continuous characters of the entry's name or continuous numbers of the entry's phone number (office, mobile or other number).

  The entries in the enabled search source lists whose name or phone number matches the characters entered will appear on the touch screen.

  You can tap the desired entry to place a call to the entry.



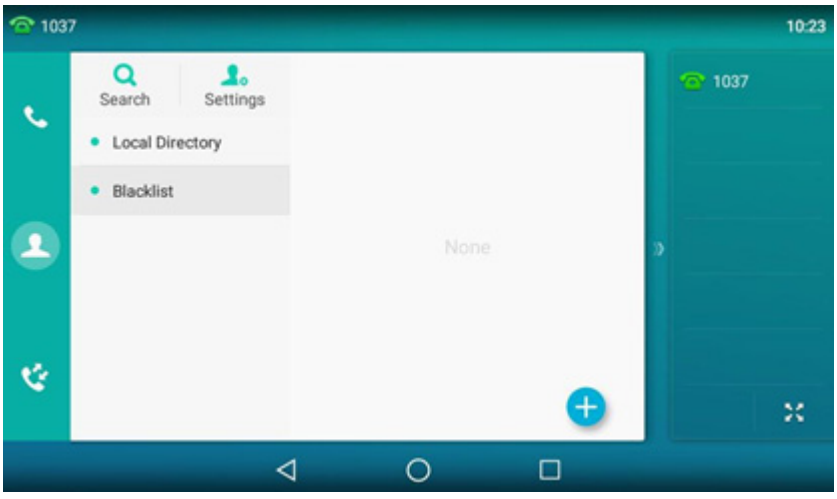
If you connect a Bluetooth-Enabled mobile phone to the IP phone, and enable the IP phone to sync phone contacts. The IP phone will first search the contacts in the mobile contacts directory then placed calls list. For more information, refer to "Mobile Contacts".

Blacklist

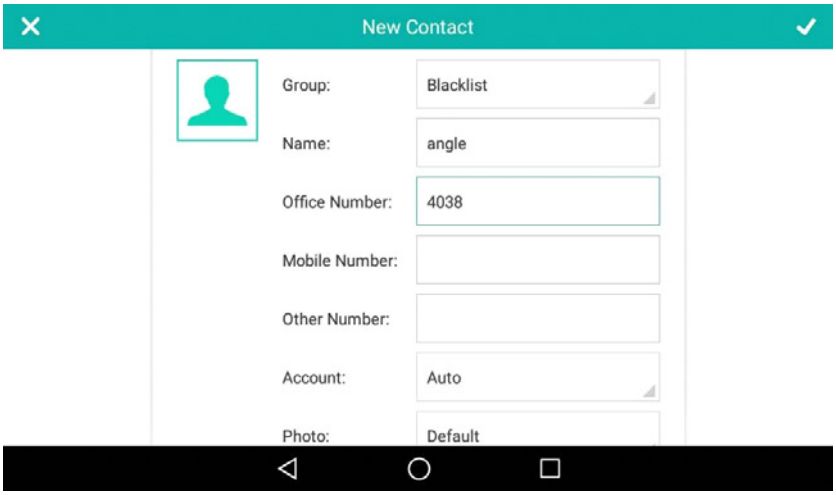
The built-in phone directory can store names and numbers for a blacklist. You can store up to 30 contacts in your phone's blacklist; add, edit, delete or search for a contact in the blacklist; even dial a contact number from the blacklist, but any incoming call from the blacklist will be rejected automatically.

To add a contact to the blacklist directory manually:

- **Tap ->Blacklist.**



- Tap .
- Enter the contact's name and the office, mobile or other numbers in the corresponding fields.



- Tap the **Account** field.
- Tap the desired account in the pop-up dialog box.

  If **Auto** is selected, the phone will use the default account when placing calls to the contact from the blacklist directory.

- Tap to accept the change or tap and then tap Discard to cancel.

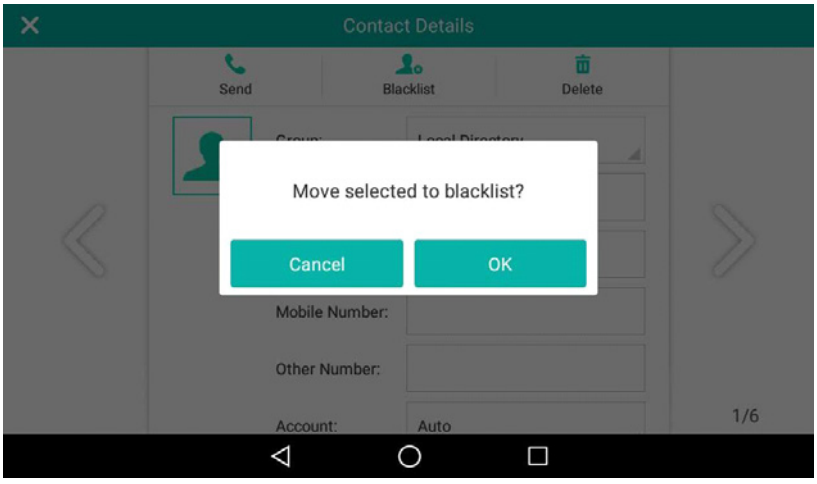
To add a contact to the blacklist directory from the local directory:

- Tap .
- Tap after the desired contact.

  If the contact was added to a specified contact group, you can tap the contact group first, and then tap after the desired contact.

- Tap **Blacklist**.

  The touch screen prompts the following warning**:**



- Tap **OK** to accept the change.

For more operating instructions on editing, deleting, placing calls to and/or searching for contacts in the blacklist, refer to the operating instructions of “Local Directory”.

Remote Phone Book

Remote phone book is a centrally maintained phone book, stored on the remote server. Users only need the access URL of the remote phone book. The IP phone can establish a connection with the remote server and download the phone book, and then display the remote phone book entries on the phone user interface.

You can add contacts to the local directory, search for a contact, or simply dial a contact number from the remote phone book.

You can configure your new phone to access up to 5 remote phone books. The phone supports up to 5000 remote phone book entries. Remote phone book can be customized.

Remote Phone book Name allows IP phones to search the entry names from the remote phone book for incoming/ outgoing calls. Search Flash Time specifies how often IP phones refresh the local cache of the remote phone book.

Procedure

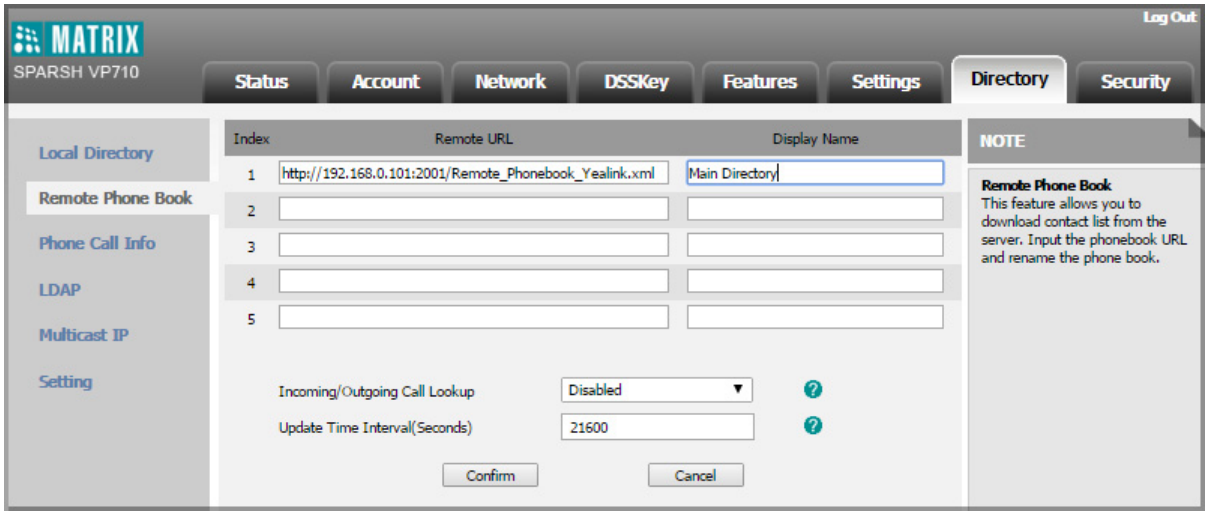
Remote phone book can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify the access URL and the display name of the remote phone book. Specify whether to query the entry name from the remote phone book for outgoing/incoming calls. Specify how often the IP phone refreshes the local cache of the remote phonebook. Specify whether to refresh the local cache of the remote phone book at a time when accessing the remote phone book. For more information, refer “Appendix D - Configuration Parameters”.

Local	Web User Interface	Specify the access URL of the remote phonebook. Specify whether to query the entry name from the remote phone book for outgoing/incoming calls. Specify how often the IP phone refreshes the local cache of the remote phonebook. **Navigate to**: http://<phoneIPAddress>/ servlet?m=m od_data&p=contacts-remote&q=load

To configure an access URL for remote phone book via web user interface:

- Click on **Directory->Remote Phone Book**.
- Enter the access URL in the **Remote URL** field.
- Enter the name in the **Display Name** field.



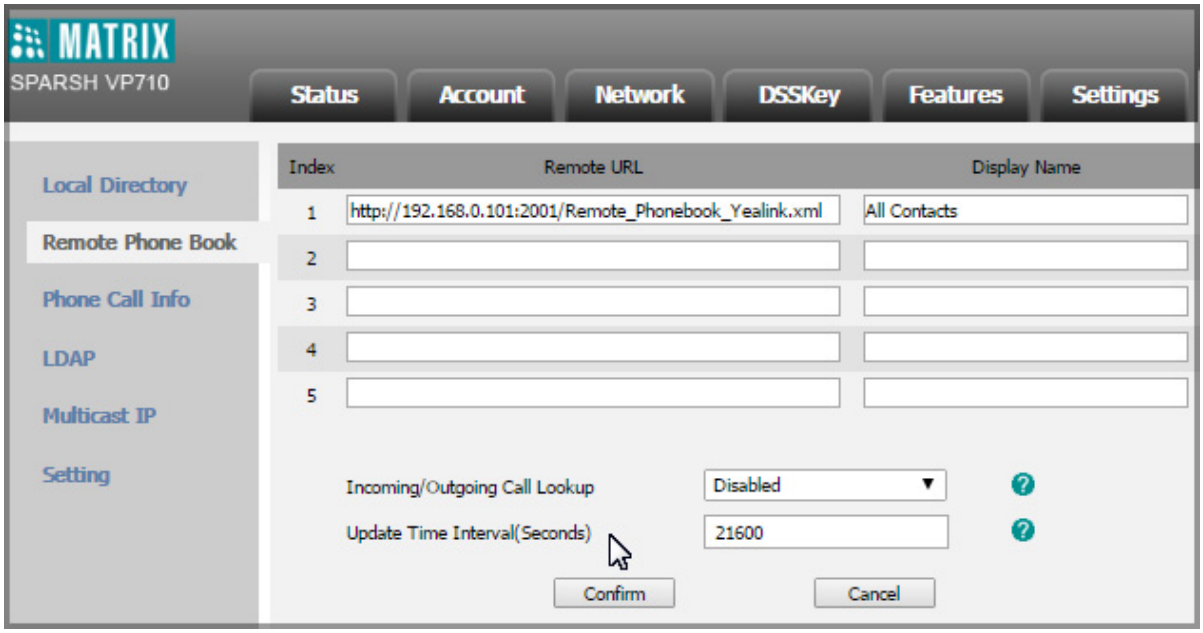
- Click **Confirm** to save the change.

An access URL for remote phone book is configurable via web user interface only.

To configure incoming/outgoing call lookup and update time interval via web user interface:

You can enable the phone to present the caller/callee identity stored in the remote phone book when receiving/ placing a call.

- Click on **Directory->Remote Phone Book.**
- Select the desired value from the **Incoming/Outgoing Call Lookup** list.
- Enter the desired time in the **Update Time Interval(Seconds)** field.

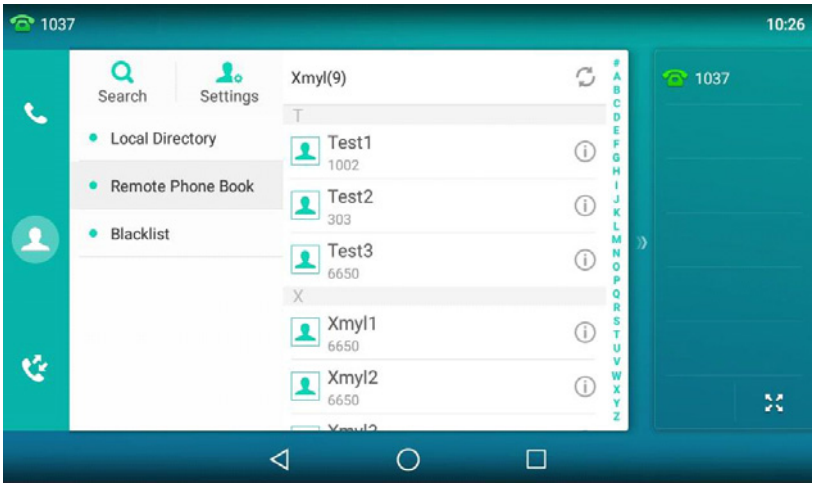


- Click **Confirm** to save the change.

To access your remote phone book via phone user interface:

- Tap  ->**Remote Phone Book**.
- Tap the desired remote phone book.

The phone then connects to the remote phone book and proceeds to load it. The contacts in the remote phone book are displayed on the touch screen.



- You can tap ◁ to go back to the idle screen.

Mobile Contacts

The first time your IP phone pairs and connects to your Bluetooth-enabled mobile phone, you will be asked if you would like to sync phone contacts temporarily. If allowed, the phone contacts will be imported to the mobile contacts directory of your IP phone.

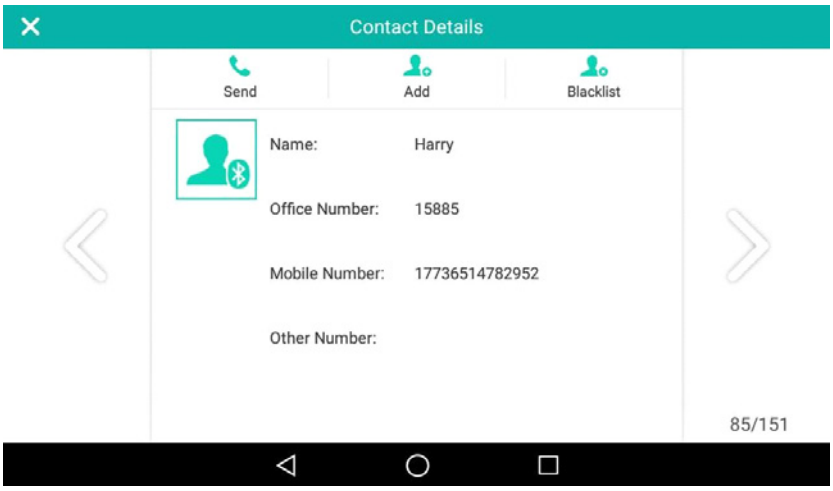
You can add a contact to the local directory or blacklist directory from mobile contacts. You cannot add a contact to the mobile contacts directory on your IP phone.

For operating instructions on placing calls to and/or searching for contacts in the mobile contacts directory, refer to the operating instructions of “Placing Calls to Contacts” and/or Searching for Contacts.

To add a contact to the local directory from mobile contacts directory:

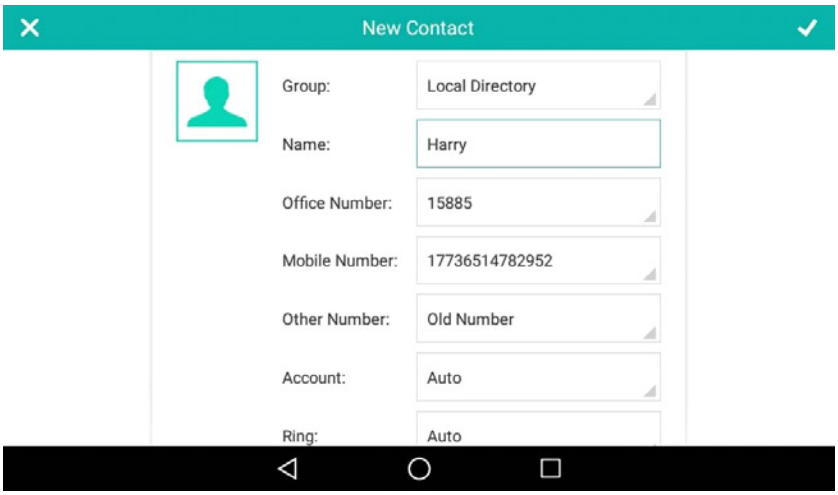
- Tap **->Mobile Contacts.**
- Drag up and down to view the list of Mobile Contacts.

- Tap ⓘ after the desired entry.



- Tap **Add**.

  Edit the corresponding fields.



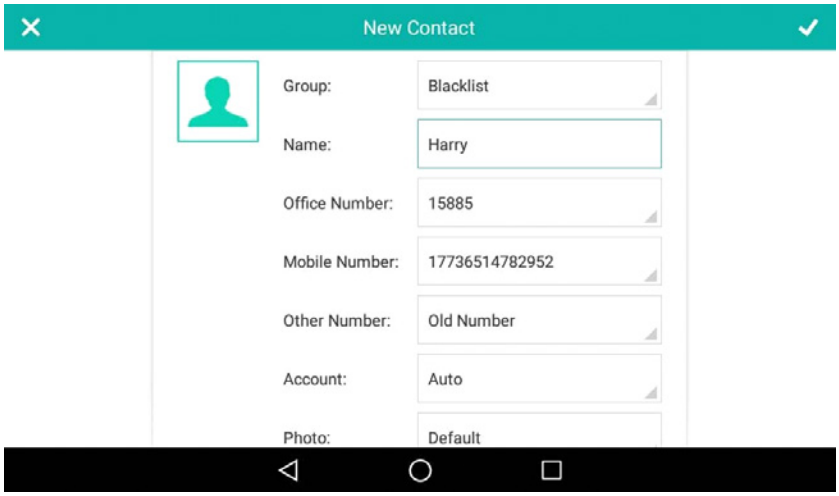
- Tap ✓ to accept the change or ◁ and then tap **Discard** to cancel.

  The entry is successfully saved to the local directory.

To add a contact to the blacklist directory from the mobile contacts directory:

- Tap 👤 **->Mobile Contacts.**
- Tap ⓘ after the desired contact.
- Tap **Blacklist**.

Edit the corresponding fields.

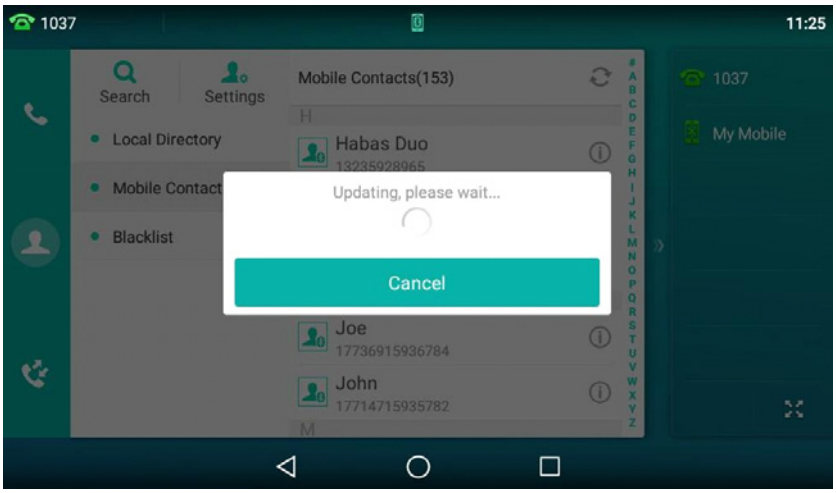


- Tap ✓ to accept the change or ◁ and then tap **Discard** to cancel.

  The entry is successfully saved to the blacklist directory

To update the mobile contacts directory:

- Tap **->Mobile Contacts.**
- Tap. .
- The touch screen will prompt “Updating, please wait…”, you need to authorize the IP phone to sync the contacts temporarily on the mobile phone.



After updating, the contacts you add, edit, delete on the mobile phone will sync to the mobile contacts directory.

If you disconnect the mobile phone from IP phone while the IP phone is updating the mobile phone book. The touch screen will prompt “Fail to download mobile contacts!”, and the mobile contacts directory will disappear from the IP phone directory list.

Call History Management

Call Log

Call log contains call information such as remote party identification, time and date, and call duration. It can be used to redial previous outgoing calls, return incoming calls, and save contact information from call log lists to the contact directory.

IP phones maintain a local call log. Call log consists of four lists: Placed Calls, Received Calls, Missed Calls and Forwarded Calls. Each Call log list supports 100 entries. History record feature is enabled by default. If you don't want to save the call history, you can disable the feature.



For specifically Missed Calls, you must separately configure an option from the web interface. Refer "Missed Call Log".

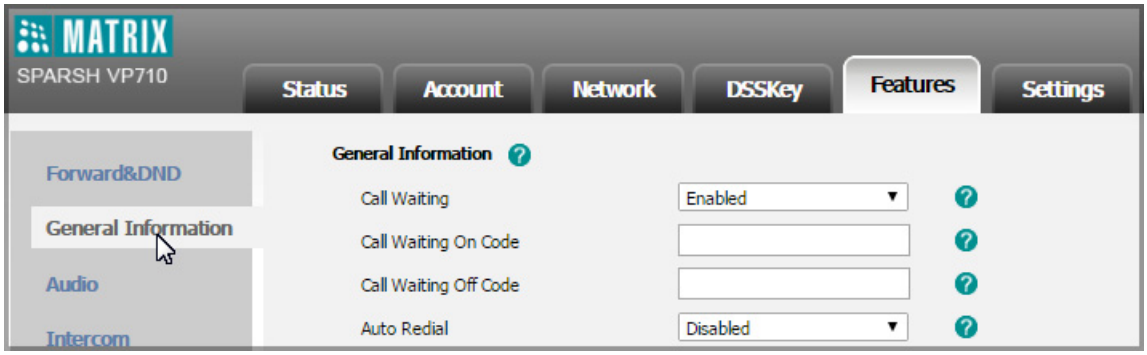
Procedure

Call log can be configured using the configuration files or locally.

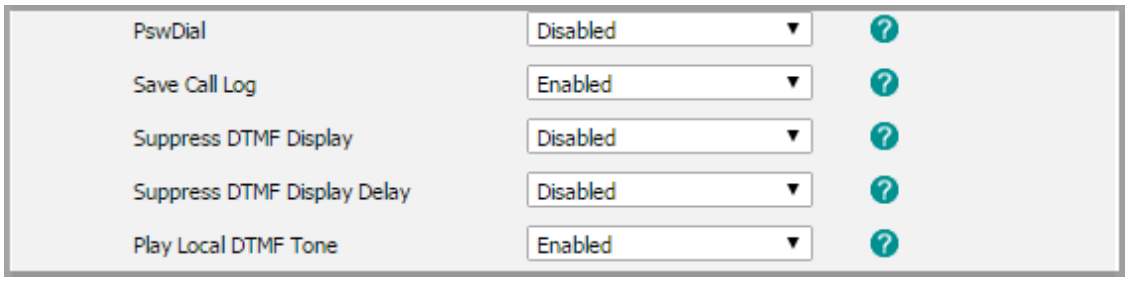
Configuration File	<MAC>.cfg	Configure call log feature. Configure call list show number. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure call log feature. Configure call list show number. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure the call log. Configure call list show number.

To configure call log feature via web user interface:

- Click on **Features->General Information**.



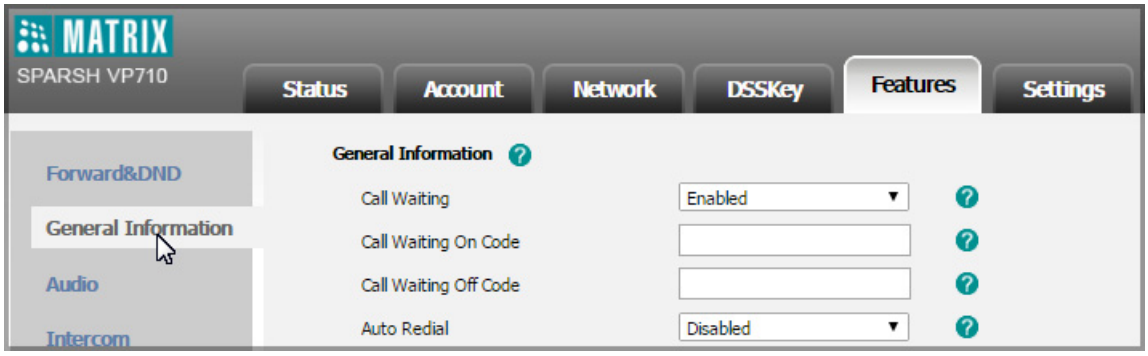
- Select the desired option from the **Save Call Log** list.



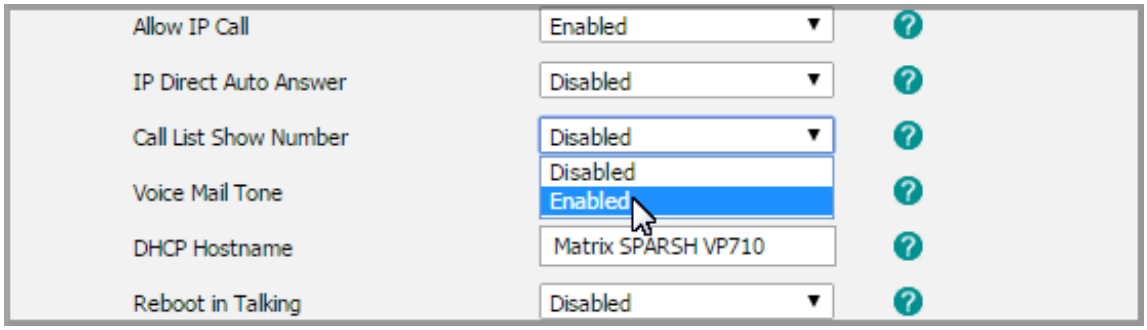
- Click **Confirm** to save the change.

To view Number instead of Name in the call history:

- Click on **Features->General Information**.



- Select **Enable** from the **Call List Show Number** list.

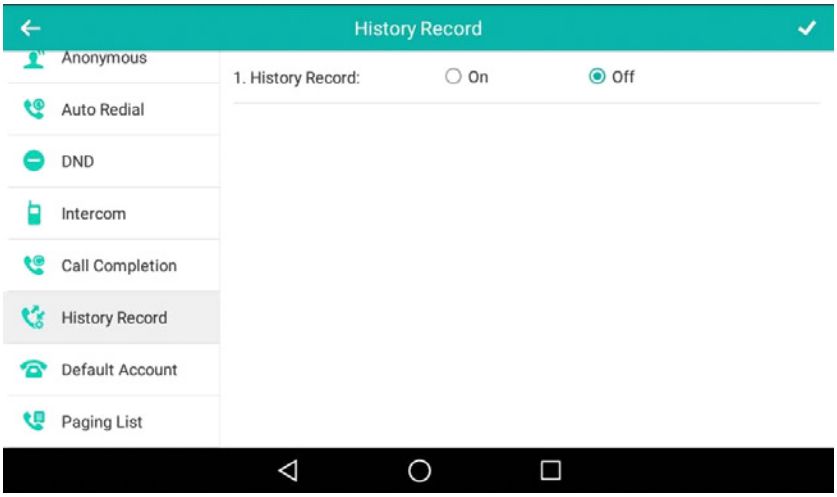


- Click **Confirm** to save the change.

  Enabling this parameter will display numbers instead of the configured names (if available) in the call history list when accessed from the IP phone.

To enable history record via phone user interface:

- Tap **Settings->Features->History Setting.**
- Tap the **On** radio box in the **History Record** field.

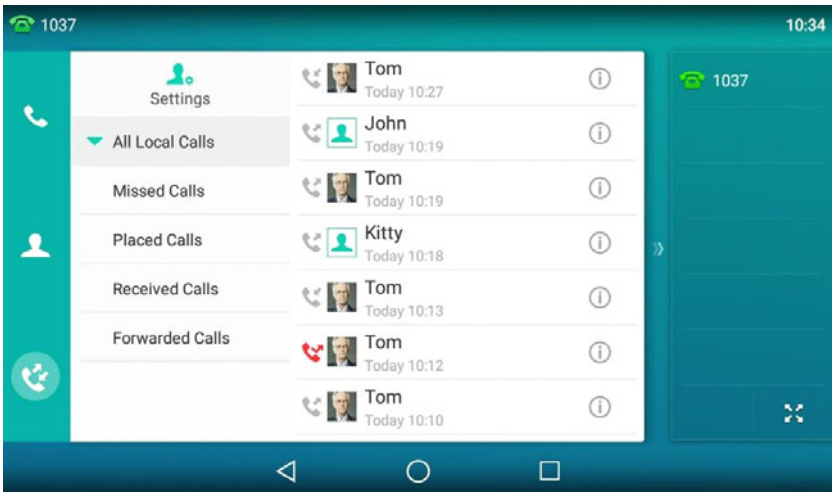


- Tap ✓ to **accept** the change.

To view call history:

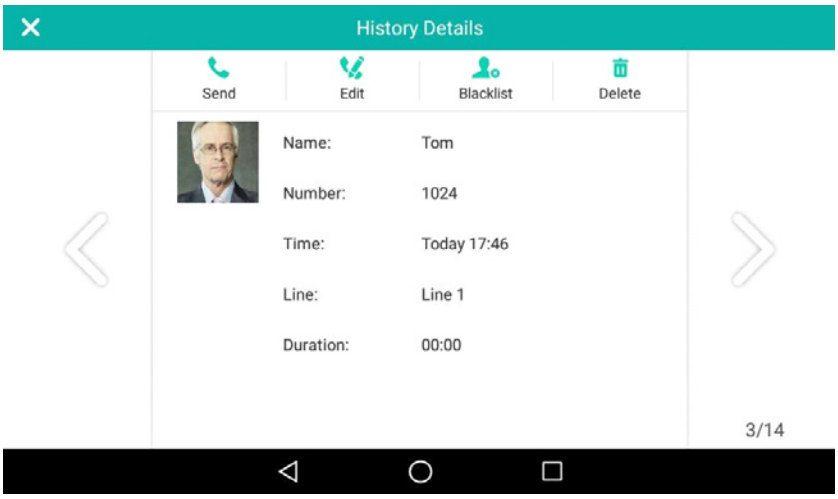
- Tap .

  The touch screen displays all call records.



- Drag up and down to scroll.You can tap **Missed Calls**, **Placed Calls**, **Received Calls** or **Forwarded Calls** to view entries in each call list directly.
- Tap after the desired entry.

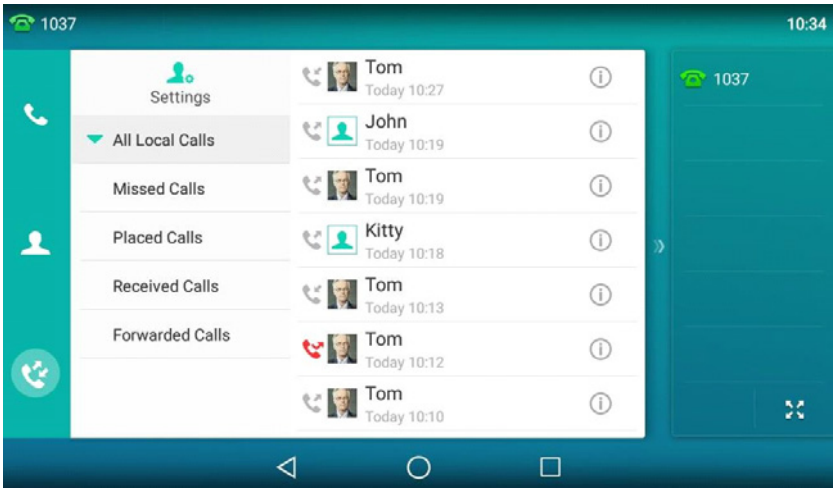
  The detailed information of the entry appears on the touch screen.



To place a call from the call history list:

- Tap .

  The touch screen displays all call records.

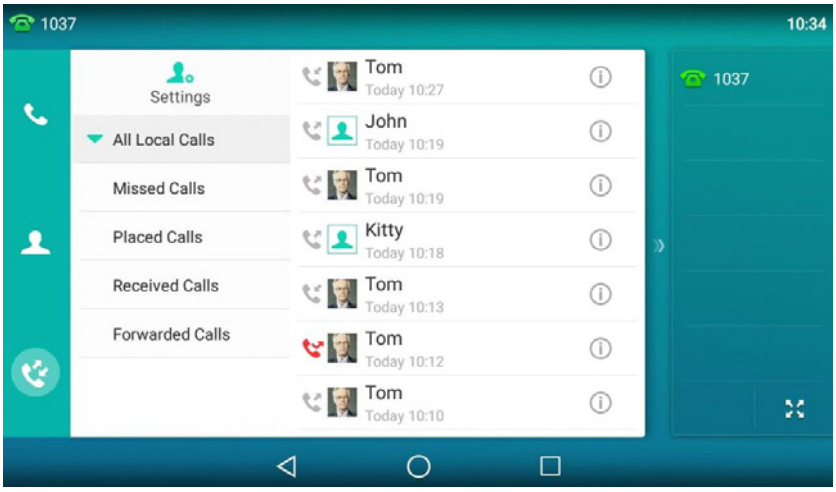


- Drag up and down to scroll.You can tap **Missed Calls, Placed Calls, Received Calls or Forwarded Calls** to view entries in each call list directly.
- Tap the desired entry.

To add a contact to the blacklist directory from the call history list:

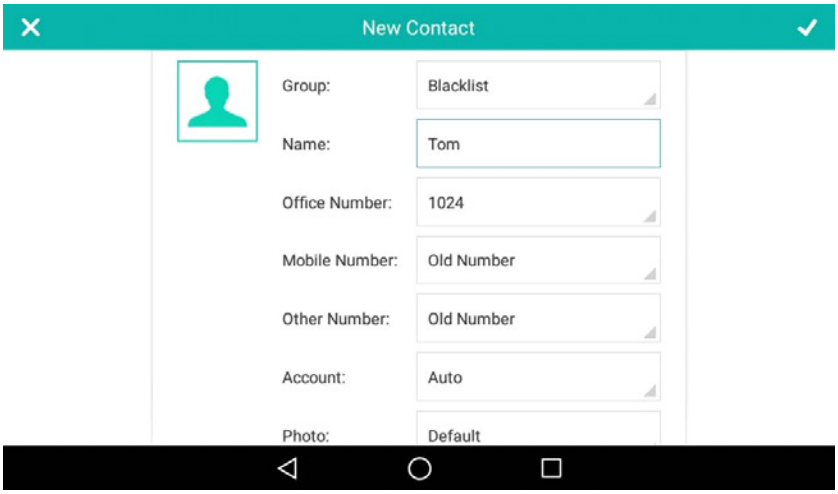
- Tap .

  The touch screen displays all call records.



- Drag up and down to scroll.You can tap **Missed Calls, Placed Calls, Received Calls or Forwarded Calls** to view entries in each call list directly.
- Tap after the desired entry.

- Tap **Blacklist**.



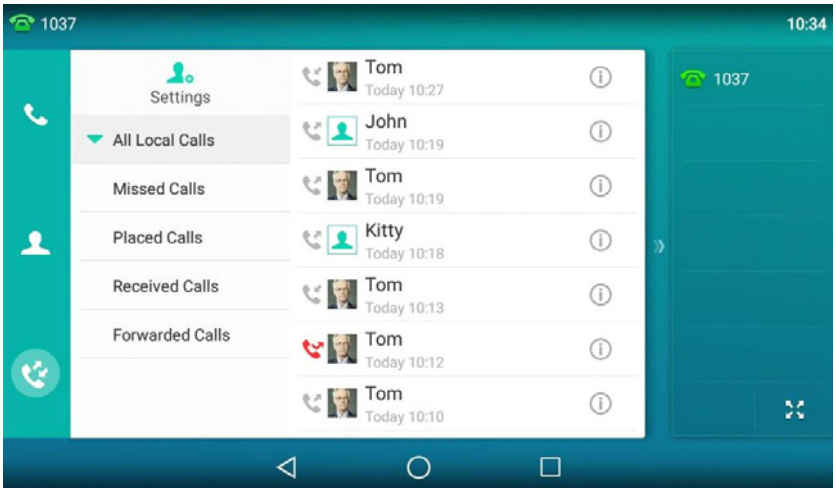
  Enter the desired values in the corresponding fields.

- Tap ✓ to accept the change or ◁ and then tap **Discard** to cancel. For more information, refer to "Blacklist"

To delete an entry from the call history list:

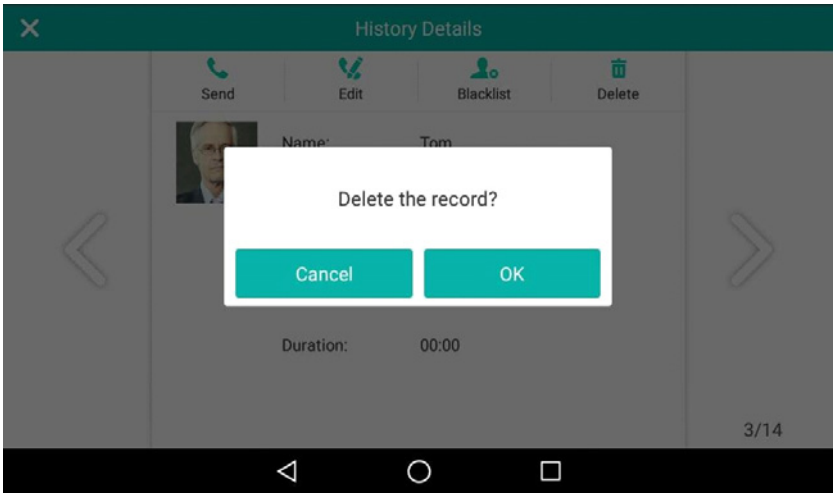
- Tap .

  The touch screen displays all call records.



- Drag up and down to scroll.You can tap **Missed Calls, Placed Calls, Received Calls or Forwarded Calls** to view entries in each call list directly.
- Tap ⓘ after the desired entry.
- Tap **Delete**.

The touch screen prompts the following warning:

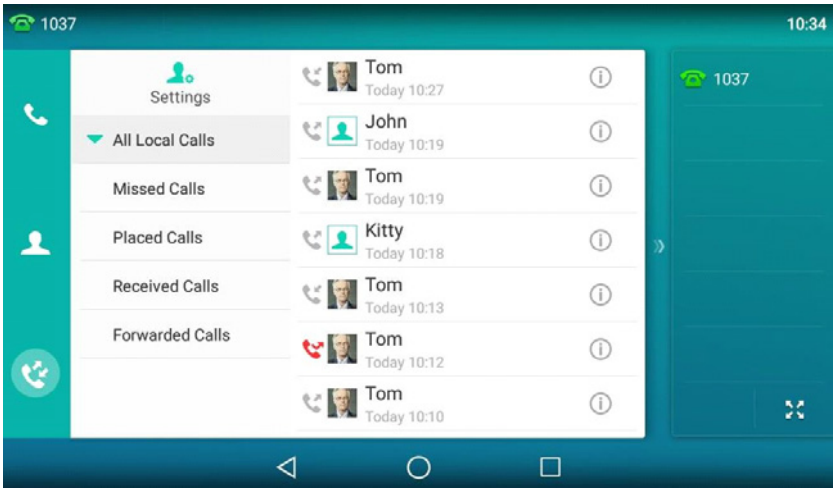


- Tap **OK** to confirm the deletion or **Cancel** to cancel.

To delete all entries from the call history list:

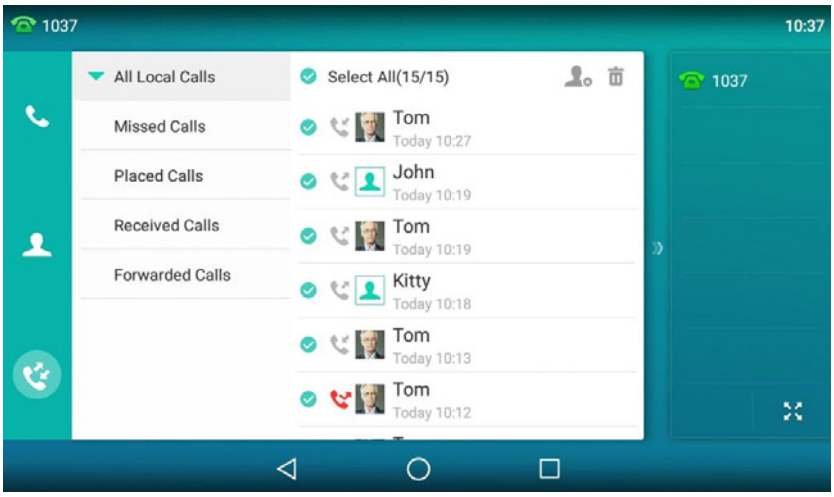
- Tap .

  The touch screen displays all call records.

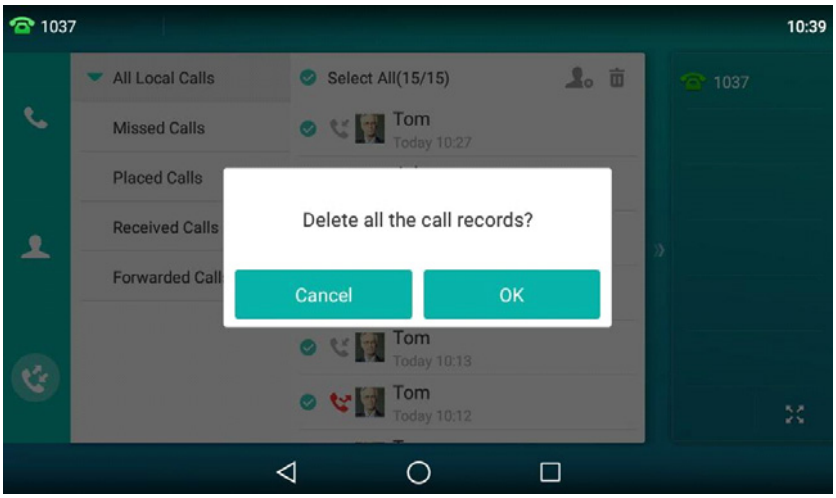


- Drag up and down to scroll. You can tap **Missed Calls, Placed Calls, Received Calls or Forwarded Calls** to view entries in each call list directly.
- Tap **Settings**.

- Tap the **Select All** radio box.



- Tap  .
- The touch screen prompts the following warning:

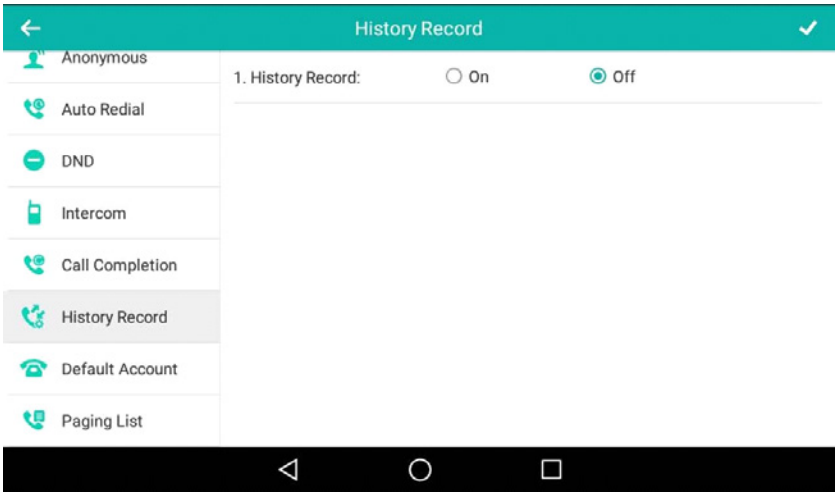


- Tap **OK** to confirm the deletion or **Cancel** to cancel.

To disable history record via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->History Record.**

- Tap the **Off** radio box in the **History Record** field.



- Tap ✓ to accept the change or ◁ to cancel.

Missed Call Log

Missed call log allows the IP phone to display the number of missed calls with an indicator icon on the idle screen, and to log missed calls in the Missed Calls list when the IP phone misses calls. Once the user accesses the Missed Calls list, the prompt message and indicator icon on the idle screen disappear.

You can configure whether to display a prompt message when missing calls. For more information, refer to "Notification Popups".

Procedure

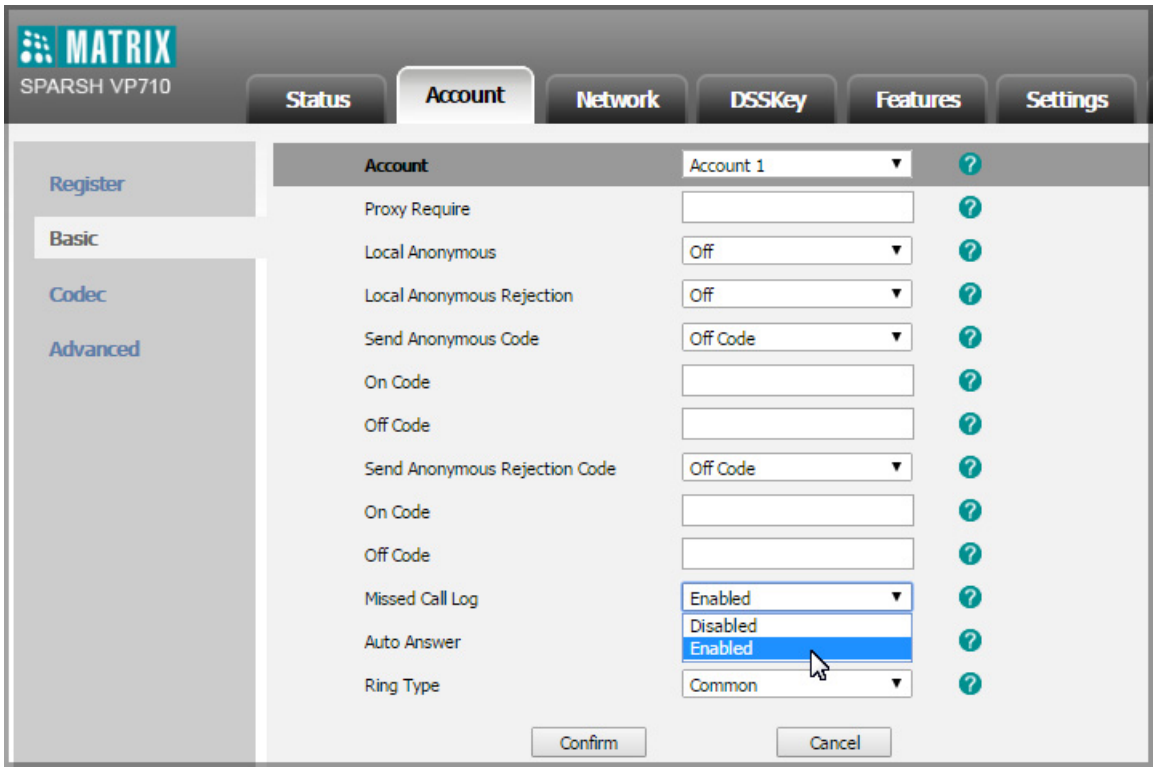
Missed call log can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure missed call log feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure missed call log feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-basic&q=load&acc=0

To configure missed call log via web user interface:

- Click on **Account-> Basic**.
- Select the desired account from the **Account** list.

- Select the desired value from the **Missed Call Log** list.



- Click **Confirm** to save the change.

System Customization

DSS Keys

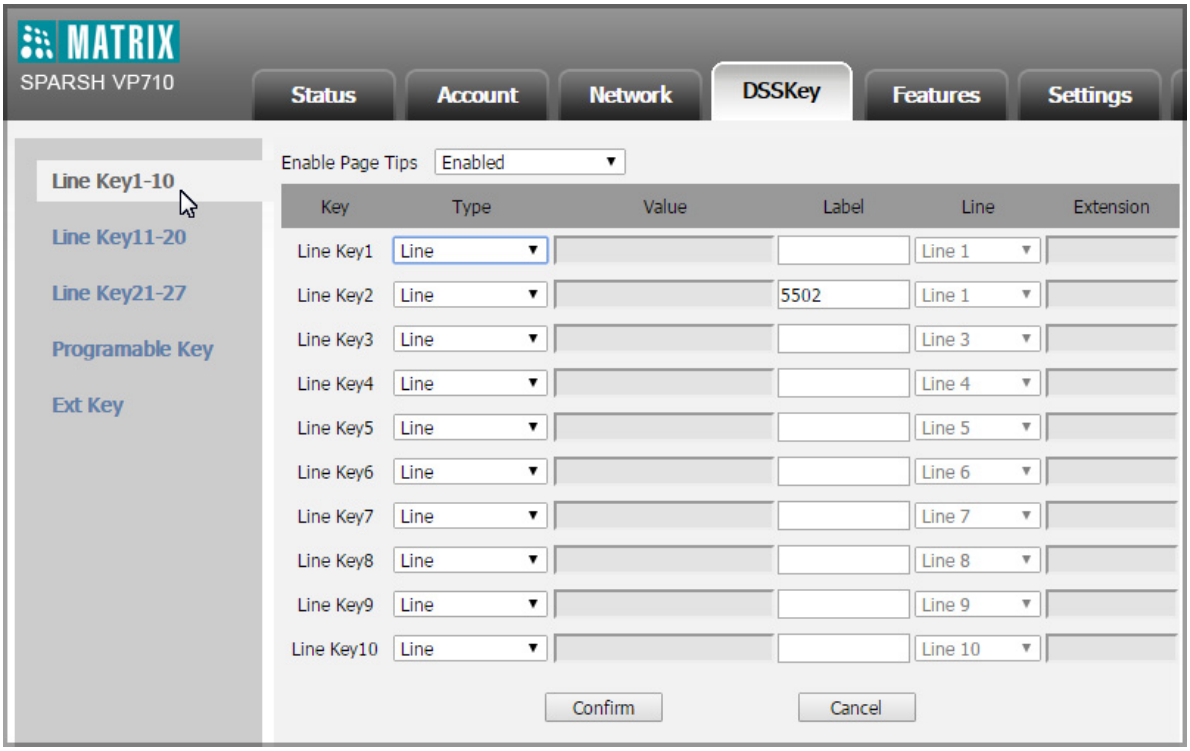
There are two types of DSS keys: Line Keys and Programable Keys. Details will be introduced in the following. The IP phone supports 27 line keys and 3 programable keys.

Line Keys

You can assign predefined functionalities to line keys. You can also define a label for a line key feature which will appear on the touch screen. Line keys allow you to quickly access features such as recall and voice mail. The line key can indicate the monitored status when the line keys are assigned with particular features, such as BLF. The default key type of line key 1-16 is Line. The default key type of line key 17-27 is N/A, which indicates that this line key provides no functionality until configuration.

To customize the line keys via web user interface:

- Click on **DSSKey->Line Key**.
- Customize specific features or assign a specific key for an account as per your requirement.

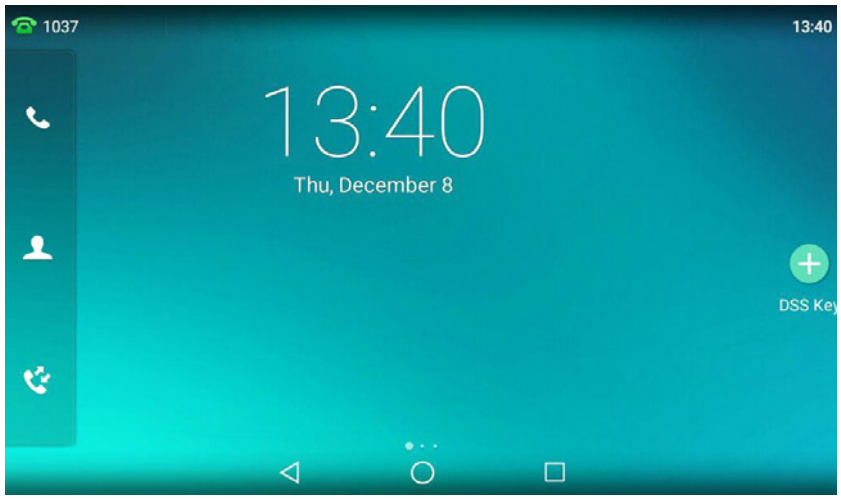


- Enter a string that will appear on the touch screen in the **Label** field (Optional).
- Click **Confirm** to save the change.

Assigning Functionality to a Line Key

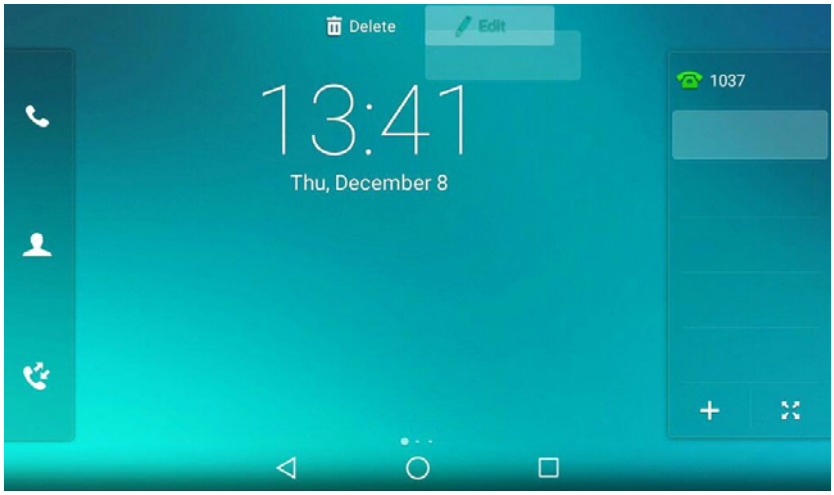
To assign functionality to a line key via phone user interface:

- Do one of the following:
  - Tap  on the home screen, when there is no line key configured on your phone.



OR

- Drag up and down to scroll through the line key list if required.
- Drag the line key to the **Edit** field.

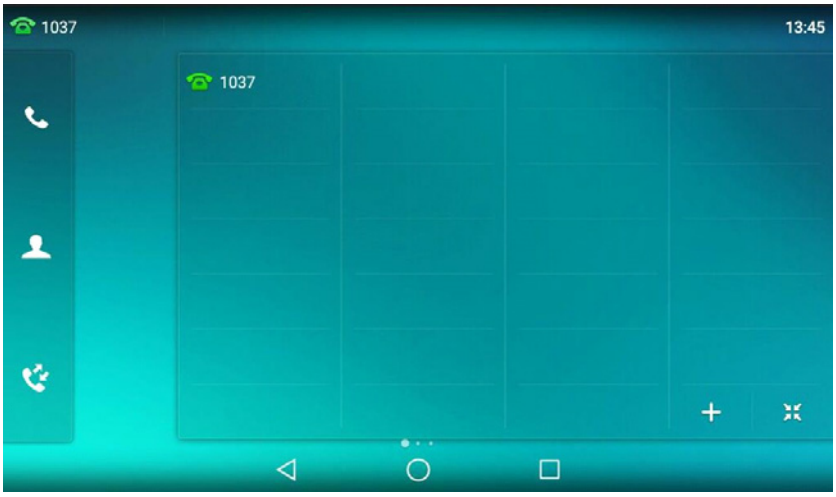


OR

- Tap on the bottom-left of the line key list.

OR

- Tap on the bottom-right of the line key list. The touch screen will display all the line keys.



- Drag the line key to the Edit field.

  OR

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Tap the desired line key.

- Select the desired key type from the **Type** field.
- Select the desired key event type from the **Key Type** field (Optional).
- Select the desired line from the **Account ID** field (Optional).
- Enter the string that will appear on the touch screen in the **Label** field (Optional).
- Enter the corresponding value in the **Value** field (Optional).
- Enter the corresponding value in the **Extension** field (Optional)
- Tap to accept the change or to cancel.

Changing the Location of the Line Keys

You can change the line key's location to a blank area on the Line keys field. You can also change the location of two line keys. And the operating instructions are almost the same. The following take how to change the location of two line keys as example.

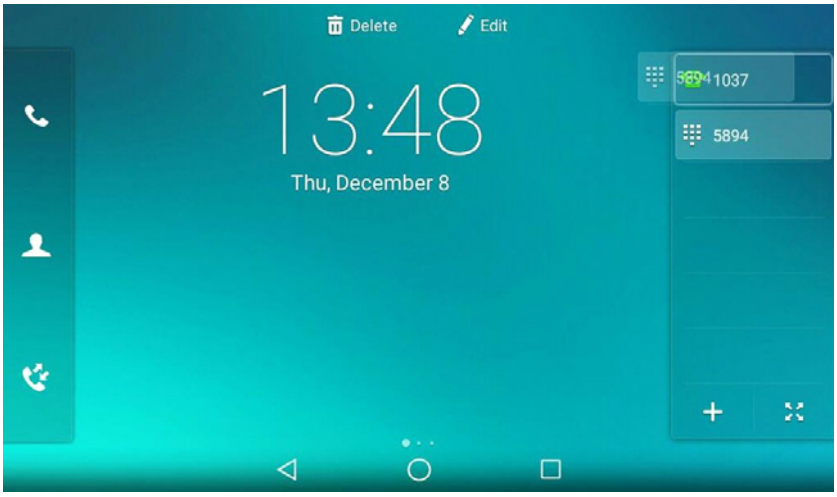
To change the locations of two line keys:

- Do one of the following:
  - Drag up and down to scroll through the line keys list if required.

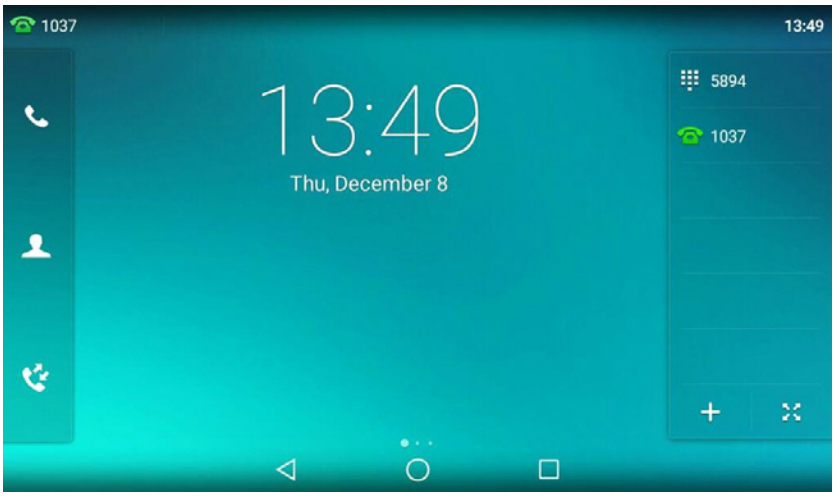
    OR

  - Tap on the bottom-right of the line key list.

- Drag a line key to the desired location at which another line key locates. For example, exchange the locations of line key 1 and line key 2:.



- The following shows the locations of two line keys after exchange.



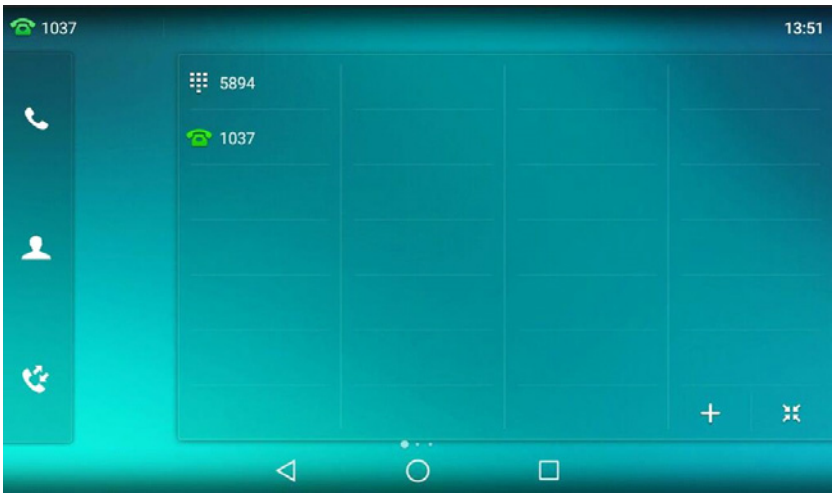
Deleting a Line Key

To delete a line key:

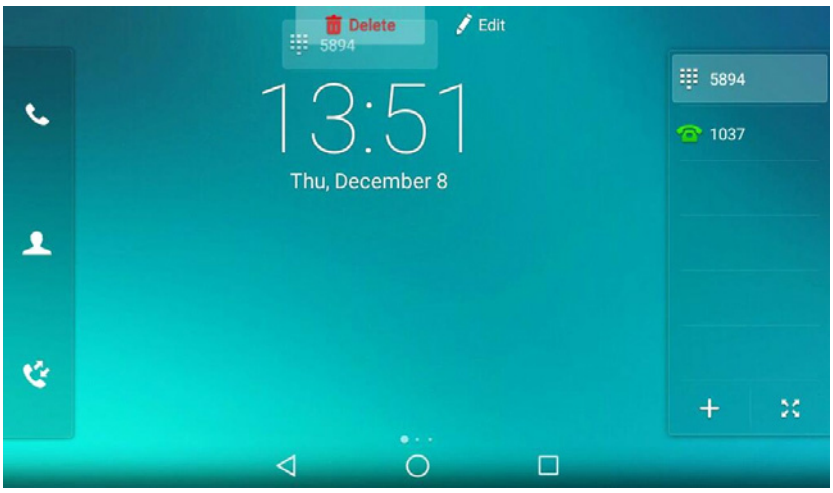
- Do one of the following:
  - Drag up and down to scroll through the line keys if required.

    OR

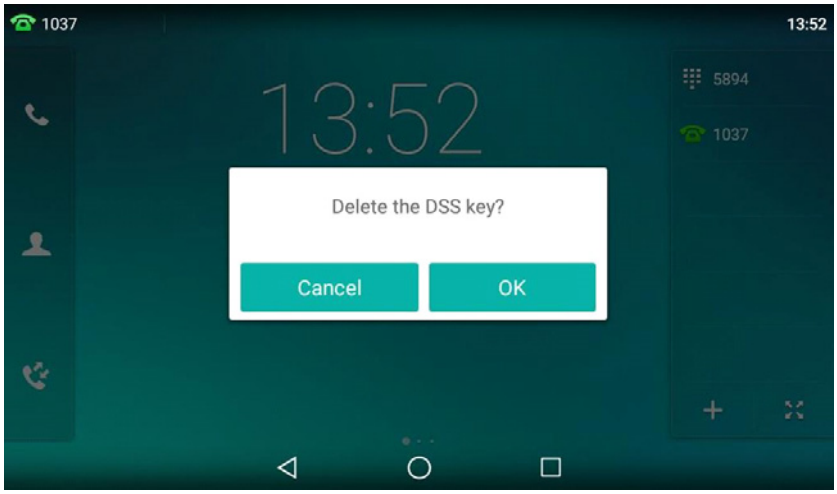
  - Tap on the bottom-right of the line key list. The touch screen will display all the line keys.



- Drag the desired line key to the **Delete** field.



- The touch screen will prompt **“Delete the DSS key?**”.



- Tap **OK** to accept the change or **Cancel** to cancel.

Line Key Features

Line key features are explained below:

- Line
- Speed Dial
- Voice Mail
- Direct Pickup
- Group Pickup
- DTMF
- Prefix
- Local Group
- XML Group
- LDAP
- Conference
- Forward
- Transfer
- Hold
- DND
- Group Listening
- Zero Touch
- URL
- Phone Lock
- Directory
- Mobile Account

Line

You can use this key feature to accept incoming calls, place active calls on hold or resume a held call.

Dependencies: Type (Line)
Account ID (the account this feature will be applied to)
Label (key label displayed on the touch screen)
Usage: When the phone receives an incoming call,

- Tap the line key to accept the incoming call.
- Tap the line key to place a new call and the active call is placed on hold.
- Tap the line key again to resume the held call.

Speed Dial

You can use this key feature to speed up dialing the numbers frequently used/hard to remember or perform blind transfer conveniently during the call.

Dependencies: Type (Speed Dial)

Account ID (the account this feature will be applied to)
Label (key label displayed on the touch screen)
Value (the number you want to dial out/transfer to)
Usage:

- When the IP phone is idle, tap the speed dial key to dial out the number specified in the **Value** field, using the account selected from the **Account ID** field.
- When there is an active call on the line selected from the **Account ID** field, drag the far-site window to the speed dial key field to perform blind transfer to the number that specified in the **Value** field.

Voice Mail

You can use this key feature to quickly connect voice mail. For more information, refer to Message.

Dependencies: Type (Key Event)
Key Type (Voice Mail)
Account ID (the account this feature will be applied to)
Label (key label displayed on the touch screen)
Value (the voice mail access code)
Usage: Tap the voice mail key to dial out the voice mail access code. Then follow the voice prompt to listen to the voice mails.

Direct Pickup

You can use this key feature to answer someone else's incoming call on the phone.

Dependencies: Type (Key Event)
Key Type (DPickup)
Account ID (the account this feature will be applied to)
Label (key label displayed on the touch screen)
Value (the directed call pickup code followed by the target phone number)
Usage: Tap the direct pickup key on your phone when the target phone number receives an incoming call. The call is then answered on your phone.

Group Pickup

You can use this key feature to answer incoming calls in a group that is associated with their own group.

Dependencies: Type (Key Event)
Key Type (GPickup)
Account ID (the account this feature will be applied to)
Label (key label displayed on the touch screen)
Value (the group pickup code)
Usage: Tap the group pickup key on your phone when a phone number in the group receives an incoming call. The call is answered on your phone.

DTMF

You can use this key feature to send the specification of arbitrary key sequences via DTMF.

Dependencies: Type (Key Event)
Key Type (DTMF)
Label (key label displayed on the touch screen)
Value (DTMF sequence)
Usage: Tap the DTMF key during an active call to send the key sequence specified in the **Value** field.



DTMF sequence can only contain “0-9”, “”, “#” and “A-D”.*

Prefix

You can use this key feature to add a specified prefix number before the dialing number.

Dependencies: Type (Key Event)
Key Type (Prefix)
Label (key label displayed on the touch screen)
Value (the prefix number)
Usage: Tap the prefix key when the phone is idle, the phone will then enter the dialing screen and display the prefix number that you specified in the **Value** field. You can enter the remaining digits and then dial out.

Local Group

You can use this key feature to quickly access a contact group in the local directory. For more information, refer to “Local Directory”
Dependencies: Type (Key Event)
Key Type (Local Group)
Local Group (the contact group name you want to access)
Label (key label displayed on the touch screen)
Usage: Tap the local group key to access the contact group specified in the **Local Group** field.

XML Group

You can use this key feature to quickly access a remote group in your remote phone book. You should configure a remote phone book in advance. For more information, refer to “Remote Phone Book”.
Dependencies: Type (Key Event)
Key Type (XML Group)
PhoneBook Name (the remote group name you want to access if the remote phone book is configured)
Label (key label displayed on the touch screen)
Usage: Tap the XML group key to access the remote group specified in the **XML Group** field.

LDAP

You can use this key feature to quickly access a LDAP search screen.

Dependencies: Type (Key Event)
Key Type (LDAP)
Label (key label displayed on the touch screen)
Usage:

- Tap the LDAP key to access the LDAP search screen.
- Enter a few continuous characters of the contact name or continuous numbers of the contact number.

  The contacts whose name or phone number matches the characters entered will appear on the touch screen.

Conference

You can use this key feature to set up a conference call. For more information, refer to “Conference”.

Dependencies: Type (Key Event)
Key Type (Conference)
Label (key label displayed on the touch screen)
Value (the number added to the conference)

Usage: Tap the conference key during an active call to set up a conference with the number specified in the **Value** field.



If the Value field is left blank, the conference key performs the same function as Conference during a call.

Forward

You can use this key feature to forward an incoming call to someone else. For more information, refer to Call Forward.

Dependencies: Type (Key Event)
Key Type (FWD)
Label (key label displayed on the touch screen)
Value (the number you want to forward to)
Usage: Tap the forward key to forward an incoming call to the number specified in the **Value** field.



If the Value field is left blank, the forward key performs the same function as the FWD soft key when receiving an incoming call.

Transfer

When there is an active call on the phone, you can use this key feature to handle the call differently depending on the transfer mode assigned to the DSS key.

Dependencies: Type (Key Event)
Key Type (Transfer)
Label (key label displayed on the touch screen)
Value (the number you want to transfer to)
Usage:

- When the transfer mode via Dsskey is **Blind Transfer**, tap the transfer key to complete the blind transfer to the number specified in the **Value** field.
- When the transfer mode via Dsskey is **Attended Transfer**, tap the transfer key to dial out the number specified in the **Value** field, and then perform the attended or semi-attended transfer.
- When the transfer mode via Dsskey is **New Call**, tap the transfer key to place a new call to the number specified in the **Value** field.



Transfer mode via Dsskey is configurable via web user interface at the path Features->Transfer->Transfer Mode via Dsskey.

*If the **Value** field is left blank, the transfer key performs the same function as the **TRANSFER** key or the **Transfer** soft key during a call. For more information, refer to "Call Transfer".*

Hold

You can use this key feature to place an active call on hold or retrieve a held call.

Dependencies: Type (Key Event)
Key Type (Hold)
Label (key label displayed on the touch screen)
Usage:

- Tap the hold key during an active call to place the call on hold.
- Tap the hold key again to retrieve the held call.

DND
You can use this key feature to enable or disable DND. You can also use this key feature to access the custom DND screen. For more information, refer to Do Not Disturb (DND).

Dependencies: Type (Key Event)
Key Type (DND)
Label (key label displayed on the touch screen)
Usage:
When DND is in phone mode:

- Tap the DND key to enable DND.
- Tap the DND key again to disable DND.

When DND is in custom mode:

- Tap the DND key to access the custom DND screen. You can enable or disable DND for one or all accounts.

Group Listening

You can use this key feature to activate the Speakerphone and Handset/Headset mode at the same time. It is suitable for group conversations which have more than one person at one side. You are able to speak and listen through the handset/headset, while the others nearby can only listen through the speaker.
Dependencies: Type (Key Event)
Key Type (Group Listening)
Label (key label displayed on the touch screen)
Usage:

- During a call, tap the group listening key to activate the group listening mode.
- You can then speak and listen through the handset/headset, while other people at your side can only listen through the speaker at the same time.
- Tap the group listening key again to deactivate the group listening mode.

Zero Touch

You can use this key feature to quickly configure auto provision and network parameters.

Dependencies: Type (Key Event)
Key Type (Zero Touch)
Label (key label displayed on the touch screen)
Usage:

- Tap the zero touch key to access the zero touch screen.
- Tap the **OK** soft key within a few seconds.
- Configure the network parameters in the corresponding fields.
- Tap the **Next** soft key.
- Configure the auto provision parameters in the corresponding fields.
- Tap the **Next** soft key.

  The phone will reboot to update configurations.

URL

You can use this key feature to trigger the phone to send an HTTP GET request containing a specific URL.

Dependencies: Type (URL)
Label (key label displayed on the touch screen)

URL (the URL contained in the HTTP GET request)
Usage: Tap the URL key to trigger the phone to send an HTTP GET request containing the URL specified in the URL field.

Phone Lock

You can use this key feature to immediately lock your phone instead of long pressing #SEND. For more information, refer to "Keypad Lock (Phone Lock)".
Dependencies: Type (Key Event)
Key Type (Phone Lock)
Label (key label displayed on the touch screen)
Usage: When the phone lock feature is enabled, tap the phone lock key to immediately lock your phone instead of long pressing #SEND.

Directory

You can use this key feature to easily access frequently used lists. For more information, refer to "Local Directory".

Dependencies: Type (Key Event)
Key Type (Directory)
Label (key label displayed on the touch screen)
Usage: Tap the directory key to immediately access frequently used lists.

The directory key performs the same function as the .

Mobile Account

You can use this key feature to accept incoming mobile calls, place active mobile calls on hold on your IP phone. For more information, refer to "Bluetooth-Enabled Mobile Phone".

Dependencies: Type (Mobile Account)
Label (key label displayed on the touch screen)
Usage: When the IP phone receives an incoming mobile call,

- Tap the mobile account key to accept the incoming call.
- Tap the mobile account key to place a new call and the active call is placed on hold.

Page Tips

Enable page tips feature allows users to enable the breathing light to indicate statuses. It is mainly used in the scenario of configuring multiple line keys (more than six).

The IP phone supports 27 line keys. The line keys display in the DSS key field at the right of the touch screen as a list. Only 6 line keys can display on one page. Users only need to drag up or down to scroll through the line keys. If enable page tips feature is enabled, the breathing light will appear at the top/bottom of the DSS key field when the status of particular feature (e.g., BLF) assigned to the line key on the non-current page changes.

The breathing light will flash red or green for different line key types:

Line Key Type	Color Type
Call Park	Red
Intercom	Red
Line	Green
BLF	Green

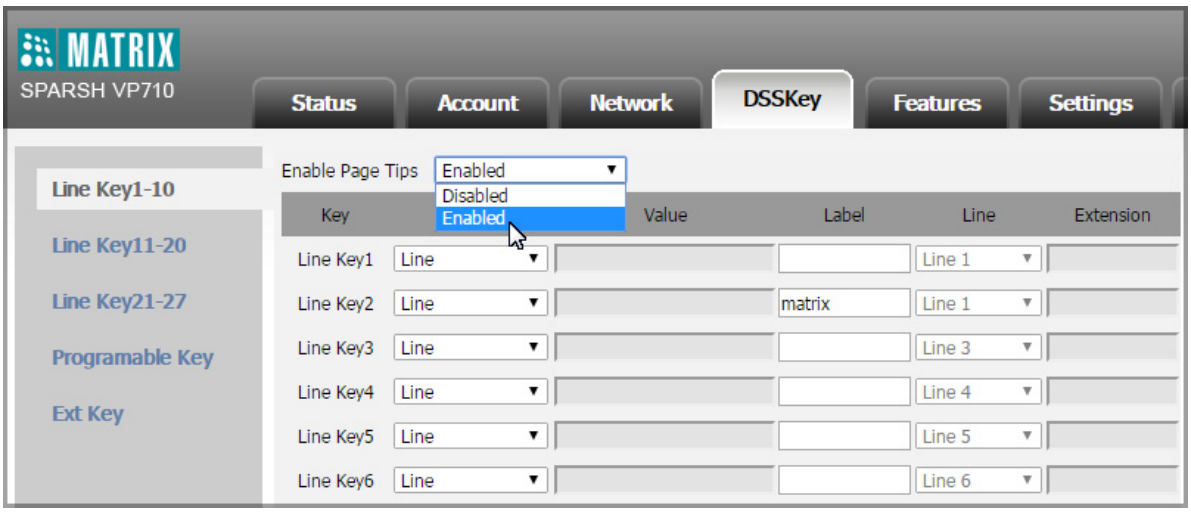
Procedure

Enable page tips can be configured locally only.

Local	Web User Interface	Configure enable page tips.. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load

To configure the page tips feature via web user interface:

- Click on **DSSKey->Line key.**
- Select Enabled from the **Enable Page Tips** list.



- Click **Confirm** to save the change.

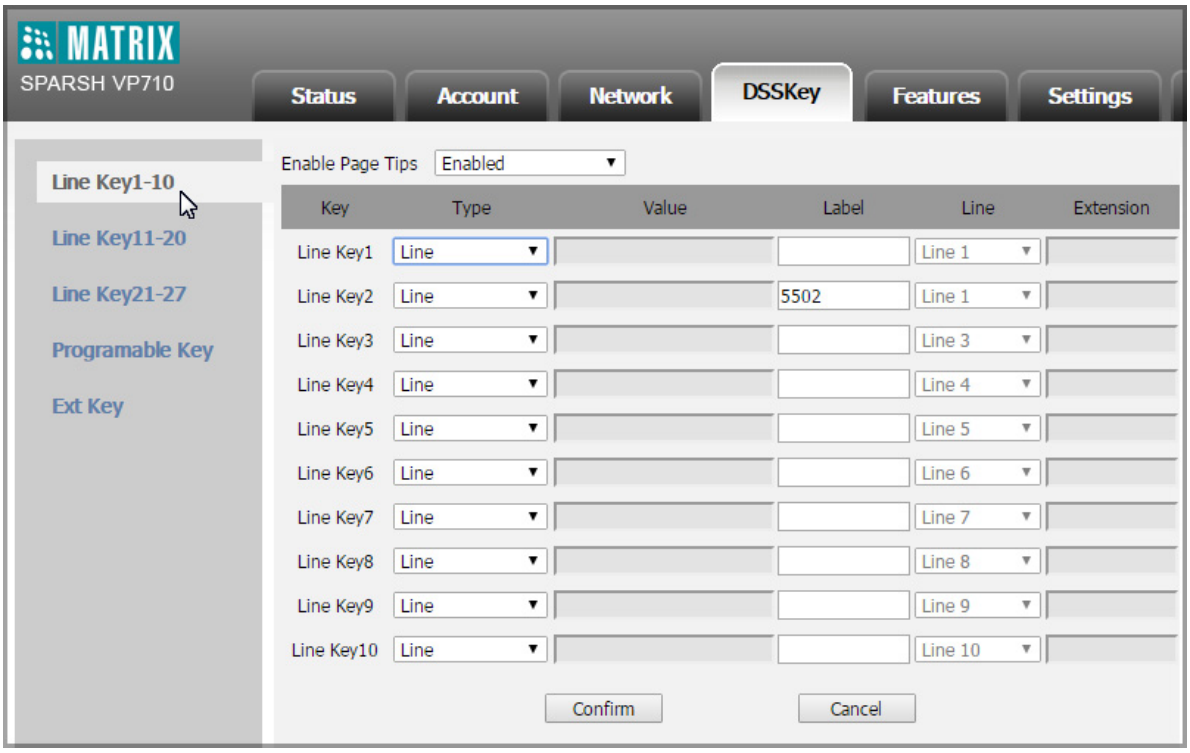
Page tips feature is configurable via web user interface only.

Programable Keys

You can customize the function keys.

To customize the programmable keys via web user interface:

- Click on **DSSKey->Programable Key.**
- Customize specific features for these keys.



- Click **Confirm** to save the change.

- You can click **Reset to default** to reset custom settings to defaults.

  Then you can press the keys on the phone to perform the features you configured.

Programmable keys are configurable via web user interface only.

Dial Plan

Regular expression, often called a pattern, is an expression that specifies a set of strings. A regular expression provides a concise and flexible means to “match” (specify and recognize) strings of text, such as particular characters, words, or patterns of characters. Regular expression is used by many text editors, utilities, and programming languages to search and manipulate text based on patterns.

Regular expression can be used to define IP phone dial plan. Dial plan is a string of characters that governs the way for IP phones to process the inputs received from the IP phone’s keypads. IP phones support the following dial plan features:

- Replace Rule

- Dial-now
- Area Code
- Block Out

The basic expression syntax you need to know:

.	The dot "." can be used as a placeholder or multiple placeholders for any string. Example: "12." would match "12**3**","12**34**","12**345**","12**abc**", etc.
x	The "x" can be used as a placeholder for any character. Example: "12x" would match "12**1**","12**2**","12**3**","12**a**", etc.
-	The dash "-" can be used to match a range of characters within the brackets. Example: "[5-7]" would match the number "**5**", "**6**" or "**7**".
,	The comma "," can be used as a separator within the bracket. Example: "[2,5,8]" would match the number "**2**", "**5**" or "**8**".
[]	The square bracket "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234"would match "91**5**1234", "91**6**1234","91**7**1234".
()	The parenthesis "( )" can be used to group together patterns, for instance, to logically combine two or more patterns. Example: "([1-9])([2-7])3" would match "**92**3", "**15**3", "**67**3", etc.
$	The "$" followed by the sequence number of a parenthesis means the characters placed in the parenthesis. The sequence number stands for the corresponding parenthesis. Example: A replace rule configuration, Prefix: "001(xxx)45(xx)", Replace: "9001$145$2". When you dial out "0012354599" on your phone, the IP phone will replace the number with "90012354599". "$1" means 3 digits in the first parenthesis, that is, "235". "$2" means 2 digits in the second parenthesis, that is, "99".

Replace Rule

You can configure one or more replace rules (up to 100) to remove the specified string and replace it with another string. You can configure a pattern with wild-cards (refer to the expression syntax in the table above), so that any string that matches the pattern will be replaced. This feature is convenient for you to dial out a long number. For example, a replace rule is configured as "Prefix: 1" and "Replace: 1234", when you try to dial out the number "1234", you just need to enter "1" on the phone and then tap Send.key.

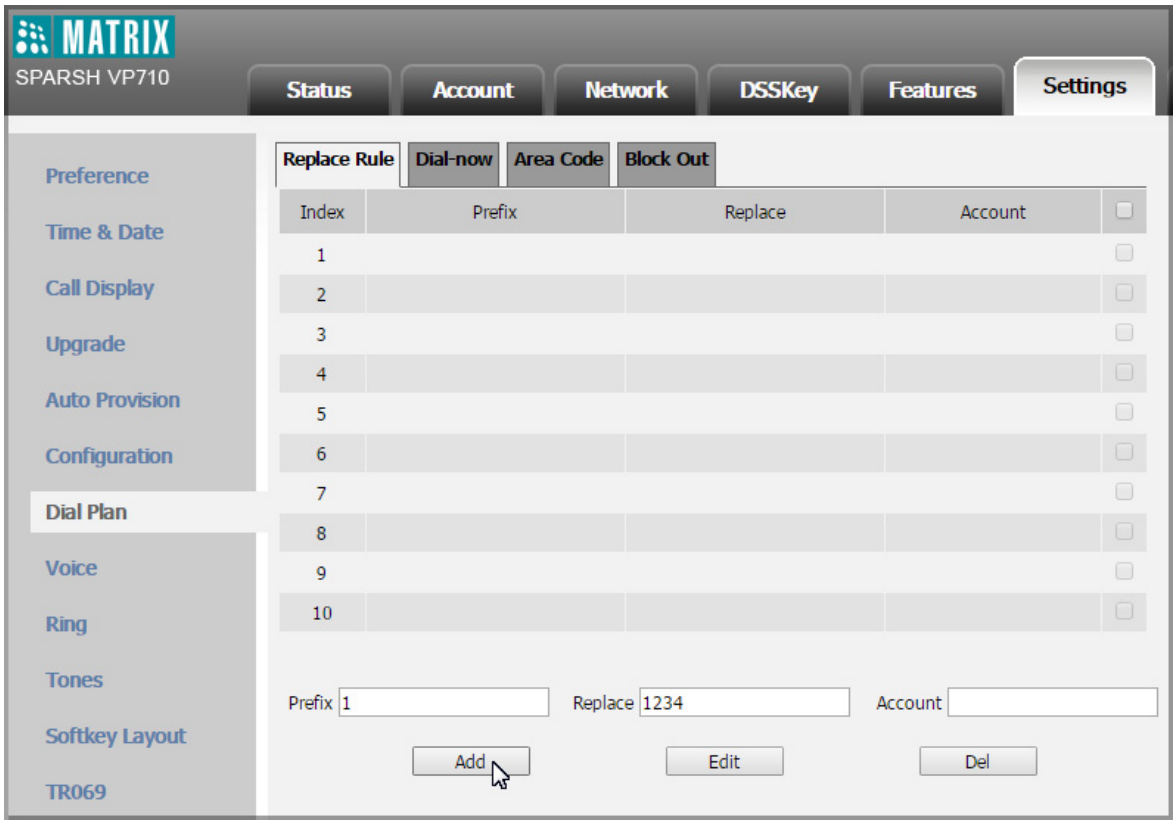
Procedure

Replace rule can be created using the configuration files or locally.

Configuration File	<MAC>.cfg	Create the replace rule for the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Create the replace rule for the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-dialplan&q=load

To add a replace rule via web user interface:

- Click on **Settings->Dial Plan->Replace Rule**.
- Enter the string (e.g.,1) in the **Prefix** field.
- Enter the string (e.g., 1234) in the **Replace** field.
- Enter the desired line ID in the **Account** field or leave it blank.
- If you leave this field blank or enter 0, the replace rule will apply to all accounts on the IP phone.



- Click **Add** to add the replace rule.

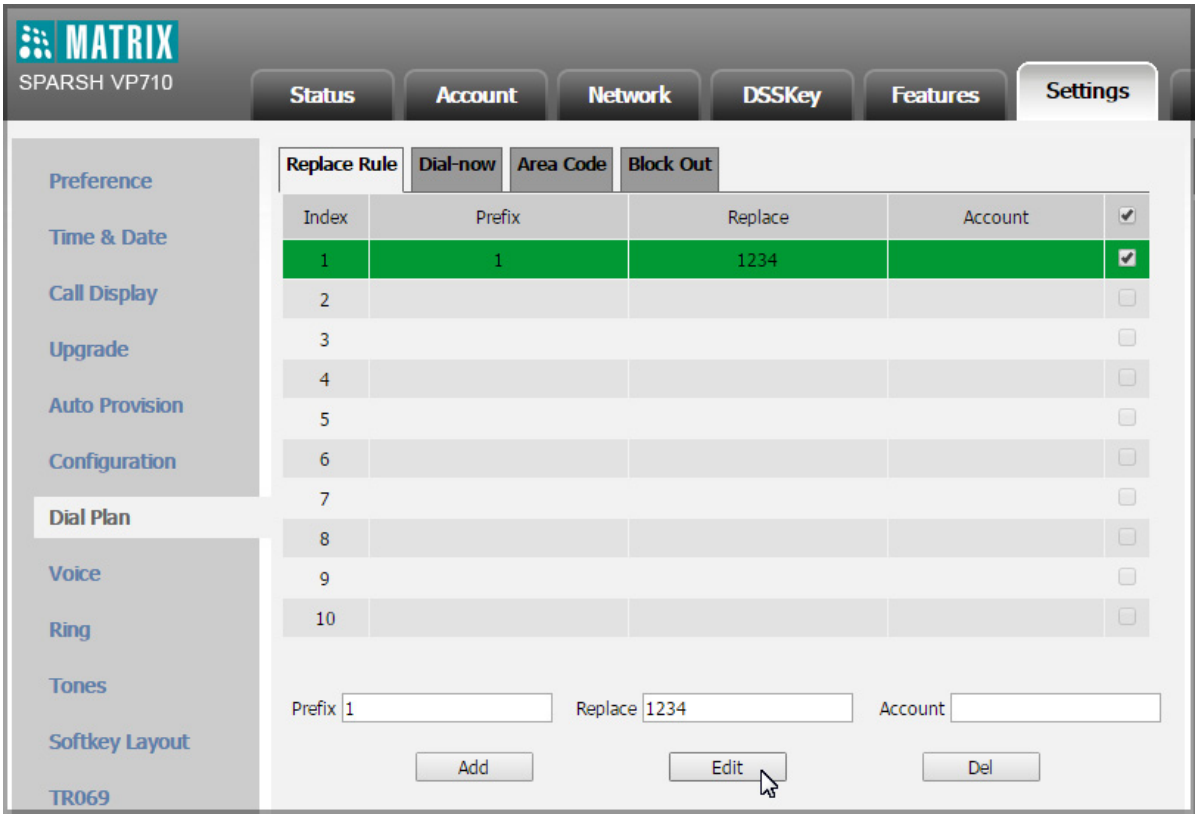
  When you enter the number “1” and then tap **Send**, the phone will dial out “1234” instead.

The valid values for the Account field can be one or more digits among 1-16. Every two digits must be separated by a comma. For example, when you enter the value “1, 2” in the Account field, this replace rule will apply to account1 and account2.

If you leave the Account field blank or enter 0, the replace rule will apply to all accounts.

To edit a replace rule via web user interface:

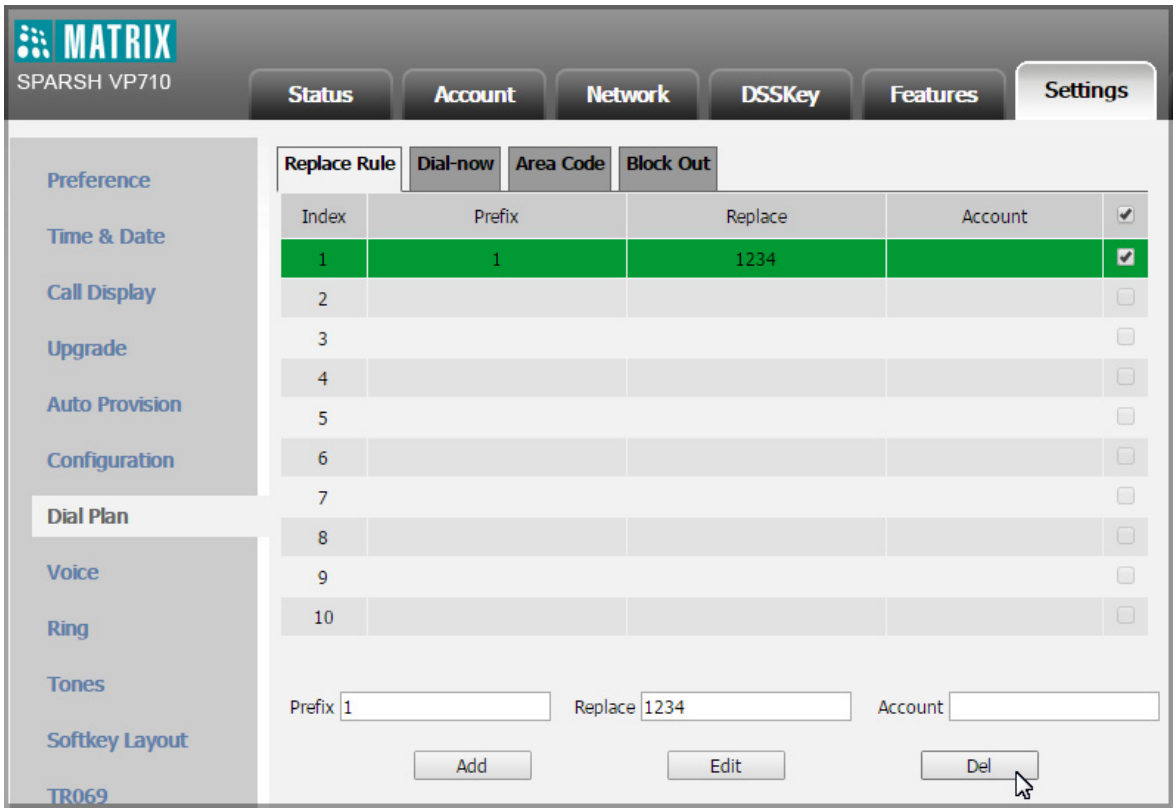
- Click on **Settings->Dial Plan->Replace Rule**.
- Select the desired replace rule by checking the check box.



- Edit the values in the **Prefix** and **Replace** fields.
- Enter the desired line ID in the **Account** field or leave it blank.
- Click **Edit** to save the change.

To delete one or more replace rules via web user interface:

- Click on **Settings->Dial Plan->Replace Rule**.
- Select one or more replace rules by checking the check box(es).



- Click **Del** to delete the replace rule(s).

Replace rule is configurable via web user interface only.

Dial-now

When the dialed out number matches the dial-now string, the number will be dialed out automatically. For example, a dial-now rule is configured as “2xx”, then any entered three-digit string beginning with 2 will be dialed out automatically on the phone.

Delay Time for Dial-now Rule

You can configure the delay time for dial-now rules. That is, you can configure your phone to dial out the phone number automatically, which matches a dial-now rule, after the specified delay time.

Procedure

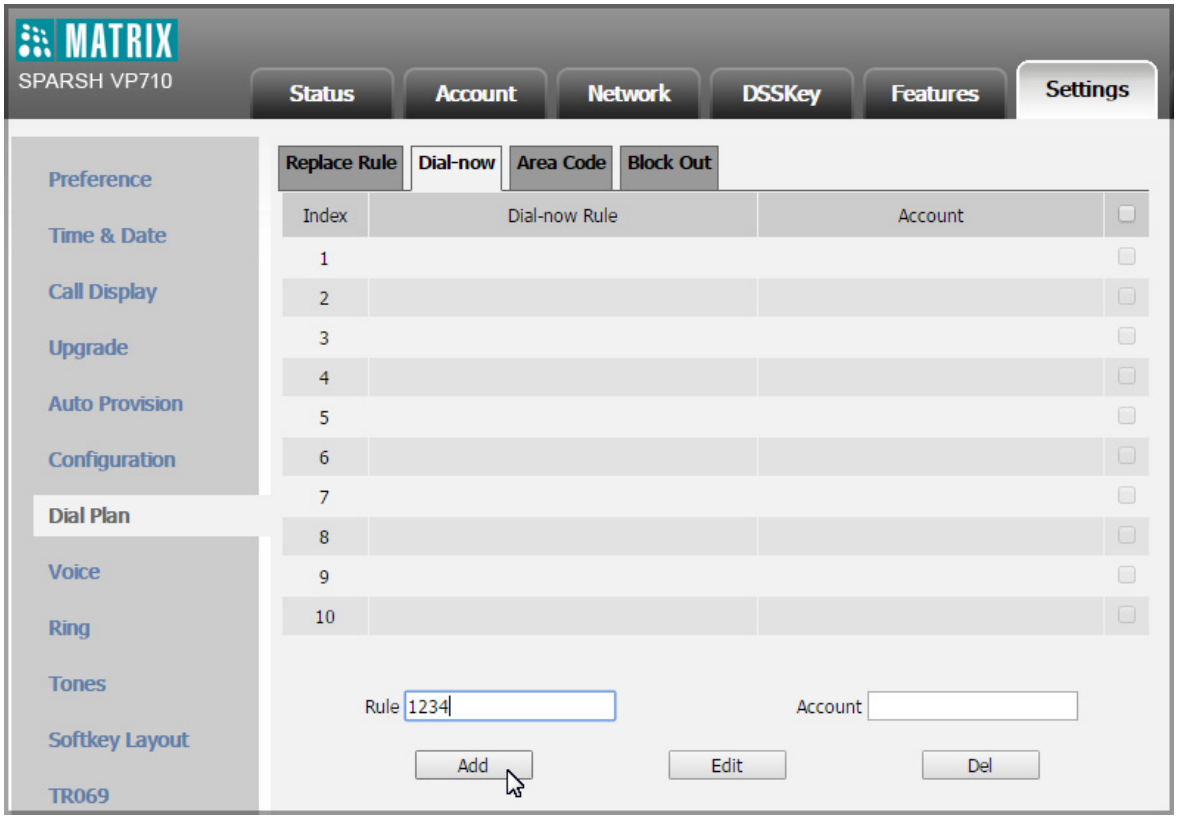
Dial-now rule can be created using the configuration files or locally.

Configuration File	<MAC>.cfg	Create the dial-now rule for the IP phone. Configure the delay time for the dial-now rule. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Create the dial-now rule for the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-dialplan&q=load&dial_page=dial-now. Configure the delay time for the dial-now rule **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To add a dial-now rule via web user interface:

- Click on **Settings->Dial Plan->Dial-now**.

- Enter the desired value (e.g.,1234) in the **Rule** field.



- Enter the desired line ID in the **Account** field or leave it blank.

  If you leave this field blank or enter 0, the dial-now rule will apply to all accounts on the IP phone.

- Click **Add** to add the dial-now rule.

  When you enter the number “1234” using the keypad, the phone will dial out “1234” automatically without pressing any key.

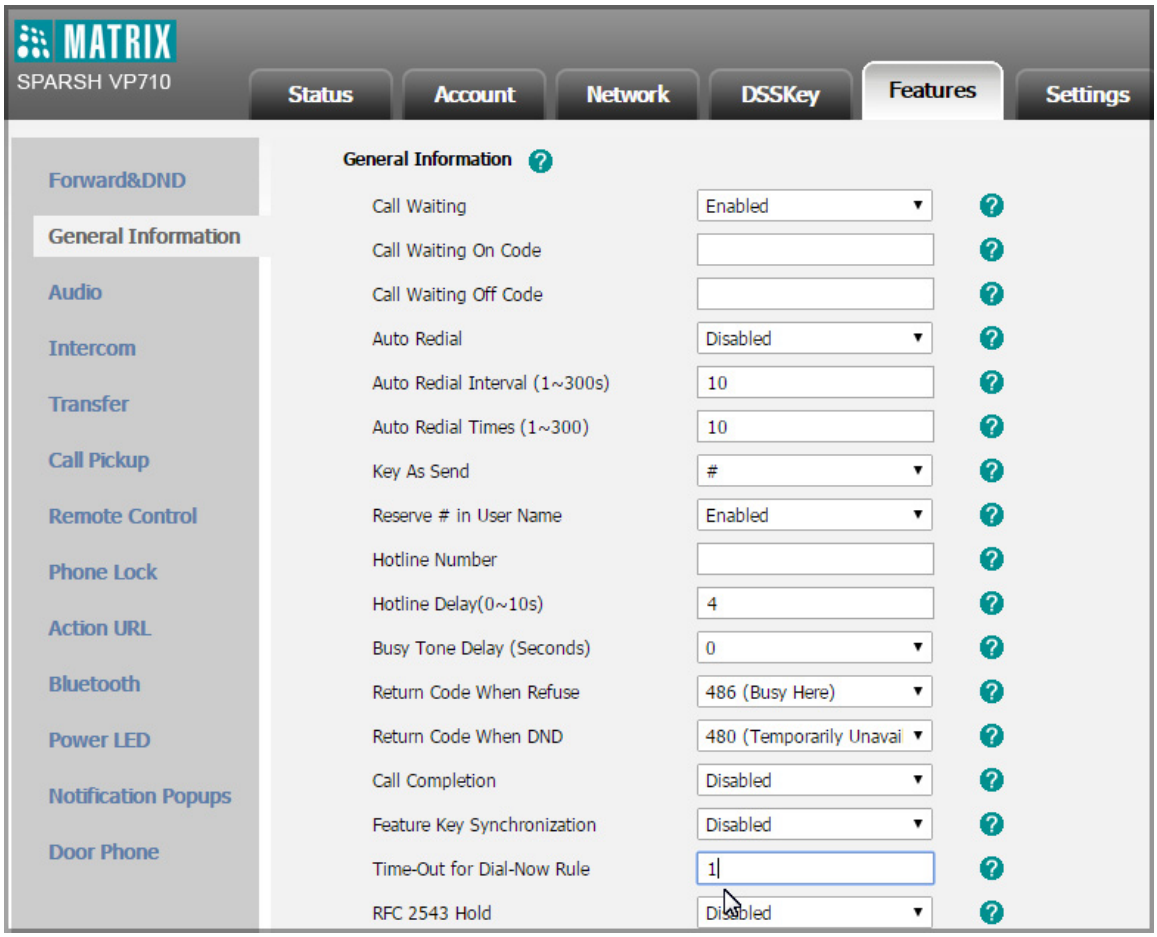
  You can also edit or delete the rule.

Dial-now rule is configurable via web user interface only.

To configure the delay time for dial-now rule via web user interface:

- Click on **Features->General Information**.

- Enter the time within 0-14 (in seconds) in the **Time-Out for Dial-Now Rule** field.



- Click **Confirm** to save the change.

Delay time for dial-now rule is configurable via web user interface only.

Area Code

Area codes are also known as Numbering Plan Areas (NPAs).They usually indicate geographical areas in a country. This feature is necessary only when dialing the number outside the code area. For example, area code is configured as “Code: 0592, Min Length: 4,Max Length: 11”, then when you dial out the number “56789”, which has the digits between 4 to 11, the phone will add the area code and dial out the number “059256789”. You can only configure one area code rule on your phone.

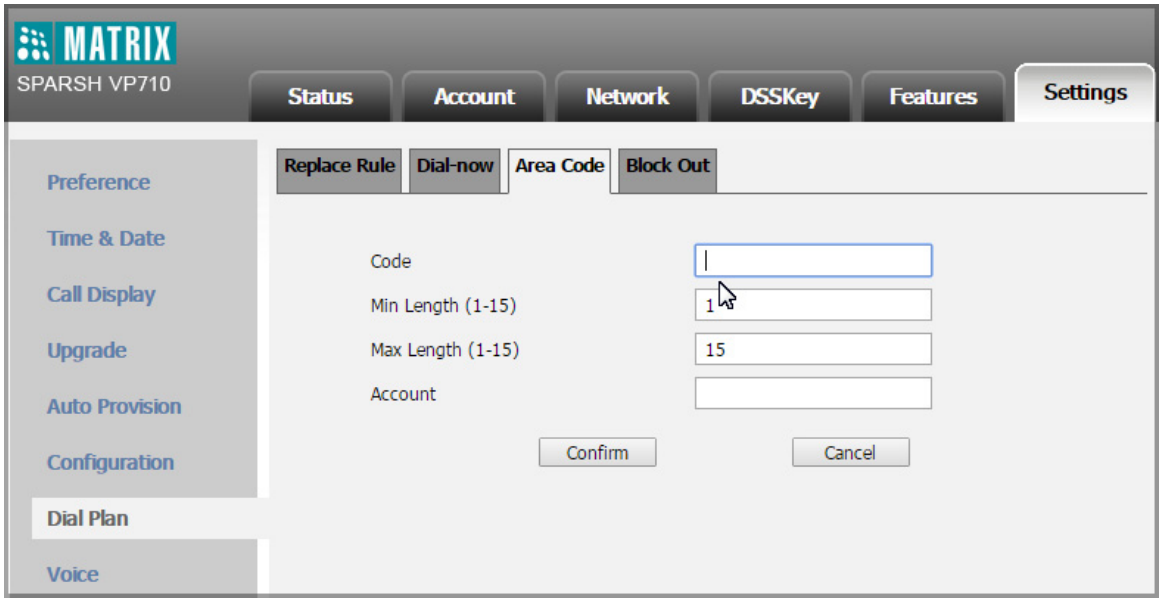
Procedure

Area code rule can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Create the area code rule and specify the maximum and minimum lengths of entered numbers. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Create the area code rule and specify the maximum and minimum lengths of entered numbers. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-dialplan&q=load&dial_page=area-code

To configure the area code via web user interface:

- Click on **Settings->Dial Plan->Area Code**.
- Enter the desired values in the **Code**, **Min Length(1-15)** and **Max Length(1-15)** fields.



- Enter the desired line ID in the **Account** field or leave it blank

  If you leave this field blank or enter 0, the area code rule will apply to all accounts on the IP phone.

- Click **Confirm** to save the change.

Area code is configurable via web user interface only.

Block Out

You can block some specific numbers from being dialed on your phone. When you dial a block out number on your phone, the dialing will fail and the touch screen will prompt “Forbidden Number”. You can add 10 block out rules at most on your phone.

Procedure

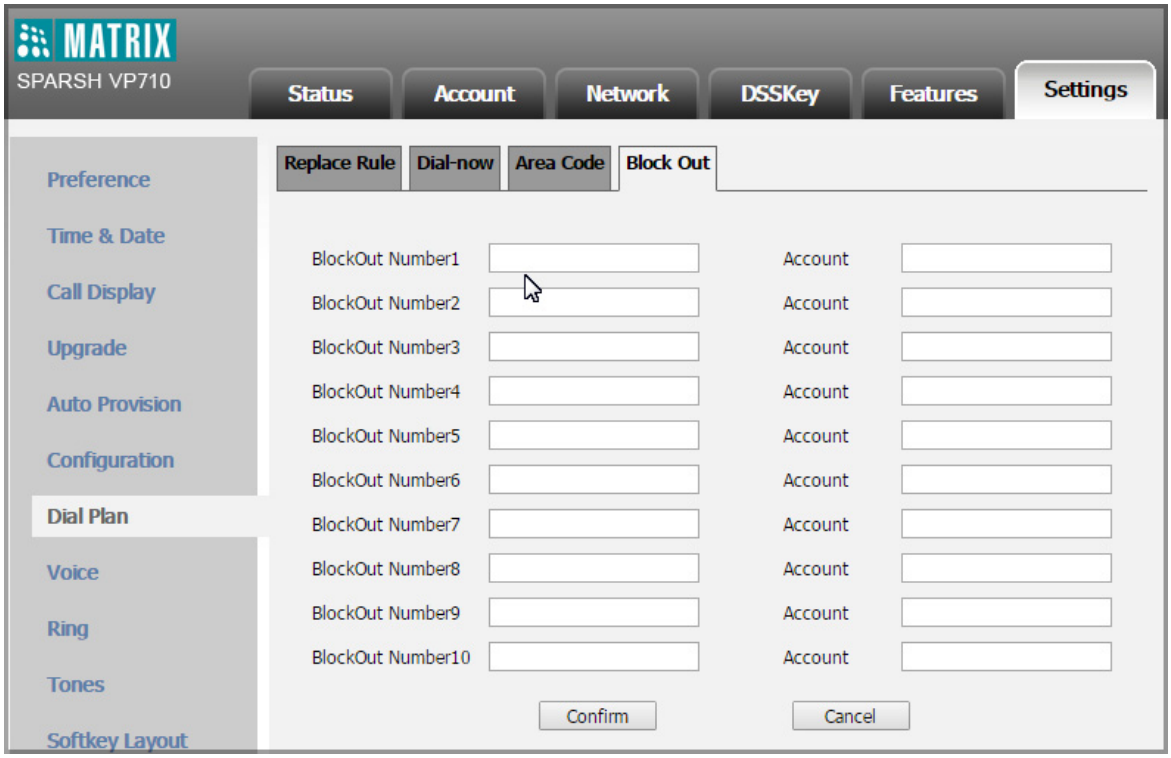
Block out rule can be created using the configuration files or locally.

Configuration File	<MAC>.cfg	Create the block out rule for the IP phone. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Create the block out rule for the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-dialplan&q=load&dial_page=block-out

To add a block out number via web user interface:

- Click on **Settings->Dial Plan->Block Out**.

- Enter the desired values in the **BlockOut Number** fields.



- Enter the desired line ID in the **Account** field or leave it blank

  If you leave this field blank or enter 0, the block out rule will apply to all accounts on the IP phone.

- Click **Confirm** to add the block out number.

Block out number is configurable via web user interface only.

Dial Plan using Digit Map String Rules

Digit maps, described in RFC 3435, are defined by a single string or a list of strings. If a number entered matches any string of a digit map, the call is automatically placed. If a number entered matches no string - an impossible match - you can specify the phone's behavior. You can specify the digit map timeout, the period of time before the entered number is dialed out. You need to know the following basic regular expression syntax when creating new dial plan:

T	The timer letter "T" indicates a timer expiry. If "T" is used alone (e.g., 123T), the default timeout value of 3 will be used. If "T" is not used alone (e.g., 123<Tx>, x can be a digit from 0 to 99), a complete match occurs when waiting x seconds after inputting 123.
x	The "x" can be used as a placeholder for any digit from 0 to 9. Example: "12x" would match "121", "122", "123", etc.
[]	The square bracket "[]" can be used as a placeholder for a single character which matches any of a set of characters. Example: "91[5-7]1234" would match "9151234", "9161234", "9171234".

-	The dash “-” can be used to match a range of digits within the brackets. Example: “[35-7]” would match the number “3”, “5”, ”6” or ”7”. Note: The digits must be concrete, e.g., [3-x] is invalid.
.	The dot “.” can be used as a placeholder or multiple placeholders, including zero, of occurrences of the preceding construct. Examples: “123.T” would match “123”, “1233”, “12333”, “123333”, etc. “x.T” would match an arbitrary number. “[x*#+].T” would match an arbitrary character. Note: If the string ends with a dot (e.g., 123.), a match will occur immediately after inputting the characters before the dot (e., 123) since the dot allows for zero occurrences of the preceding construct. So we recommend you to add a letter “T” after the dot (e.g., 123.T) for inputting more characters.
R	The letter “R” indicates that certain matched strings are replaced. Using a RRR syntax, you can replace the digits between the first two Rs with the digits between the last two Rs. Example: “R12R234R” would replace 12 with 234.
<:>	The letter “:” in the angle bracket indicates that certain matched strings are replaced. Using the <:> syntax, you can replace the digits before the colon with the digits after the colon. Example: “<12:234>” would replace 12 with 234. It is the same with R12R234R.
!	The exclamation mark “!” can be used to prevent users from dialing out specific numbers. It can only be put last in each string of the digit map. Example:“235x!” would match “2351”, “2352”, “2353”, etc. The number starting with 235 will be blocked to dial out.
,	The comma “,” can be used as a separator to generate secondary dial tone. Example: “<9,:55>xx”, after entering digit “9”, secondary dial tone plays andyou can complete the remaining two-digit number. Note: The secondary dial tone can be customized. For more information, refer to Tones on page 594.

Procedure

Digit Map can be created using the configuration files.

Configuration File	<MAC>.cfg	Configure digit map on a phone basis. Configure digit map on a per-line basis. For more information, refer “Appendix D - Configuration Parameters”.

Emergency Dialplan

The IP Phone support dialing emergency telephone numbers when the phone is locked (refer to Phone Lock). Due to the fact that the IP phone must have a registered account or a configured SIP server, it may not meet the need of dialing emergency telephone number at anytime.

Emergency dialplan allows users to dial the emergency telephone number (emergency services number) at any time when the IP phone is powered on and has been connected to the network.It is available even if your phone keypad is locked or no SIP account is registered.

Emergency Dial Plan

Users can configure the emergency dial plan on the phone (e.g., emergency number, emergency routing). The phone determines if this is an emergency number by checking the emergency dial plan configured on the phone. When placing an emergency call, the call is directed to the configured emergency server. Multiple emergency servers may need to be configured for emergency routing, avoiding that emergency calls couldn't get through because of the server failure. If the phone is not locked, it checks against the regular dial plan (refer to Dial Plan). If the phone is locked, it checks against the emergency dial plan.

Emergency Location Identification Number (ELIN)

The IP Phones support Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED). LLDP-MED allows the phone to use the location information, Emergency Location Identification Number (ELIN), sent by the switch, as a caller ID for making emergency calls. The outbound identity used in the P-Asserted-Identity (PAI) header of the SIP INVITE request is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN). The administrator can customize the outbound identity. The custom outbound identity will be used if the phone fails to get the LLDP-MED ELIN value.

The following is an example of the PAI header:
P-asserted-identity: <sip: 1234567890@abc.com > (where 1234567890 is the custom outbound identity.)

P-Access-Network-Info (PANI)

When placing an emergency call, the MAC address of the phone/connected switch should be added in the P-Access-Network-Info (PANI) header of the INVITE message. It helps the aid agency to immediately identify the caller's location, improving rescue efficiency.

The following is an example of the PANI header:
P-Access-Network-Info: IEEE-802.3; eth-location="00:15:65:74:b1:6e" (where 00156574B16E is the phone's MAC address.)

Procedure

Emergency dialplan can be created using the configuration files.

Configuration File	<MAC>.cfg	Configure the emergency dialplan. For more information, refer "Appendix D - Configuration Parameters".

Emergency Number

Public telephone networks in countries around the world have a single emergency telephone number (emergency services number), that allows a caller to contact local emergency services for assistance when necessary. The emergency telephone number may differ from country to country. It is typically a three-digit number so that it can be easily remembered and dialed quickly. Some countries have different emergency numbers for different emergency services.

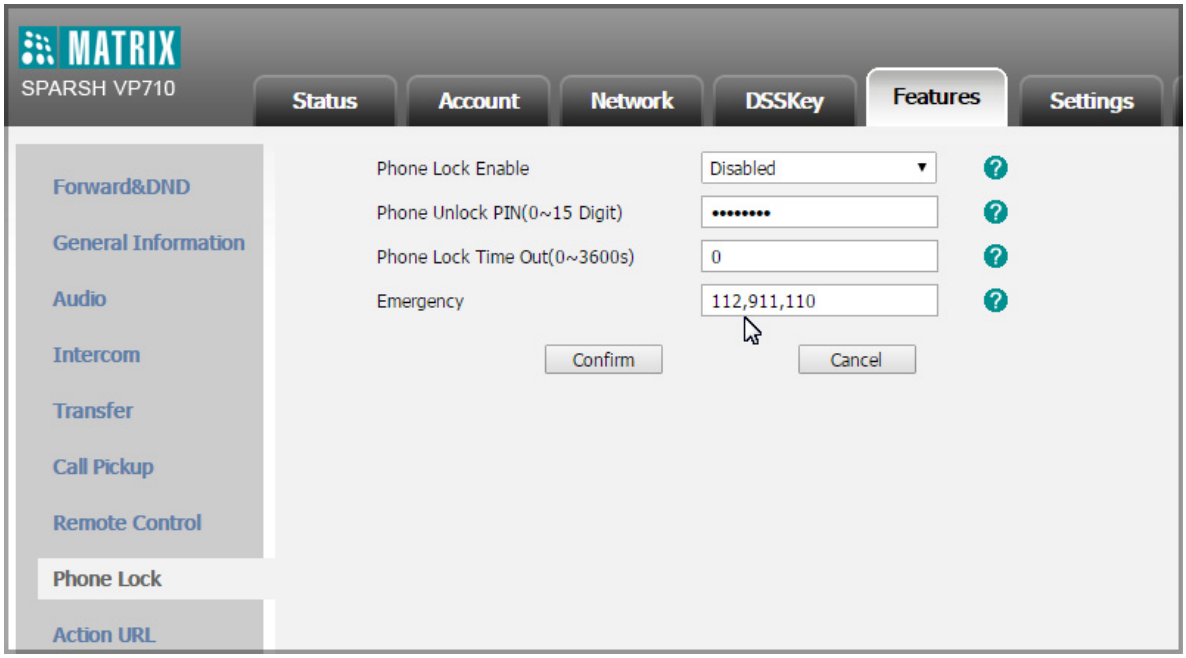
You can specify the emergency telephone number on the IP phone for contacting the emergency services in an emergency situation. You can dial these numbers when the phone is locked.

Contact your local phone service provider for available emergency numbers in your area.

To specify emergency numbers via web user interface:

- Click on **Features->Phone Lock**.
- Enter the emergency number in the **Emergency** field.

  For multiple numbers, enter a comma between every two emergency numbers. There may be default emergency numbers configured in the IP phone.



- Click **Confirm** to save the change.

Emergency number is configurable via web user interface only.

Live Dialpad

You can enable live dialpad on your phone, which enables it to automatically dial out the phone number without pressing the send key. You can also configure a delay, where the phone will dial out the phone number automatically after the specified period of time.

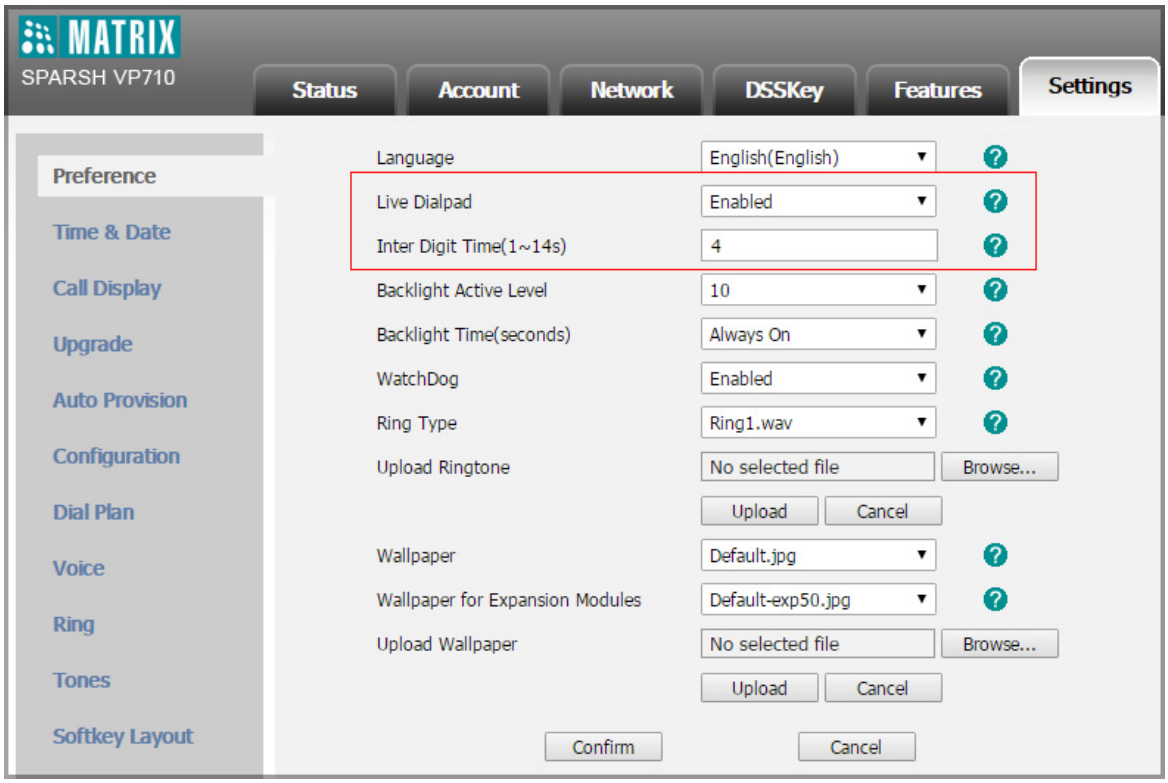
Procedure

Live dialpad can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure live dialpad. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure live dialpad. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-preference&q=load

To configure live dialpad via web user interface:

- Click on **Settings->Preference**.
- Select **Enabled** from the **Live Dialpad** list.
- Enter the desired delay time in the **Inter Digit Time (1~14s)** field.



- Click **Confirm** to save the change.



Live dialpad is configurable via web user interface only.

Hotline

You can dial a hotline number immediately upon lifting the handset or pressing the Speakerphone key. You can also configure a delay, where the phone will dial out the hotline number automatically after the specified period of time.

Procedure

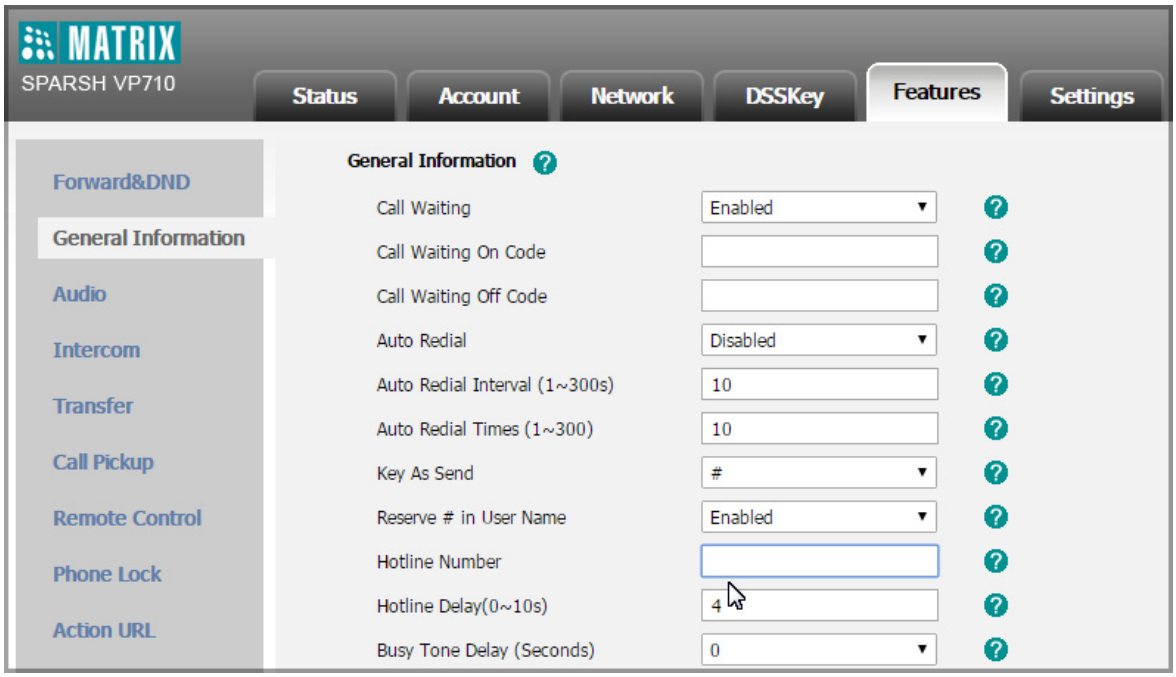
Hotline can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the hotline number. Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the hotline number. Specify the time (in seconds) the IP phone waits before automatically dial out the hotline number. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure the hotline number. Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number.

To configure hotline via web user interface:

- Click on **Features->General Information**.
- Enter the hotline number in the **Hotline Number** field.

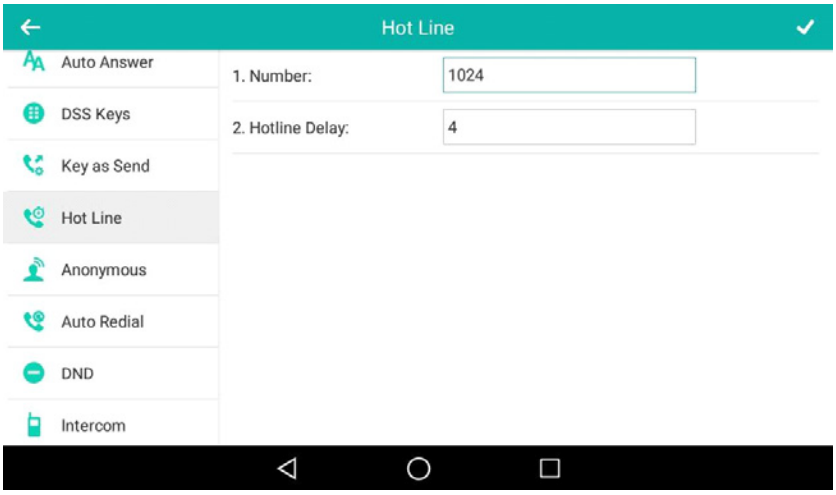
- Enter the delay time in the **Hotline Delay(0~10s)** field.



- Click **Confirm** to save the change.

To configure the hotline number via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Hot Line.**
- Enter the desired number in the **Number** field.
- Enter the desired delay time in the **Hotline Delay** field.



- Tap ✓ to save the change or ◁ to cancel.

CHAPTER 7

System Applications

The system applications are pre-installed on your phone and they cannot be uninstalled. This chapter provides basic operating instructions for using the system applications. Topics include:

- Managing Applications and Widgets
- Managing Running Applications
- APP Accounts
- FileManager
- Gallery
- Recorder
- Camera
- Email
- Browser
- Calendar
- Calculator
- Clock

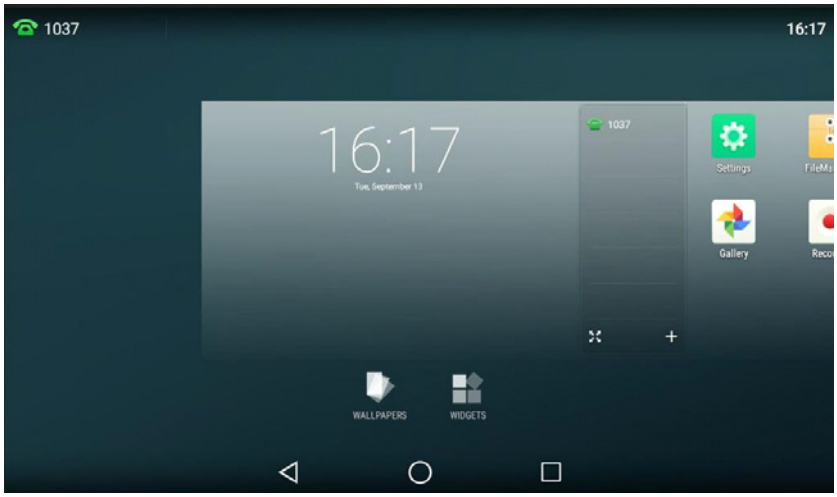
Managing Applications and Widgets

Adding a Widget to the Idle Screen

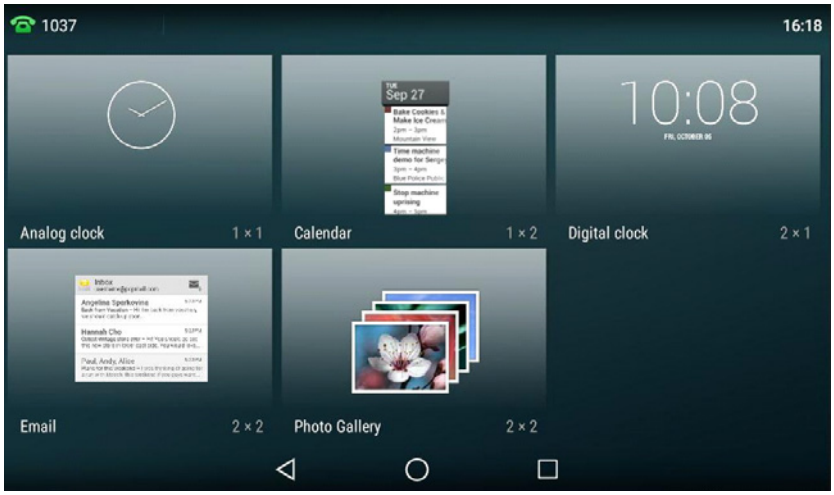
The IP phone provides widgets like analog clock, calendar, digital clock, email and photo gallery. You can conveniently add these widgets to the idle screen as preferred.

To add a widget to idle screen:

- Long tap the empty spot on the idle screen.

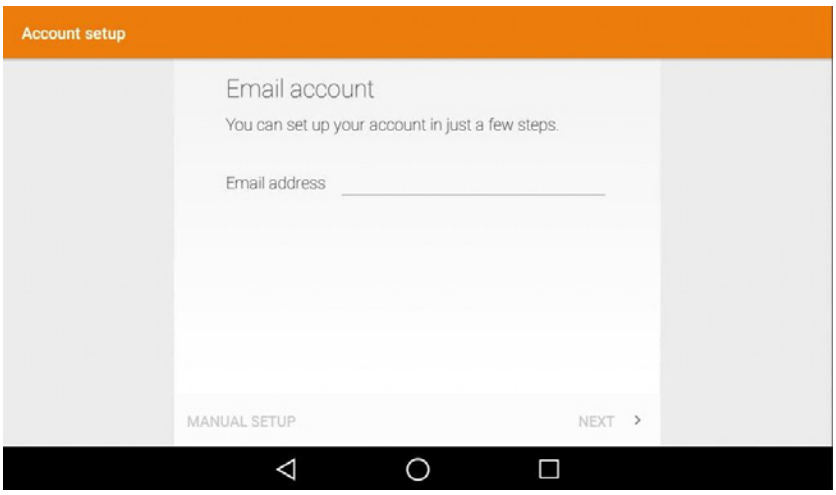


- Tap **WIDGETS**.

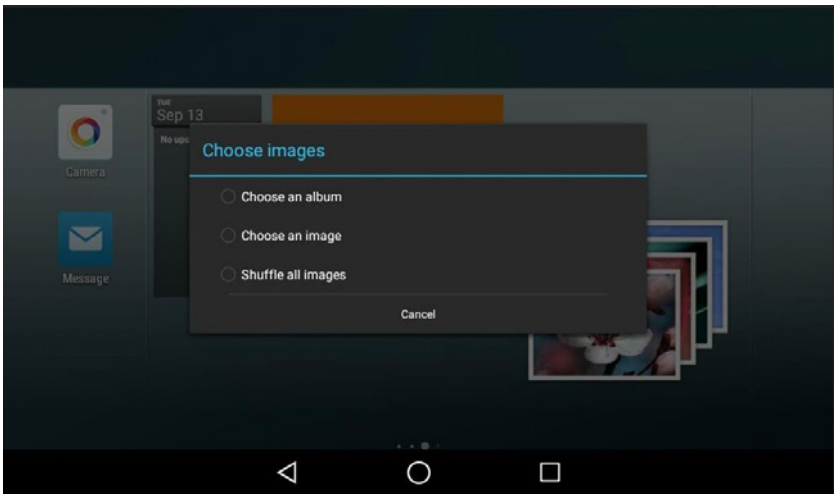


- Drag the desired widget up/down/left/right to the destination spot on the idle screen.

If you want to add an email widget, and you have not registered an account on the Email or Calendar application, a wizard will be provided for you to set up the email account step by step. For more information, refer to “Adding APP Accounts”.



If you want to add a photo gallery widget, you will be required to choose the desired images to display.



Before adding a widget to the idle screen, make sure the idle screen have enough space for the widget. Otherwise the widget cannot be added and the touch screen will prompt “No more room on this Home screen”.

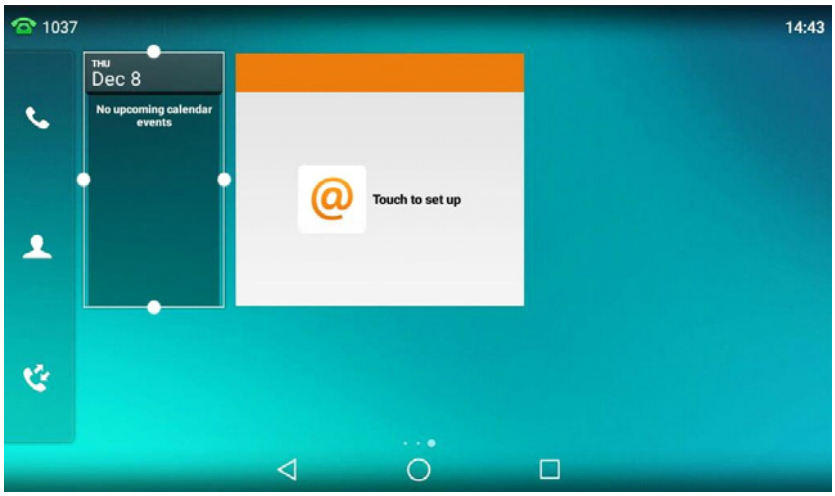
Resizing a Widget

Certain widgets such as calendar, digital clock and email can be resized on the idle screen.

To resize a widget:

- Long tap the desired widget for about 2 seconds, and then lift your finger from the screen.

This will bring up the white frame as shown below:



- Drag the border of the frame to adjust the size.

Moving an Application or Widget

You can move an application or widget to a different spot on the idle screen or to a different idle screen.

Before moving an application or widget, you should know the following:

- You can drag an application or widget to the rightmost of three idle screens to add an idle screen.
- If there is only one application or widget on the screen, you cannot drag the application or widget to the rightmost of three idle screens to add an idle screen.
- The idle screen (except the home screen) will be deleted automatically if there is no application or widget on the screen.

To move an application or widget:

- Long tap the desired item for about 2 seconds.
- Drag the item up/down/left/right to the destination spot on the screen.



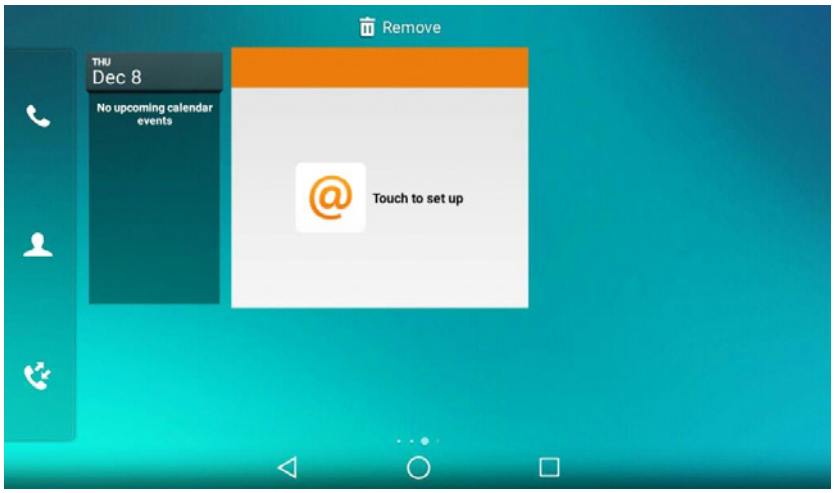
Removing a Widget from the Idle Screen

Removing operation only removes the widget of the application on the screen. The application itself will not be uninstalled.

To remove a widget from the idle screen:

- Long tap the desired item for about 2 seconds.

  The **Remove** field will appear on the top of the touch screen.



- Drag the item to the **Remove** field to remove it.

Creating a Folder

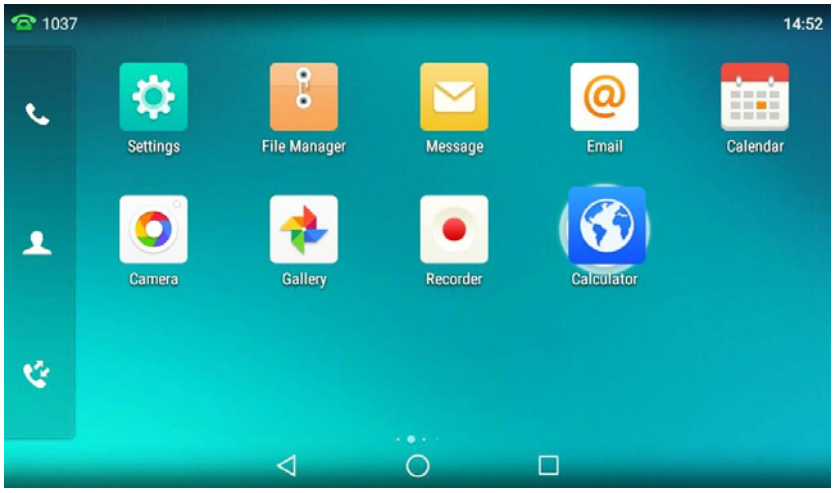
You can create a folder to organize your applications. For example, you can place the same kind of applications in a folder.

To create a folder:

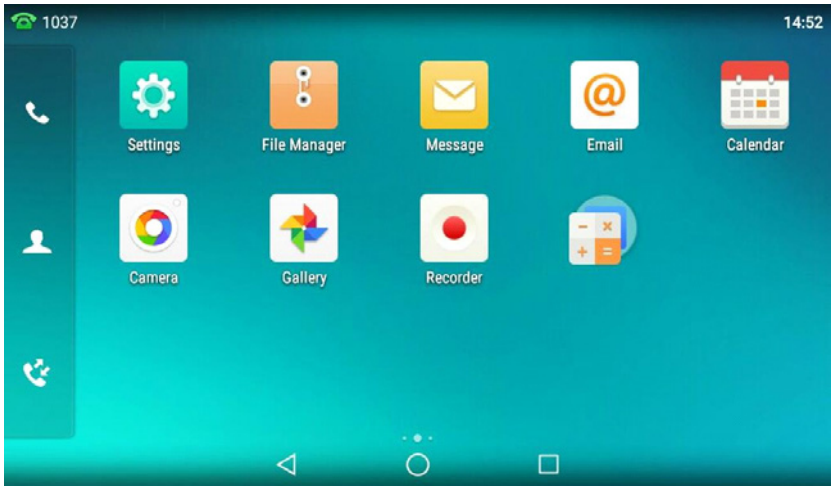
- Drag an application icon over another application icon.



- Lift your finger when a folder frame appears around the applications.



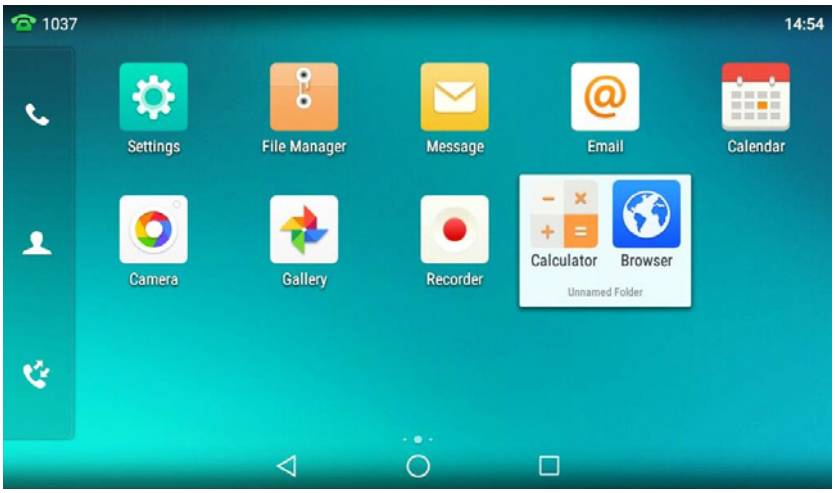
- A new folder containing the selected applications is automatically created.



- You can do the following:
  - Drag another application icon to the folder to add more applications into the folder.
  - Tap the folder and then drag an application icon in the folder to the destination spot on the idle screen to move the application from the folder to the idle screen.
  - The folder will be deleted automatically if there is only one application in the folder.

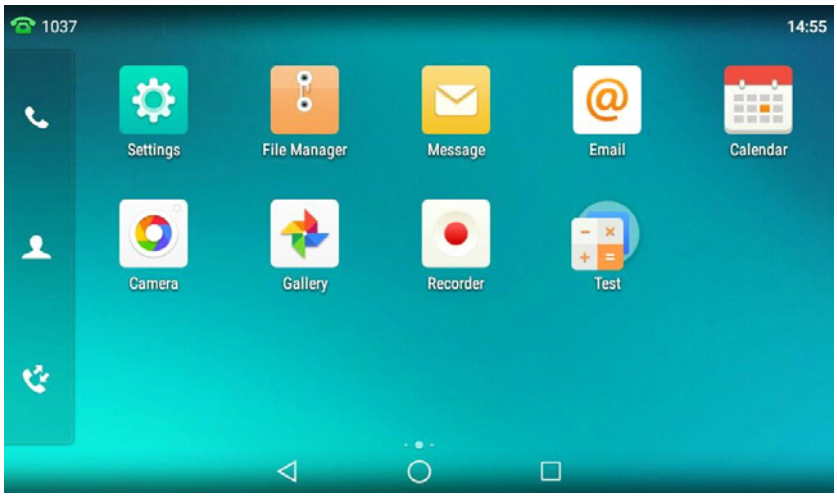
To rename a folder:

- Tap the desired folder to expand the folder.



- Tap the **Unnamed Folder** field.
- Enter the desired folder name.
- Tap **Done** on the onscreen keyboard.
- Tap the blank area of the touch screen.

  The folder name is renamed.

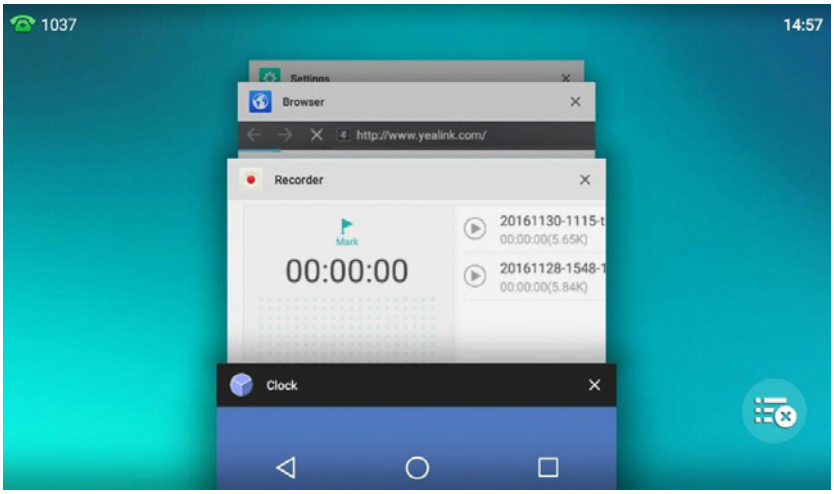


Managing Running Applications

You can tap ☐ to view and manage the list of recently used applications that are still running in the background. You can do the following:

- Swipe the desired application left or right to stop the running application.
- Tap  to stop all running applications.

- Tap the desired application to access the application.



If there are no running applications, the touch screen will prompt “Your recent screens appear here”.

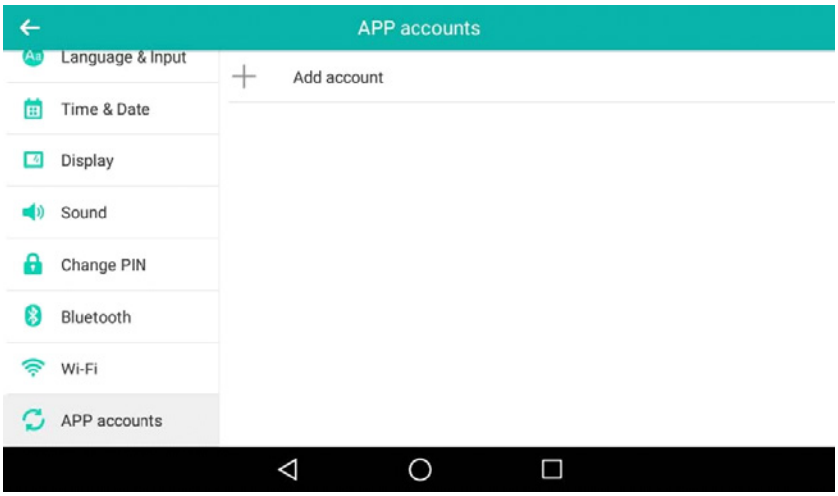
APP Accounts

Email and Calendar applications require a registered account. You can create accounts to have the best experience with your phone. You can also sync the account information or remove the account.

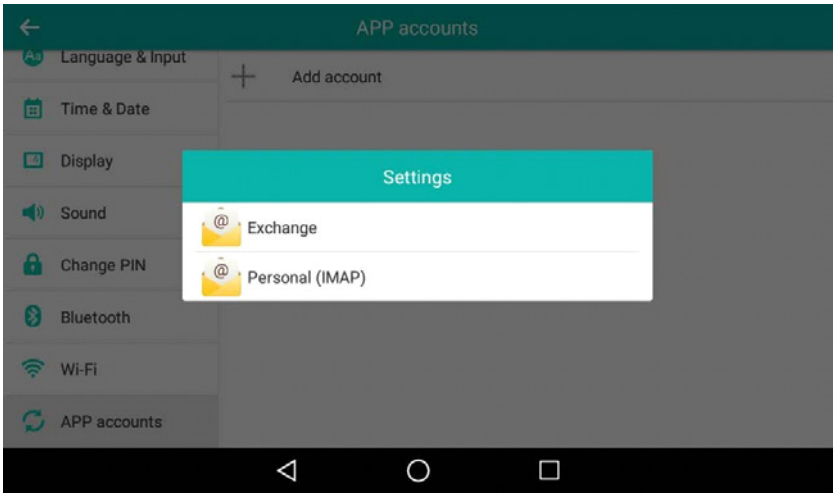
Adding APP Accounts

To add an app account via phone user interface:

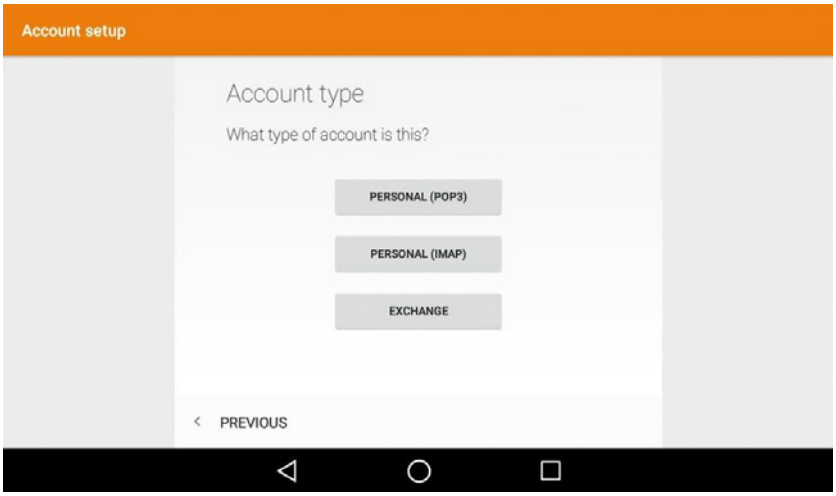
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Basic->APP accounts.**



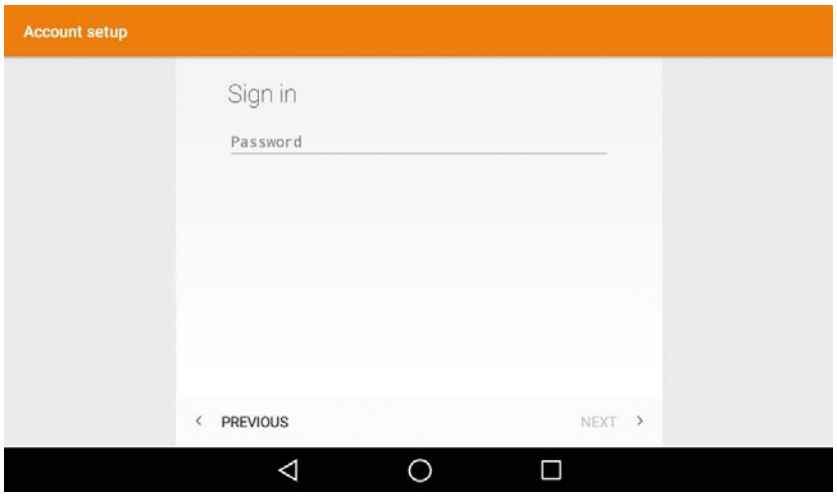
- Tap **Add** account.



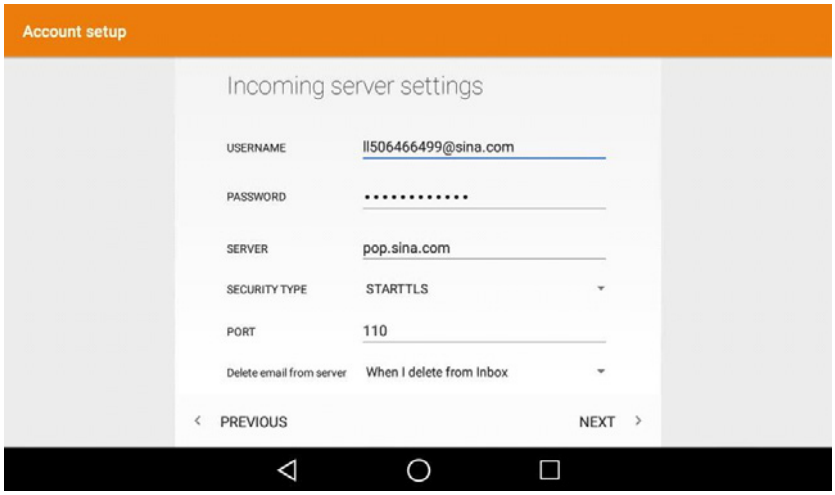
- Tap a desired account type.
- Enter the email address.
- Tap **Next** or **Manual** setup.
- Choose the email account type.



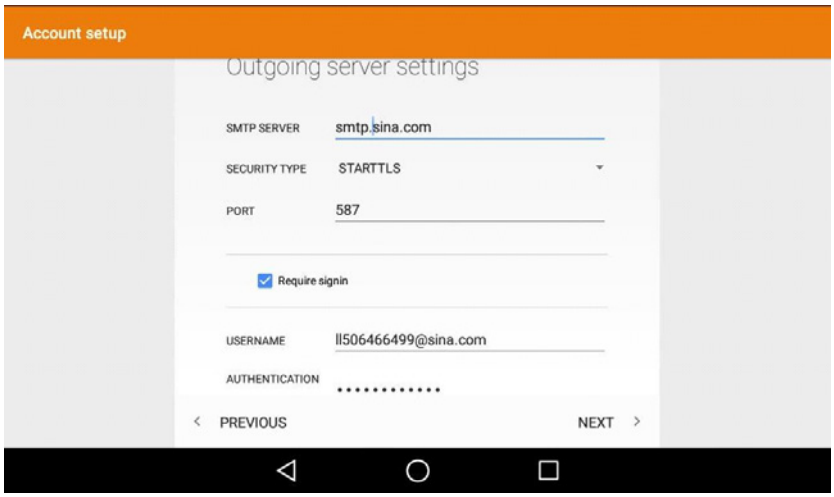
- **POP3:** When using POP3, all emails are stored locally instead of the server side. Please check with your network administrators for the correct email account settings for POP3.
- **IMAP:** When using IMAP, all emails are stored on the server side and users can see them from the IP phone or other mail clients as well. Please check with your network administrators for the correct email account settings for IMAP.
- **Exchange:** When using Exchange, all emails are stored in the central Exchange mail server and can be checked using capable mail client such as Microsoft Outlook. Please check with your network administrators for the correct email account settings for Exchange.



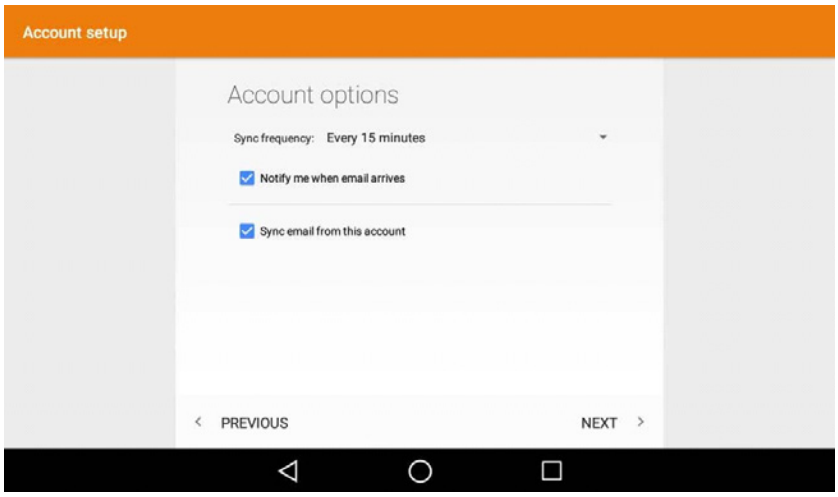
- Enter the password and then tap **NEXT**.
- Configure the incoming server settings.



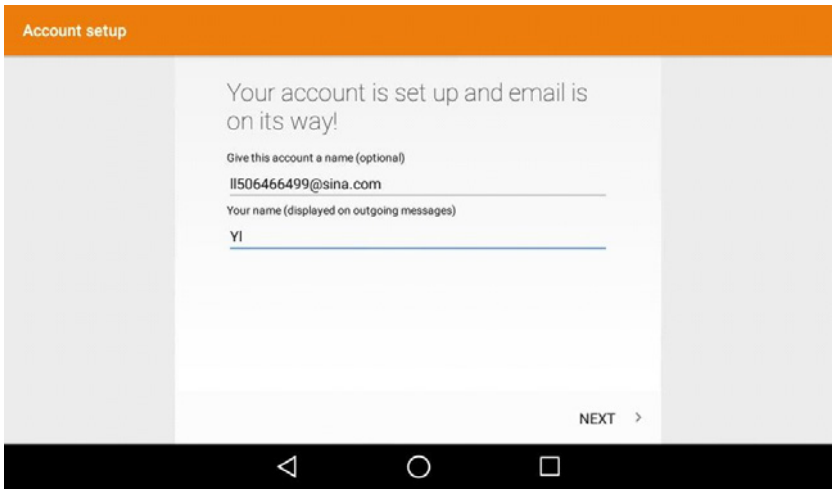
- Tap **NEXT**.
- Configure the outgoing server settings.



- Tap **NEXT**.
- Configure the account options.



- Tap **NEXT**.
- Enter the string in the **Your name** field as the sender name.



- Tap **NEXT**.

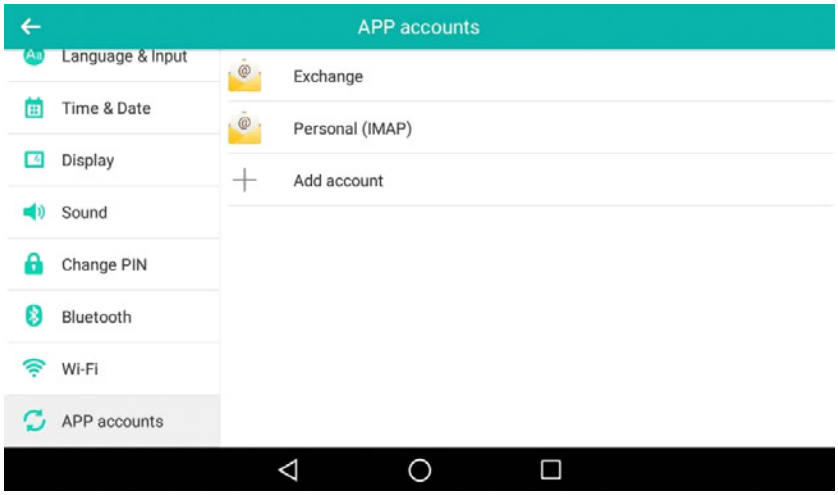
  You are able to start using the email account now.

Syncing APP Account Information

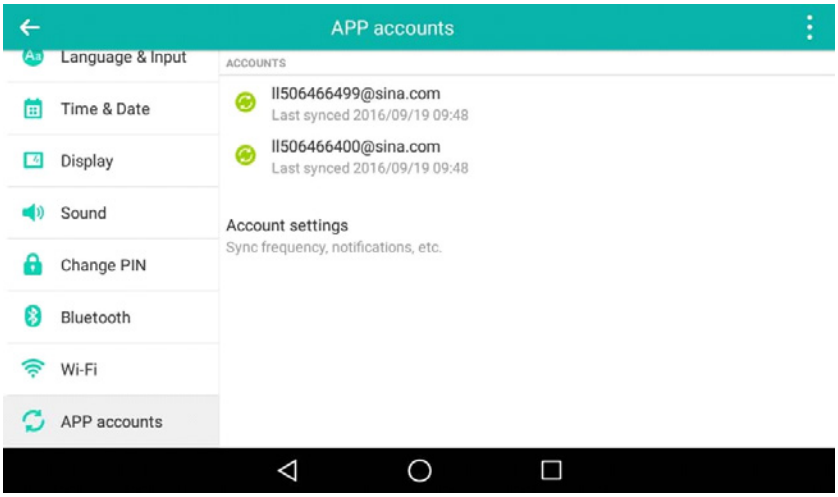
To sync account information via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.

- Tap **Settings->Basic->APP accounts.**

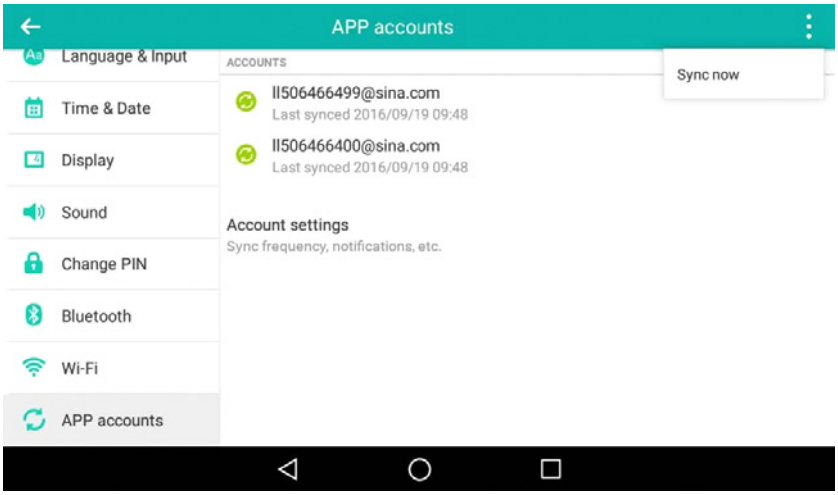


- Tap the desired account type.



- Do one of the following:
  - Tap and then tap **Sync now**.

In this example, it will sync both ll50466499@sina.com and ll506466400@sina.com information.

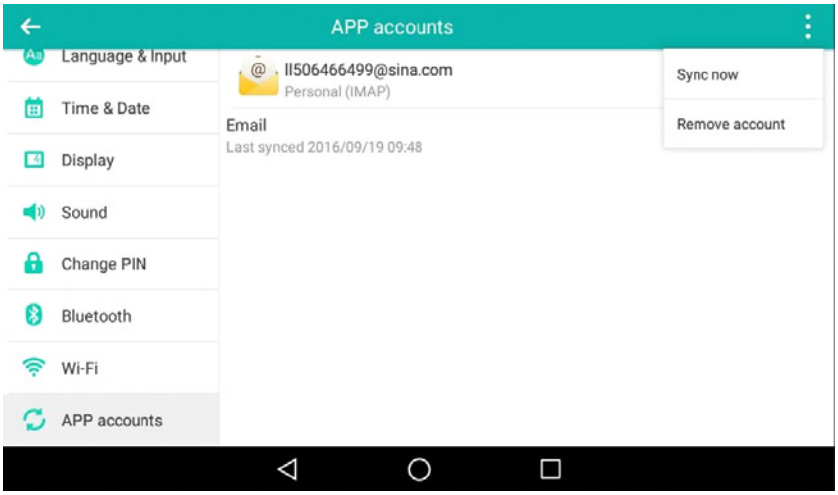


OR

- Tap a desired account.

  Tap ⋮ and then tap **Sync now**.

  In this example, it will only sync ll50466499@sina.com information.

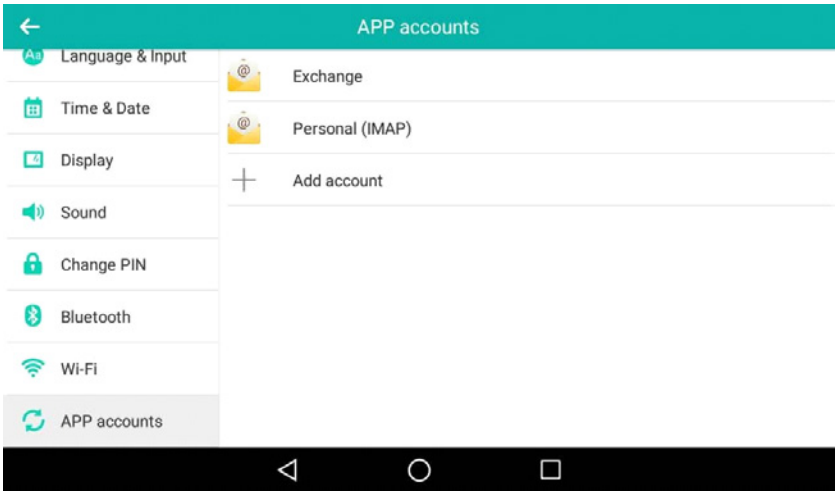


Removing APP Accounts

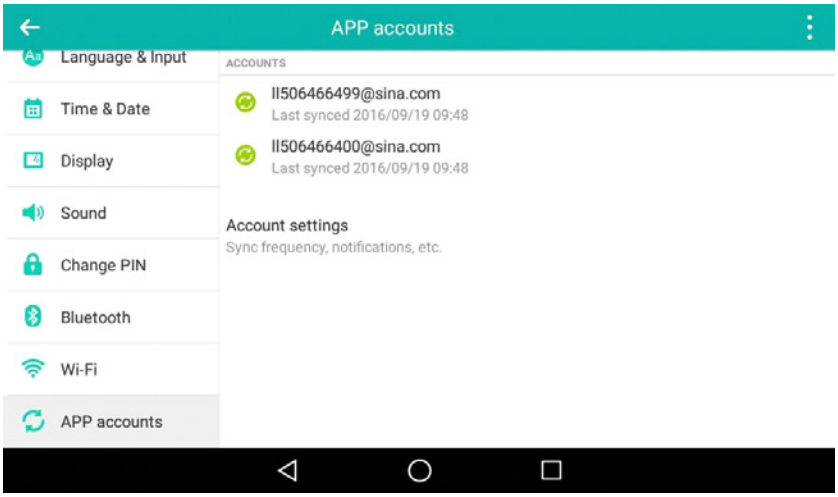
To remove an app account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.

- Tap **Settings->Basic->APP accounts**.

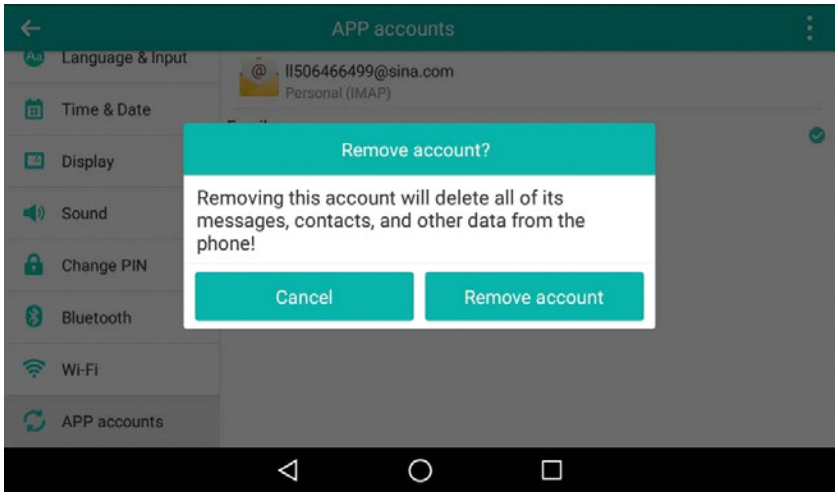


- Tap a desired account type.



- Tap a desired account.
- Tap and then tap **Remove** account.

The following prompt will appear on the touch screen of the phone:



- Tap **Remove account** to save the change or **Cancel** to cancel.

FileManager

FileManager is a tool that allows you to view, search, delete, copy or move photo/video/audio files. You can also share files via Bluetooth or email. It helps you to access and manage files in internal SD card as well as USB flash drive. For more information on how to connect the USB flash drive to your phone, refer to “Phone Installation”.

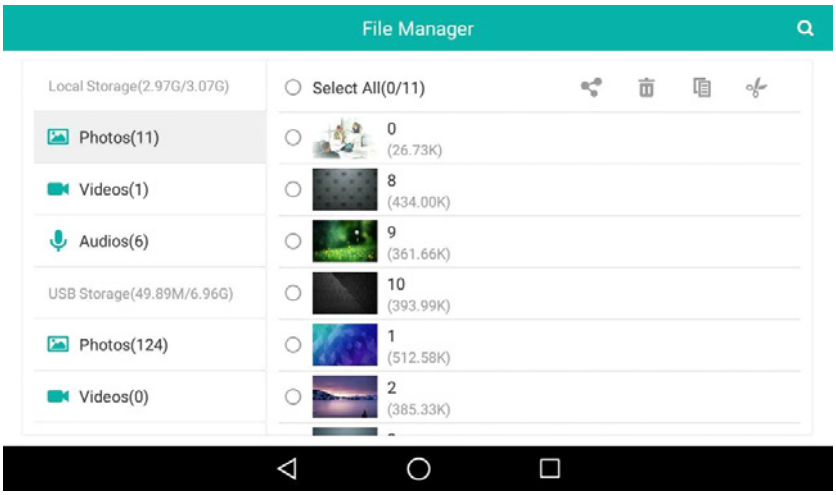
Viewing Files

To view pictures:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **FileManager** application.
- Tap **Photos** in the **Local Storage** field.

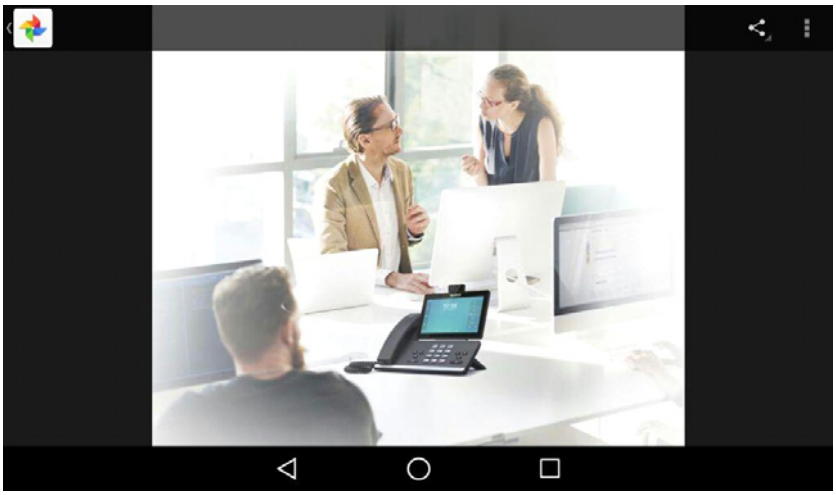
  If a USB flash drive has been connected to your phone, you can tap **Photos** in the **USB Storage** field to view the pictures stored in USB flash drive.

Your pictures will display as a list.



- Tap the desired picture to view.

  The IP phone will automatically launch **Gallery** application to preview the picture.



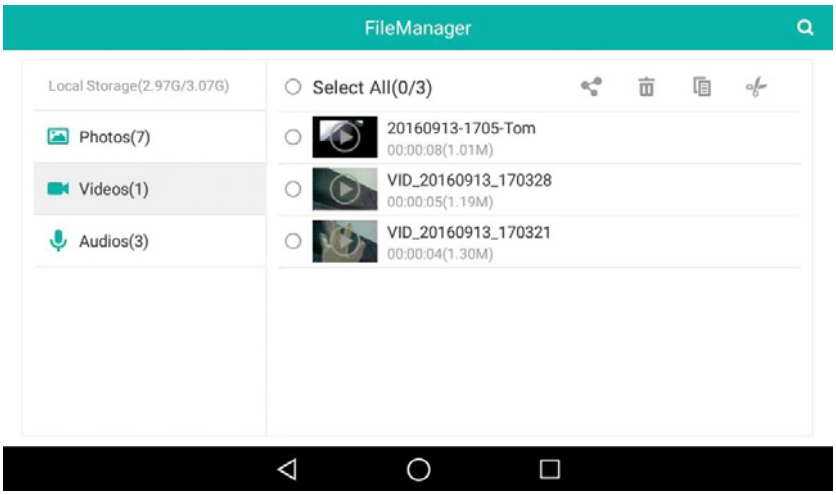
- You can set the picture as wallpaper, print the picture and/or view detail information of the picture. For more information, refer to “”, “Printing Pictures” and “Viewing Detail Information of the Pictures/Videos”.

To view videos:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **FileManager** application.
- Tap **Videos** in the **Local Storage** field.

  If a USB flash drive has been connected to your phone, you can tap **Videos** in the **USB Storage** field to view the videos stored in USB flash drive.

- Your videos will display as a list.

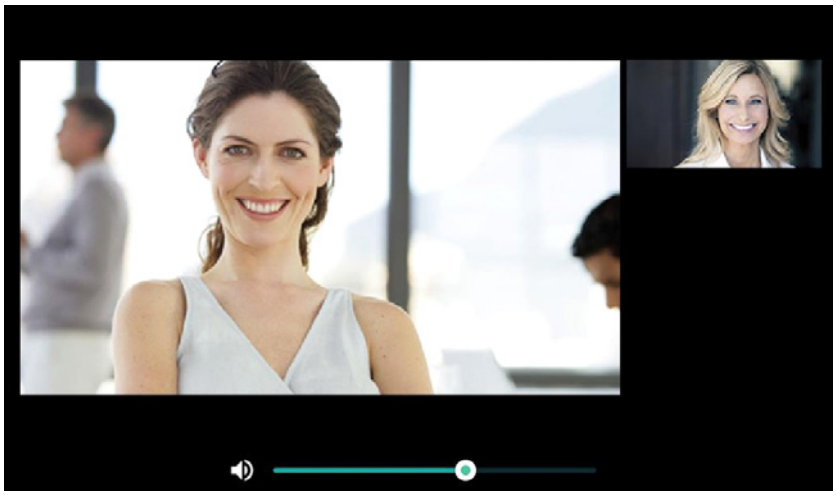


- Tap the desired video to play.

  The IP phone will automatically launch **Gallery** application to view the video.



- During the playing, you can do the following:
  - Tap the touch screen. Then  appears on the touch screen.
  - Tap  to pause the playing. Then  appears on the touch screen.
  - Tap  to continue the playing.
- Drag the slider to skip forward the playing or rewind the playing.
- Press  to adjust the media volume.



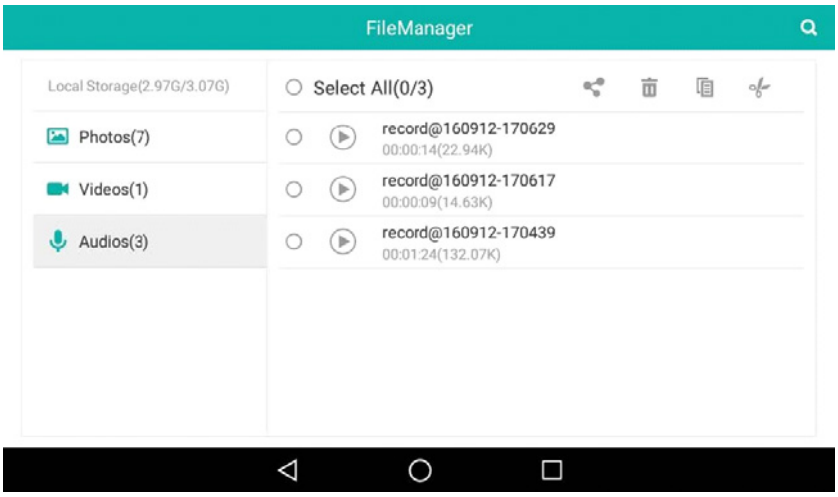
The menu keys and Android keys will disappear within 5 seconds, you may tap the touch screen to show them again.

To view audios:

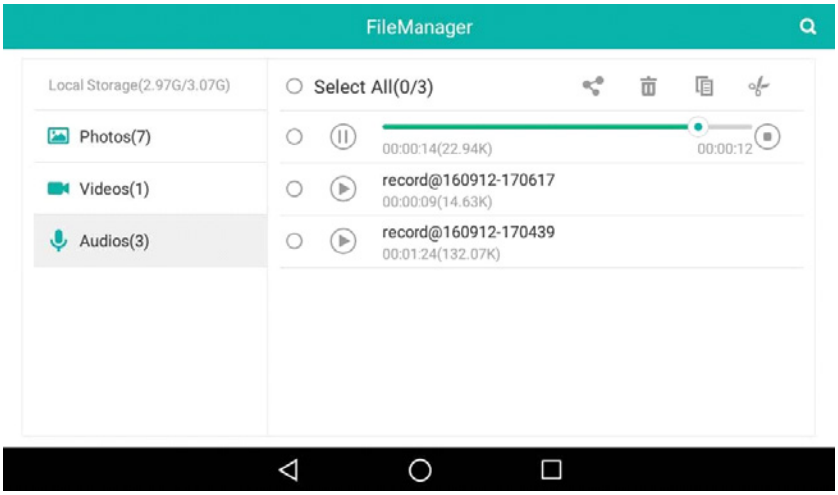
- Swipe left or right to go to the second idle screen.
- Tap to launch **FileManager** application.
- Tap **Audios** in the **Local Storage** field.

  If a USB flash drive has been connected to your phone, you can tap **Audios** in the **USB Storage** field to view the audios stored in USB flash drive.

  Your audios will display as a list.



- Tap the desired audio to play.



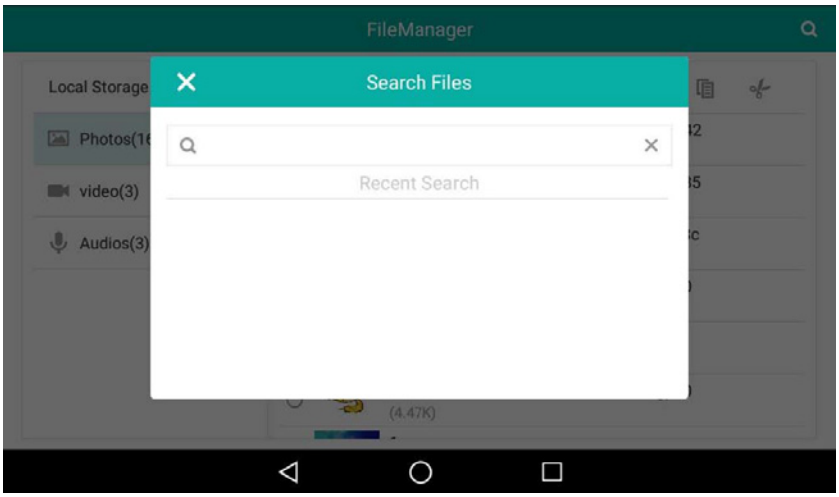
- During the playing, you can do the following:
  - Tap to pause the playing. Then appears on the touch screen.
  - Tap to continue the playing.
  - Drag the slider to skip forward the playing or rewind the playing.
  - Press to adjust the volume of the speakerphone.
  - Tap to stop the playing.

    You can also swipe down from the top of the screen to enter the control center, and then tap **USB** to launch **FileManager** application to view the files if you have connected a USB flash drive to the phone.

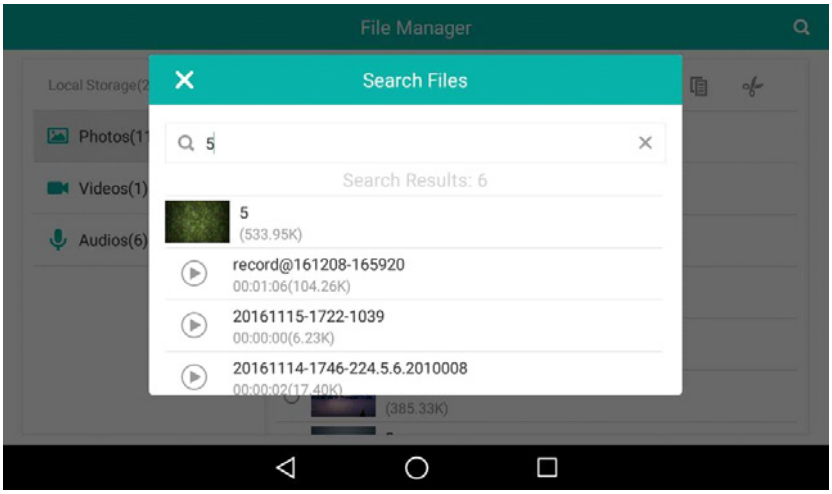
Searching Files

To search files:

- Swipe left or right to go to the second idle screen.
- Tap to launch FileManager application.
- Tap on the top-right of the screen.



- Enter one or more continuous characters of the file name.



The file whose name matches the entered characters will appear on the touch screen. You can drag up and down to scroll through the searching results, and then tap the desired result to view the file.

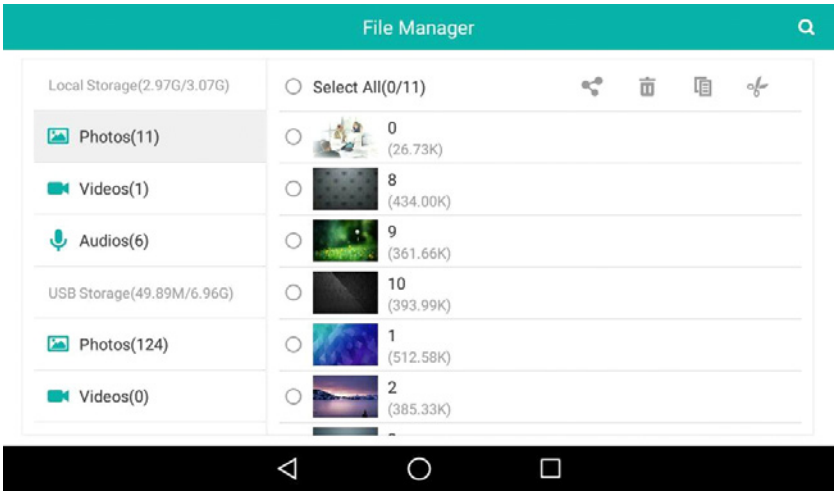
Copying Files

You can copy the files that stored in the USB flash drive to internal SD card, or copy the files that stored in internal SD card to the USB flash drive.

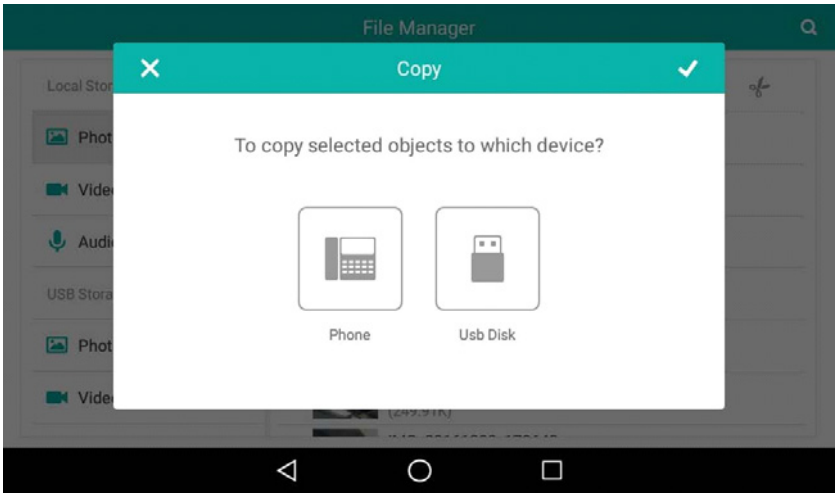
To copy files:

- Swipe left or right to go to the second idle screen.
- Tap to launch **FileManager** application.
- Tap **Photos/Videos/Audios** in **Local Storage** field.
- If a USB flash drive has been connected to your phone, you can tap **Photos/Videos/Audios** in the **USB Storage** field to view the files stored in USB flash drive.

Your files will display as a list.

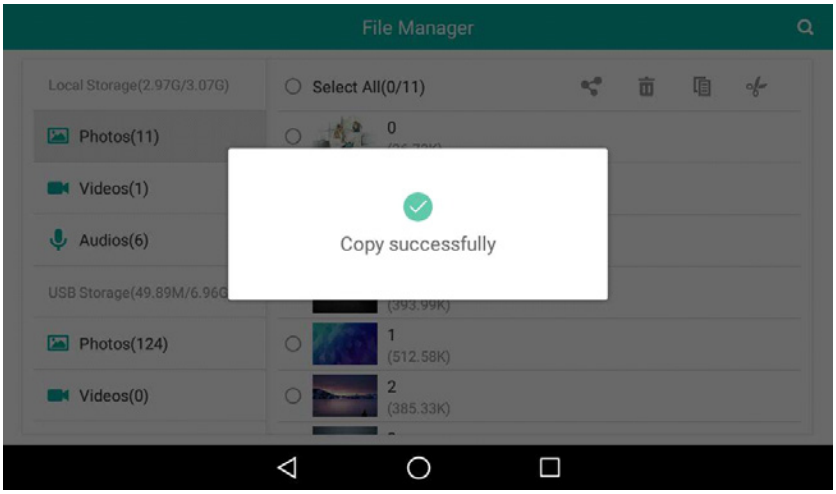


- Drag up and down to scroll through the files if required.
- Tap the checkbox(es) before the desired file(s).
- Tap  .
- The following prompt will appear on the touch screen of the phone:



- Tap the desired storage device to copy the file to.
- Tap  to save the change or  to cancel.

- If the file has been copied successfully, the touch screen will prompt “Copy successfully”.



- You can also tap the **Select All** checkbox to copy all files.

Cutting Files (Moving Files)

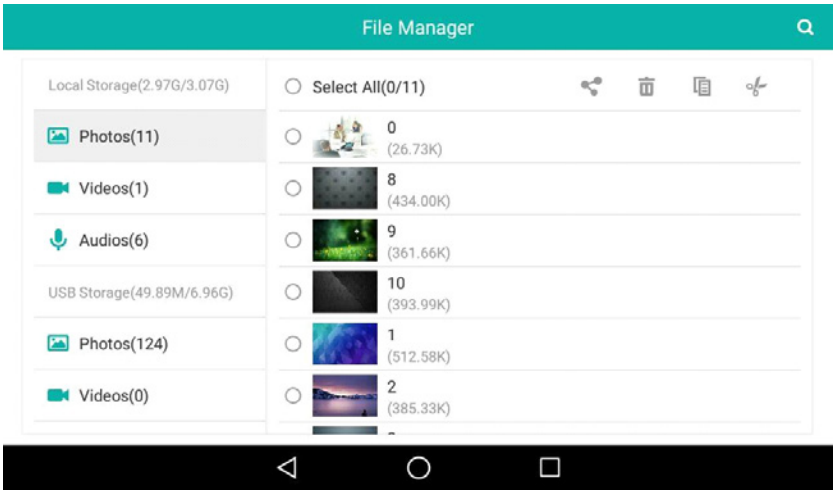
The operating instructions on moving a file are similar to copying a file. The difference is that once the file is moved, the file will not be kept in the source storage device.

To cut files:

- Swipe left or right to go to the second idle screen.
- Tap  to launch FileManager application.
- Tap **Photos/Videos/Audios** in **Local Storage** field.

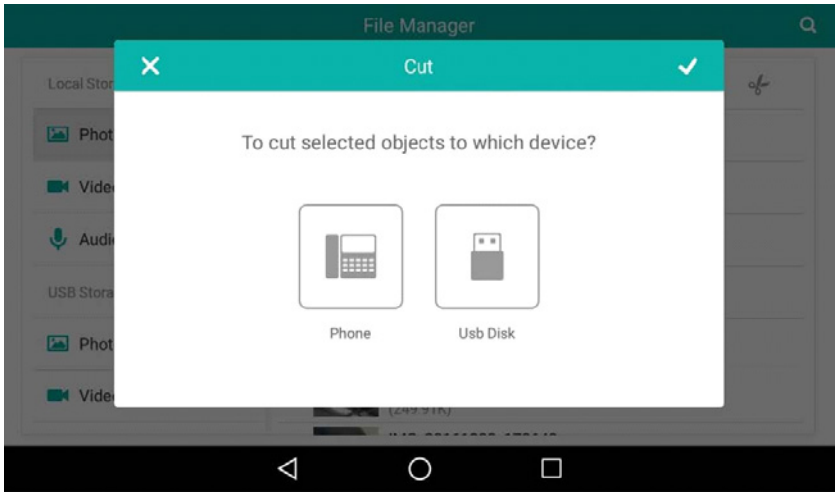
  If a USB flash drive has been connected to your phone, you can tap **Photos/Videos/Audios** in the **USB Storage** field to view the files store in USB flash drive.

  Your files will display as a list.



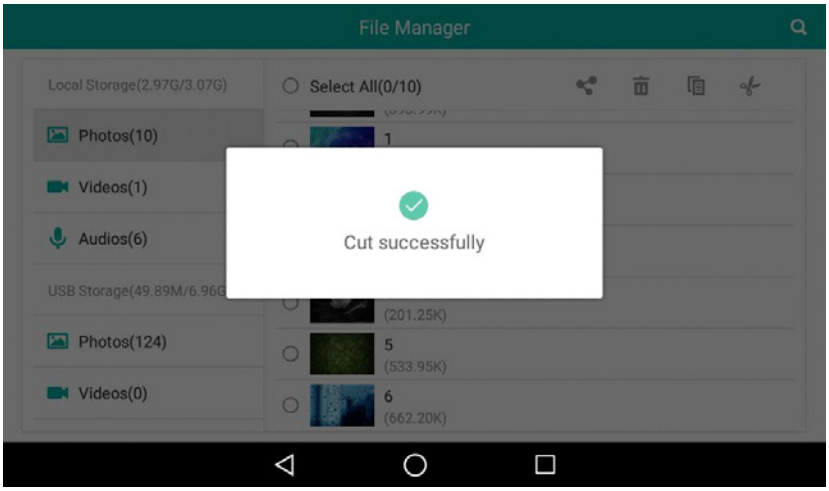
- Drag up and down to scroll through the files if required.
- Tap the checkbox(es) before the desired file(s).
- Tap .

  The following prompt will appear on the touch screen of the phone:



- Tap the destination storage device.
- Tap to save the change or to cancel.

  If the file is cut successfully, the touch screen will prompt “Cut successfully”.



- You can also tap the **Select All** checkbox to cut all files.

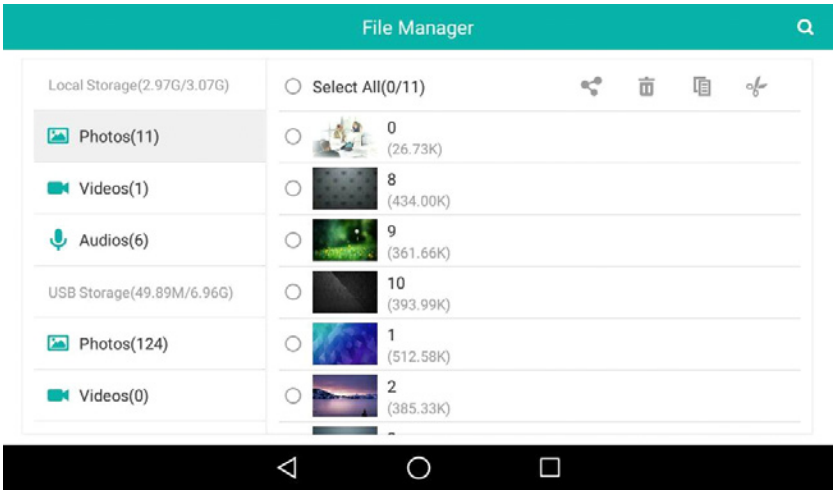
Deleting Files

To delete files:

- Swipe left or right to go to the second idle screen.
- Tap to launch **FileManager** application.
- Tap **Photos/Videos/Audios** in **Local Storage** field.

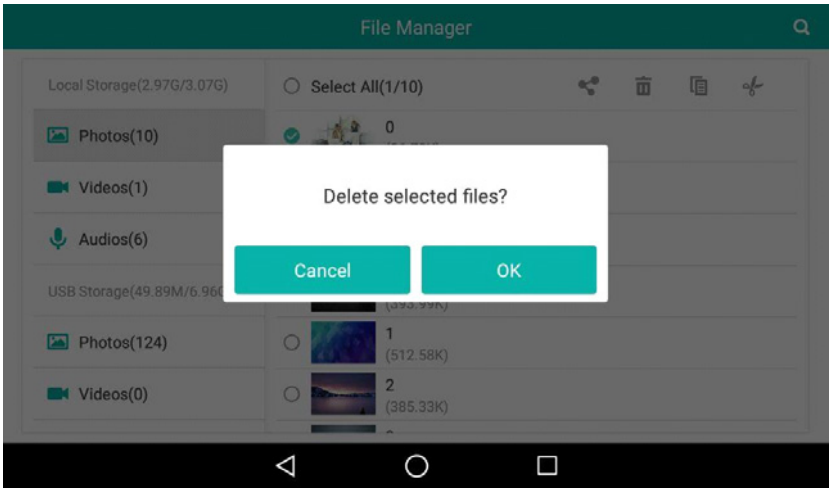
If a USB flash drive has been connected to your phone, you can tap **Photos/Videos/Audios** in the **USB Storage** field to view the files store in USB flash drive.

Your files will display as a list.



- Drag up and down to scroll through the files if required.
- Tap the checkbox(es) before the desired file(s).
- Tap  .

The following prompt will appear on the touch screen of the phone:



- Tap **OK** to accept the change or **Cancel** to cancel.
- You can also tap the **Select All** checkbox to delete all files.

Sharing Pictures/Videos/Audios

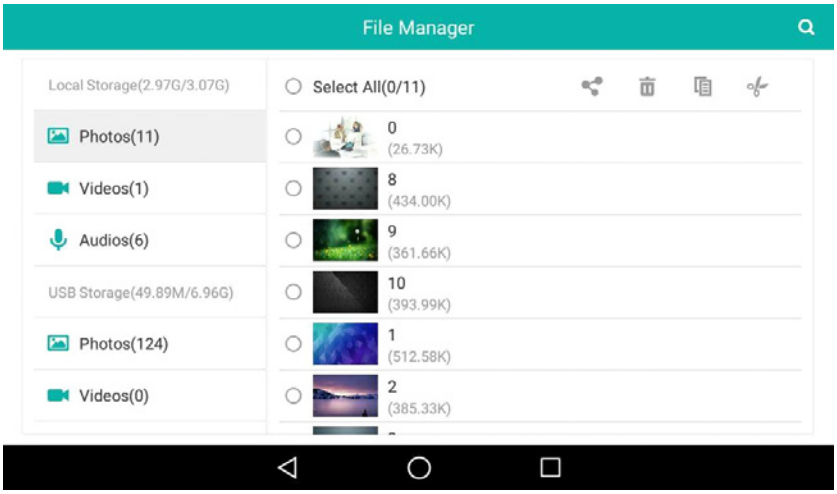
You can share pictures/videos/audios with other device via Bluetooth or email. Before sharing pictures/videos/audios via Bluetooth, make sure the Bluetooth device is discoverable. The operating instructions on sharing pictures, video and audios are the same, the following take sharing pictures as example.

To share picture(s) via Bluetooth:

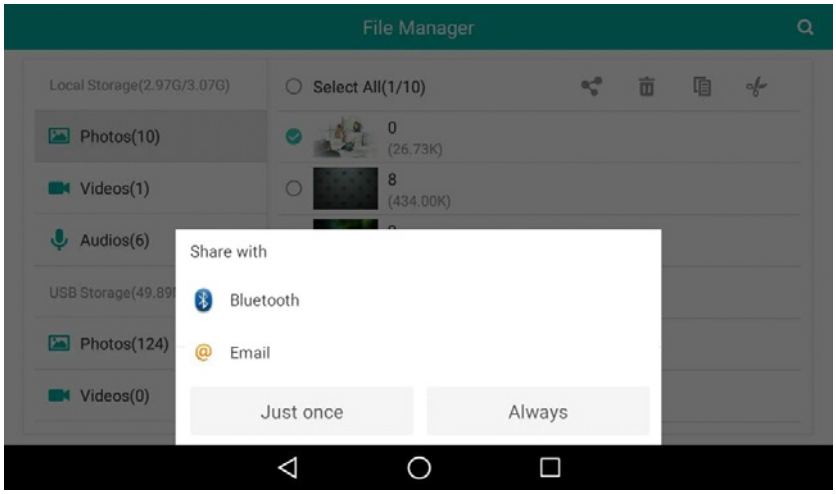
- Swipe left or right to go to the second idle screen.
- Tap to launch FileManager application.
- Tap **Photos** in the **Local Storage** field.

  If a USB flash drive has been connected to your phone, you can tap Photos in the USB Storage field to view the pictures stored in USB flash drive.

  Your pictures will display as a list.

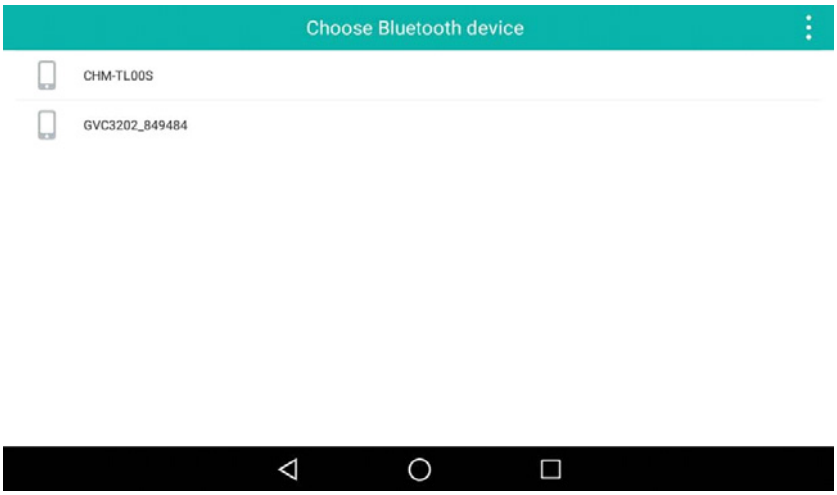


- Drag up and down to scroll through the files if required.
- Tap the checkbox(es) before the desired picture(s).
- Tap .



- Tap **Bluetooth** and then tap **Always** or **Just once**.

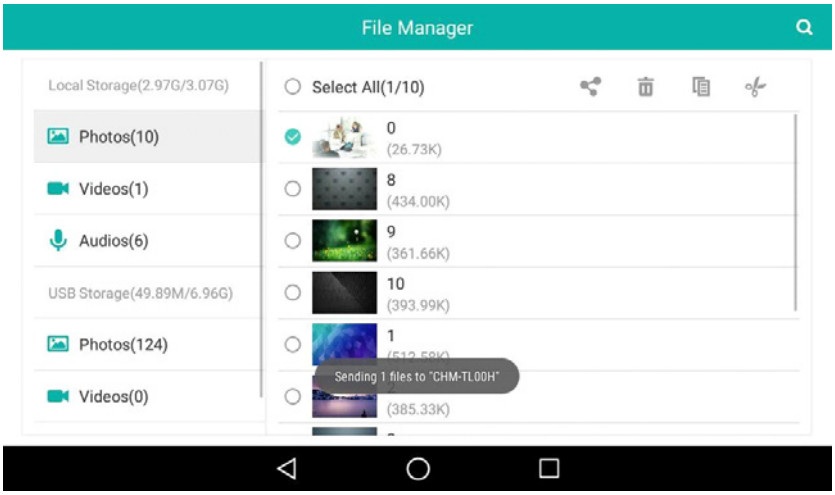
  The IP phone scans and displays the available Bluetooth device automatically.



- If there is no Bluetooth device found on the touch screen, tap ⋮ and then **Refresh** to search for the Bluetooth devices.

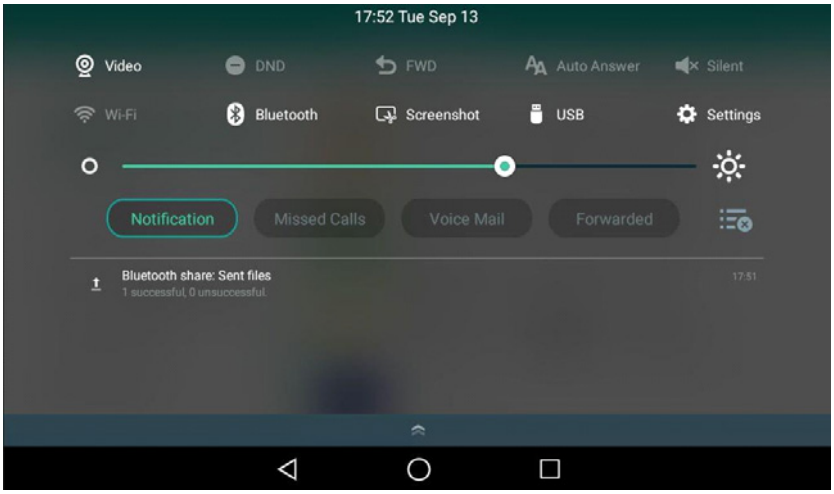
If the Bluetooth mode of the IP phone is deactivated, the touch screen will prompt "To use Bluetooth services, you must first turn on Bluetooth. Turn on Bluetooth now?". Tap Turn on to activate the Bluetooth mode.

- Tap the desired Bluetooth device to share the pictures.
- The touch screen will prompt "Sending n file to "XXX"" (n represents the number of pictures that is being send).



- Accept the Bluetooth file transfer request on the Bluetooth device.

If the pictures are successfully transferred, the notification center will display a notification “Bluetooth share: Sent files n successful, n unsuccessful.” (n represents the number of files that are sent successfully or unsuccessfully).



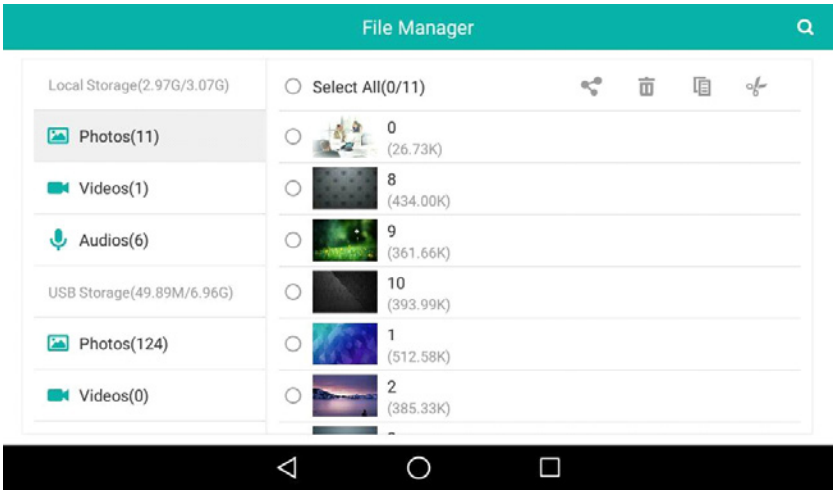
You can tap the notification to view the detail information about the shared pictures.

To share picture(s) via email:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Gallery** application.
- Tap **Photos** in the **Local Storage** field.

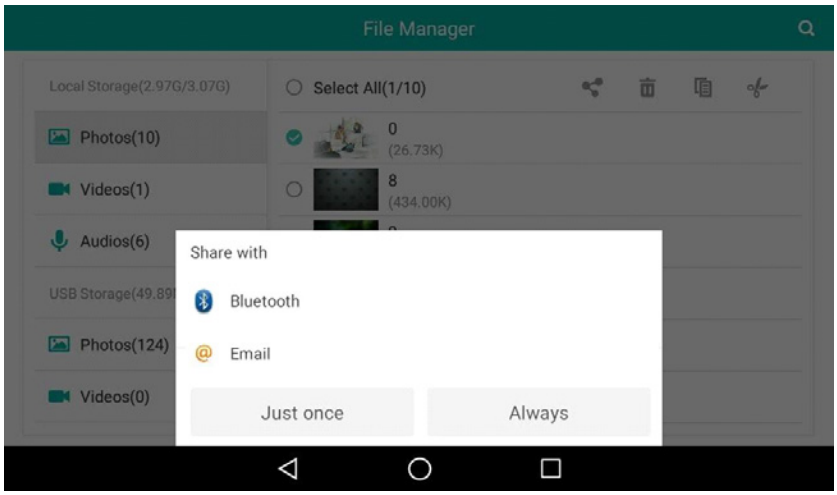
  If a USB flash drive has been connected to your phone, you can tap **Photos** in the **USB Storage** field to view the pictures stored in USB flash drive.

- Your pictures will display as a list.

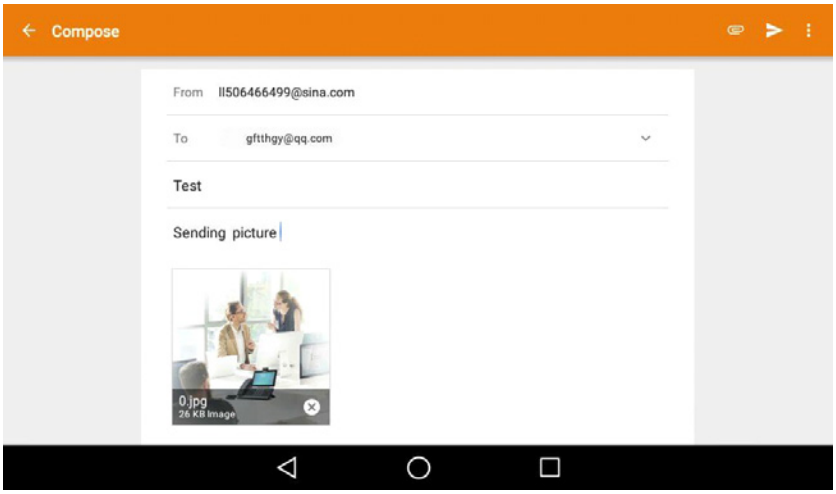


- Drag up and down to scroll through the files if required.
- Tap the checkbox(es) before the desired picture(s).

- Tap .



- Tap **Email** and then tap **Always** or **Just once**.
- Sign in an email account if required (refer to “Adding APP Accounts” (Optional).
- Compose the Email.
- Enter the email account you want to send the email to in the **To** field.
- Enter the email subject in the **Subject** field (Optional).



- Tap to send the email.
- The pictures are transferred as mail attachment. The attachment files size must be within 5MB. You can also share pictures with other devices via Bluetooth or email when previewing the pictures. And the operating instructions are the same with sharing pictures by Gallery application. For more information, refer to “Sharing Pictures/Videos”.

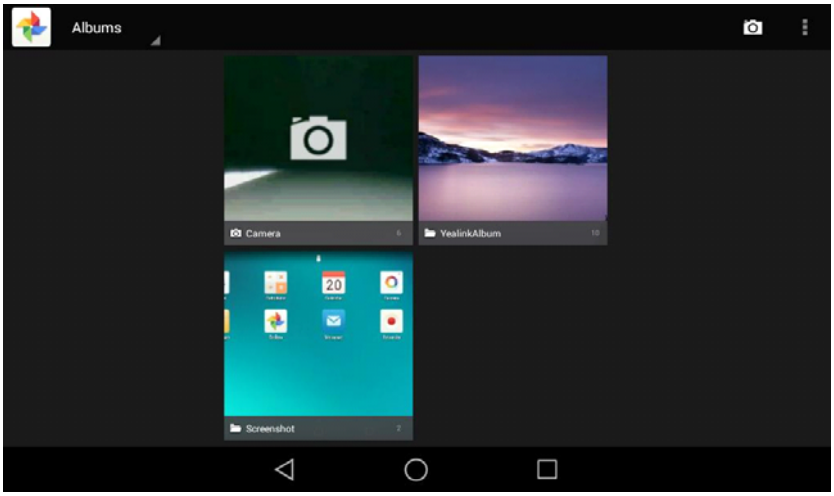
Gallery

You can use Gallery application to view pictures/videos stored in internal SD card/USB flash drive or take pictures/videos. You can also perform basic editing tasks on pictures or set the desired picture as the wallpaper. The pictures/videos can be shared with friends via Bluetooth or email.

Viewing Pictures/Videos

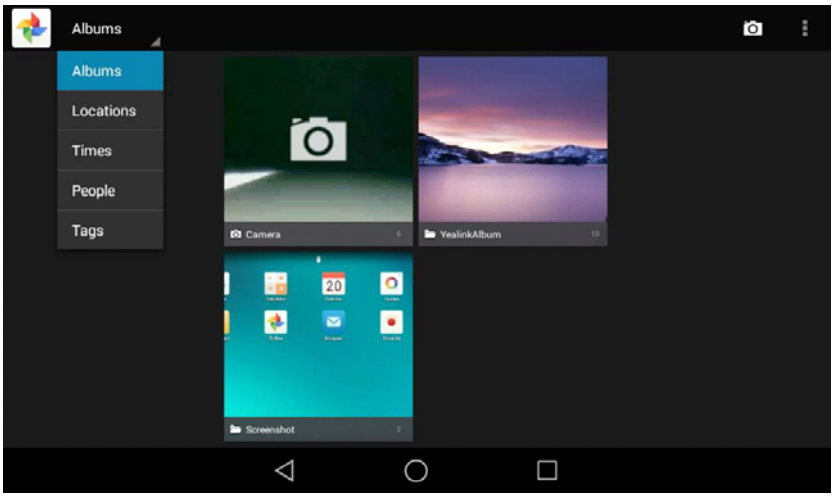
To view pictures/videos:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Gallery** application.
- By default, the pictures/videos are displayed as thumbnails in an albums arrangement mode.



- Tap **Albums** (Optional).

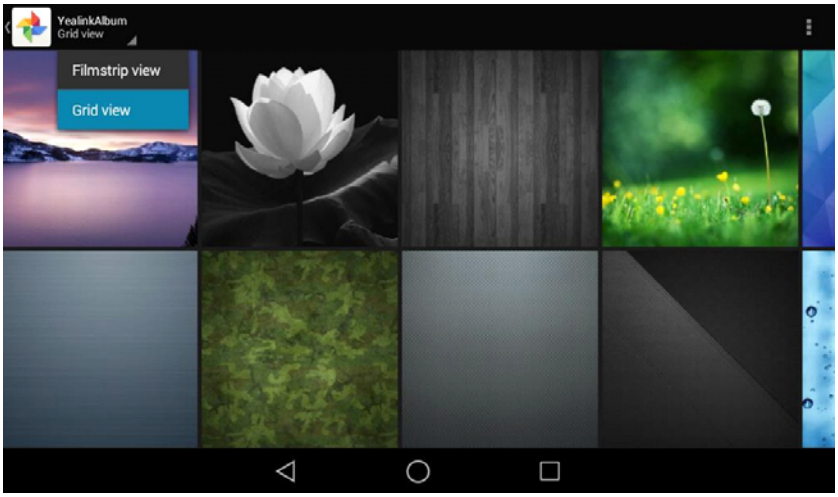
  You can tap **Albums/Locations/Times/People/Tags** to view the pictures/videos in different arrangement modes.



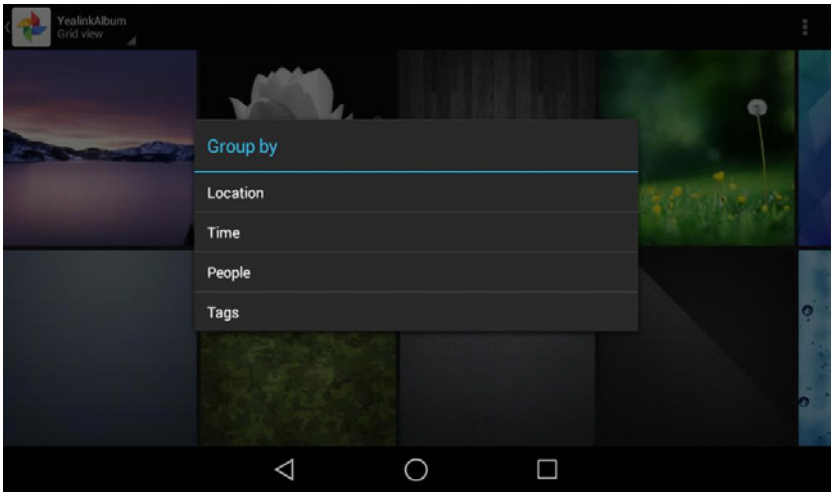
- Swipe left/right to view all albums if required.

- You can do the following:
  - If you tap a picture album, you can do the following:
    - Tap the album name on top-left of the touch screen.

      Tap **Filmstrip view** or **Grid view** to view the pictures in different modes.



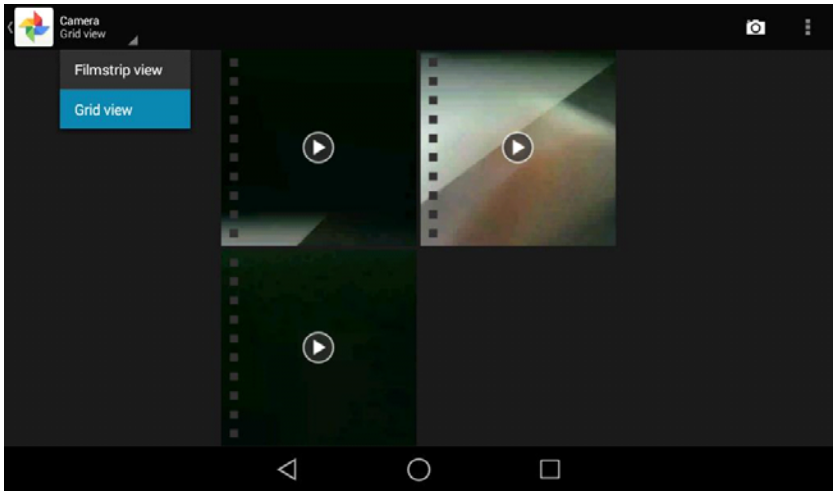
    - Tap and then tap **Slideshow** to start a slide show of the pictures in the album.
    - Tap and then tap **Group by**.



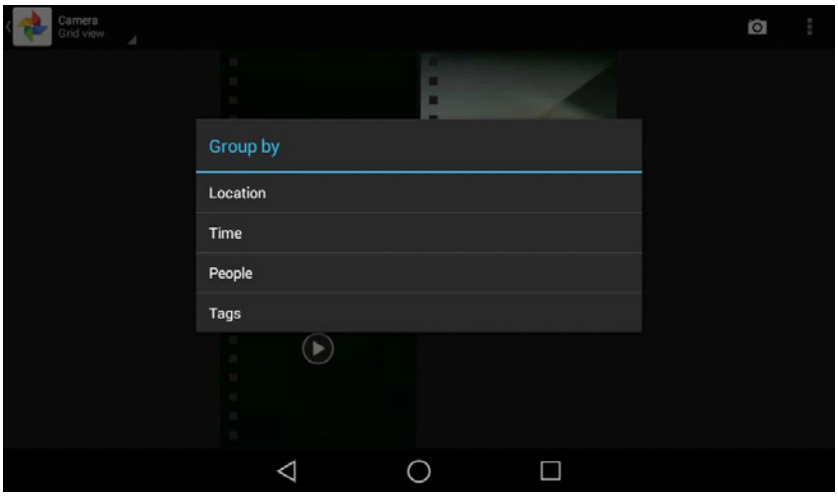
      Tap a desired group type in the pop-up dialog box to view the pictures in different arrangement modes.

    - Tap a desired picture to preview the picture.
  - If you tap a video album, you can do the following:
    - Tap the album name on top-left of the touch screen.

Tap **Filmstrip view** or **Grid view** to view the videos in different mode.



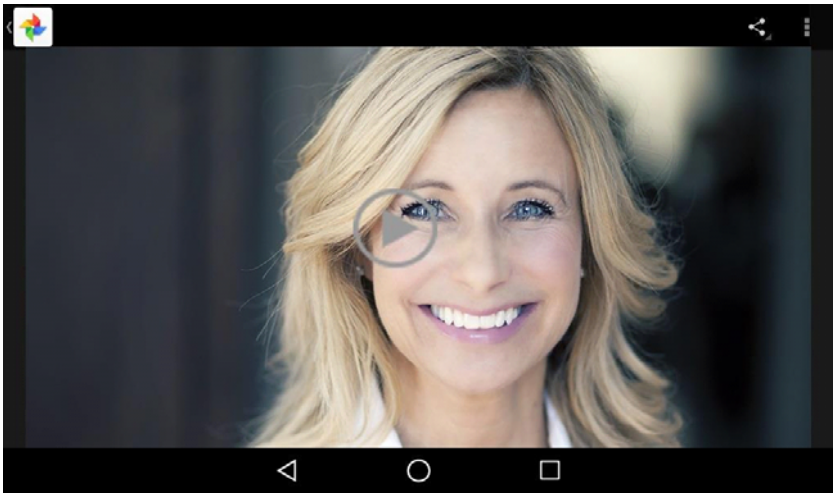
- Tap and then tap **Group by**.



Tap a desired group type in the pop-up dialog box to view the videos in different arrangement modes.

- Tap a desired video.

Videos show the icon on the preview thumbnail. Tap to play the video.



Editing Pictures

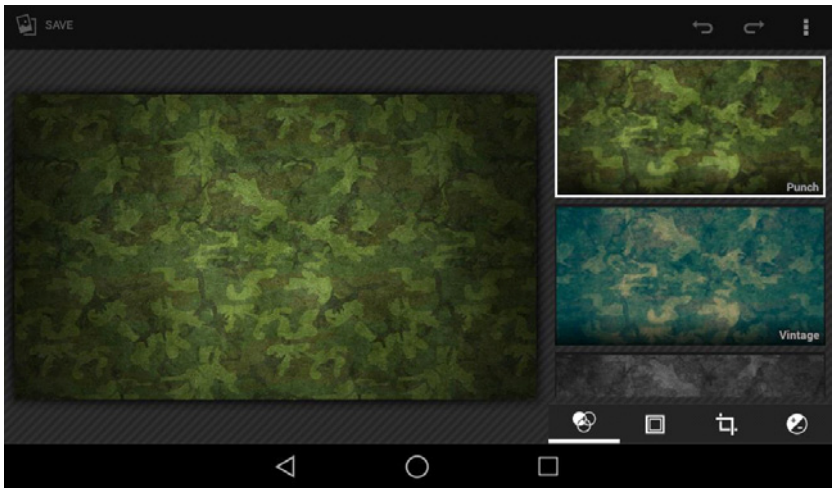
To edit a picture:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Gallery** application.
- Tap a desired picture album.
- Tap the desired pictures.
- You can do the following:
  - Tap and then tap **Rotate left/Rotate right** to rotate the picture left/right.

    OR
  - Tap and then tap **Crop** to resize the picture.

    OR
  - Tap and then tap **Edit**.

The touch screen will display as below



- You can do the following:
  - Tap  to add filters.

    OR
  - Tap  to add a picture frame.

    OR
  - Tap  .
    - Tap **Crop** to resize the picture.
    - Tap **Straighten** to straighten the picture.
    - Tap **Rotate** to rotate the picture clockwise.
    - Tap **Mirror** to flip the picture symmetrically.
    - Tap **Draw** to draw the picture.

    OR
  - Tap  to adjust the exposure, contrast, saturation, etc.
- Tap **SAVE** to accept the change.

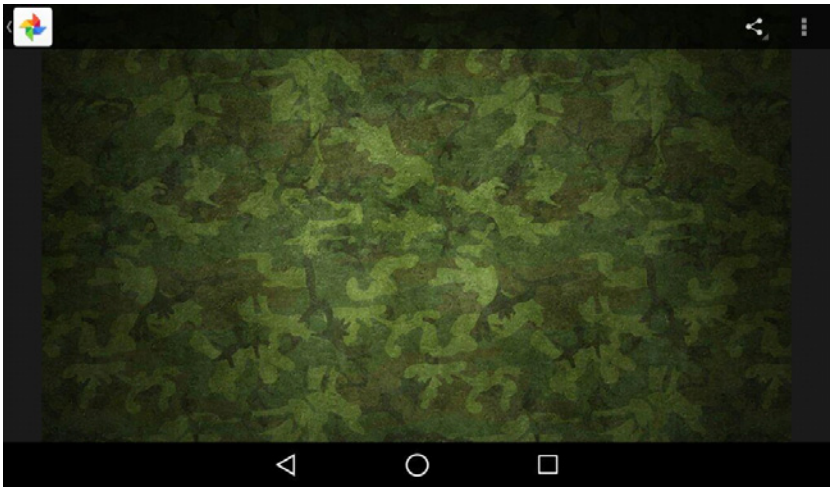


The menu keys and Android keys will disappear within 5 seconds, you may tap the touch screen to show them again.

Setting a Picture as Wallpaper

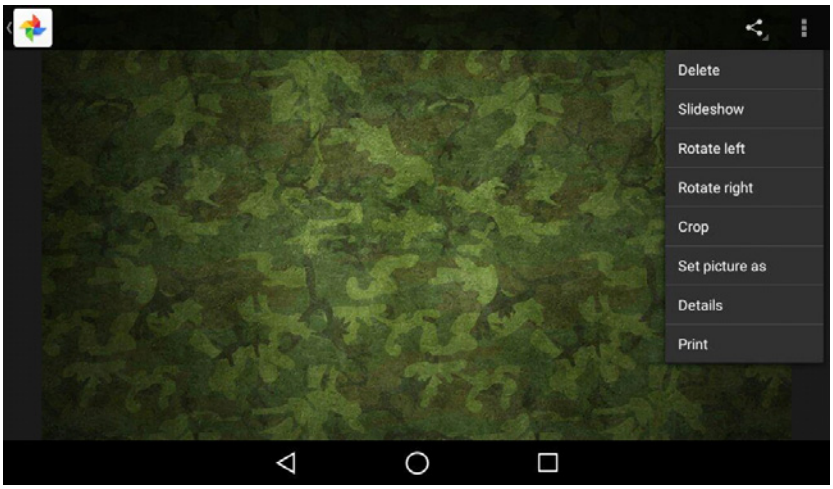
To set a picture as wallpaper:

- Preview the picture (refer to “Viewing Pictures/Videos”).

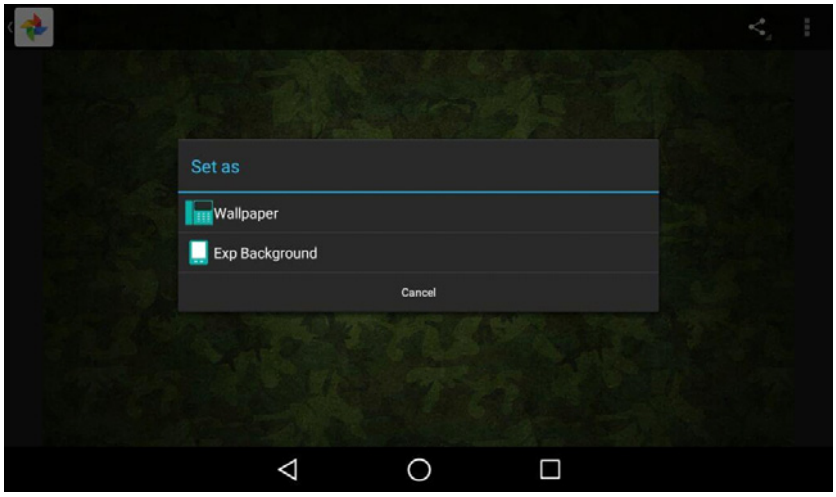


- Tap .

  The touch screen will display as below:



- Tap **Set picture** as.



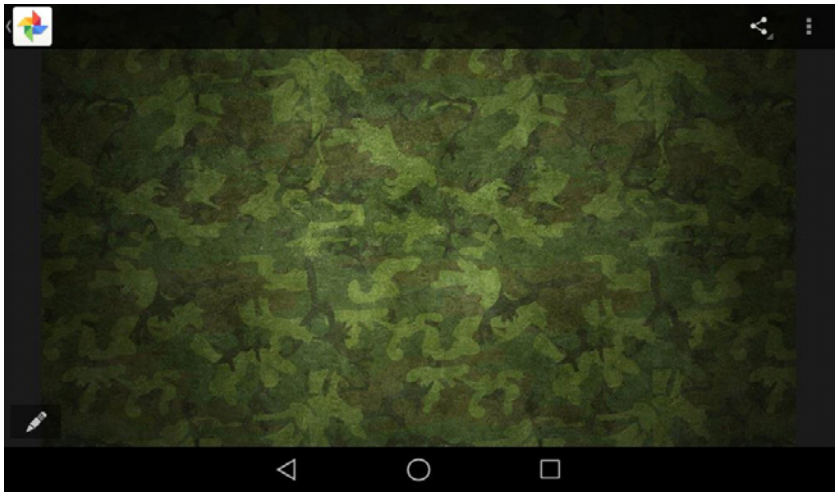
- Tap **Wallpaper** and then tap **Set wallpaper t**o set the picture as phone wallpaper.

The menu keys and Android keys will disappear within 5 seconds, you may tap the touch screen to show them again.

Printing Pictures

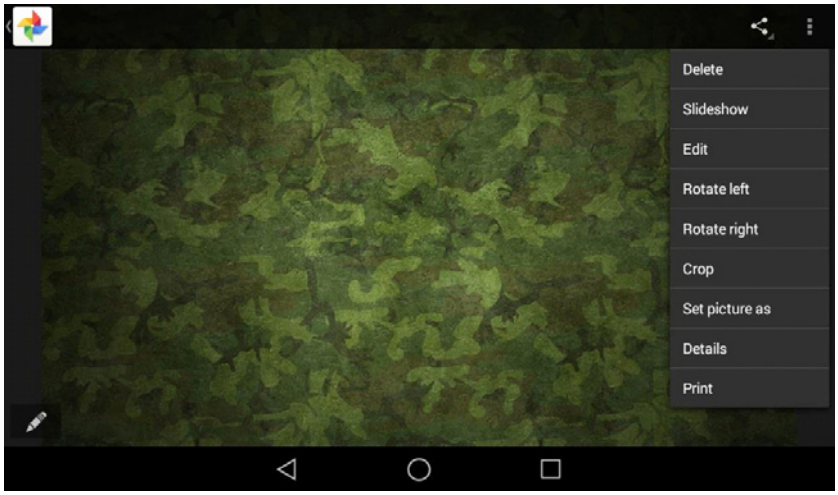
To print a picture:

- Preview the picture (refer to “Viewing Pictures/Videos”).



- Tap .

The touch screen will display as below:



- Tap **Print** to print the picture.

  Before printing the picture, make sure your phone has connected to the network printer. For more information, refer to the documentation from the printer manufacturer.

Viewing Detail Information of the Pictures/Videos

The operating instructions on viewing detail information of pictures and videos are the same, the following take viewing detail information of pictures as example.

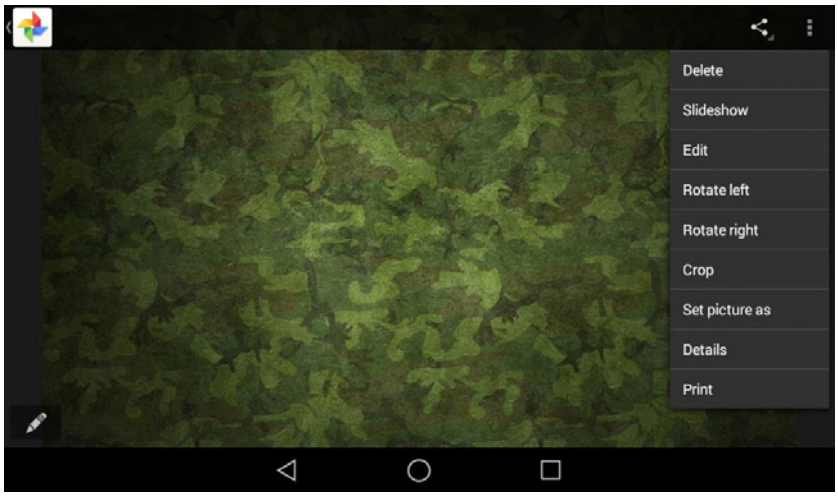
To view detail information of the picture:

- Preview the picture (refer to “Viewing Pictures/Videos”).

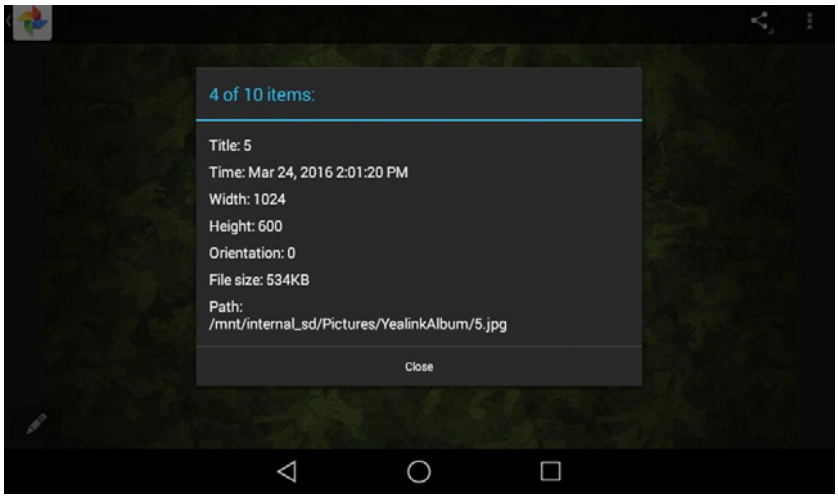


- Tap .

The touch screen will display as below:



- Tap **Details** to view the detail information of the picture.



- Tap **Close** to close the pop-up box.

Deleting Pictures/Videos

The operating instructions on deleting pictures and videos are the same. Deleting pictures is taken as an example.

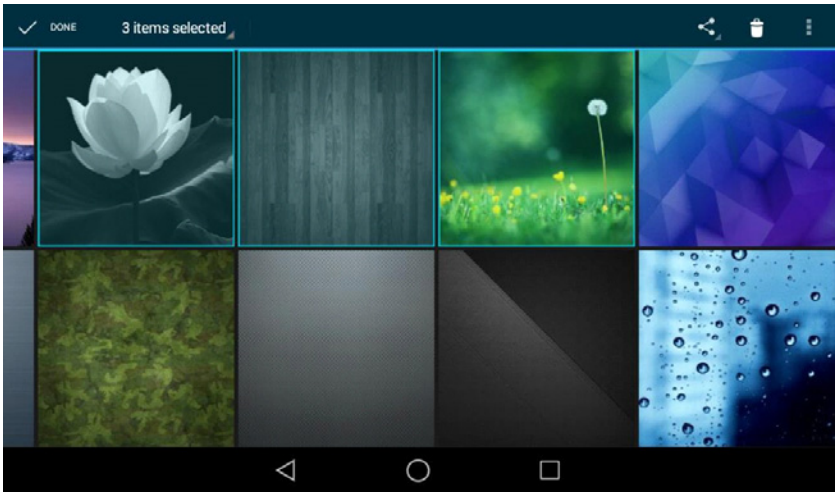
To delete picture(s):

- Swipe left or right to go to the second idle screen.
- Tap to launch **Gallery** application.
- Tap a desired picture album.
- Do one of the following:
  - Long tap one of the pictures.
    Tap on other pictures to select multiple pictures.

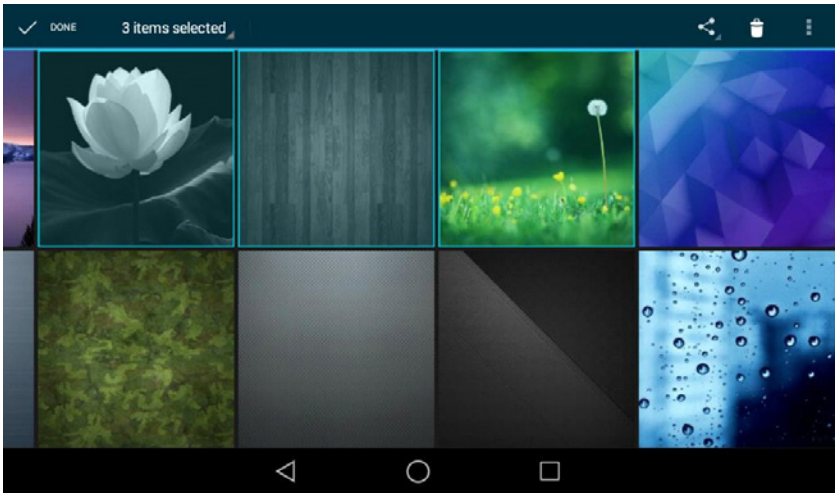
  OR

  - Tap and then tap **Select item**.

Tap the desired picture(s) to select multiple pictures.

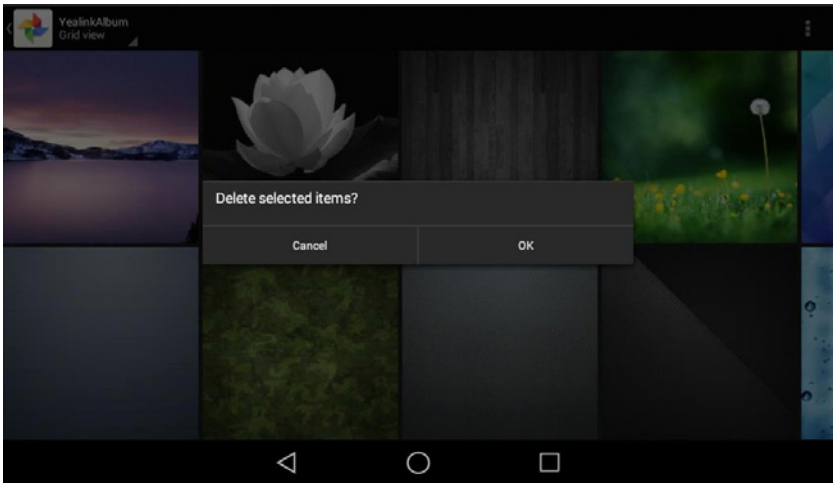


You can also tap **n items selected** (n represents the number of pictures you selected) and then tap **Select all** to select all pictures.



- Tap  to delete the selected picture(s).

  The touch screen prompt “Delete selected item?”.



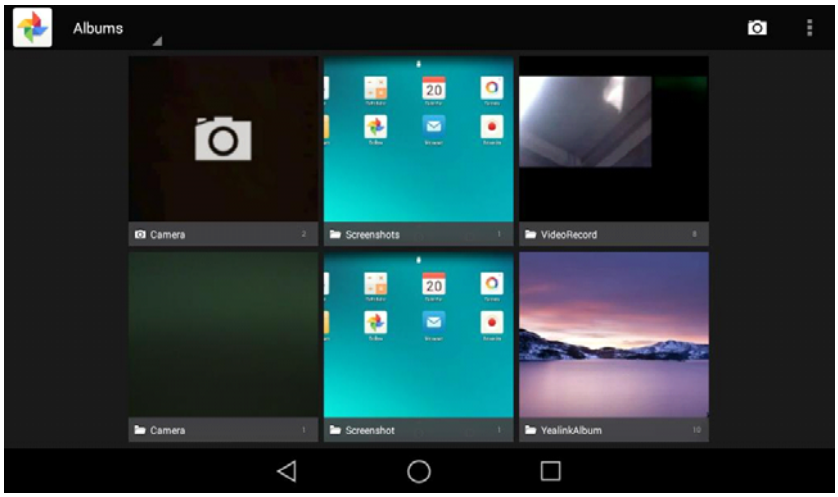
- Tap **OK** to save the change or **Cancel** to cancel.

Taking a Picture/Video

Before taking a picture/video, you should connect the camera to the IP phone. For more information on how to connect the camera, refer “Phone Installation”.

To take a picture/video:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Gallery** application.
- Tap on the top-right of the touch screen.



- Take a picture or video (refer to “Taking Pictures” or “Recording Videos”).

Sharing Pictures/Videos

You can share pictures/videos with other Bluetooth devices via Bluetooth or email. Before sharing pictures/videos via Bluetooth, make sure the Bluetooth device is discoverable. The operating instructions on sharing pictures and sharing video are the same, the following take sharing pictures as example.

To share picture(s) via Bluetooth:

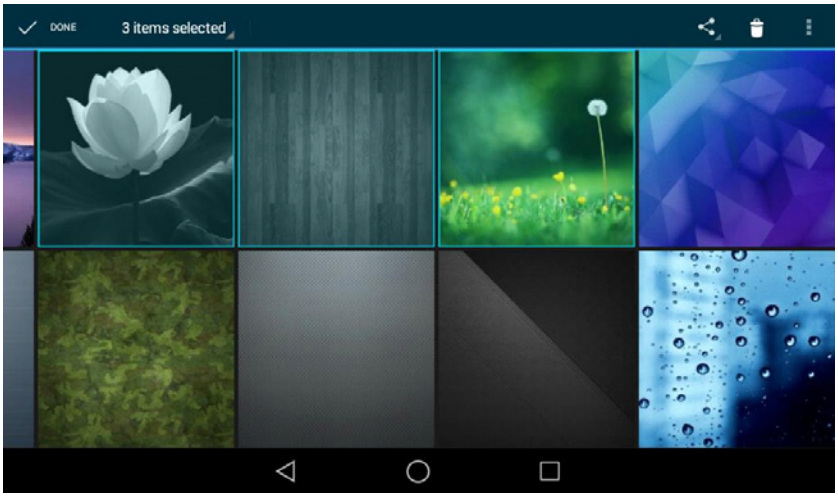
- Swipe left or right to go to the second idle screen.
- Tap to launch **Gallery** application.
- Tap a desired picture album.
- Do one of the following:
- Long tap one of the pictures.

  Tap on other pictures to select multiple pictures.

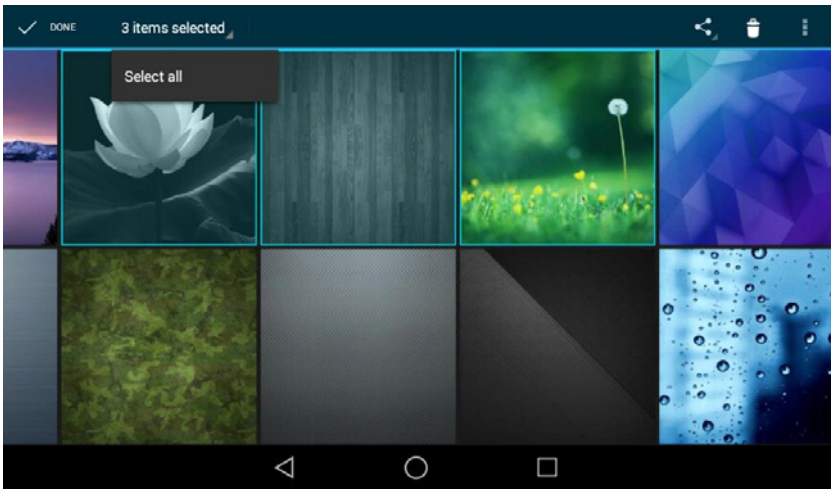
  OR

- Tap and then tap **Select item**.

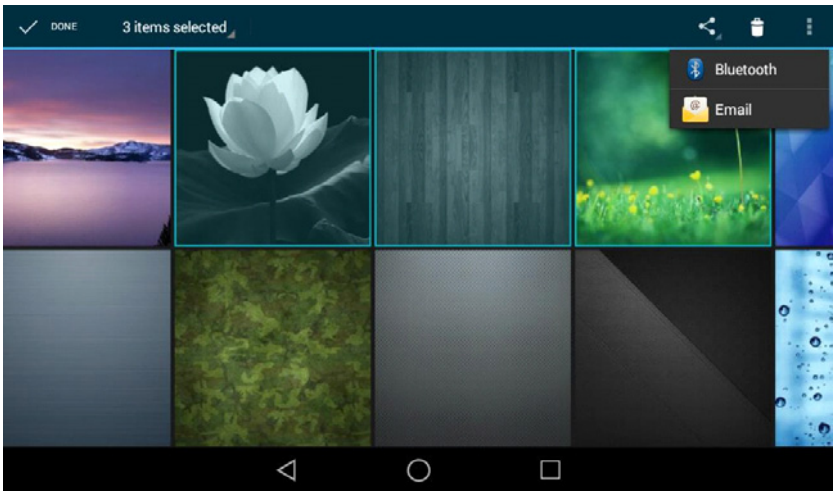
  Tap the desired picture(s) to select multiple pictures.



You can also tap **n items selected** (n represents the number of pictures you selected) and then tap **Select all** to select all pictures.

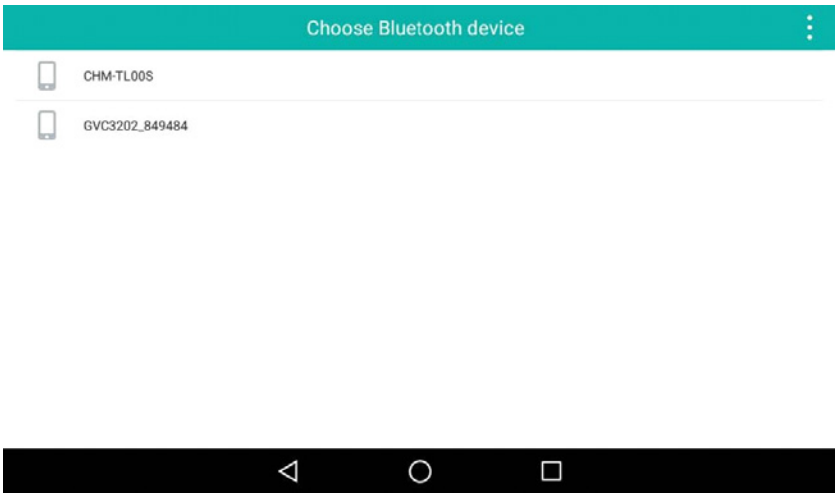


- Tap  .



- Tap **Bluetooth**.

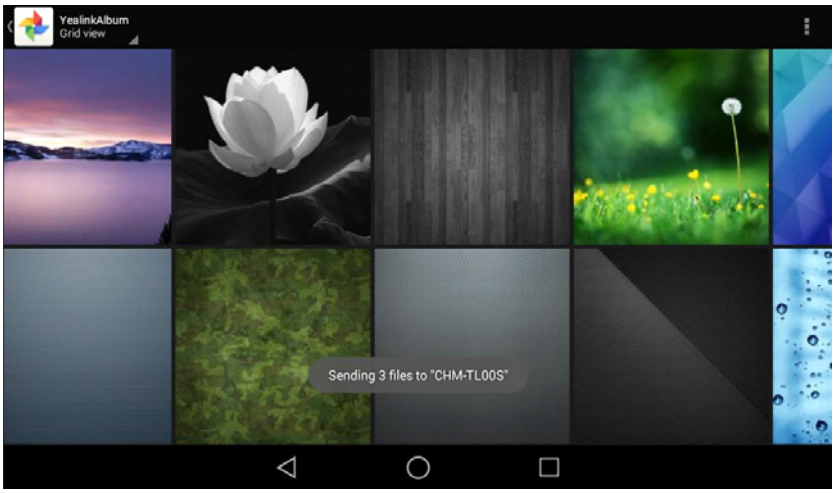
  The IP phone scans and displays the available Bluetooth device automatically. If there is no Bluetooth device found on the touch screen, tap and then **Refresh** to search for the Bluetooth devices.



If the Bluetooth mode of the IP phone is deactivated, the touch screen will prompt "To use Bluetooth services, you must first turn on Bluetooth. Turn on Bluetooth now?". Tap Turn on to activate the Bluetooth mode.

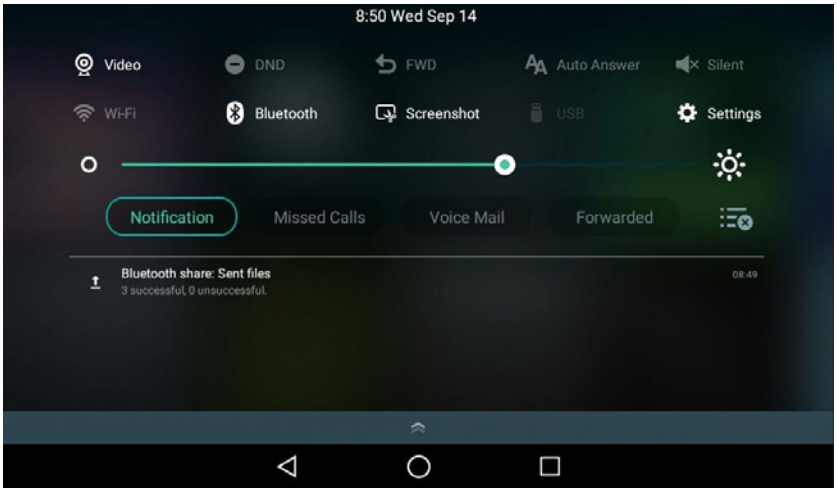
- Tap the desired Bluetooth device to share the pictures.

The touch screen will prompt “Sending n file to “XXX”” (n represents the number of pictures that is being send).



- Accept the Bluetooth file transfer request on the Bluetooth device.

  If the pictures are successfully transferred, the notification center will display a notification “Bluetooth share: Sent files n successful, n unsuccessful.” (n represents the number of files that are sent successfully or unsuccessfully).



You can tap the notification to view the detail information about the shared pictures.

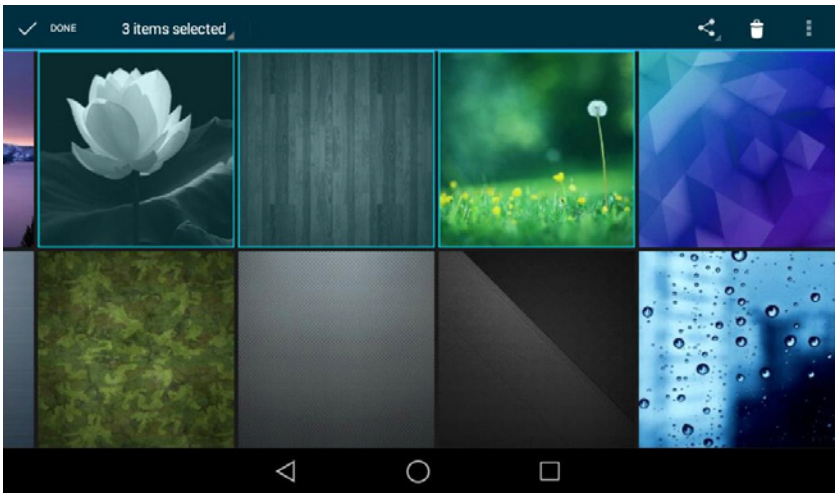
To share picture(s) via email:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Gallery** application.
- Tap a desired picture album.
- Do one of the following:
  - Long tap one of the pictures.
    Tap on other pictures to select multiple pictures.

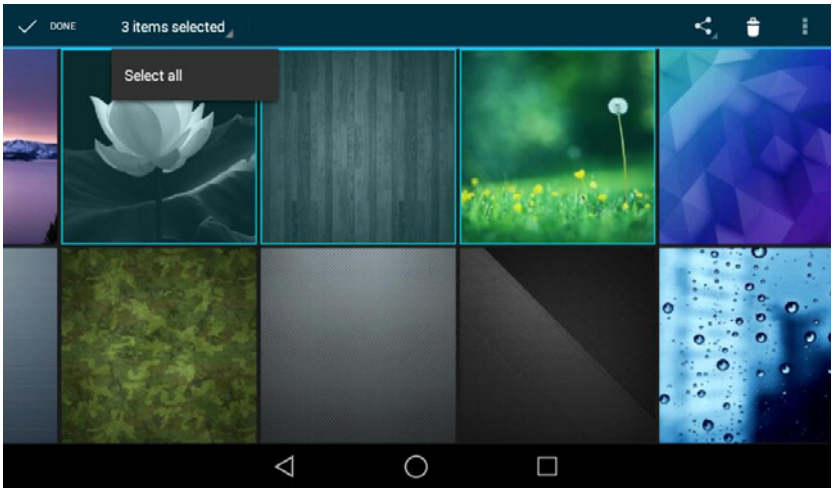
  OR

- Tap and then tap **Select item**.

  Tap the desired picture(s) to select multiple pictures.

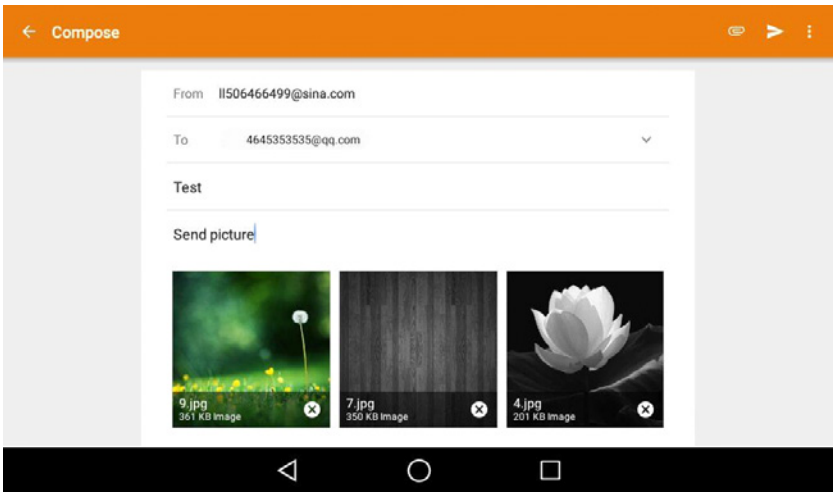


You can also tap **n items selected** and then tap **Select all** to select all pictures.



- Tap .
- Tap **Email**.
- Sign in an email account if required (Optional).Refer to “Signing In Email”.
- Compose the Email.
- Enter the email account you want to send the email to in the **To** field.

- In the **Subject** field.(optional), enter the email subject.



- Tap **SEND** to send the email.

  The pictures are transferred as mail attachment. The attachment files size must be within 5MB.

Recorder

You can use **Recorder** application to record audios and manage the recorded audios. During recording, you can mark specific moments so that you can quickly find and listen to them later. There is a time limit (12 hours) on the recorder. The recorded audios are saved with the name consisting of a prefix “record@” and date & time stamp, for example, record@20160303-092310 was created on March 3, 2016, at 09:23:10.

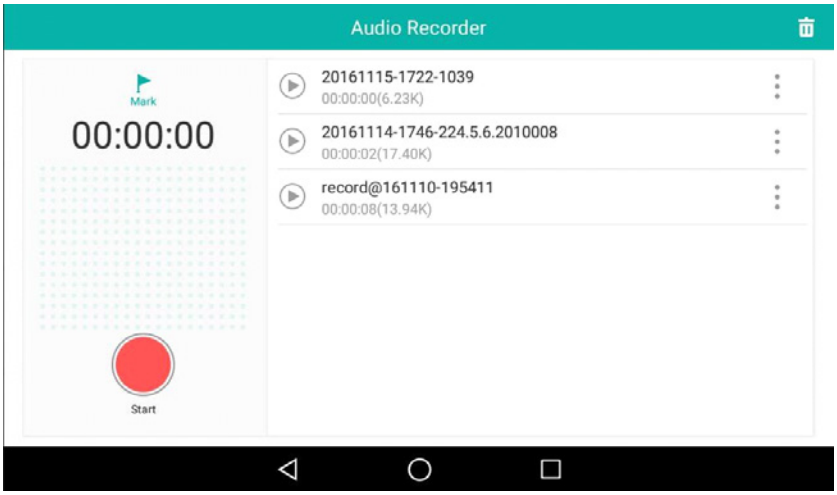
By default, the recorded audios are saved in the internal SD card. But if there is a USB flash drive connected to your phone, the recorded audios will be saved according to the priority: USB flash drive>internal SD card.

*The recorded audios are saved in *.amr format. You can play the recorded audios on either the phone itself or on a computer using an application capable of playing *.amr files.*

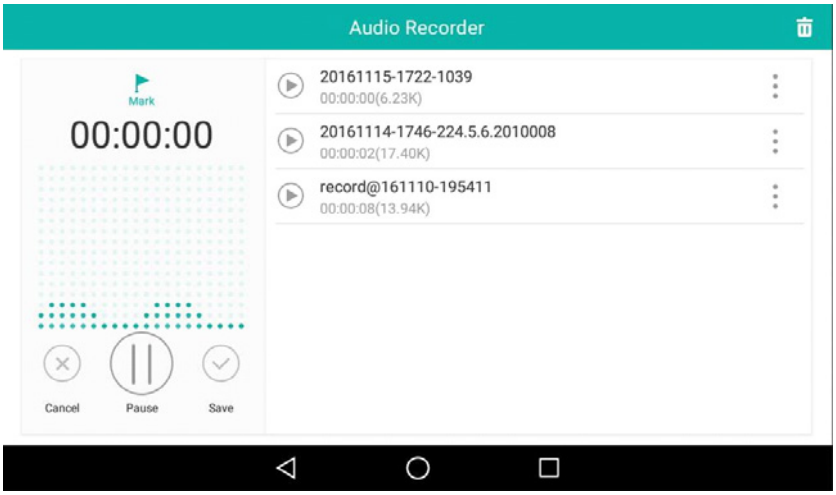
Recording Audios

To record an audio:

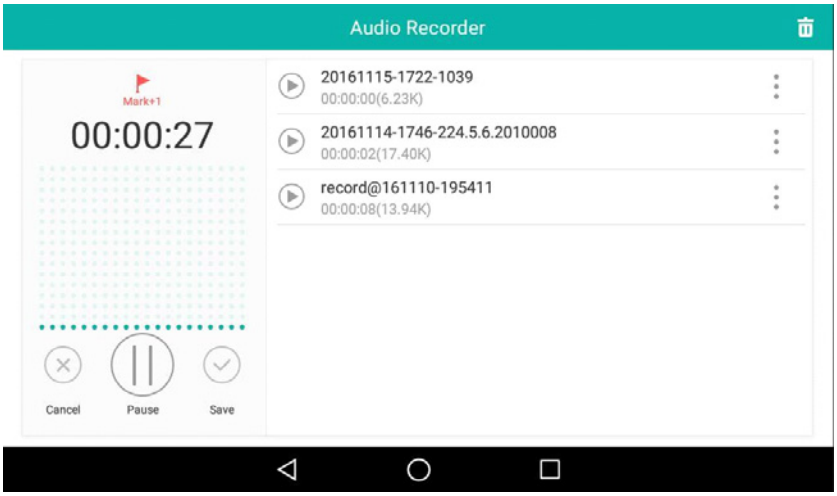
- Swipe left or right to go to the second idle screen.
- Tap  to launch **Recorder** application.



- Tap  to start recording.
- You can do one of the following:
  - Tap  to pause the recording, and then tap  to resume the recording.



- Tap Mark to set a mark.



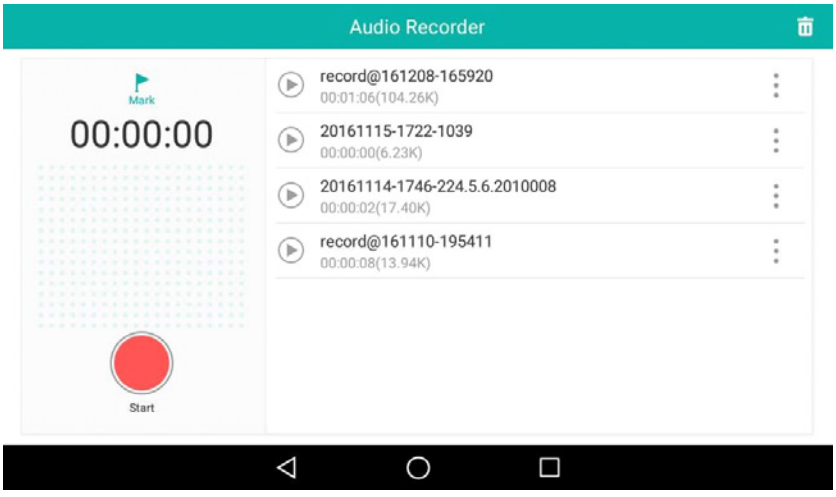
- The icon changes to Mark+1, the number will increase if you set more marks. And there is a red flag on this moment when playing the recorded audio.
- Tap to end the recording and save the recorded audio, or tap to cancel the recording.
- The recorded audio will be displayed on the right list automatically.

Managing Recorded Audios

You can also use the Recorder application to play/delete/rename the recorded audios.

To play the recorded audio:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Recorder** application.



The recorded audios will display as a list.

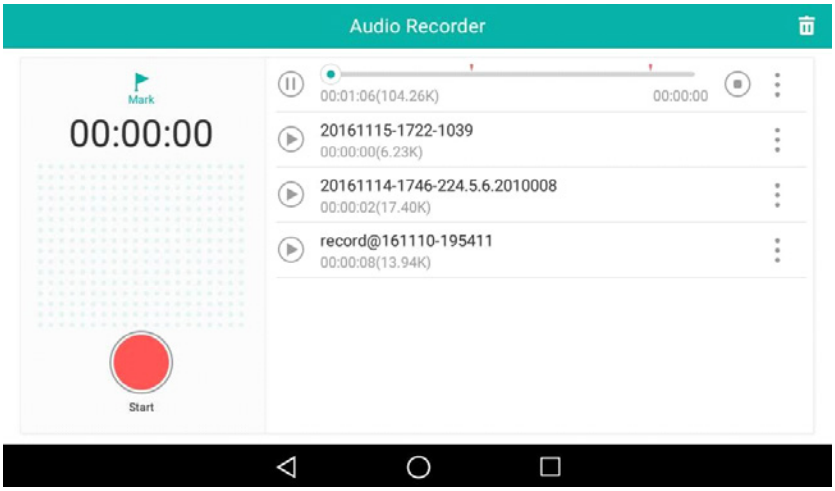
- Tap before the desired recorded audio to play the recorded audio.

During the playing, you can do the following:

- Tap to pause the playing.

To continue the playing, tap .

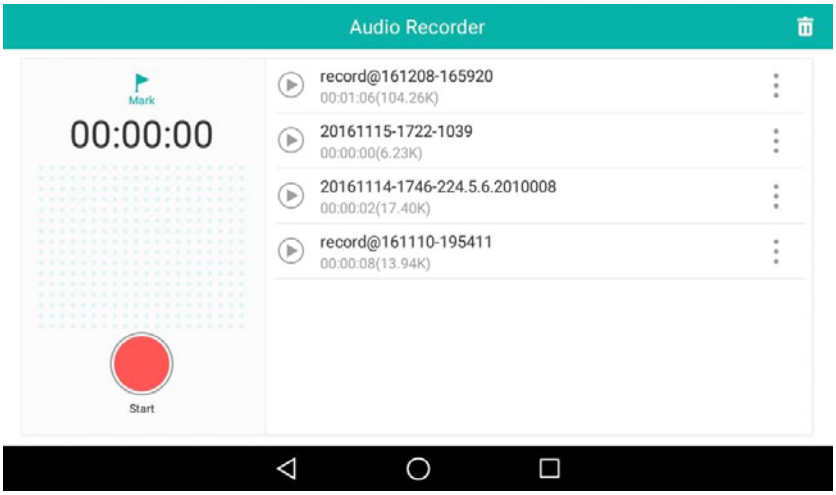
- Drag the slider to skip forward the playing or rewind the playing.
- Drag the slider to the position with red flag to listen to the marked moments.
- Press to adjust the volume of the speakerphone.
- Tap to stop the playing.



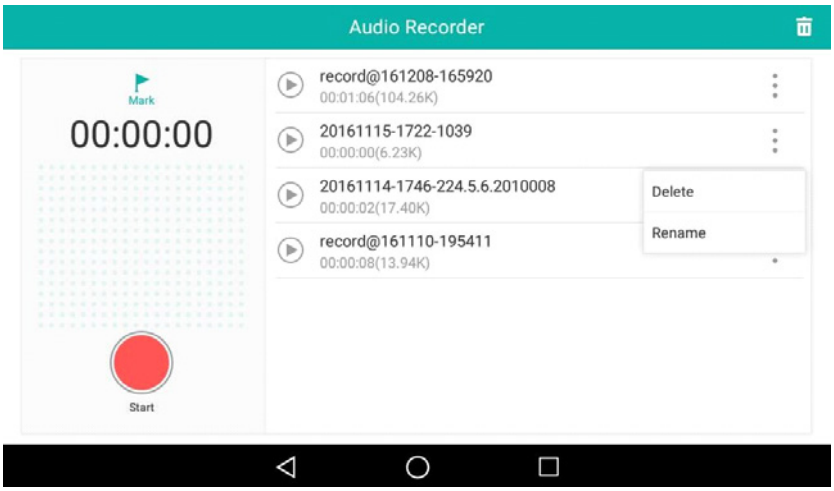
To delete the recorded audio:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Recorder** application.

  The recorded audios will display as a list.

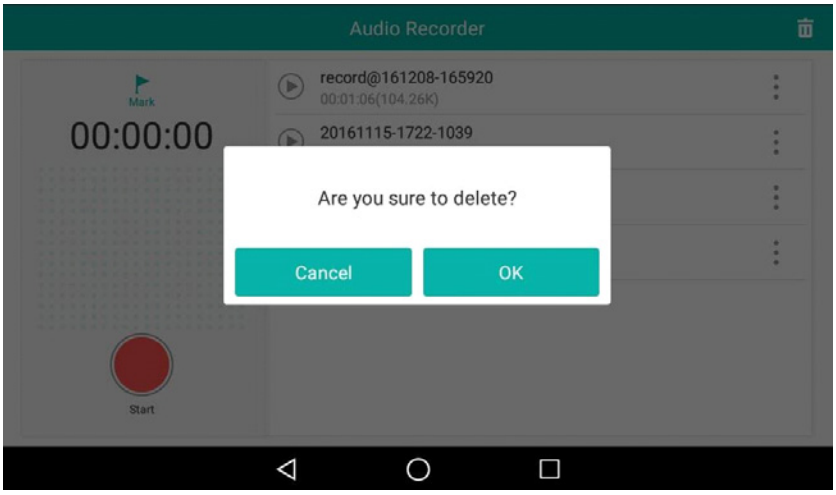


- Tap after the desired recorded audio.



- Tap **Delete** to delete the recorded audio.

  The touch screen prompt “Are you sure to delete?”.

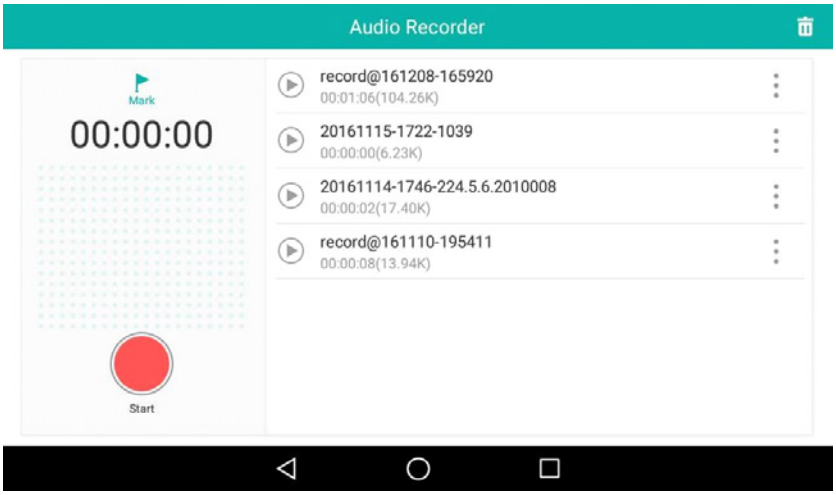


- Tap **OK** to save the change or **Cancel** to cancel.
- If you want to delete all recorded audios, tap  on the top-right of the touch screen.

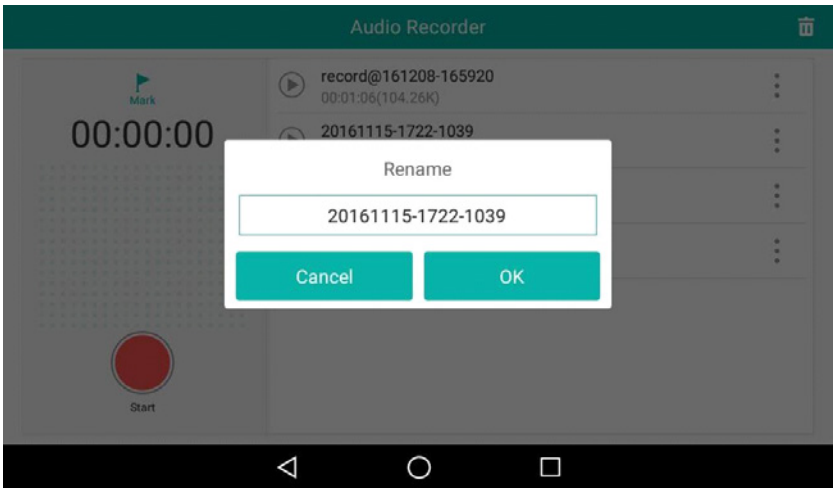
To rename a recorded audio:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Recorder** application.

- The recorded audios will display as a list.



- Tap ⋮ after the desired recorded audio.
- Tap **Rename**.



- Enter the desired file name.
- Tap **OK** to save the change or **Cancel** to cancel.

Camera

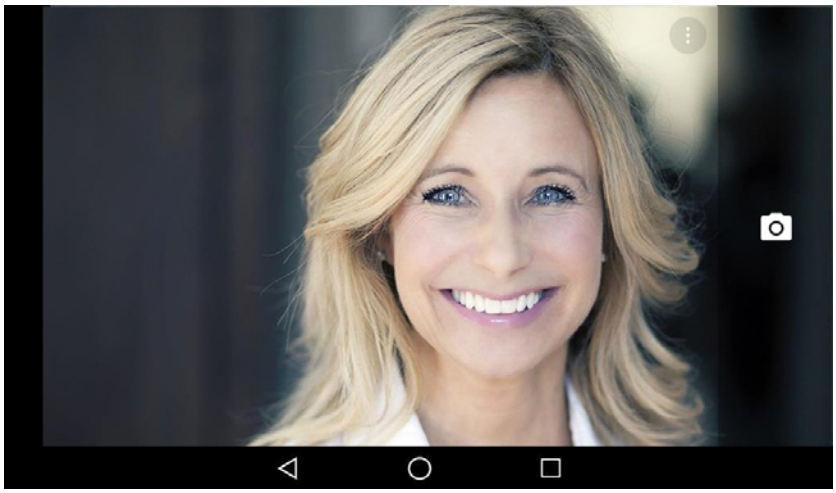
The IP phone supports taking pictures and recording videos via camera. For more information on how to connect the camera, refer “Phone Installation”. The pictures/videos taken by camera are saved in the internal SD card. And the pictures are saved with the name consisting of a prefix “IMG” and date & time stamp, for example, IMG_20160303-111413 was created on March 3, 2016, at 11:14:13. The videos are saved with the name consisting of a prefix “VID” and date & time stamp, for example, VID_20160303-112328 was created on March 3, 2016, at 11:23:28.

*The pictures/videos taken by camera are saved in *.jpg/*.mp4 format. You can view the pictures/videos on either the phone itself or on a computer using an application capable of viewing *.jpg/*.mp4 files.*

Taking Pictures

To take a picture:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Camera** application.



- Tap and then do the following (Optional):
  - Tap to turn on the camera grid.
  - Tap to configure the camera timer.
- Tap to take a picture.

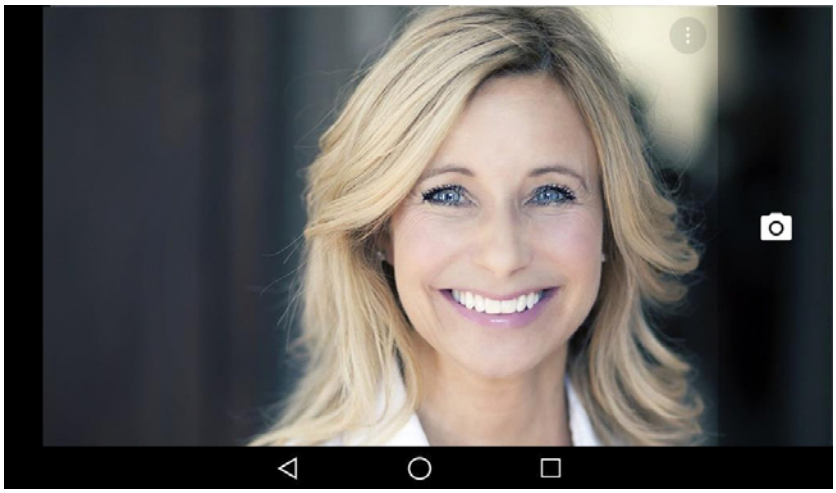
  The IP phone will play a tone. You can swipe left to view all pictures taken by the camera.

Recording Videos

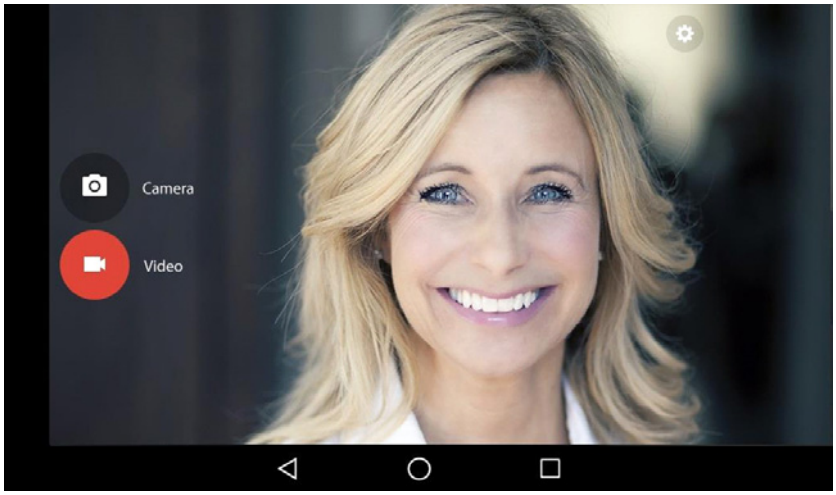
To record a video:

- Swipe left or right to go to the second idle screen.

- Tap to launch **Camera** application.



- Swipe right and then tap **Video**.



- Tap and then tap to turn on the camera grid (Optional)
- Tap to start recording.

  The IP phone will play a tone and recording time will appear on the left-top of the touch screen.

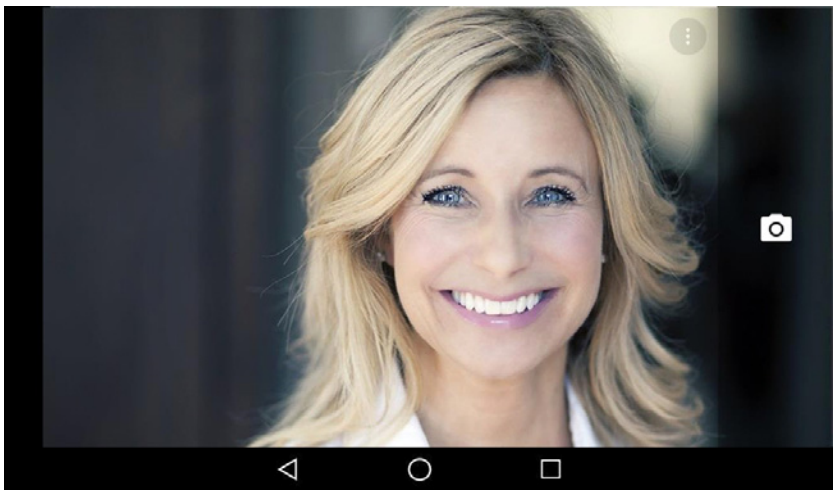
- Tap to stop recording.

  The IP phone will play a tone. You can swipe left to view all videos taken by the camera.

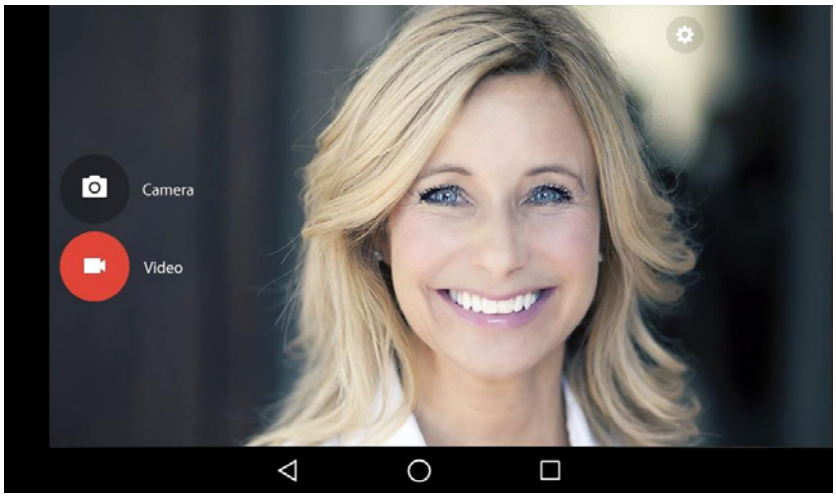
Camera Settings

To configure camera settings:

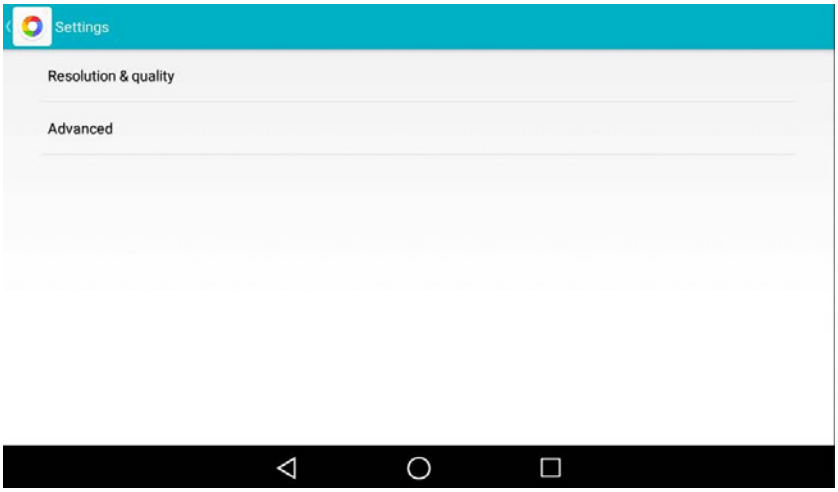
- Tap  to launch **Camera** application.



- Swipe right.



- Tap  .



- You can do the following:
  - Tap **Resolution & quality** to configure the resolution of the camera.
  - Tap **Advanced** and then tap **Manual exposure** to configure whether to use manual exposure.

Email

You can use **Email** application to access personal or business email account, and send/receive emails. Before signing in email, you should configure the type of your email account.

Signing In Email

When launching the **Email** application for the first time, a wizard will be provided for you to set up the email account step by step.

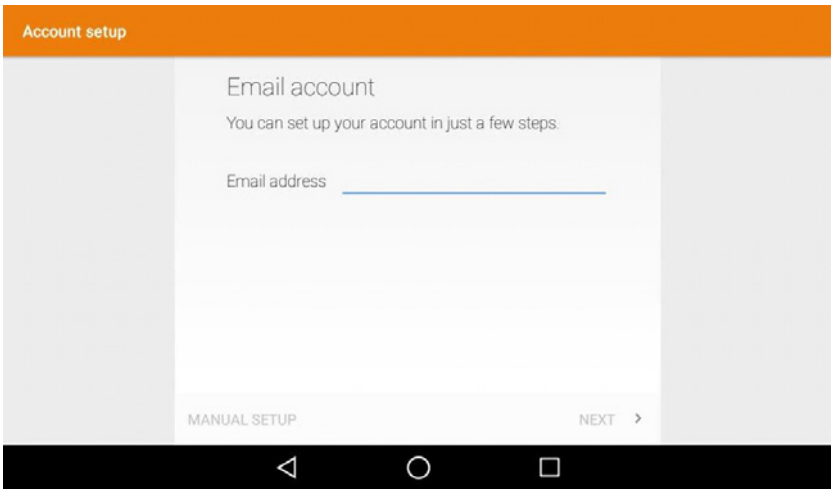
To sign in email:

- Do one of the following:
  - Swipe left or right to go to the second idle screen.

    Tap  to launch **Email** application.

    OR
  - Swipe left or right to go to the third idle screen.

Tap the email widget.

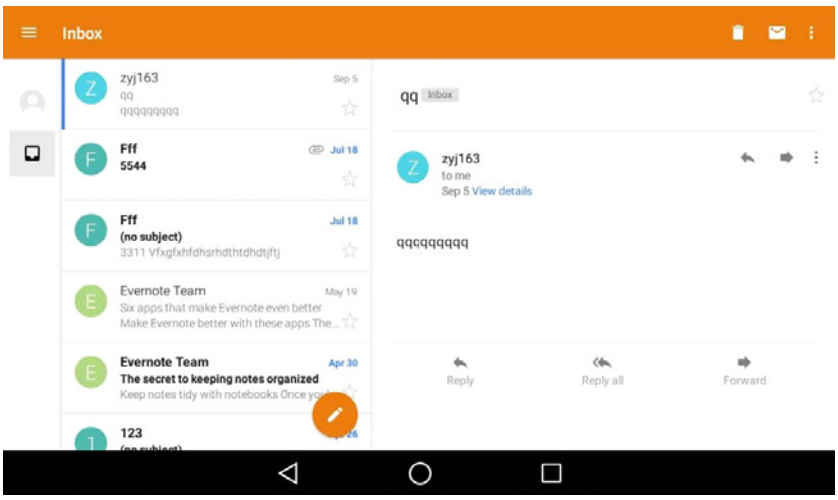


- Sign in email, refer to “Adding APP Accounts”.

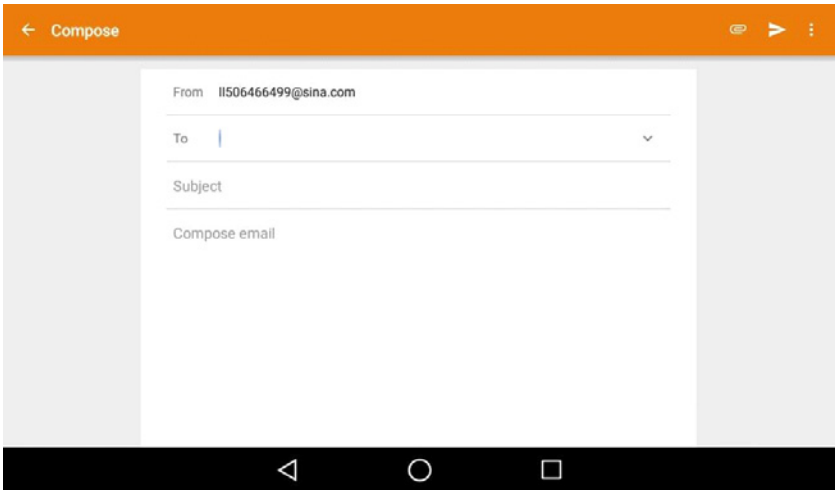
Sending Emails

To send an email:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Email** application.

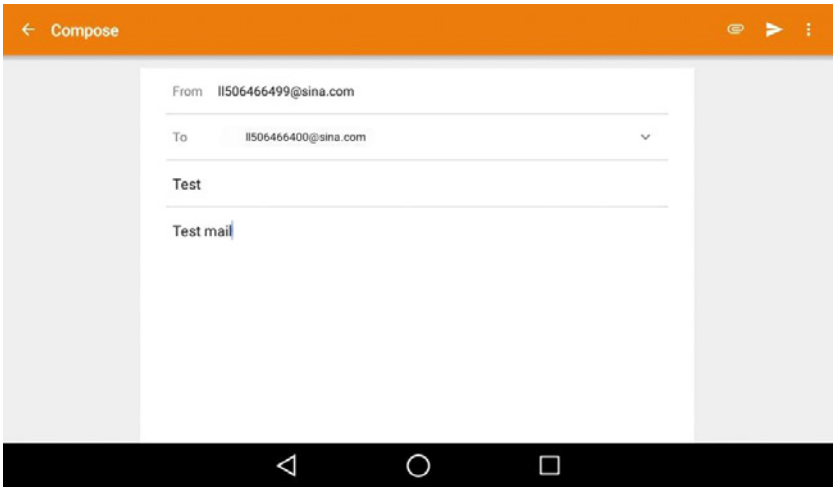


- Tap  .



- Compose the email.
- Enter the email account you want to send the email to in the **To** field.
- In the **Subject** field (optional), enter the email subject.
- Tap  and then tap **Attach file** to add a file as mail attachment (Optional).

  The attachment file size must be within 5MB.



- Tap  to send the email.

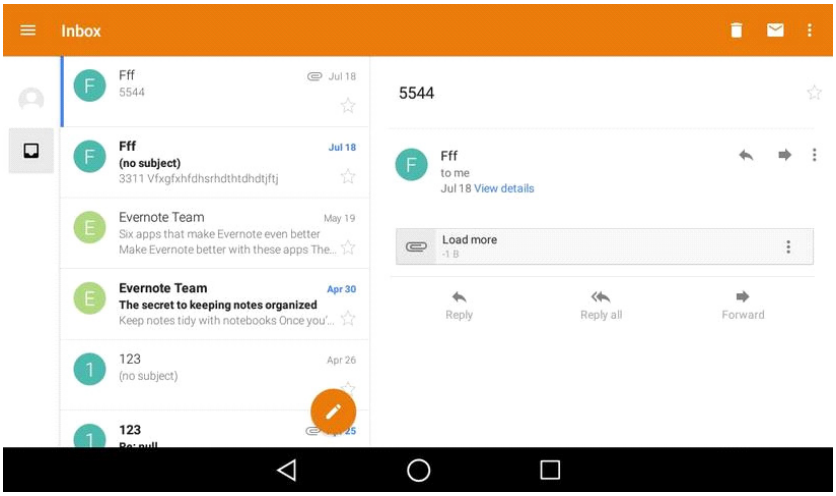
Reading Emails

If there is an incoming email, the IP phone will play a notification sound. And if there is an incoming email notification on the notification center, you can tap the notification to read the email.

To read an email:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Email** application.

  If the email you want to read is not in the inbox, you can tap  on the top-left of the touch screen to select the folder that contains the email you want to read.



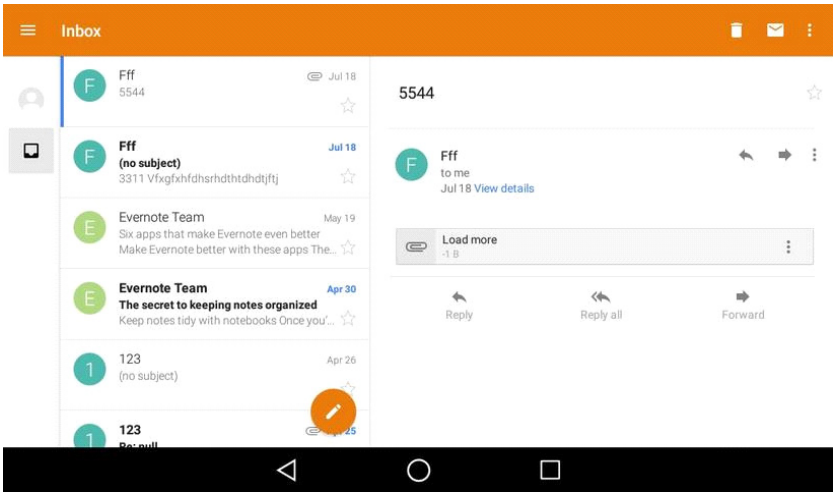
- Tap the desired email to read.

Deleting Emails

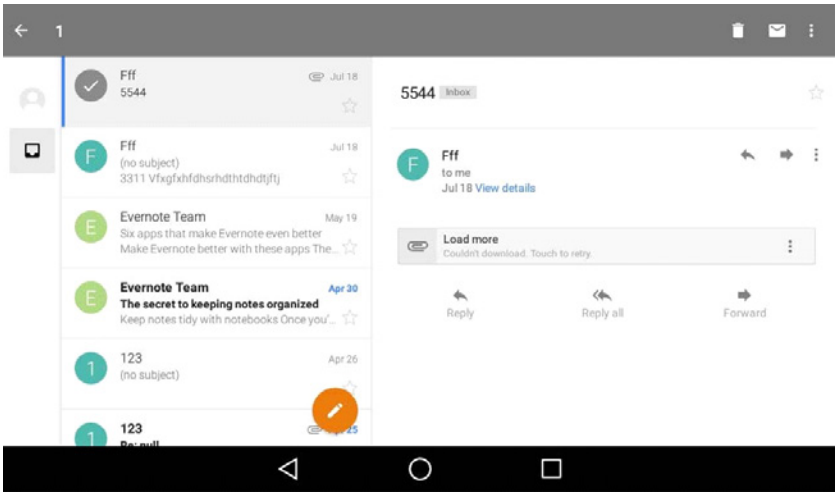
To delete an email:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Email** application.

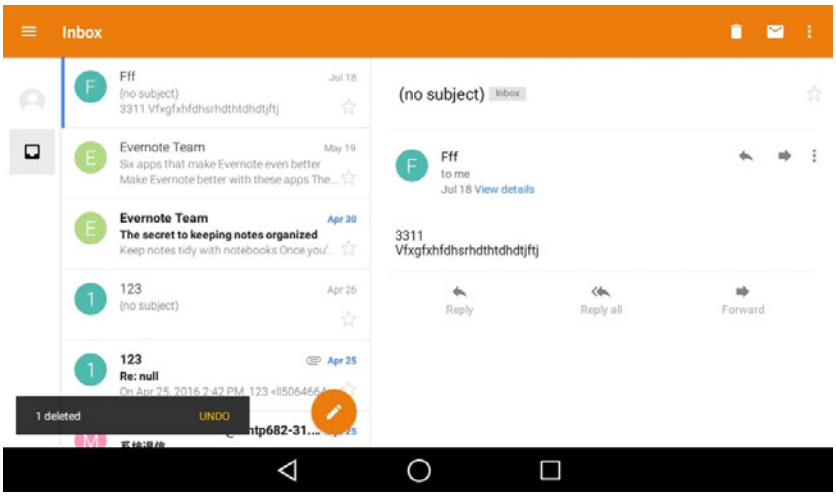
If the email you want to delete is not in the inbox, you can tap on the top-left of the touch screen to select the folder that contains the email you want to delete.



- Long tap the desired email.



- Tap to delete the email.



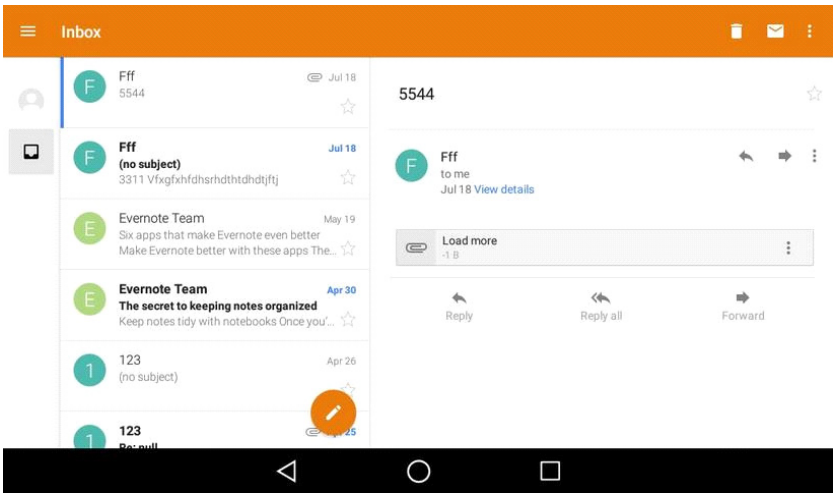
- You can tap **UNDO** to cancel the deletion.

Email Options

To use the email options:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Email** application.

  If the email you want to configure is not in the inbox, you can tap  on the top-left of the touch screen to select the folder that contains the email you want to configure.



- You can do the following:
  - Tap  /  to mark the email as unread/read.
  - Tap and then tap **Print** to print the email.
  - Long tap the desired email.

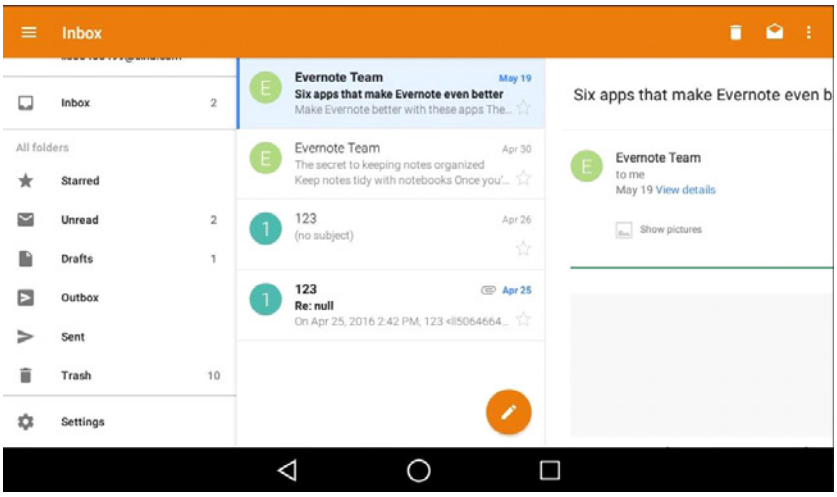
    Tap and then tap **Add star** to star the email.

Multiple Email Accounts

You can add more than one email account on the IP phone.

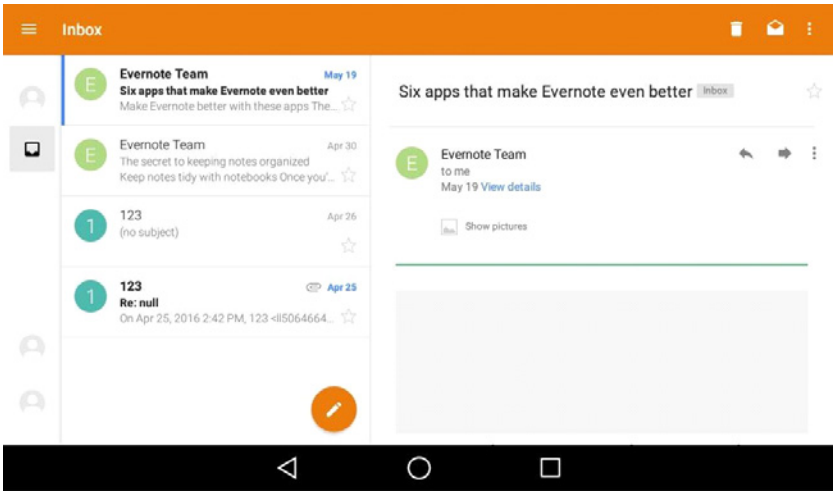
To add another email account on the IP phone:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Email** application.
- Tap  on the top-left of the touch screen.



- Tap **Settings->Add account.**
- A new email account setup wizard will show up. For more information, refer to “Adding APP Accounts”.

  Once another email account is successfully added, the touch screen will display as below:



- You can tap on the bottom-left of the touch screen to switch between the email accounts.

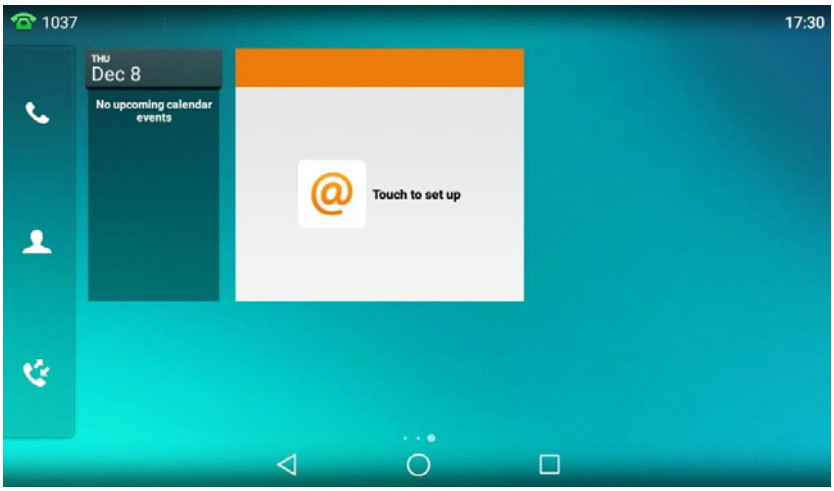
Email Widget

You should sign in the email if you have not registered an email account. For more information, refer “Adding APP Accounts”.

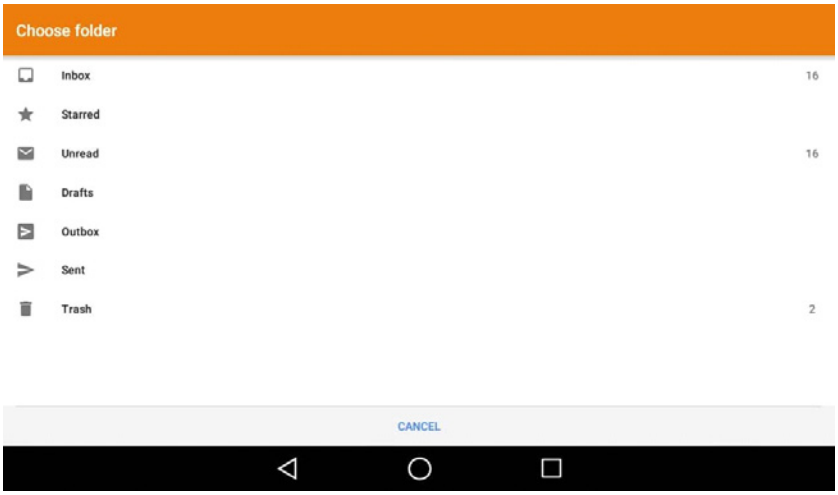
After you sign in the email, you can choose the desired folder to display on the widget. If you add an email widget to display the unread emails, then you want the widget to display the incoming emails, you should add a new widget.

To use the email widget:

- Swipe left or right to go to the third idle screen.

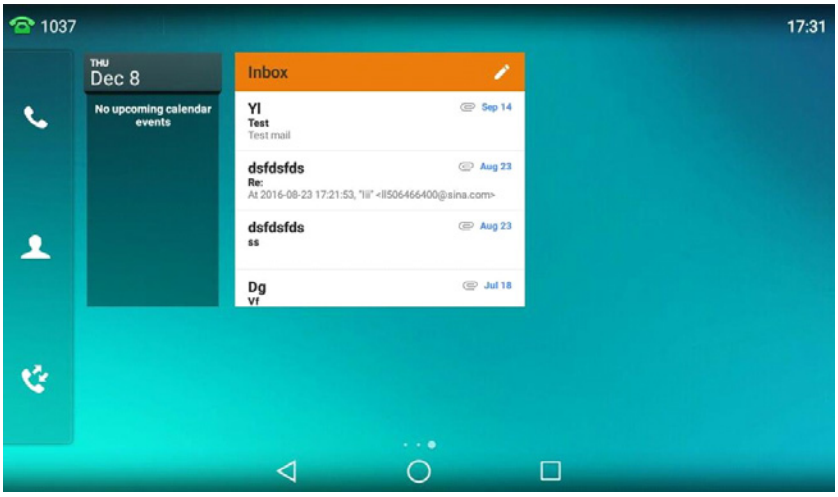


- Tap the email widget.
- Tap the desired account, if you have more than one email account.



- Tap the desired folder to display on the widget.

- If you select **Inbox**, the email widget will display as below:



You can drag up and down to scroll through emails in inbox folder, and tap the desired email to view. You can also tap  on the top-right of the widget to compose and send an email.

Sending Files via Email

The Gallery and FileManager applications support data transfer via Email. You can send pictures, audios or videos as mail attachments to other account. For more information, refer to “” and “Sharing Pictures/Videos”.

Browser

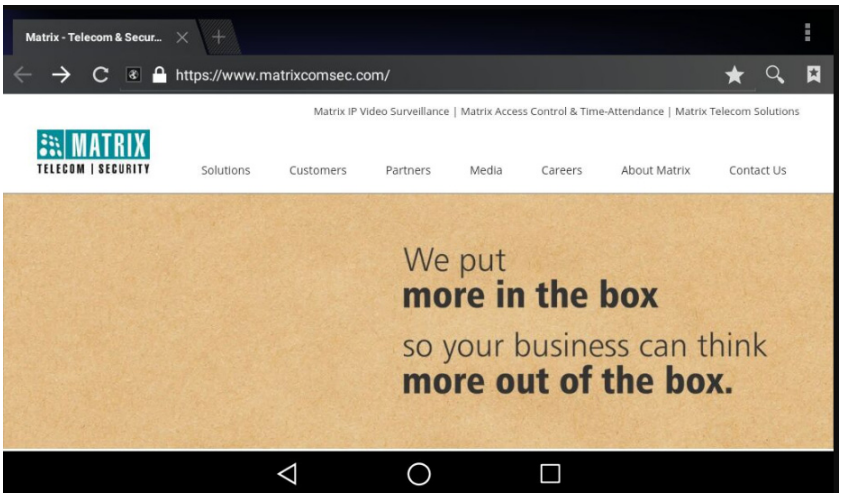
The Browser application could be used in a similar way as a PC-based web browser, i.e., browse web pages, bookmarks, set home pages, clear history and etc. The default home page is the Matrix website.

Browsing a Web Page

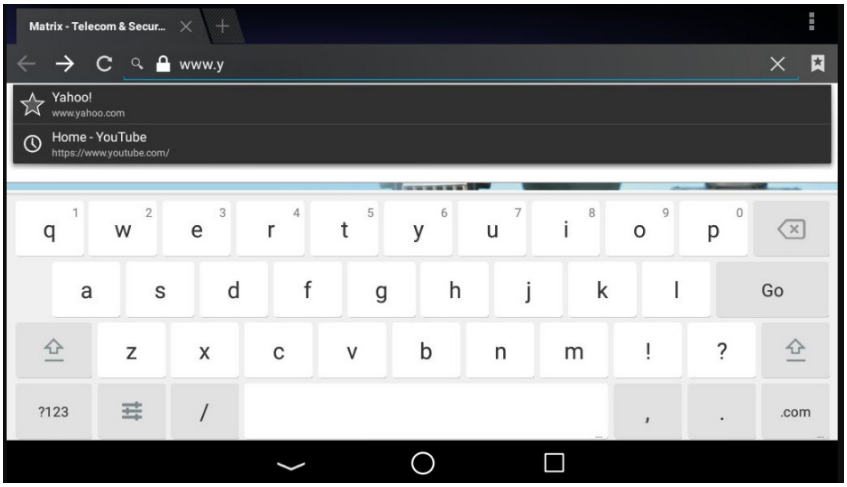
To browse a web page:

- Swipe left or right to go to the second idle screen.

- Tap  to launch **Browser** application.



- Enter a part of URL in the address input field of the browser.
- The phone will automatically retrieve and list similar websites.



- Tap the website you want to visit or enter the full URL then tap **Go** on the onscreen keyboard to browse the web page.
- You can do the following:
  - Scroll up/down to view the web page.
  - Pinch open or pinch close the web page.

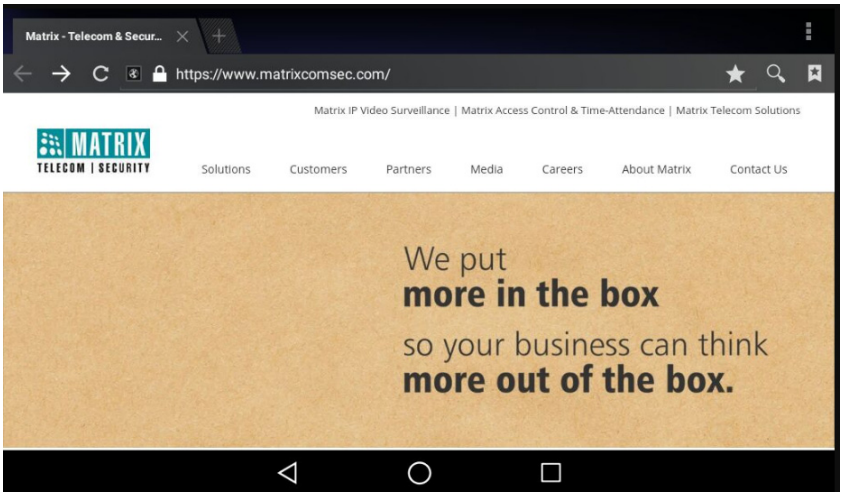
Managing Bookmarks

You can bookmark the web pages that you frequently visit.

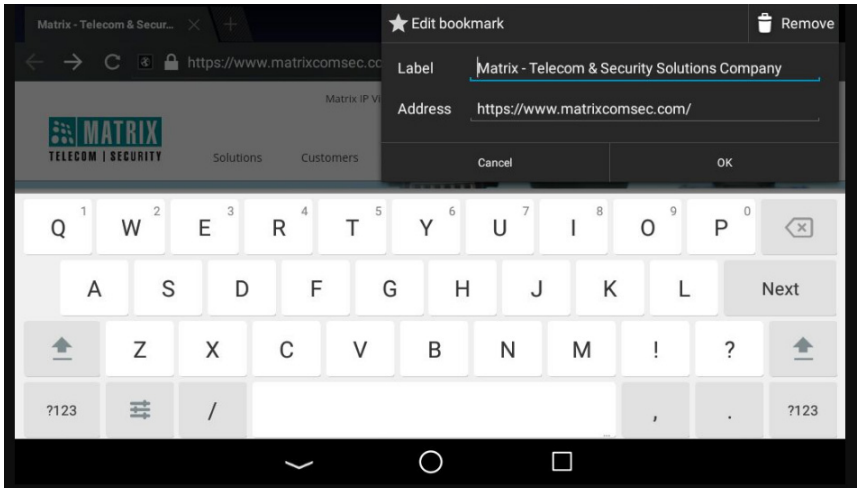
To add a web page to the bookmarks list:

- Swipe left or right to go to the second idle screen.

- Tap  to launch **Browser** application.



- Open the web page you want to bookmark.
- Tap  .

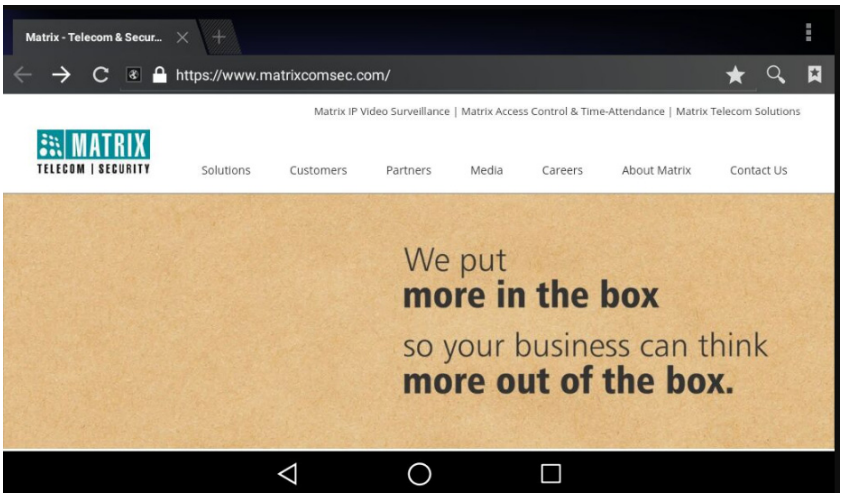


- Edit the bookmark information in corresponding fields.
- Tap **OK** to accept the change.
- Then the web page is added to the **BOOKMARKS** list.

To open a web page from the bookmarks list:

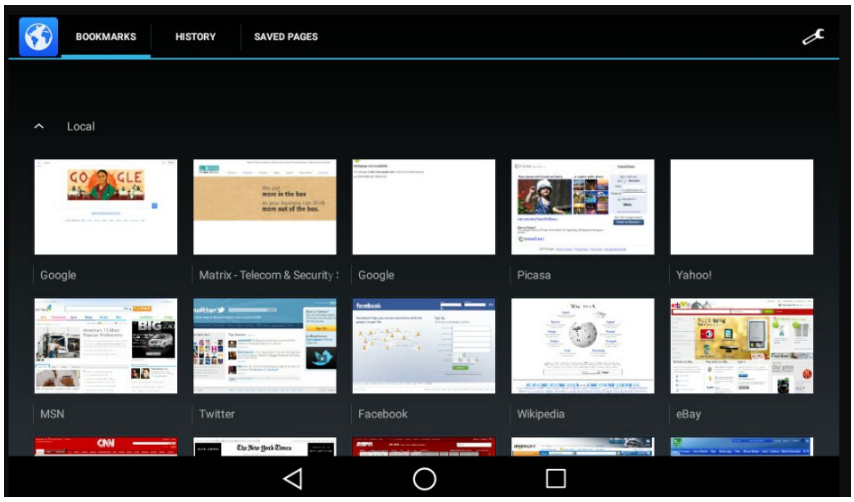
- Swipe left or right to go to the second idle screen.

- Tap  to launch **Browser** application.



- Tap  to access the **BOOKMARKS** list.

  There are some pre-configured bookmarks by default.

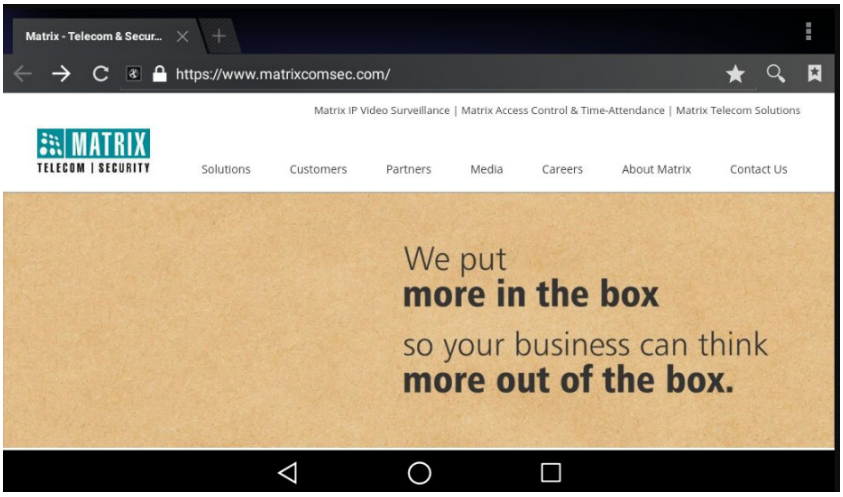


- Tap the web page you want to visit.

To edit a bookmark:

- Swipe left or right to go to the second idle screen.

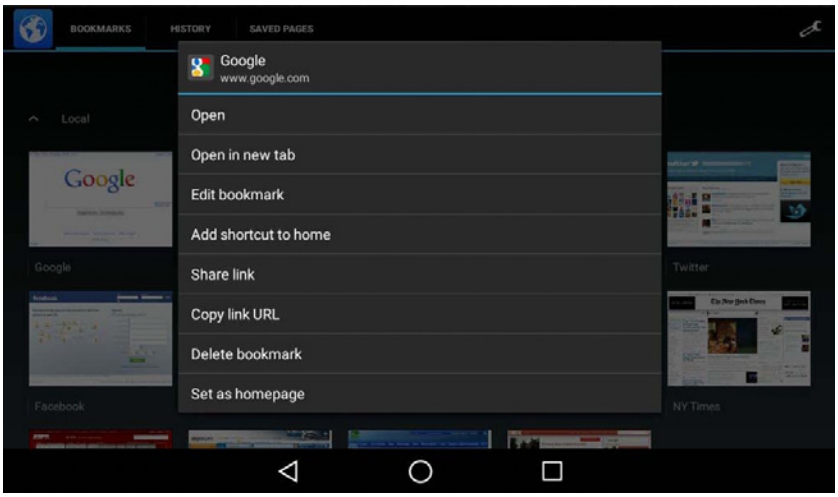
- Tap  to launch **Browser** application.



- Tap  to access the **BOOKMARKS** list.

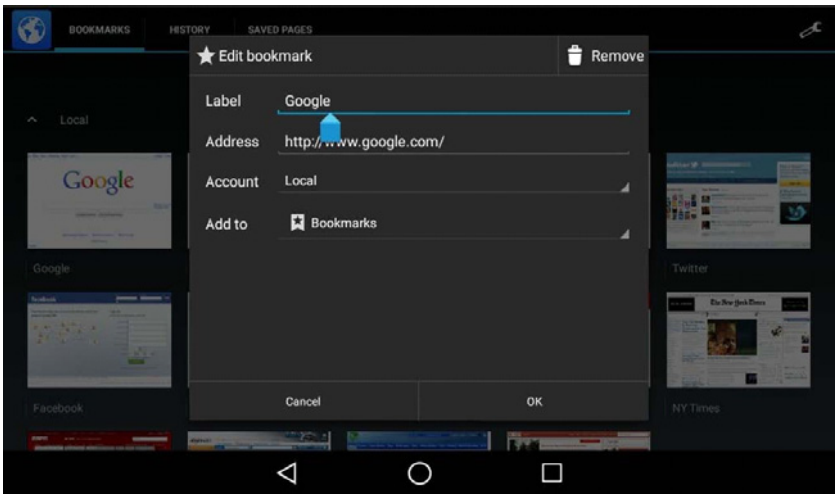
  There are some pre-configured bookmarks by default.

- Long tap the bookmark you want to edit.



- Tap **Edit bookmark** in the pop-up dialog box.

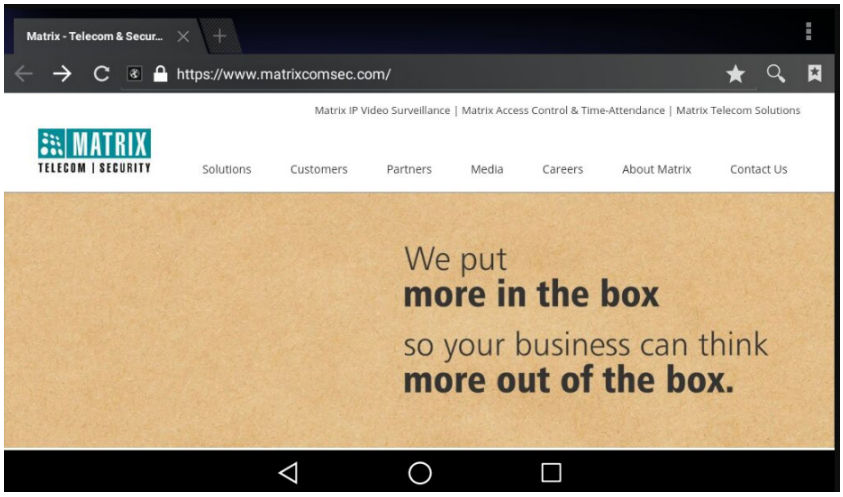
- Edit the bookmark.



- Tap **OK** to save the change or **Cancel** to cancel.

To delete a bookmark:

- Swipe left or right to go to the second idle screen.
- Tap  to launch **Browser** application.



- Tap  to access the **BOOKMARKS** list.

  There are some pre-configured bookmarks by default.

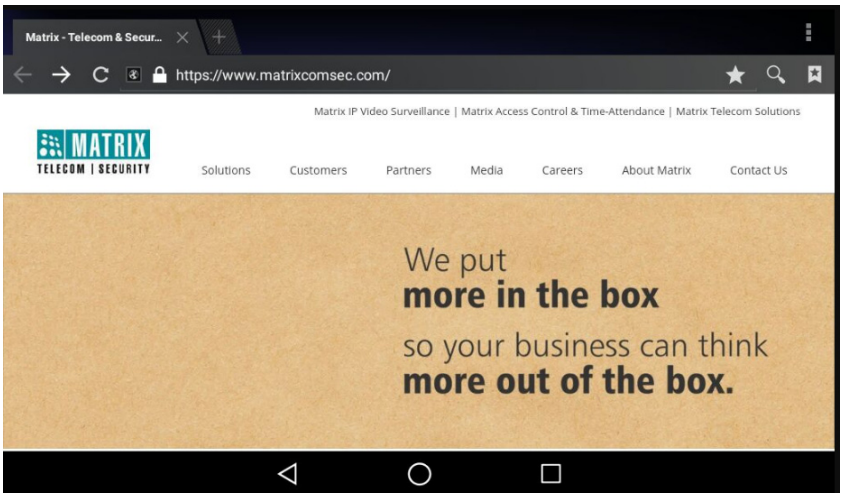
- Long tap the bookmark you want to delete.
- Tap **Delete bookmark** in the pop-up dialog box.
- Tap **OK** to save the change or **Cancel** to cancel.

Setting a Home Page

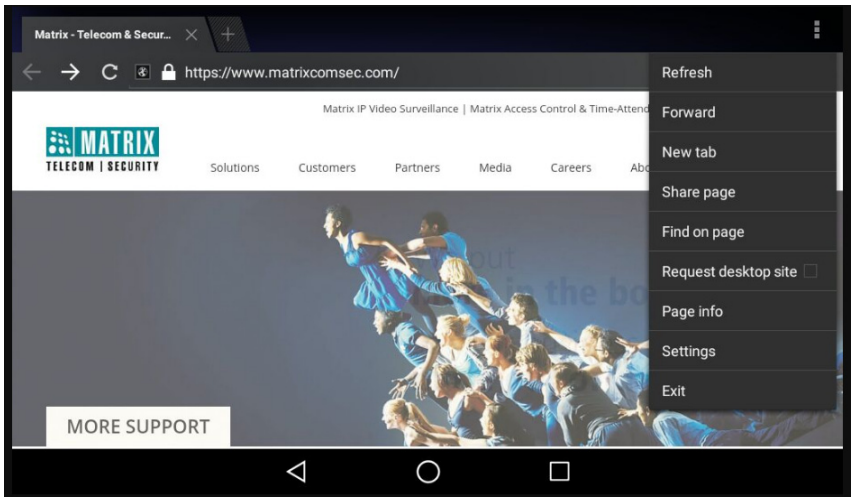
You can set the web page you visit most frequently as the home page. Then, each time you open the browser, this web page is automatically opened.

To set a home page:

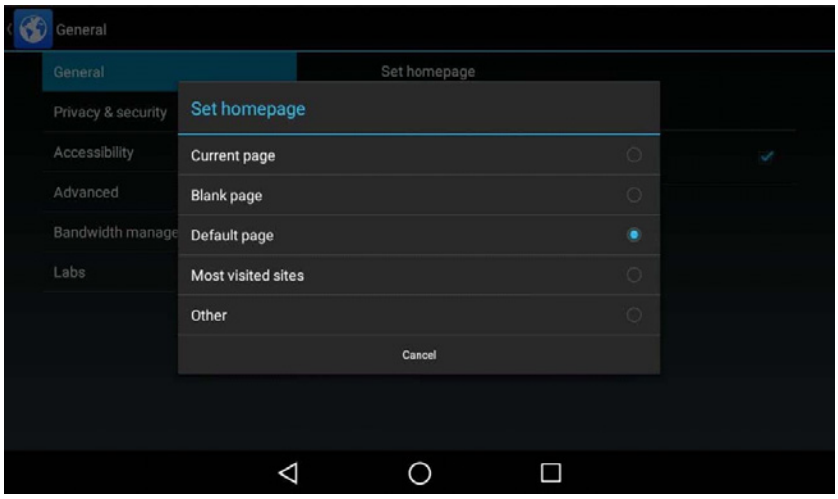
- Swipe left or right to go to the second idle screen.
- Tap  to launch **Browser** application.



- Tap .



- Tap **Settings->General->Set homepage.**



- Tap the web page you want to set as the home page in the pop-up dialog box.

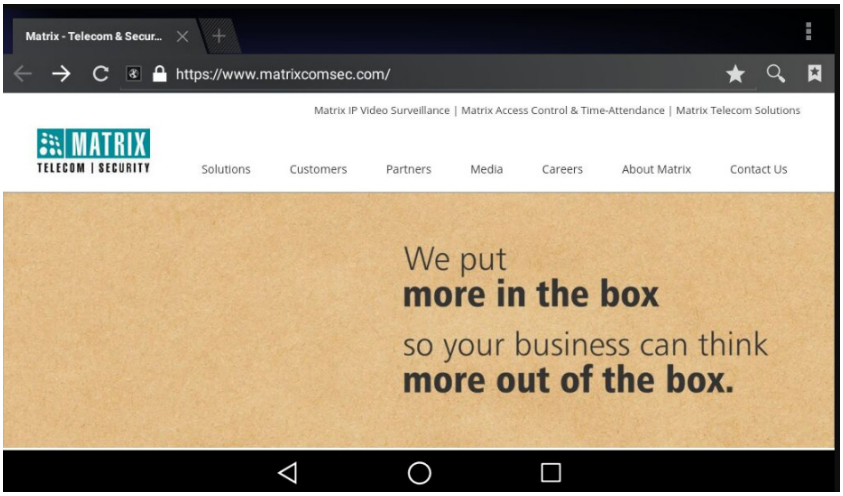
  You can also long tap a web page in **BOOKMARKS** list and then tap **Set as homepage**.

Clearing the Navigation History

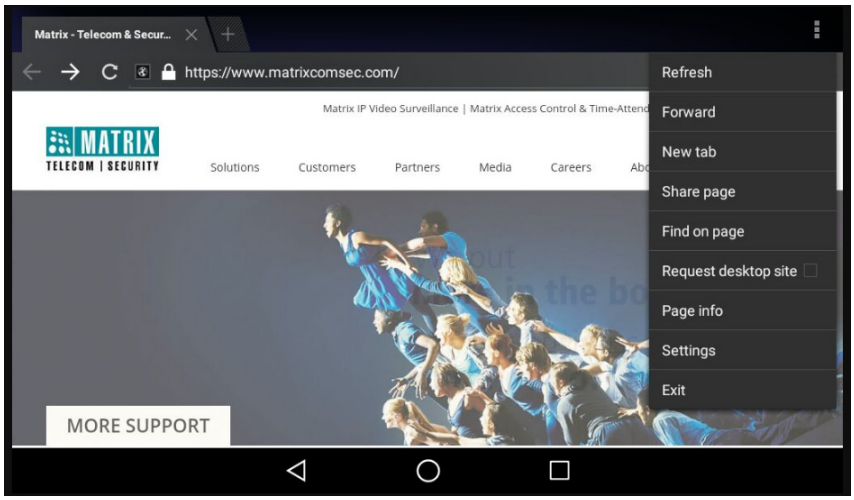
You can periodically clear your navigation history to protect your privacy and prevent information disclosure.

To clear the navigation history:

- Swipe left or right to go to the second idle screen.
- Tap  to launch Browser application.

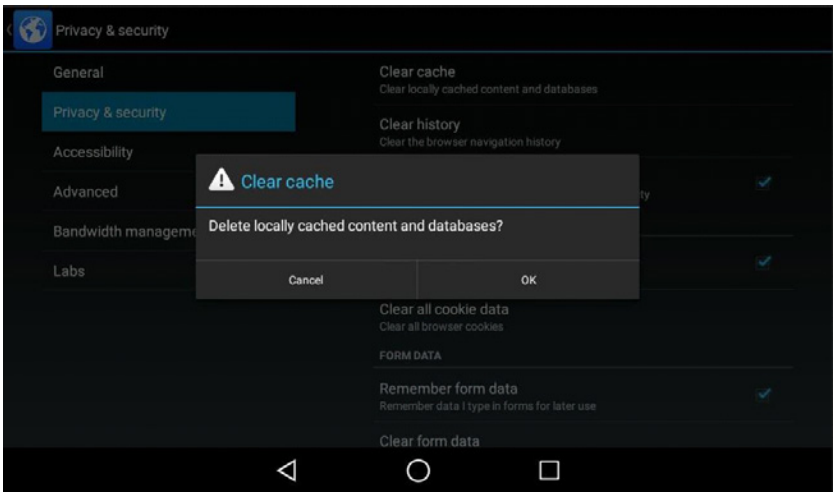


- Tap .



- Tap **Settings->Privacy & security->Clear history.**

  The touch screen prompt “Delete the browser navigation history?”.



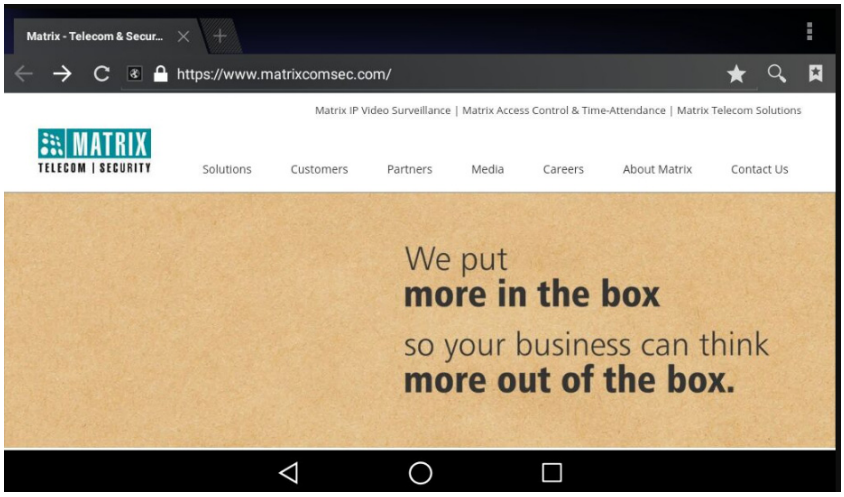
- Tap **OK** to save the change or **Cancel** to cancel.

Modifying Browser Settings

To modify browser settings:

- Swipe left or right to go to the second idle screen.

- Tap  to launch Browser application.



- Tap .
- Tap **Settings**.

  Customize the browser style you prefer.

Calendar

You can use the Calendar application to synchronize events from email account as well as creating, modifying and viewing the events. The IP phone also provides notification for the upcoming events from the Calendar.

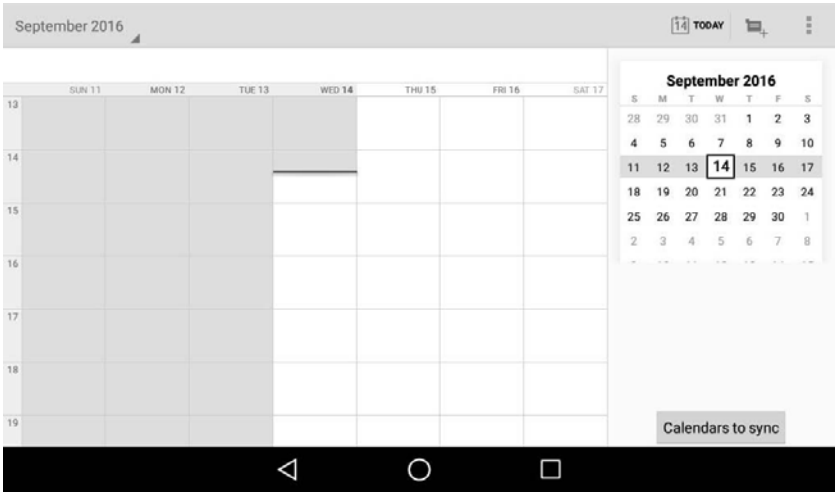
Signing In Calendar

Before creating an event, you should sign in an email account first. If you have registered an account on the Email application already, you can skip this operation.

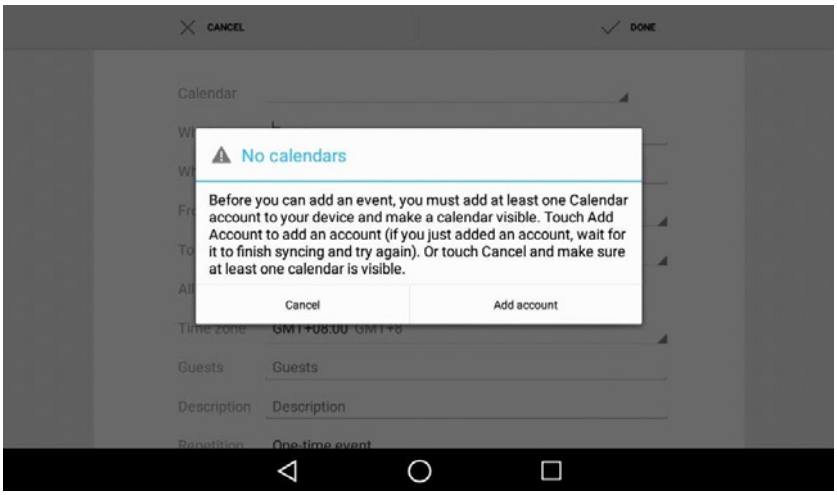
To sign in calendar:

- Swipe left or right to go to the second idle screen.

- Tap  to launch Calendar application.



- Tap .



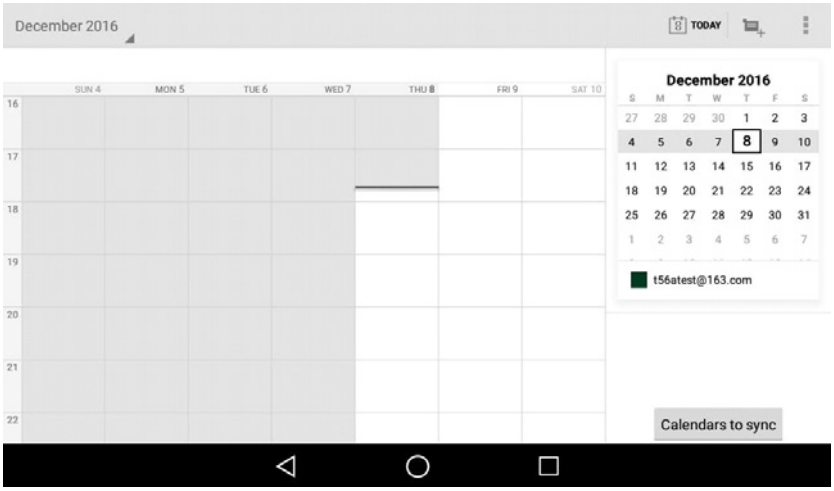
- Tap **Add account**.
- Sign in the Calendar, refer to “Adding APP Accounts”.

Creating an Event

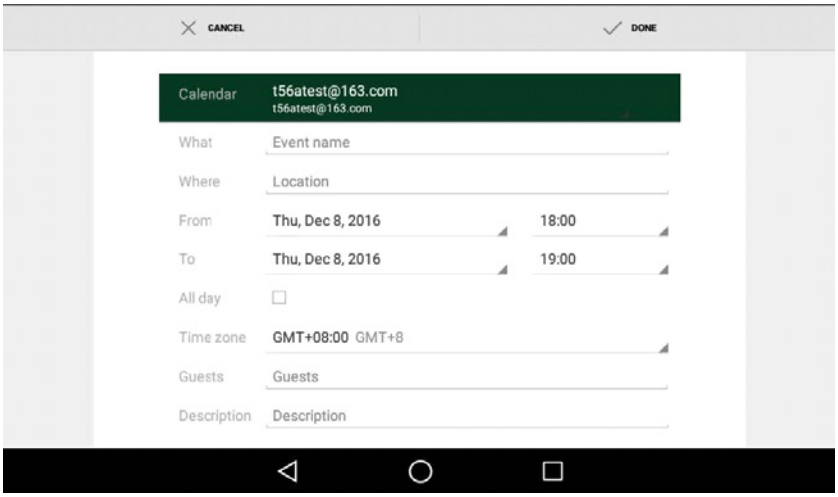
To create an event:

- Swipe left or right to go to the second idle screen.

- Tap  to launch Calendar application.



- Tap a desired date.
- Tap .



- Edit the event.
- Tap **DONE** to save the change or **CANCEL** to cancel.

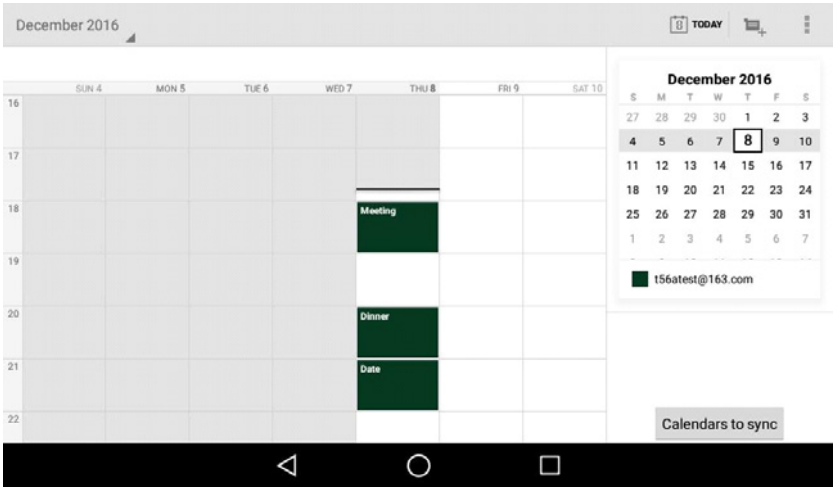
Checking Events

On the calendar screen, you can check the information about an event you have added.

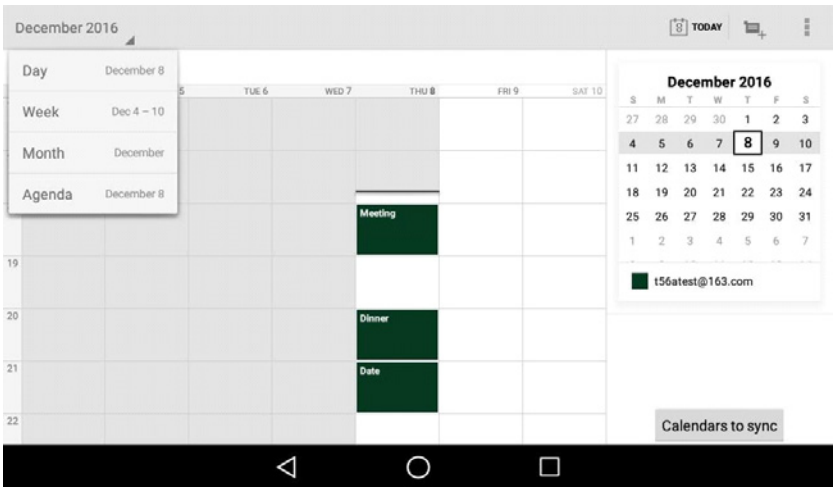
To view calendar:

- Swipe left or right to go to the second idle screen.

- Tap  to launch Calendar application.

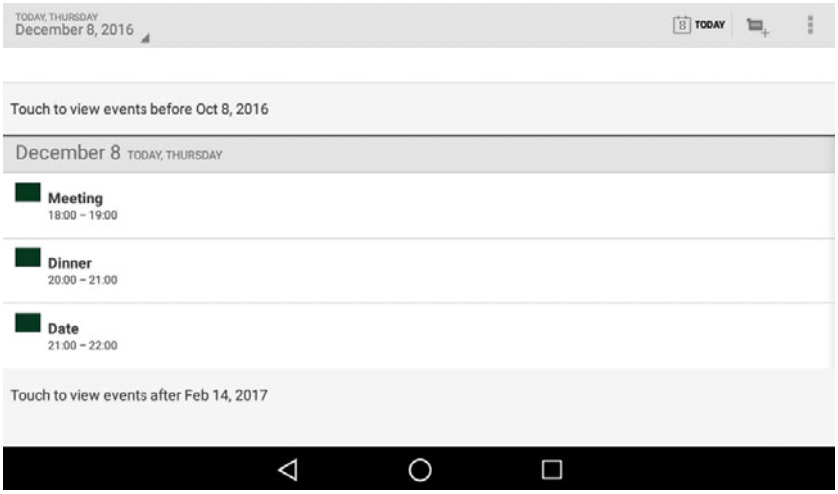


- Tap December 2016 on the top-left of the touch screen (Optional).



Tap **Day/Week/Month/Agenda** to select different time format to view the calendar.

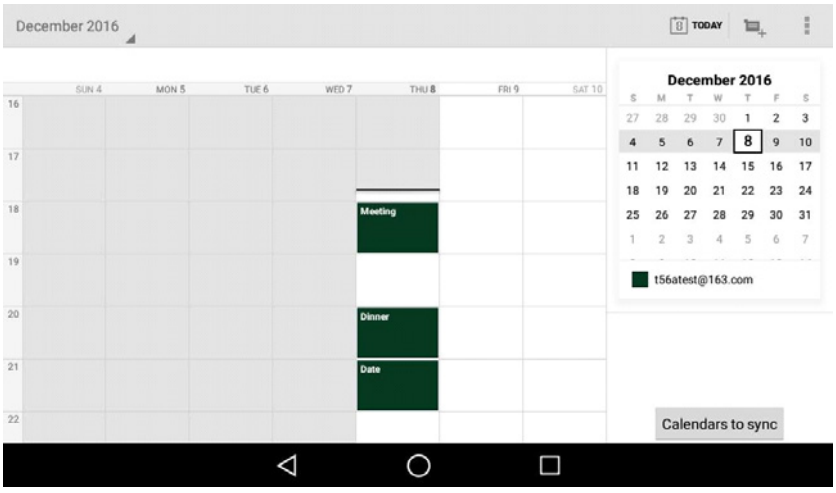
In the agenda view, the events you have added are displayed as a list. You can tap **Touch to view events before XXX/Touch to view events after XXX** to view all events.



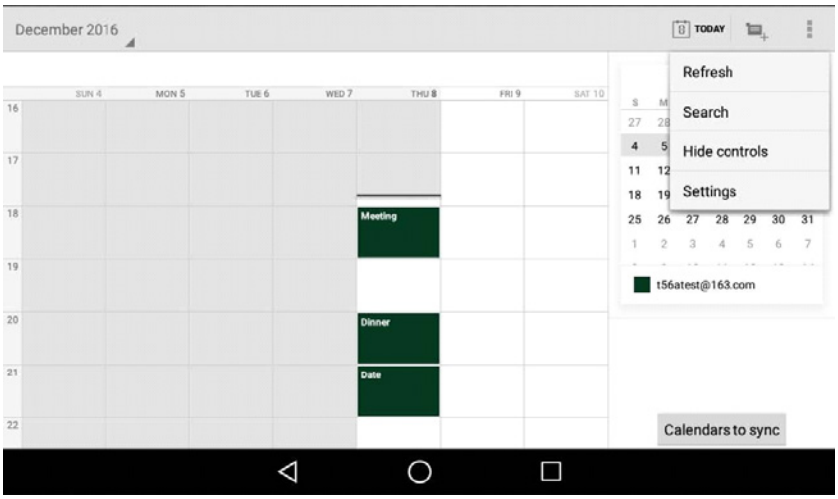
Modifying the Calendar Settings

To modify the calendar settings:

- Swipe left or right to go to the second idle screen.
- Tap  to launch Calendar application.



- Tap .



- Tap **Settings**.
- Customize the general settings and account settings.

Calculator

You can use the Calculator application to perform simple or complex calculations.

To access the calculator application:

- Swipe left or right to go to the second idle screen.
- Tap to launch Calculator application.



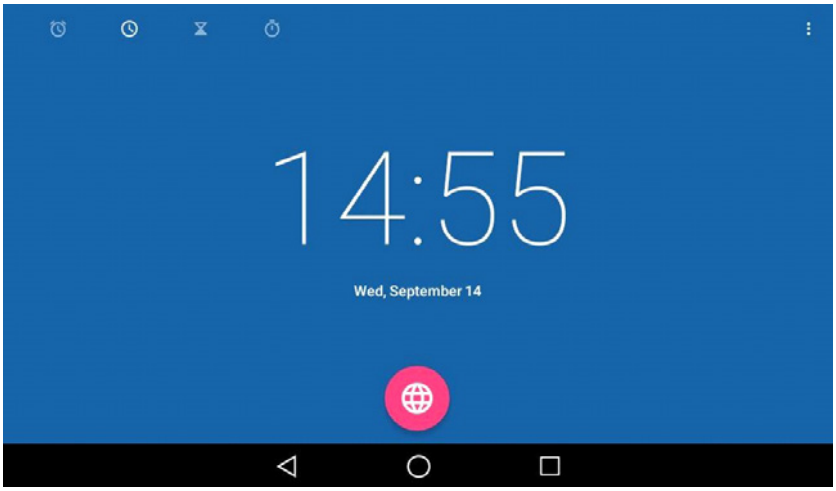
Clock

You can use the Clock application to set an alarm, check the time of other cities in the world, calculate the lasting time of an event and set a timer.

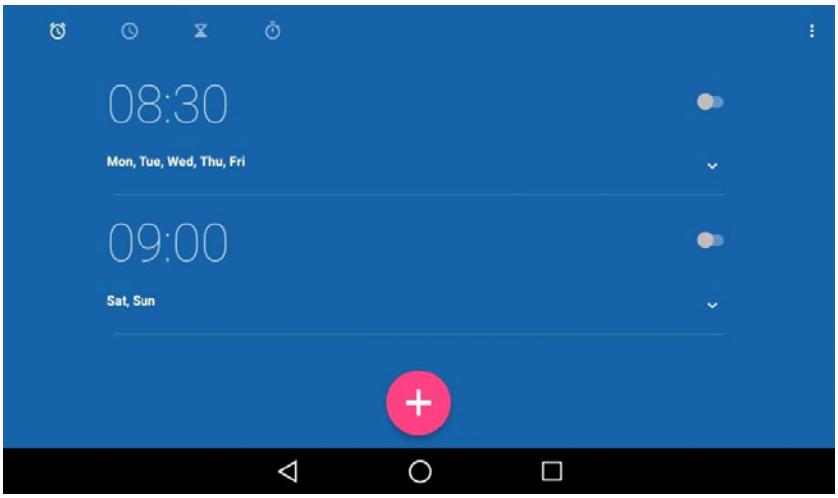
Alarm

To set an alarm clock:

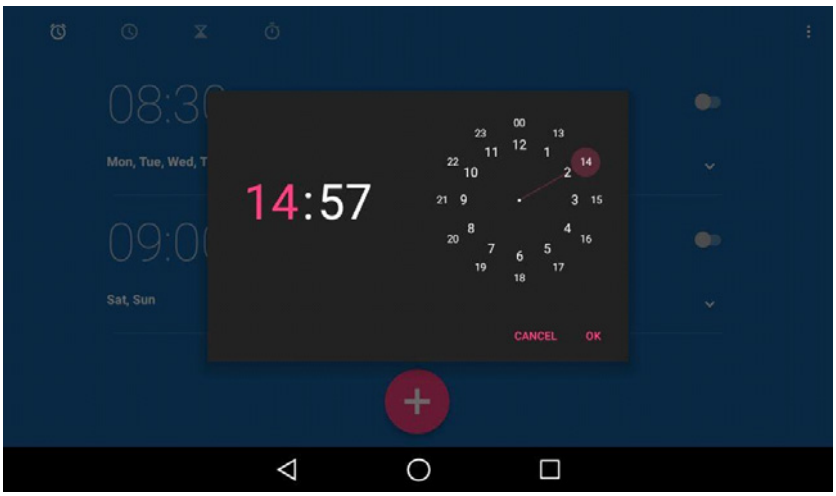
- Tap the digital clock widget on the home screen.



- Tap  on the top-left of the touch screen.

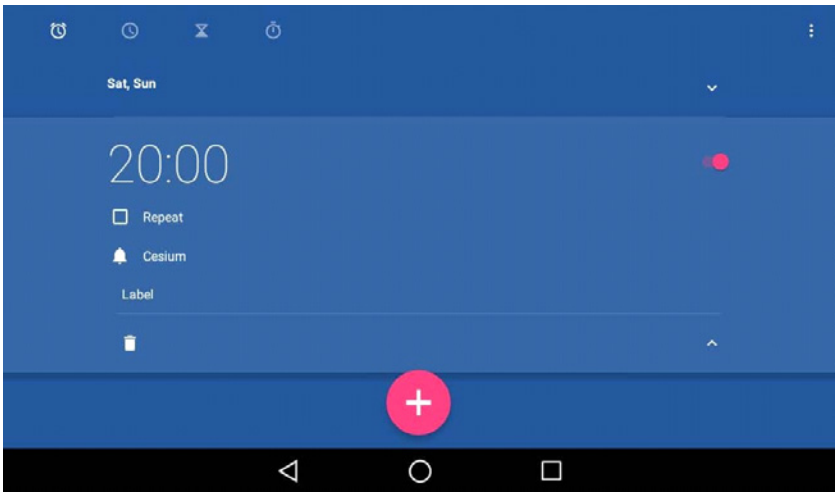


- Tap .



- Tap the desired number to enter the alarm time.
- Tap **OK**.

  The alarm clock you have set is displayed on the touch screen.

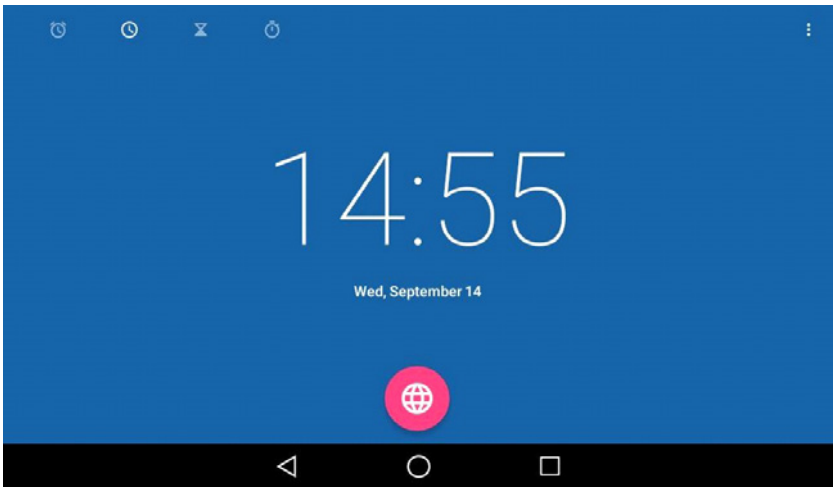


- Tap  to enable/disable an alarm.
- Tap the **Label** field and then enter a label for an alarm.
- Tap the **REPEAT** radio box and then tap the repeat day of the alarm.
- Tap  and then select a desired ringtone for alarm.

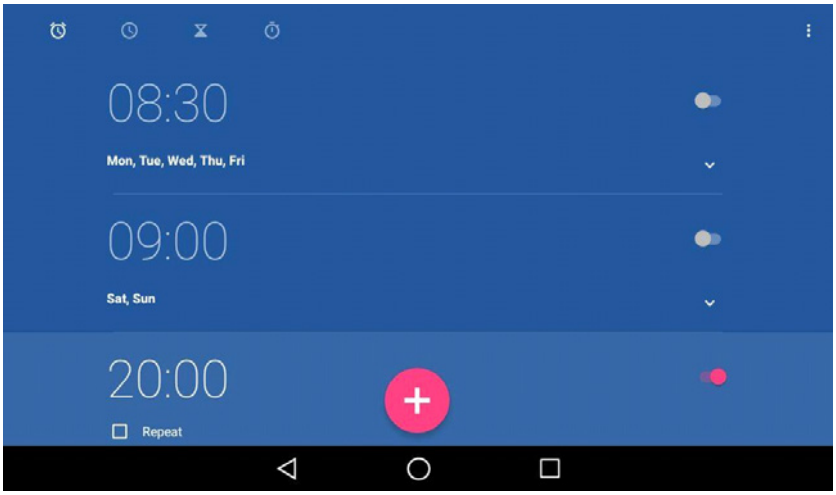
If you have set the time format as 12 Hour, you need to select AM or PM when setting the alarm time. For more information on time format, refer to "Time & Date".

To edit an alarm clock:

- Tap the digital clock widget on the home screen.



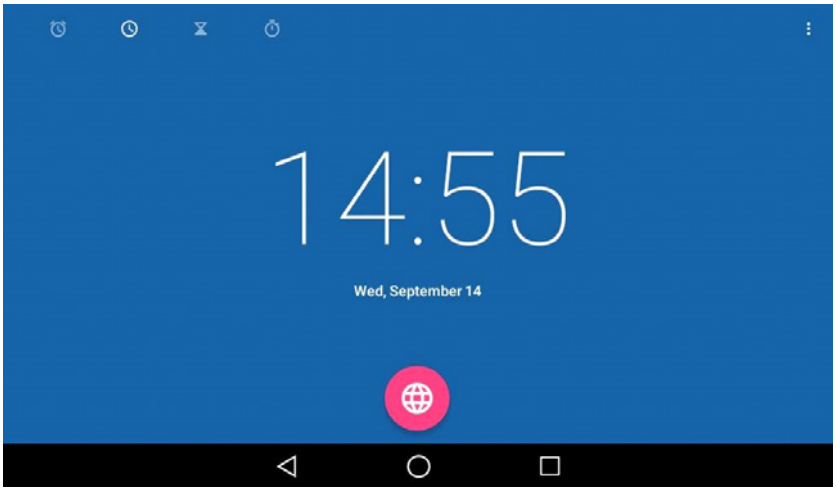
- Tap  on the top-left of the touch screen.



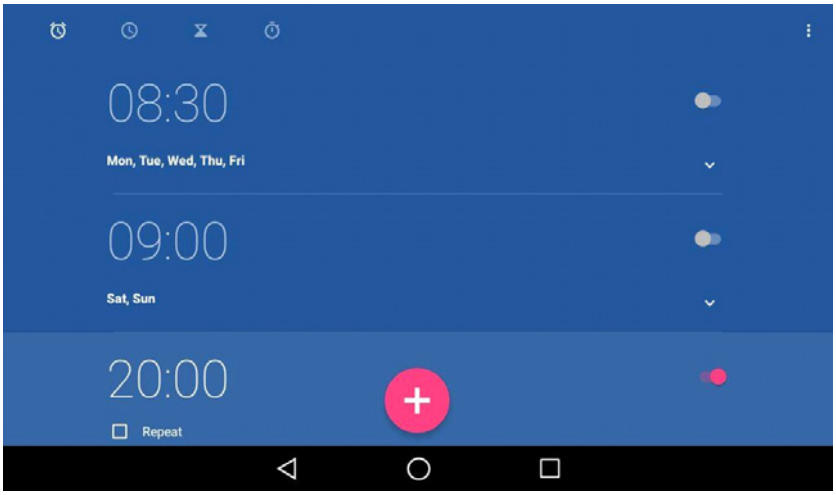
- Drag up and down to scroll through alarms.
- Select a desired alarm.
- Tap the desired option to edit the alarm clock.

To delete an alarm clock:

- Tap the digital clock widget on the home screen.



- Tap  on the top-left of the touch screen.

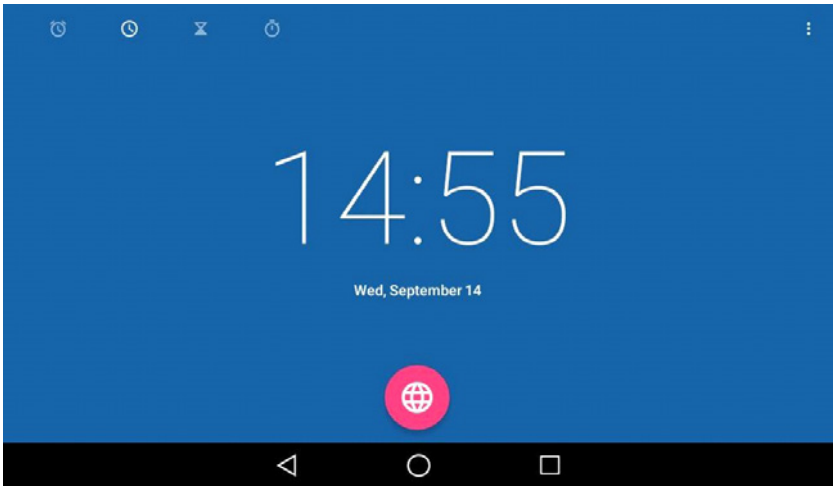


- Select a desired alarm.
- Tap  on the bottom-left of the alarm to delete the desired alarm.

You can tap **UNDO** to cancel the deletion.

To configure the alarm sound parameters:

- Tap the digital clock widget on the home screen.



- Tap  on the top-left of the touch screen.
- Tap and then tap **Settings**.
- You can do the following:
  - Tap the Silence after field, and then tap the desired lasting time of the alarm bell in the pop-up dialog box. The alarm clock will automatically stop after this specified period of time.

  OR
  - Tap the **Snooze length** field, and then tap the desired snooze time after the alarm clock rings in the pop-up dialog box.

  OR
  - Tap the **Alarm volume** field, and then drag the slider to adjust the alarm volume.

  OR
  - Tap the **Volume buttons** field, and then configure the desired function for the Volume key when the alarm is ringing.

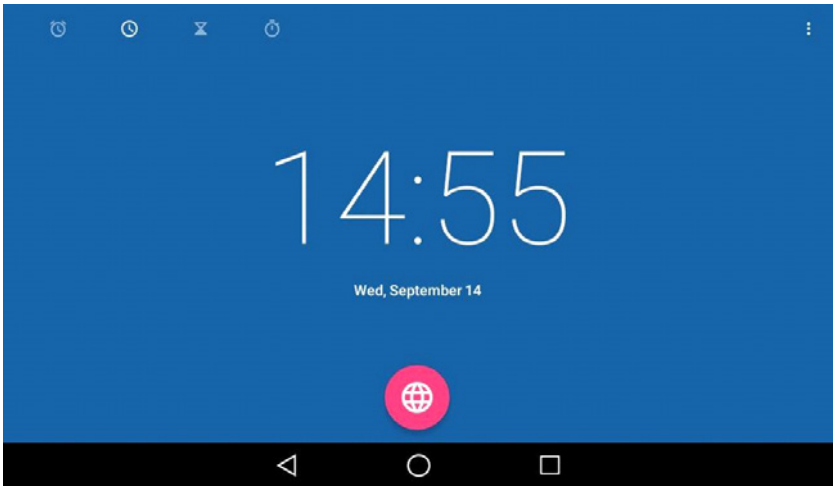
    - **Snooze:** Press  when the alarm is ringing, the alarm will stop and ring after a specified period of time.
    - **Dismiss:** Press  when the alarm is ringing, the alarm will stop.
    - **Do nothing:** Press  when the alarm is ringing, the alarm will still ring.

Clock

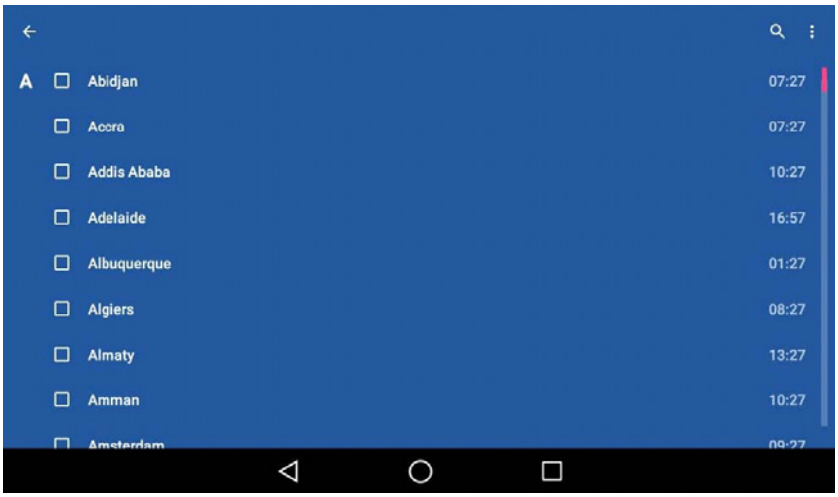
To add the clock of other cities in the world:

- Tap the digital clock widget on the home screen.

The touch screen displays the current date and time.

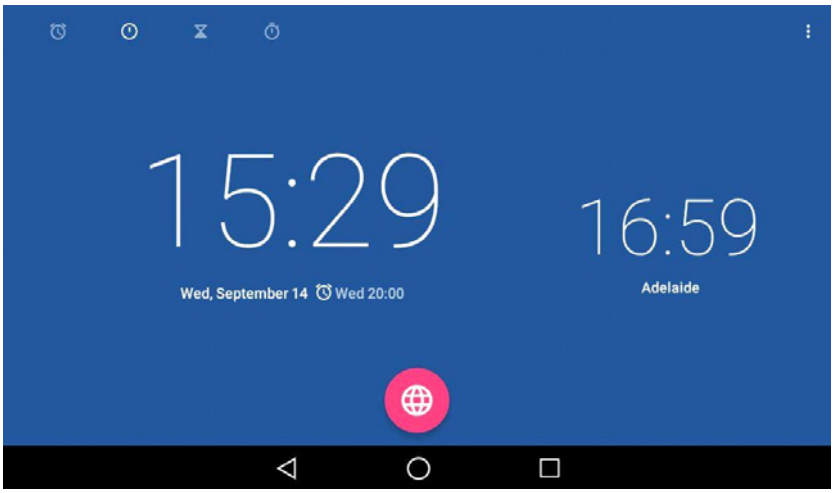


- Tap .



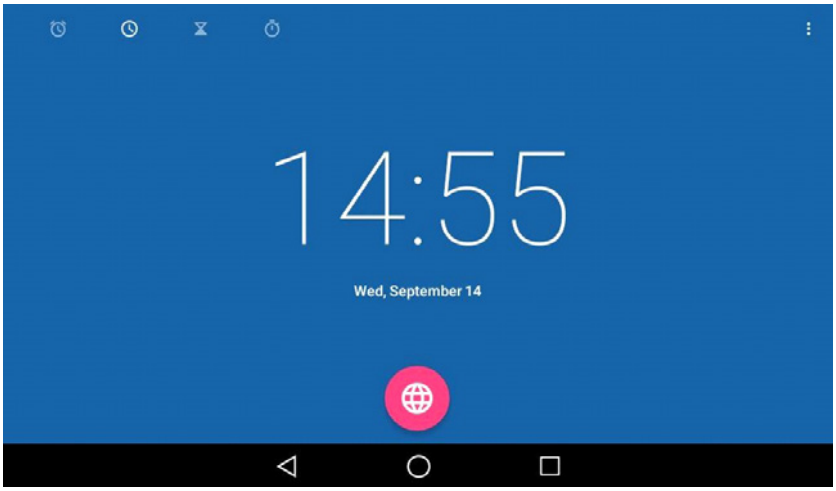
- Drag up and down to scroll through the cities.
- Tap the checkbox before the desired city.
- Tap to return to the clock screen.

The added clock of the desired city displays on the touch screen.



To configure the clock display mode:

- Tap the digital clock widget on the home screen.

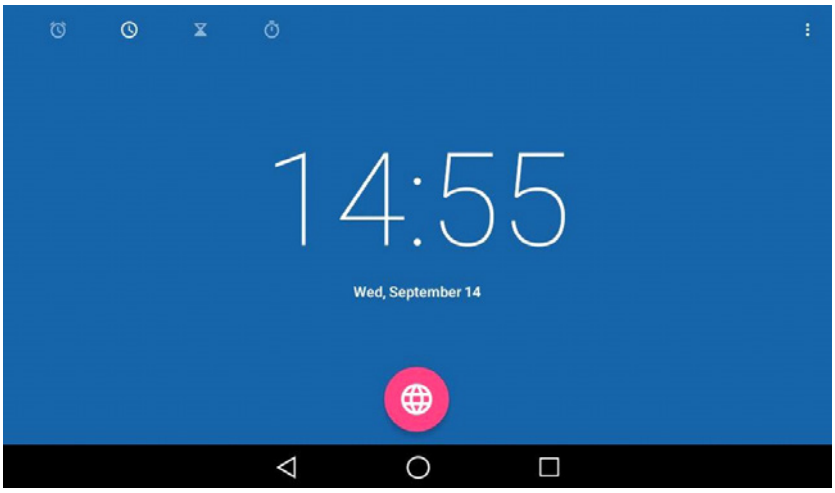


- Tap and then tap **Settings.**
- You can do the following:
  - Tap the **Style** field.
    Tap **Analog** or **Digital** in the pop-up dialog box.
  - Mark the Automatic **home clock** radio box.
    - Tap the **Home time zone** field.
      Tap a desired time zone in the pop-up dialog box for the home clock.

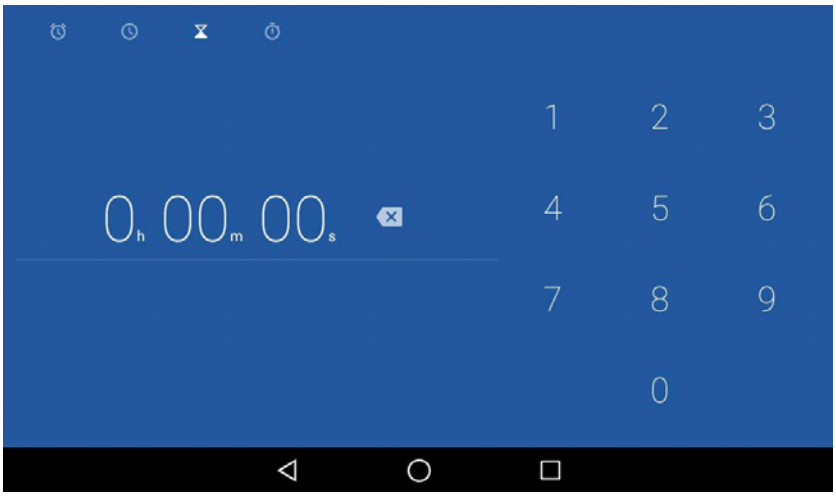
Timer

To set a timer:

- Tap the digital clock widget on the home screen.

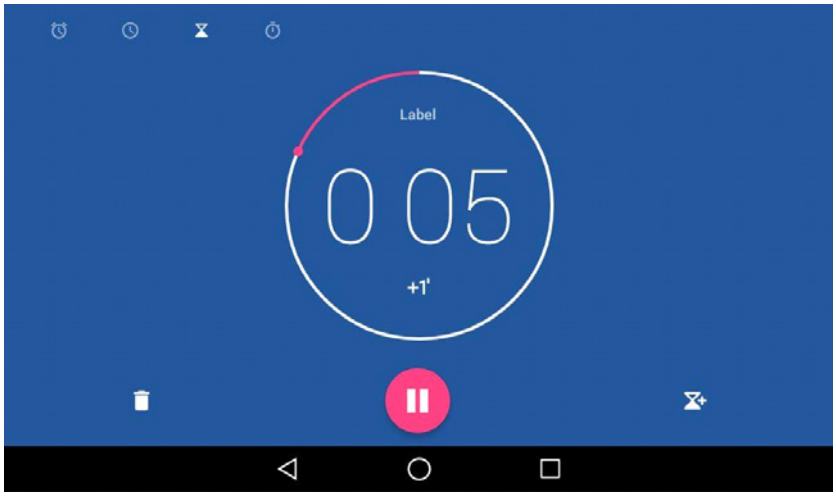


- Tap .



- Enter the time by tapping the digits on the right of the screen.
- You can tap  to delete the entered numbers.

- Tap  .



When the timer is running, you can do the following:

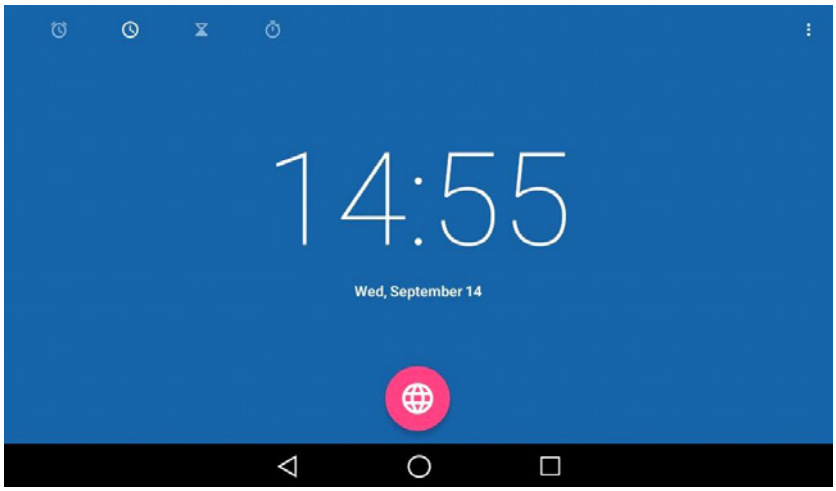
- Tap  to add an additional minute.
- Tap  to pause the timer, then you can do the following:
  - Tap  to resume the timer.
  - Tap  to restart the timer.
- Tap  to delete the current timer.

The timer beeps when the time is up. It keeps beeping until you tap to stop.

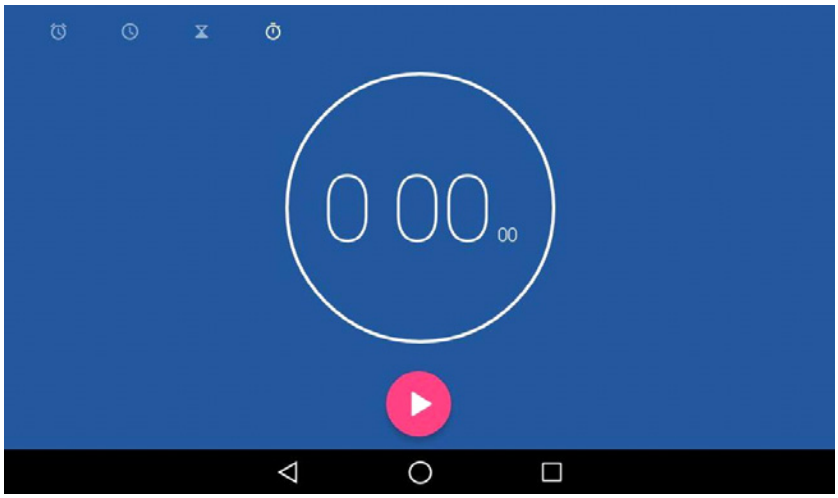
Stopwatch

To set a stopwatch:

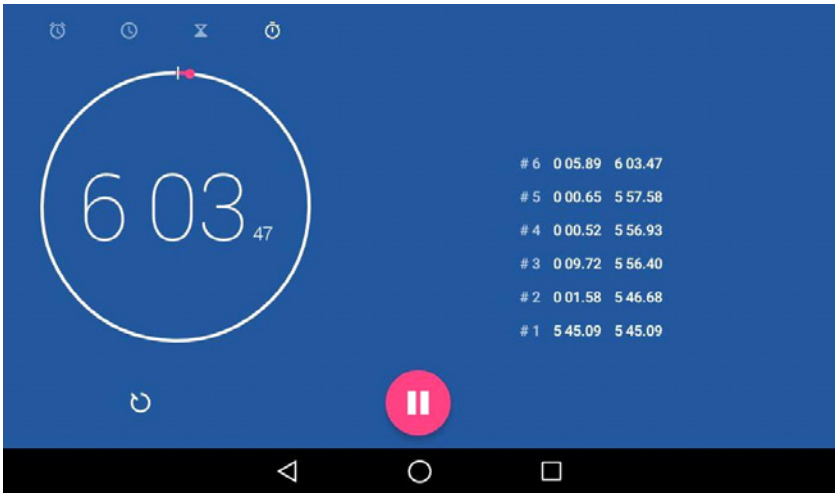
- Tap the digital clock widget on the home screen.



- Tap  .



- Tap  to start the stopwatch.
- You can do the following:
  - Tap  to pause the stopwatch.
    - Tap  to reset the stopwatch.
    - Tap  and then tap Bluetooth or Email to share the record via Bluetooth or email. For more information, refer to “Sending Files via Bluetooth” and “Sending Files via Email”.
  - Tap  to add laps while the stopwatch is running.



CHAPTER 8

Account Management

Account Registration

Registering a SIP account makes it easier for the IP phone to receive an incoming call, dial an outgoing call. The IP phone supports registering multiple accounts on a phone; each account requires an extension or phone number. The IP phone support up to 16 accounts. You can register the SIP Account with some ITSP's (Internet Telephony Service Provider) proxy server or you can configure the IP phone to make/receive Peer-to-Peer (P2P) calls (or, direct IP calling) without the intervention of any SIP proxy.

The IP phone support SIP server redundancy for account registration. For more information, refer to "Server Redundancy". If you want to customize multiple DSS keys to associate with an account, refer to "Multiple Line Keys per Account".

Configuration Proxy and Peer-to-Peer Calling

Proxy Calling

For Proxy Calling you need to register an account with an ITSP For more information refer "Accept SIP Trust Server Only".

Procedure

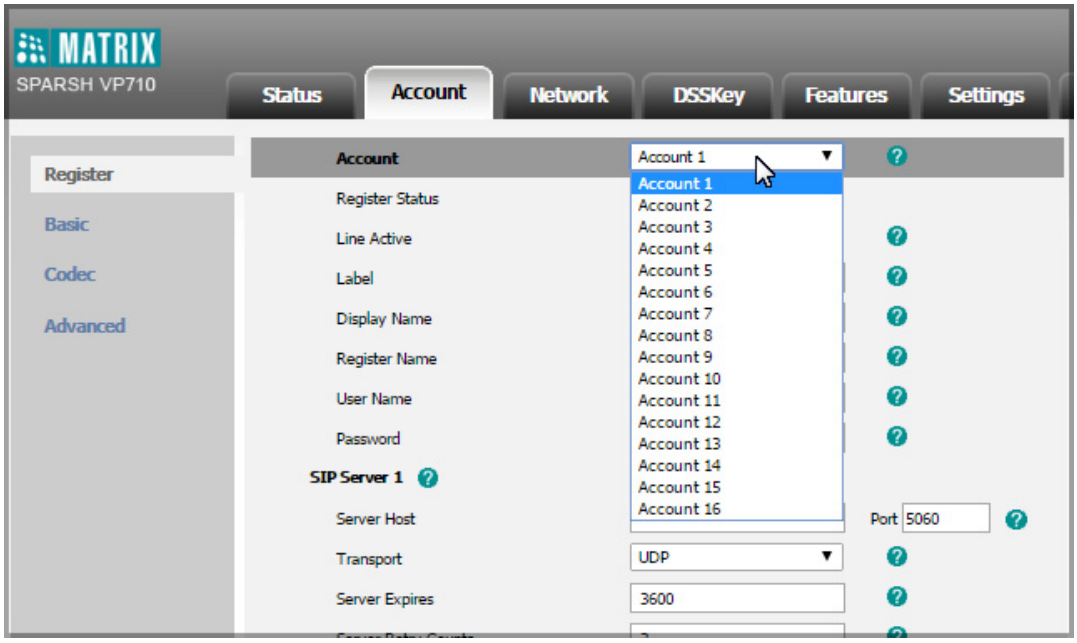
Account Registration can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the account registration information. Configure the interval for the IP phone to retry to re-register when registration fails. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure the account registration information.Configure the account registration information. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-register&q=load&acc=0 Configure the interval for the IP phone to retry to register when registration fails. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0
	Phone User Interface	Configure the account registration information

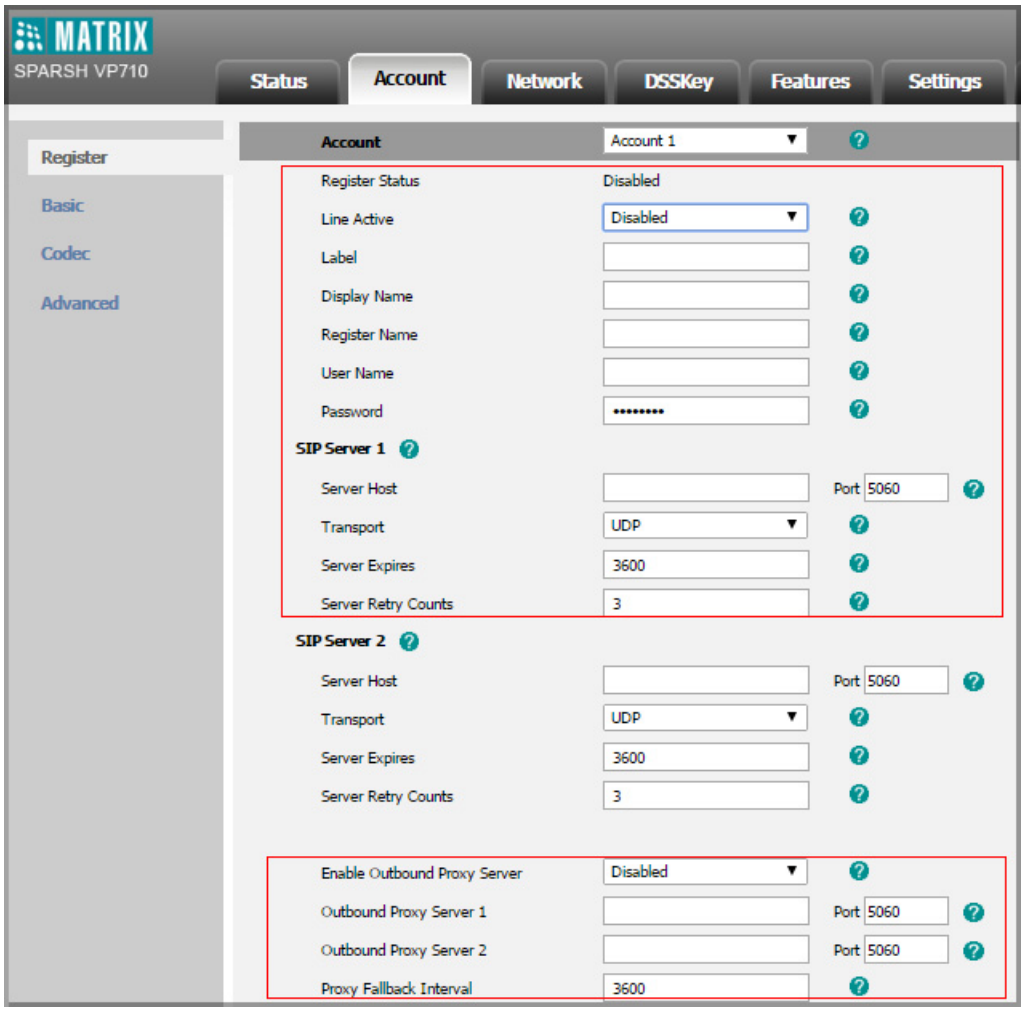
To register an account via web user interface:

- Click **Account->Register.**
- Select the desired account from the **Account** list.



- Select **Enabled** from the **Line Active** list.
- Enter the desired value in **Label, Display Name, Register Name, User Name, Password** and **SIP Server1/2** in the respective field.
- If you use outbound proxy servers, do the following:

- Select **Enabled** from the **Enable Outbound Proxy Server** list.
- Enter the desired IP address or domain name in the **Outbound Proxy Server 1/2** field and the desired port of the outbound proxy server 1/2 in the **Port** field respectively.
- Enter the desired interval in the **Proxy Fallback Interval** field.

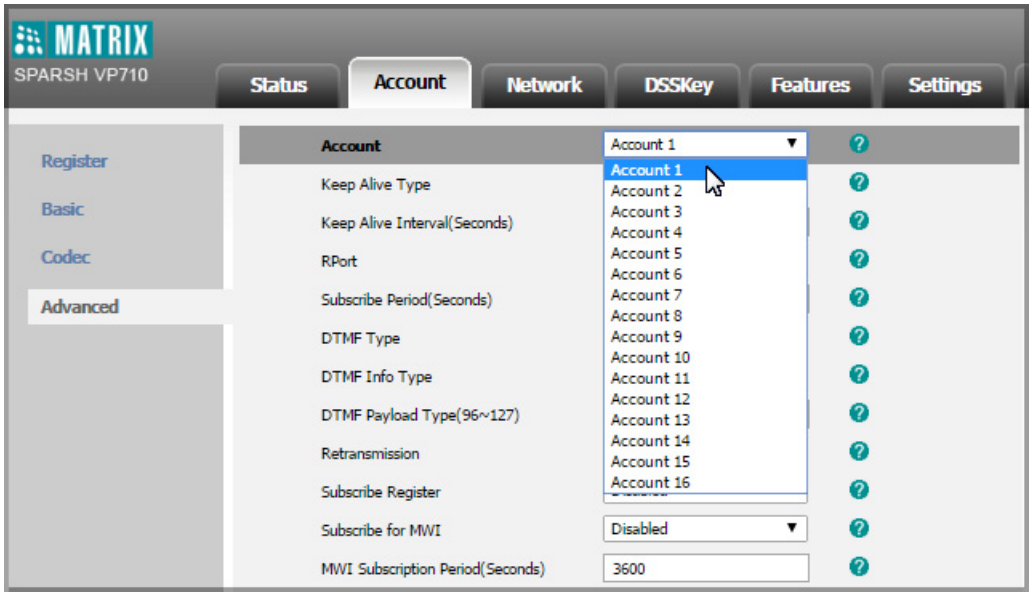


- Click **Confirm** to save the change.

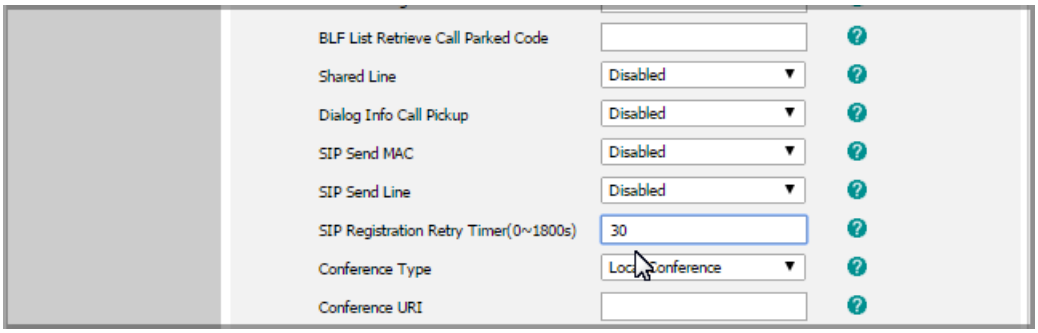
To configure the interval for re-register when registration fails via web user interface:

- Click **Account->Advanced.**

- Select the desired account from the **Account** list.



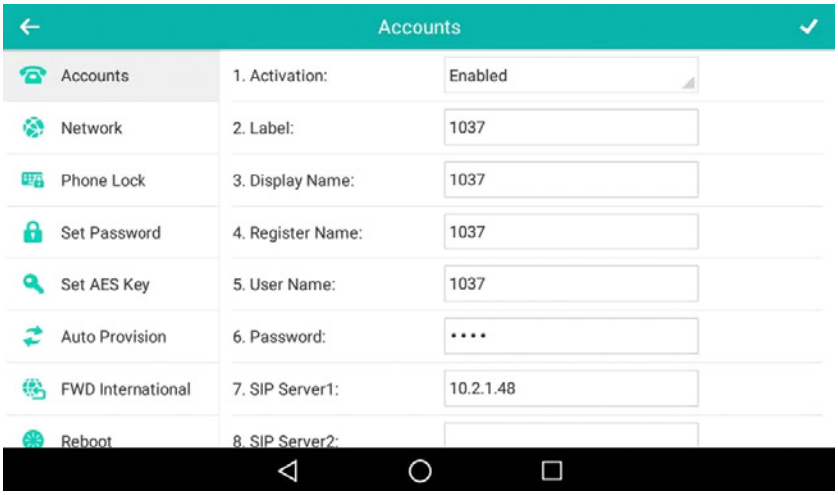
- Enter the desired interval in the **SIP Registration Retry Timer(0~1800s)** field.



- Click **Confirm** to save the change.

To register an account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced (default password: admin) ->Accounts.**
- Tap the desired account.
- Tap the **Activation** field.
- Tap **Enabled** in the pop-up dialog box.
- Enter the desired value in the **Label, Display Name, Register Name, User Name, Password and SIP Server1/2** field respectively. Contact your system administrator for more information.



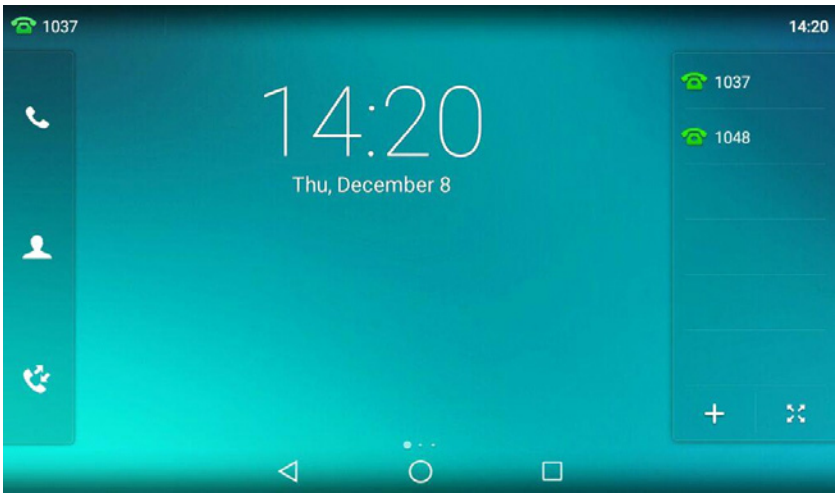
- If you use the outbound proxy servers, do the following:
  - Tap the **Outbound Status** field.
  - Tap **Enabled** in the pop-up dialog box.
  - Enter the desired value in the **Outbound Proxy1/2** and **Proxy Fallback Interval** field respectively. Contact your system administrator for more information.
- Tap ✓ to save the change or ◁ to cancel.
- You can repeat steps 3 to 8 to register more accounts.

The following figures demonstrate single or multiple accounts registered on the phone:

Single account:



Multiple accounts:



To disable an account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced (default password: admin) ->Account.**
- Tap the desired account.
- Tap the **Activation** field.
- Tap **Disabled** in the pop-up dialog box.
- Tap  to save the change or  to cancel.

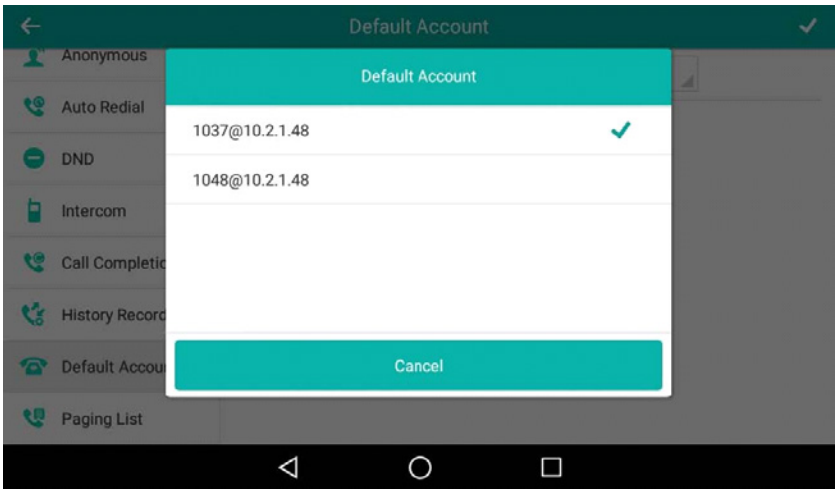
Peer-to-Peer (P2P) Calling

Matrix SPARSH VP710 supports Peer-to-Peer (P2P) calling and Direct IP calling using the phone dial pad without registering to the SIP server. For more information refer “Allow IP Call” and “IP Direct Auto Answer”.

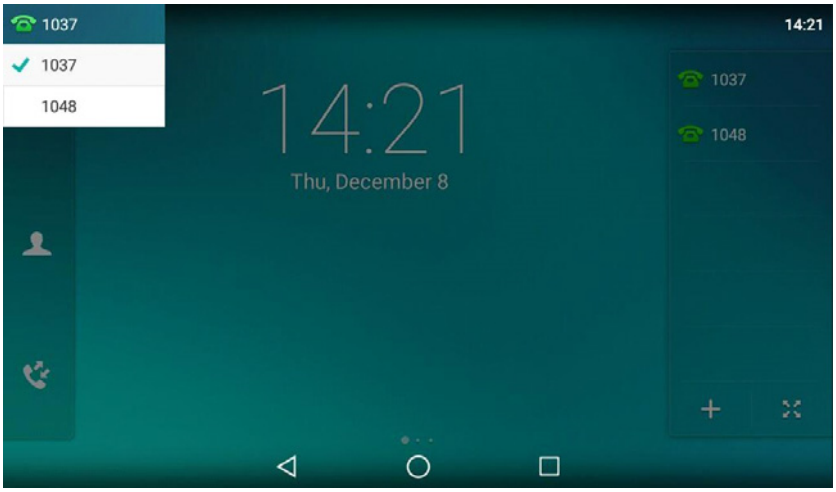
Default Account

To configure the default account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Default Account.**
- Tap the **Default Account** field.
- Tap the desired account in the pop-up dialog box.



- Tap ✓ to save the change or ◁ to cancel.
- The label of the default account is displayed on the left of the status bar. The IP phone will use this account by default when dialing out.
- You can also tap the label of the default account, and then tap the desired account to be the new default account.



Default account is configurable via phone user interface only.

Multiple Line Keys per Account

You can configure multiple line keys to associate with an account. This enhances call visualization and simplifies call handling.

You can customize the number of DSS keys to be automatically assigned with Line type. It means multiple DSS keys will associate with an account. It is useful for managing a high volume of calls to a line. For more information on how to register accounts, refer to "Account Registration".

The number of the DSS keys associated with an account in case of SPARSH VP710 can be less than or equal to 27.

The following shows two line keys associated with a registered account 1037:



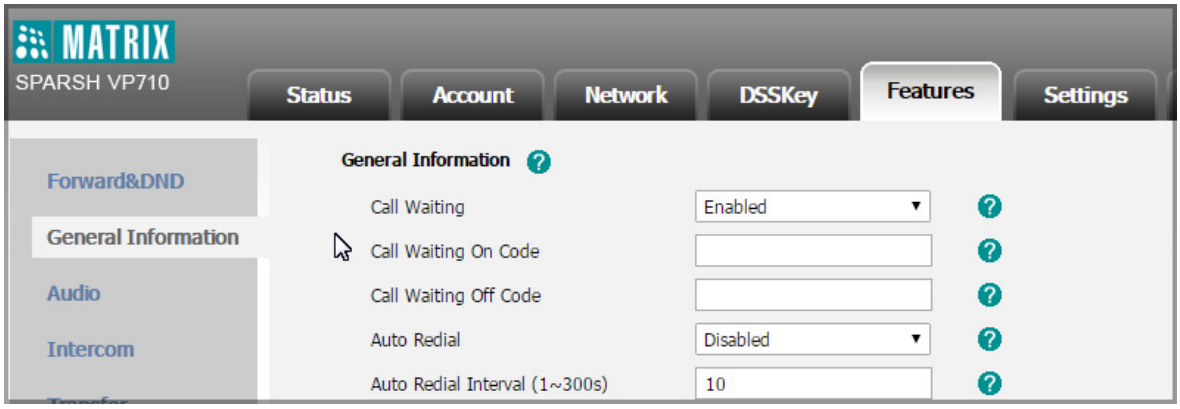
Procedure

Multiple line keys per account can be configured using the following methods.

Configuration File	<MAC>.cfg	Configure auto linekeys. Configure the number of DSS keys to be assigned automatically. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure auto linekeys. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load Configure the number of DSS keys to be assigned automatically. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

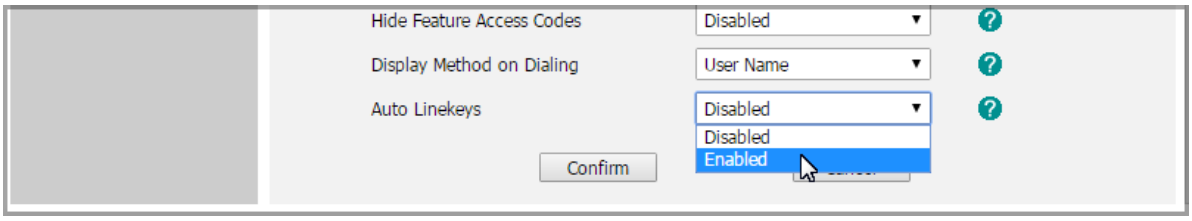
To configure auto linekeys feature via web user interface:

- Click on **Features->General Information.**



- Select **Enabled** from the **Auto Linekeys** list.

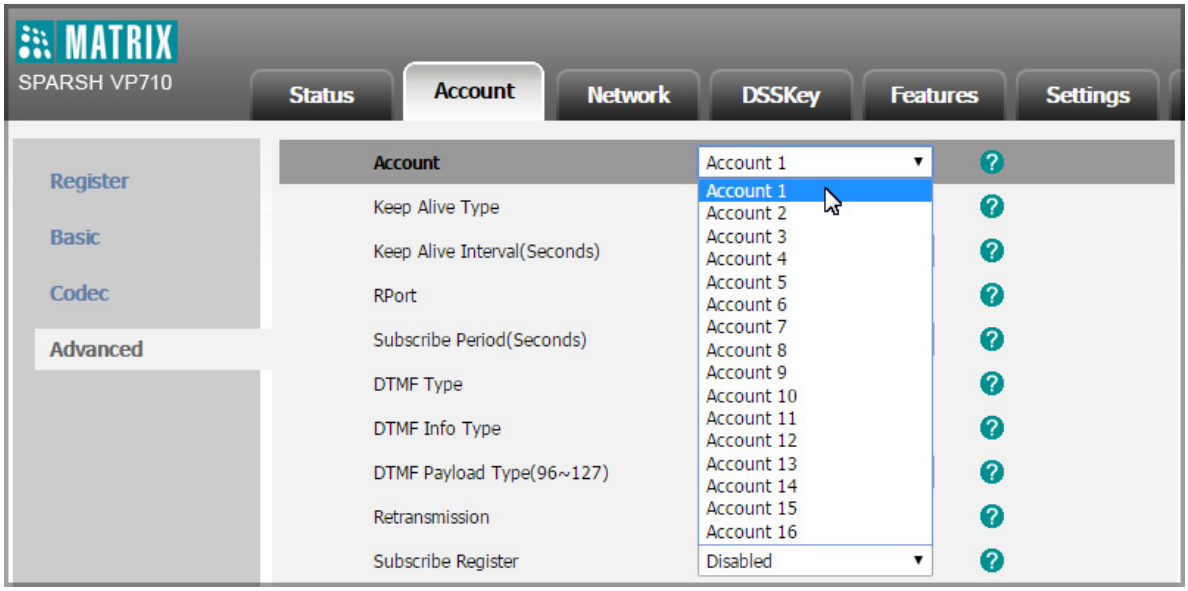
  If Auto LineKeys is enabled, you can automatically assign multiple DSS keys with Line type for a registered line on the phone.



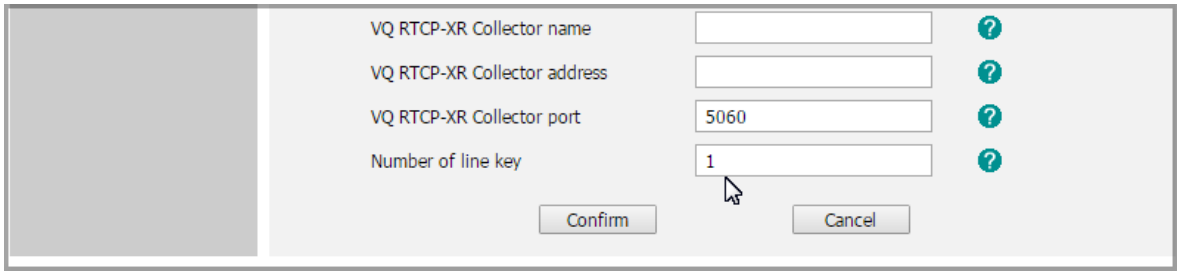
- Click **Confirm** to save the change.

To configure the number of line keys via web user interface:

- Click **Account->Advanced.**
- Select the desired account from the **Account** list.

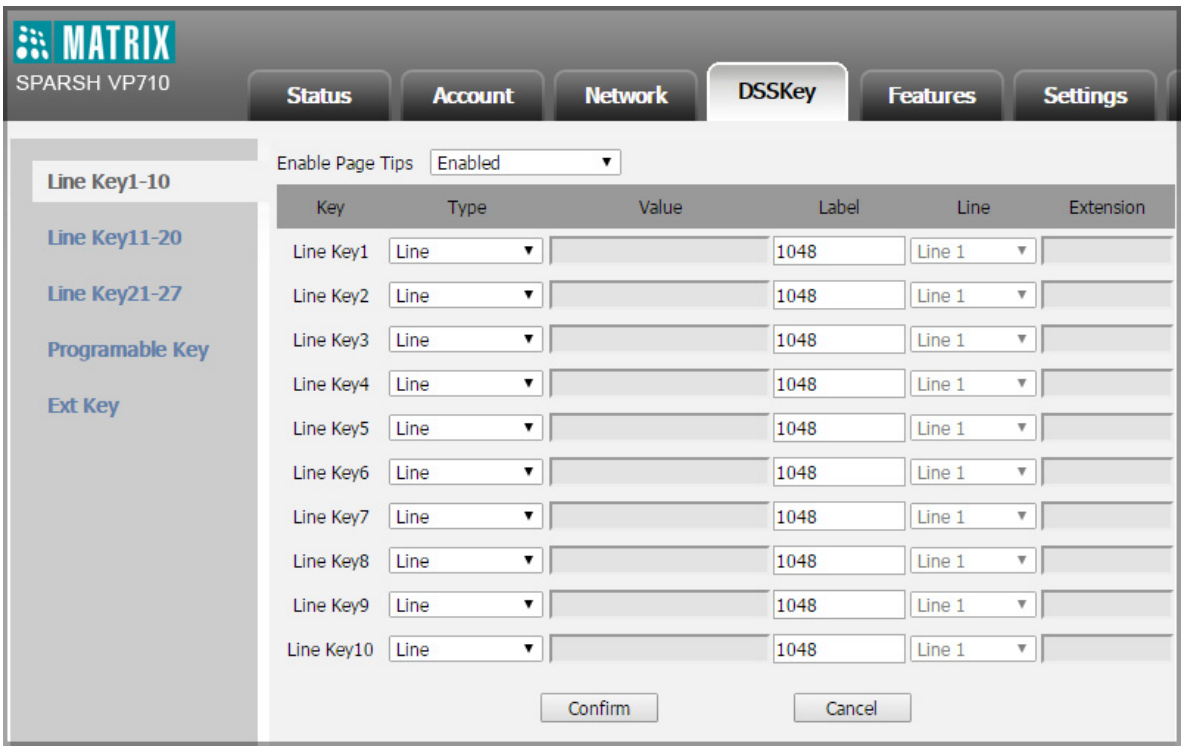


- Enter the desired number in the **Number of line key** field. This field appears only if **Auto Linekeys** is enabled.



- Click **Confirm** to save the change.

Once you have defined the number of line keys (example 10) to be associated with an account (example 1048), on the DSS Keys page these are automatically reserved for the account 1048.



If this is the case, the touch screen will resemble the following figure:



Incoming calls to this account will be distributed evenly among the available line keys. Outgoing calls will be distributed similarly.

Your phone can be configured to have a combination of accounts with a single line key and accounts with multiple line keys.

Mobile Account

The IP phone supports connecting a Bluetooth-enabled mobile phone. After connection, the IP phone will automatically find an available DSS key and assign the DSS key for Mobile Account.

The Mobile Account key's default label is "My Mobile". If there is no available DSS key, you may assign it manually.

The Mobile Account key can be used in the following scenarios:

- Accept the incoming mobile call if there is an incoming call to your mobile phone.
- Make a call through a mobile phone. But the IP phone acts as a hands free device for your mobile phone.
- Reconnect the last paired Bluetooth-Enabled mobile phone if the distance between mobile phone and IP phone is more than 10 meters or the Bluetooth mode on the mobile phone is deactivated.

For more information on how to use your phone in conjunction with Bluetooth-enabled mobile phone. For details, refer "Bluetooth".

Procedure

Mobile account key can be configured using the following methods.

Configuration File	<MAC>.cfg	Assign a mobile account key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Assign a mobile account key. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-preference&q=load
	Phone User Interface	Assign a mobile account key.

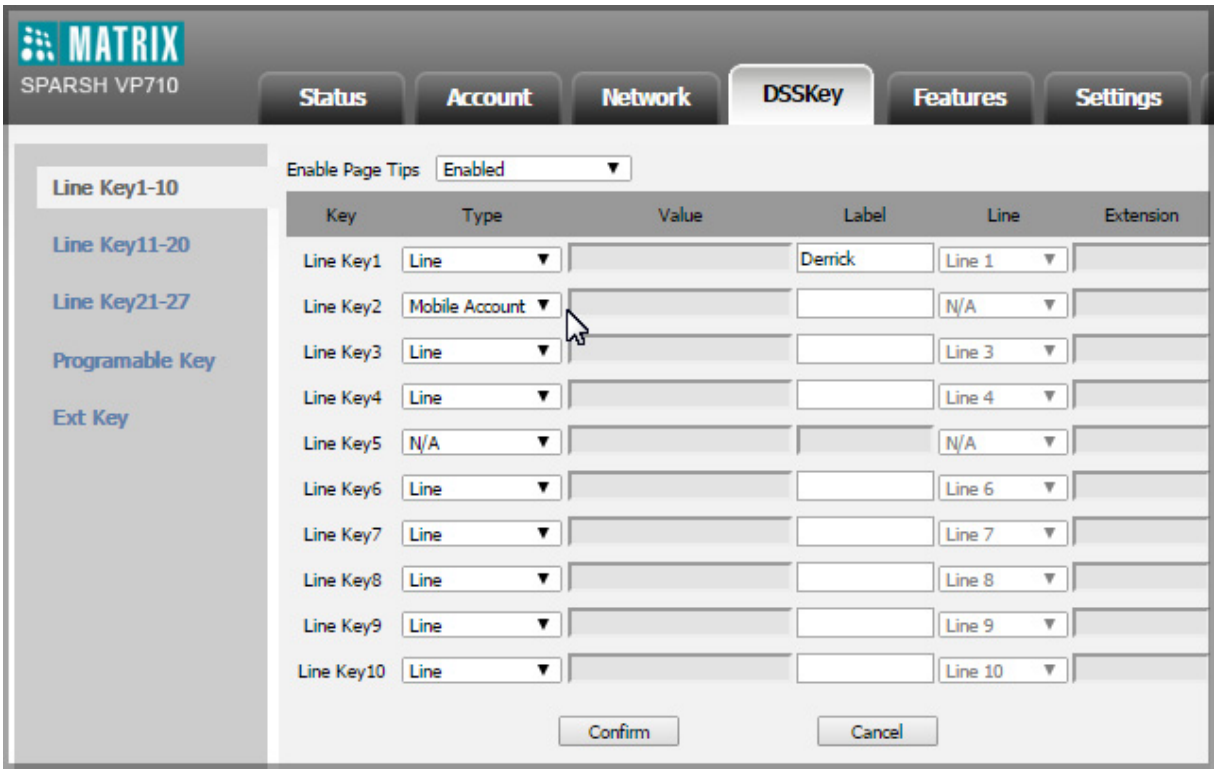
For more information on how to configure the DSS Key, refer "DSS Keys" and "Configuring DSS Key".

Mobile account key can be configured only if the mobile phone has been connected successfully.

To configure a mobile account key via web user interface:

- Click on **DSSKey->Line Key (or Programable Key).**
- In the desired DSS key field, select **Mobile Account** from the **Type** list.

- Enter the string that will appear on the touch screen in the **Label** field.



- Click **Confirm** to save the change.

To configure mobile account key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Select the desired DSS key.
- Tap the **Type** field.
- Tap **Mobile Account** in the pop-up dialog box.
- Enter the string that will appear on the touch screen in the **Label** field.
- Tap ✓ to save the change or ◁ to cancel.

CHAPTER 9

Accessing Basic Features

The SPARSH VP710 IP phone is designed to be easily used like a regular phone on a public switched telephone network (PSTN). You can place calls, answer calls, transfer a call to someone else, or conduct a conference call.

This chapter provides basic operating instructions for the IP phone. Topics include:

- Placing Calls
- Answering Calls
- Ending Calls
- Performing Operations during a Call
- Redialing Numbers
- Accept SIP Trust Server Only
- Allow IP Call
- Anonymous Call
- Anonymous Call Rejection
- Auto Answer
- Auto Redial
- Call Completion
- Call Forward
- Call Hold/Resume
- Call Number Filter
- Call Park
- Call Pickup
- Call Recording
- Call Timeout
- Call Transfer
- Call Waiting
- Calling Line Identification Presentation (CLIP)
- Connected Line Identification Presentation (COLP)
- Conference
- Door Phone
- Do Not Disturb (DND)
- Feature Key Synchronization

- Hide Feature Access Code
- Intercom
- IP Direct Auto Answer
- IPv6 Support
- Logon Wizard
- Message
- Music on Hold
- Mute
- Notification Popups
- Password Dial
- ReCall
- Recent Call In Dialing
- Transfer Mode via DssKey

Placing Calls

You can place a call in three ways using your IP phone:

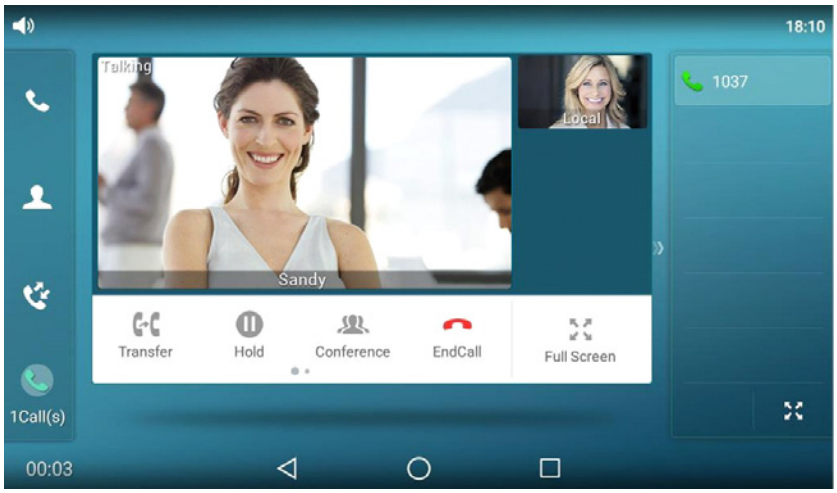
- Using the handset
- Using the speakerphone
- Using the headset

You can also dial the number first, and then choose the way you want to speak to the other party. You can also search and dial a contact from call history, local directory or remote phone book. For more information, refer to “Contact Management” and “Call History Management”.

By default, you can set up a video call with the other party that is video-enabled. The IP phone can set up video calls with only two parties (even if more than two parties that set up a call with you can send video). If you do not connect the camera to the phone, the IP phone will not transmit the near-site video. For more information on how to connect the camera, refer to “Phone Installation”. You can also place an audio-only call with the other party that is video-enabled. For more information, contact your system administrator.

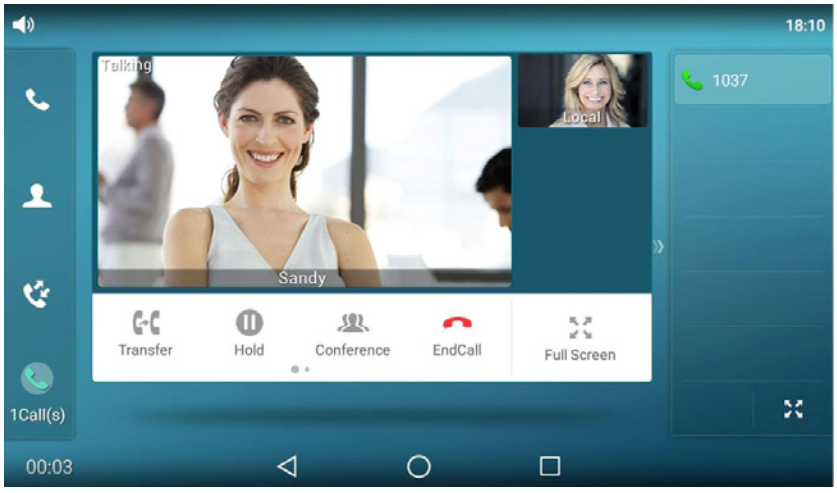
During a call, you can alternate between Speakerphone, Headset and Handset modes by pressing the Speakerphone key, the Headset key, or picking up the handset.

The call duration of the call and far-site's information (name or phone number) are visible on the touch screen. In the figure below, the video call to "Sandy" has lasted 3 seconds.



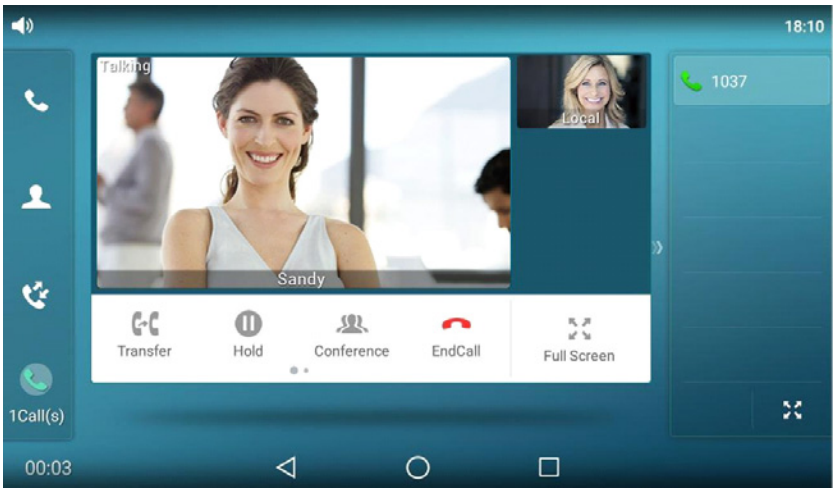
The phone will enter the video call full-screen within 5 seconds during a call, you may tap or blank area of the touch screen to see the soft keys.

In the figure below, the audio-only call to "Jerry" (the phone number: 1039) has lasted 7 seconds.



To place a call using the handset:

- Pick up the handset.
- Enter the desired number using the keypad or the dial pad on the touch screen. Tap to see the onscreen keyboard, and then you can enter the number using the onscreen keyboard.



- Press #SEND, or tap **Send**.

The # key is configured as a send key by default. You can also set the * key as the send key, or set neither. For more information, refer to the "Key as Send".

You can also tap or to search for an entry or contact after picking up the handset, and then tap the desired entry or contact to place a call.

You can also dial using the SIP URI or IP address. To obtain the IP address of a phone, swipe down from the top of the screen or swipe left/right to go to the second idle screen and then tap ***Settings->Status****. The maximum SIP URI or IPv4 address length is 32 characters. For example, SIP URI: 3606@sip.com, IP: 192.168.1.15.*

Your phone may not support direct IP dialing. Contact your system administrator for more information.

To place a call using the hands-free (speakerphone) mode:

Do one of the following:

- With the handset on-hook, press or tap the line key to obtain a dial tone.

  Enter the desired number using the keypad or the dial pad on the touch screen.

  Press #SEND or tap **Send**.
- With the handset on-hook, enter the desired number using the keypad or the dial pad on the touch screen.

  Press , #SEND or tap **Send**.

To place a call using the headset:

Do one of the following:

- With the optional headset connected, press  to activate the headset mode.

  Tap the line key to obtain a dial tone.
  Enter the desired number using the keypad or the dial pad on the touch screen.

  Press  or tap **Send**.

- With the optional headset connected, press  to activate the headset mode.

  Enter the desired number using the keypad or the dial pad on the touch screen.

  Press  or tap **Send**.



To permanently use the headset mode, refer to "Headset Prior".

The IP phone can handle multiple calls at a time. However, only one active call (the call that has audio associated with it) can be in progress at any time. The IP phone can handle a maximum of 50 calls at one time.

To place multiple calls:

You can have more than one call on your IP phone. To place a new call during an active call, do one of the following:

- Tap the line key. The active call is placed on hold.

  Enter the desired number using the keypad or the dial pad on the touch screen.

  Press  or tap **Send**.

- Press  or tap the Hold soft key to place the original call on hold.

  Tap the **NewCall** soft key.
  Enter the desired number using the keypad or the dial pad on the touch screen.

  Press  or tap **Send**.

  You can tap the desired call, and then tap the **Resume** soft key to retrieve it.



If multiple accounts are registered on the phone, you can first tap the desired line key on the idle screen or tap the desired line from the status bar in dialing screen, and then you can use the selected account to place a call.

Answering Calls

When you are not in another call, you can answer a call in three ways:

- Using the handset
- Using the speakerphone
- Using the headset

- *You can ignore incoming calls by pressing the **Reject** soft key or the **Silence** soft key. You can also activate Do Not Disturb mode to ignore all incoming calls without ring on your phone. For more information, refer "Do Not Disturb (DND)".*

- *You can forward incoming calls to someone else by pressing the **Fwd** soft key. For more information, refer "Call Forward".*

Answering When Not in Another Call

Call duration and destination will always appear on the touch screen for the active call.

To answer a call using the handset:

- Pick up the handset.

To answer a call using the hands-free speakerphone mode:

Do one of the following:

- Press .

  OR
- With the handset on-hook and headset mode deactivated, press the **Answer** soft key.

  OR
- With the handset on-hook and the headset mode deactivated, tap the line key.

To answer a call using the headset:

Do one of the following:

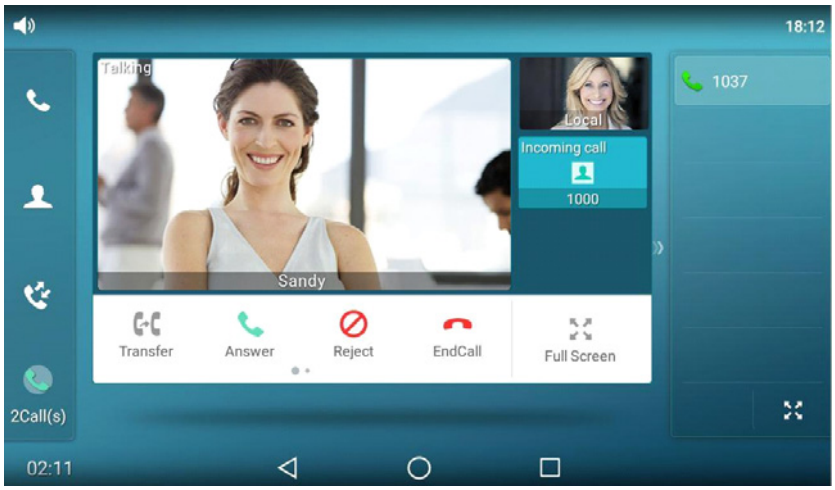
- Press .

  OR
- With the headset mode activated, press the **Answer** soft key.

  OR
- With the headset mode activated, tap the line key.

Answering When in Another Call

If you have an active call, and an incoming call arrives on the phone, The incoming call information is displayed on the touch screen.

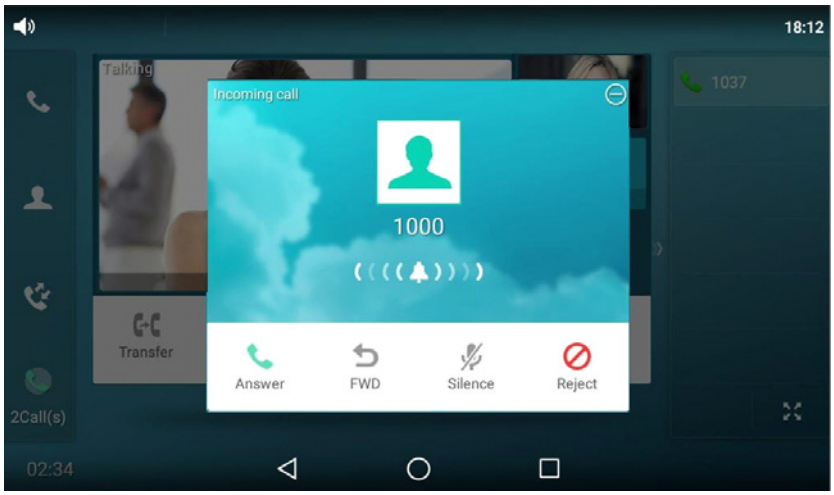


Do one of the following:

- Press the **Answer** soft key.

  OR
- Tap the incoming call avatar.

  Tap **Answer**.



The incoming call is answered and the original call is placed on hold.

The phone will not receive a new incoming call when there are two calls on the phone.

Ending Calls

To end a call:

Do one of the following:

- If you are using the handset, tap the **EndCall** soft key or hang up the handset.

OR

- If you are using the headset, tap the **EndCall** soft key.

OR

- If you are using the speakerphone, press  or tap the **EndCall** soft key.

*To end a call placed on hold, you can press the **EndCall** soft key to end the call directly, or press the **Resume** soft key to resume the call before ending it.*

Performing Operations during a Call

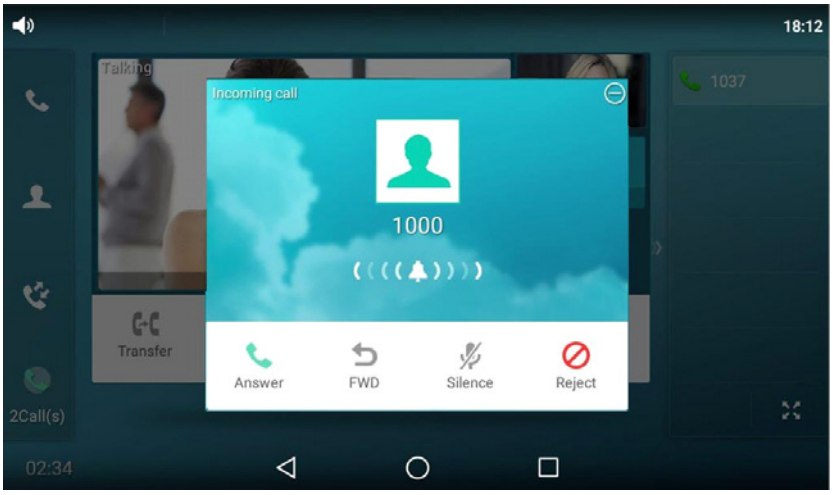
During the call, you may want to see if there is an event on schedule or check if there is a new email in the inbox and so on, you can exit the talking screen to access the corresponding application. And after the operation, you can return to the call conveniently.

The following operations are not allowed during the call:

- Record by Recorder application.
- Take pictures.
- Record videos.
- Playback audios/videos.
- Launch Camera application.

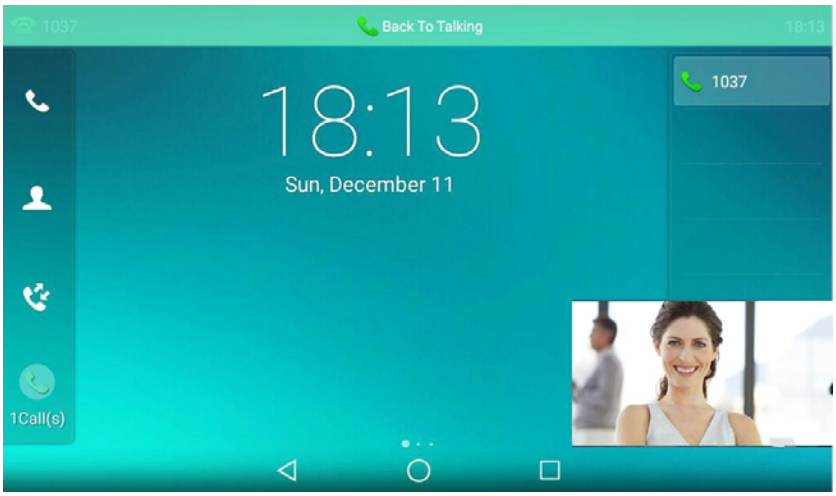
Going to the Idle Screen during a Call

To go to the idle screen during a call:



- Tap ,  or  to go to the idle screen.

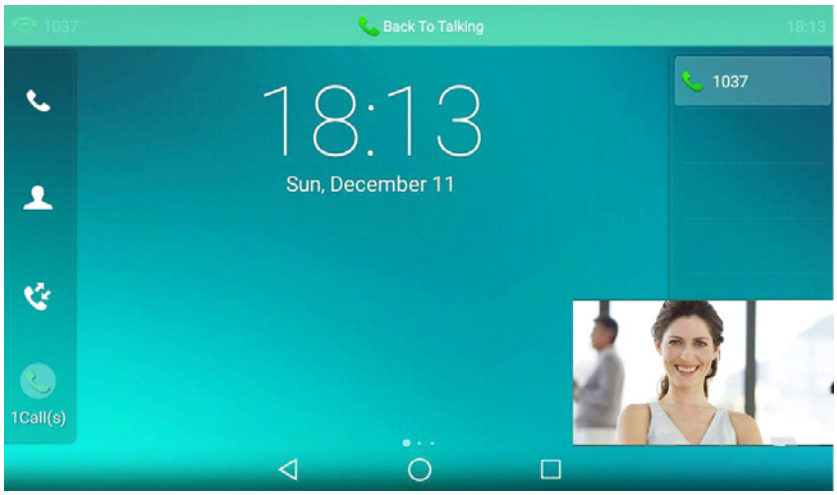
When the phone exits the talking screen, the touch screen will display as below:



By default, a floating video window (far site) will display on the bottom-right of the touch screen. If you are in a conference video call, the floating video window will display all participants video image. If you hold the video call or you are in an audio-only call, the floating video window will not display. You can drag the floating video window to the desired location.

Returning to the Talking Screen

To return to the talking screen:



- Tap .
- Tap the floating video window.
- Tap **Back To Talking** on the top of the touch screen.

Changing Video Call Settings

When you update video settings, you can change the way for the current video call to display on your IP phone.

You can update the following video settings on your phone:

- Stop Transmitting Video during a Call
- Change the Video Layout during Calls
- Preview the Near-Site Video

Stopping Transmitting Video during a Call

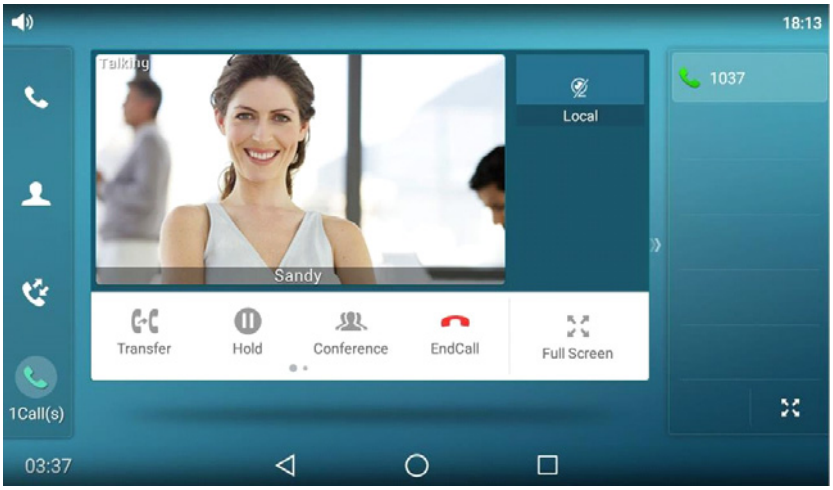
To stop transmission,

- Unplug the camera

  OR
- Swipe down from the top of the screen in the video call screen
- Tap Video to stop your phone from automatically transmitting your video during a video call.

You cannot turn off far-site video.

If you stop transmitting video, the icon will display on the near-site video, as shown below:



Your camera will not transmit video to your contact and the far site cannot see you.

Stopping transmitting video does not create an audio-only call. Even if you stop transmitting video, you can see the other party, and the call is still a video call. To place an audio-only call, contact your system administrator.

Changing the Video Layout during Calls

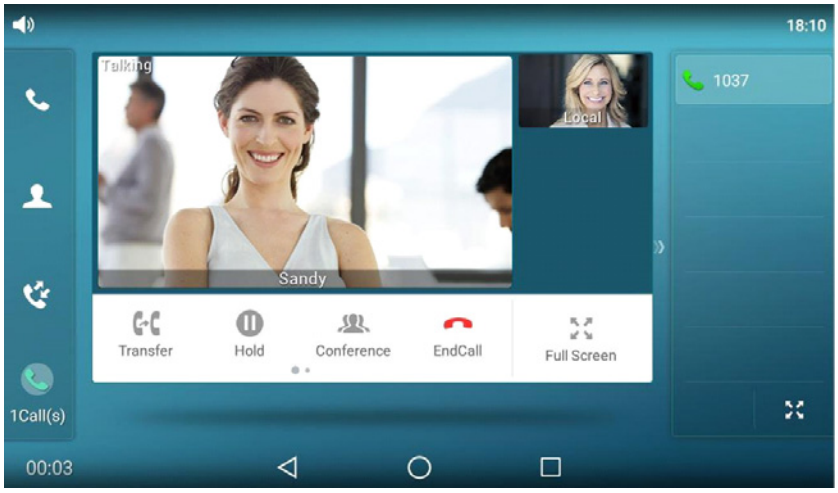
During video calls, near-site and far-site video images are displayed on the touch screen. You can change the video layout.

The system supports three video layouts:

- One large, other small: One video image displays in a large size, the other video images along the right side of the screen displays in small sizes.
- Full screen: One video image displays in full size, the other video images are hidden.
- Same size: All video images display in the same size.

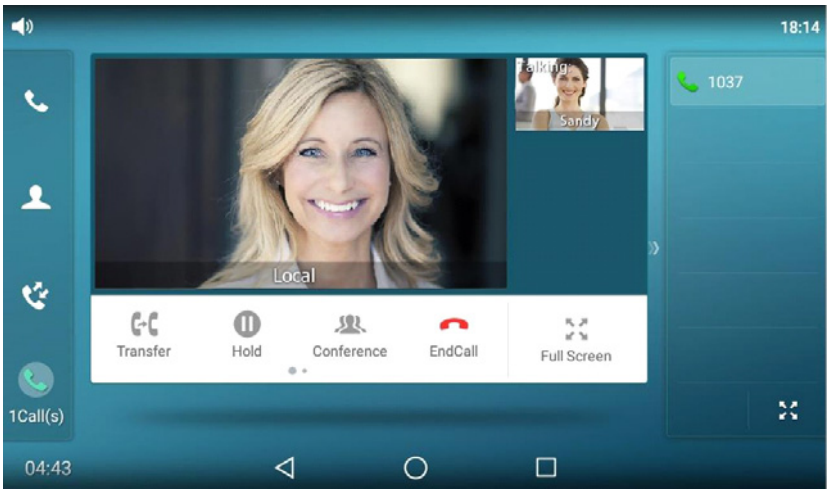
To swap video during a video call via phone user interface:

- Tap the near-site (small window) video during a video call.



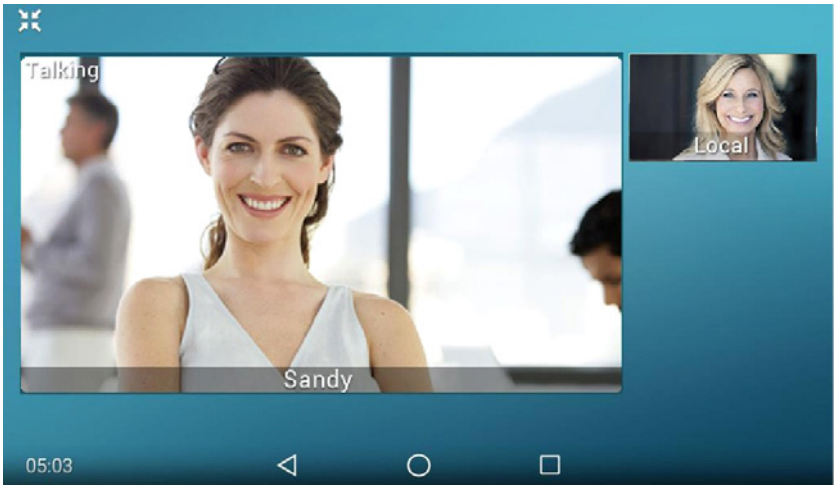
The near-site video will display in the center of the touch screen, and the far-site video will display on the top-right of the touch screen.

The following figure shows what the touch screen looks like after you tap the near-site video:

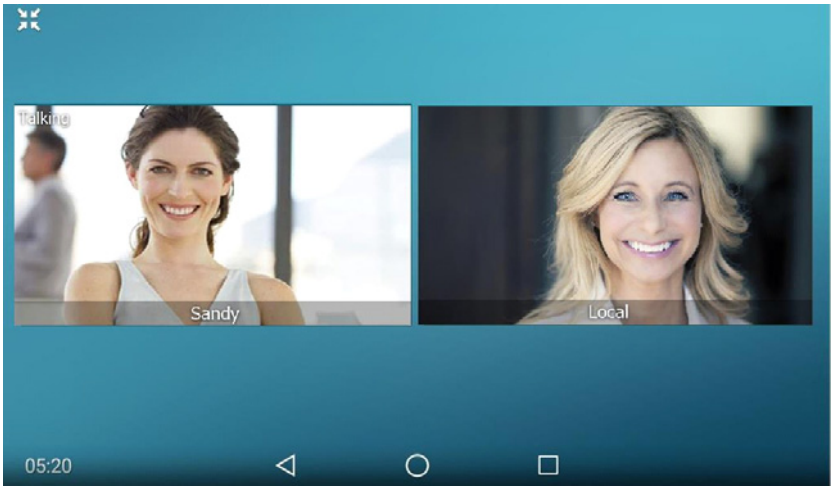


To change the video layout to the same size mode during a video call via phone user interface:

- From the video call full-screen, pinch close the large window.



The following figure shows what the touch screen looks like in the same size mode:

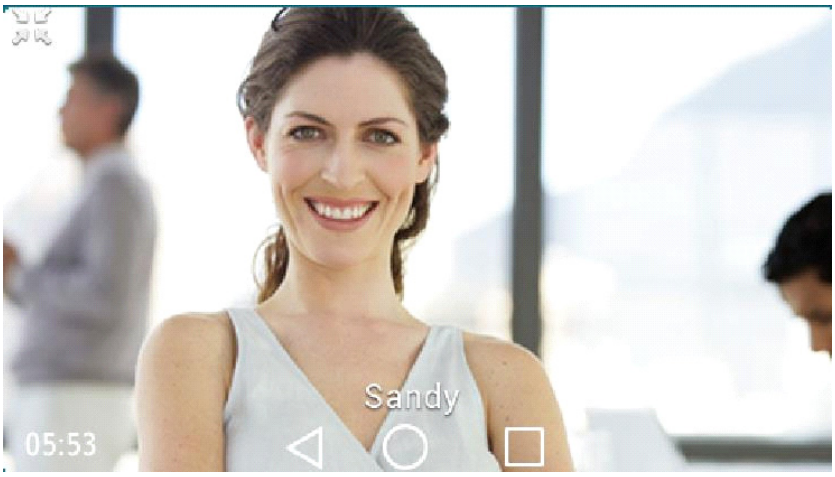


You can tap / or the blank area of the touch screen to go back to video call screen (refer to"Video Call Screen Display") or tap the desired image to go back to "one big, other small" layout.

To change the video layout to the full screen mode during a video call via phone user interface:

- From the video call full-screen, do one of the following:
  - Tap the large window.
    OR
  - Pinch open the large window.

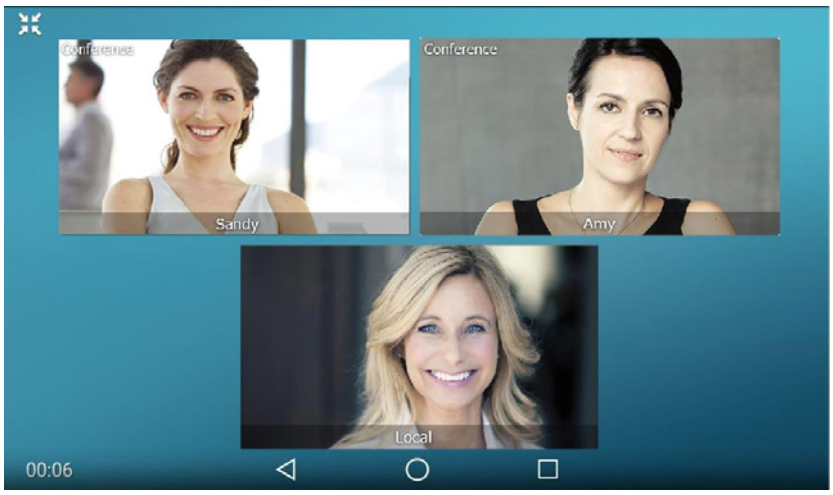
The small window is hidden, and the following figure shows what the touch screen looks like in full screen mode:



You can tap ◁ or  to go back to video call screen (refer to “Video Call Screen Display”) or tap the video area to go back to “one big, other small” layout.

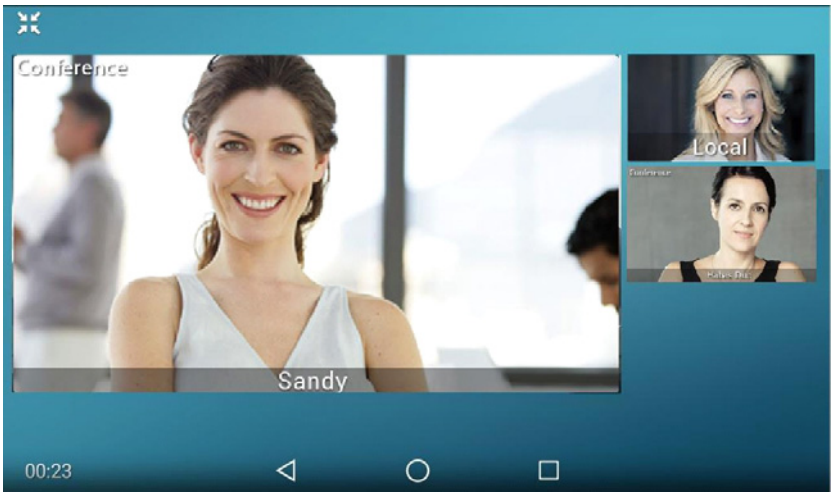
To change the video layout during a video conference call via phone user interface:

The default screen layout during a video conference call is the same size mode. The following figure shows what the touch screen looks like during a three-way video conference call:



You can tap ◁ /  or the blank area of the touch screen to go back to video call screen (refer to “Video Call Screen Display”) or tap one of the conference party to go back to the “one big, other small” layout.

The following figure shows what the touch screen looks like in the “one big, other small” layout during a three-way video conference call:



Previewing the Near-Site Video

If you have connected the camera to the phone, you can preview the video of yourself that you display during video calls. For more information on how to connect the camera, refer “Phone Installation”. During previewing, you can also configure camera settings. For more information, refer to “Camera Settings”.

To preview the near-site video when the phone is idle, do one of the following:

- Swipe down from the top of the screen to enter the control center and tap Video.
- Swipe left or right to go to the second idle screen and tap to launch Camera application.

Redialing Numbers

To redial the last dialed number from your phone:

- Press twice.

  A call to your last dialed number is attempted.

To redial a previously dialed number from your phone:

- Press when the phone is idle.
- Tap the desired entry from the placed calls list.

Accept SIP Trust Server Only

Accept SIP trust server only enables the IP phone to accept the SIP message only from the SIP server and outbound proxy server you are registered with.

It can prevent the IP phone from receiving ghost calls from random numbers like 100, 1000 etc.To stop this, you need to disable allow IP call feature. For more information on allow IP call, refer to "Allow IP Call".

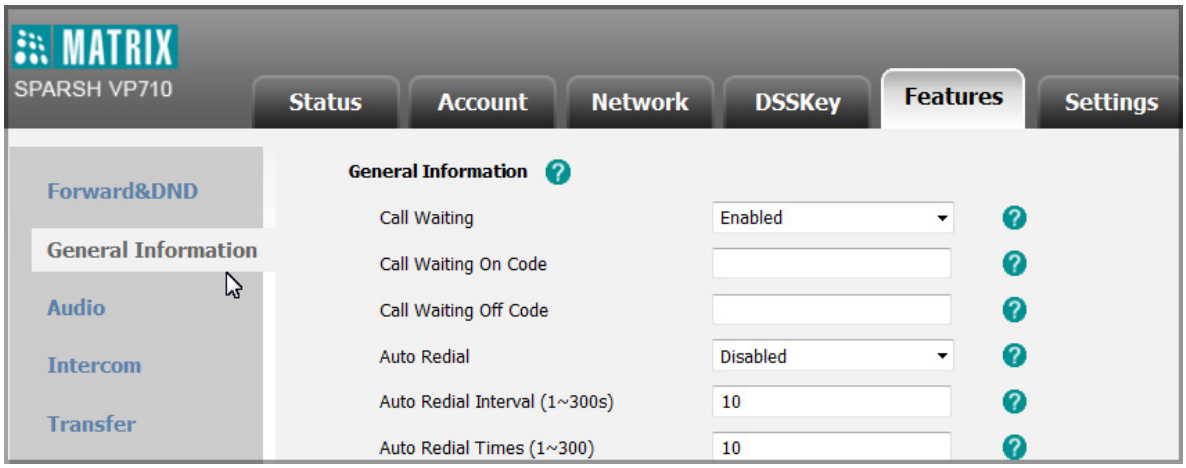
Procedure

Accept SIP trust server only can be configured using the configuration files or locally.

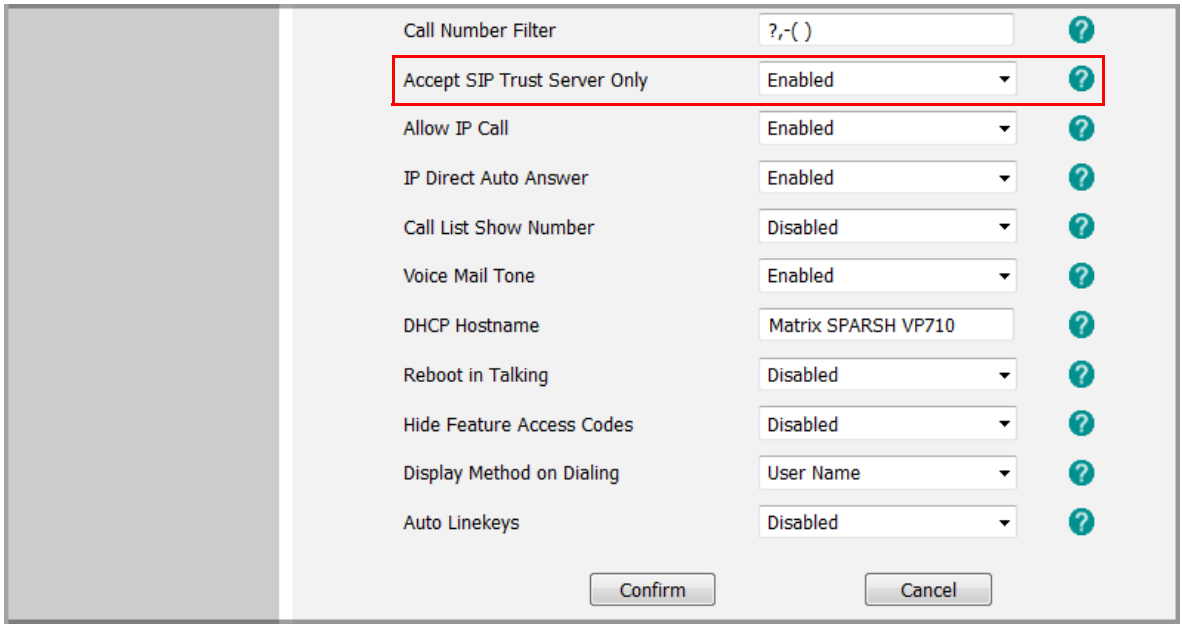
Configuration File	<MAC>.cfg	Configure accept SIP trust server only. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure accept SIP trust server only. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure accept SIP trust server only via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Accept SIP Trust Server Only** list.



- Click **Confirm** to save the change.

Allow IP Call

Allow IP Call allows IP phone to receive or place an IP address call. If allow IP call is disabled, you can neither receive nor place an IP address call.

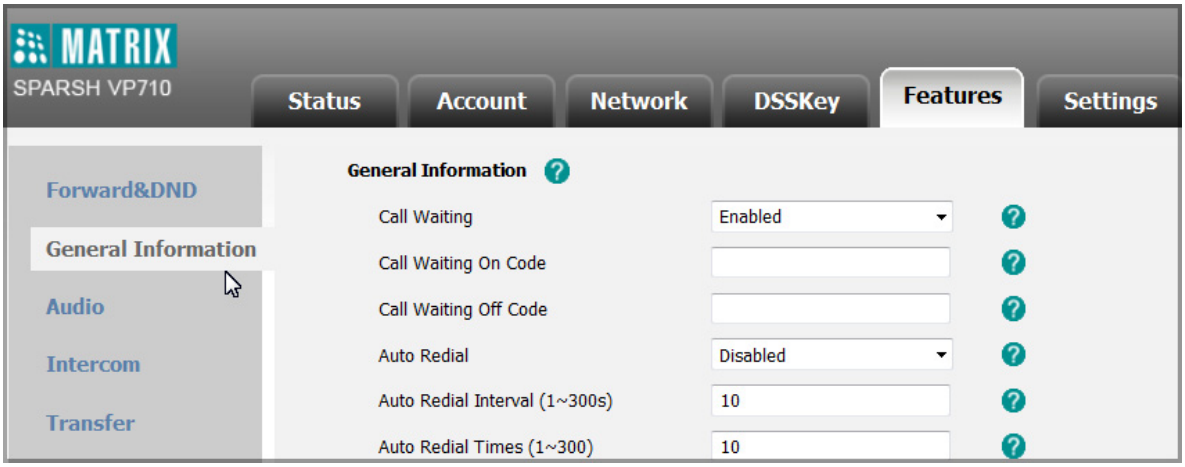
Procedure

Allow IP call can be configured using the configuration files or locally.

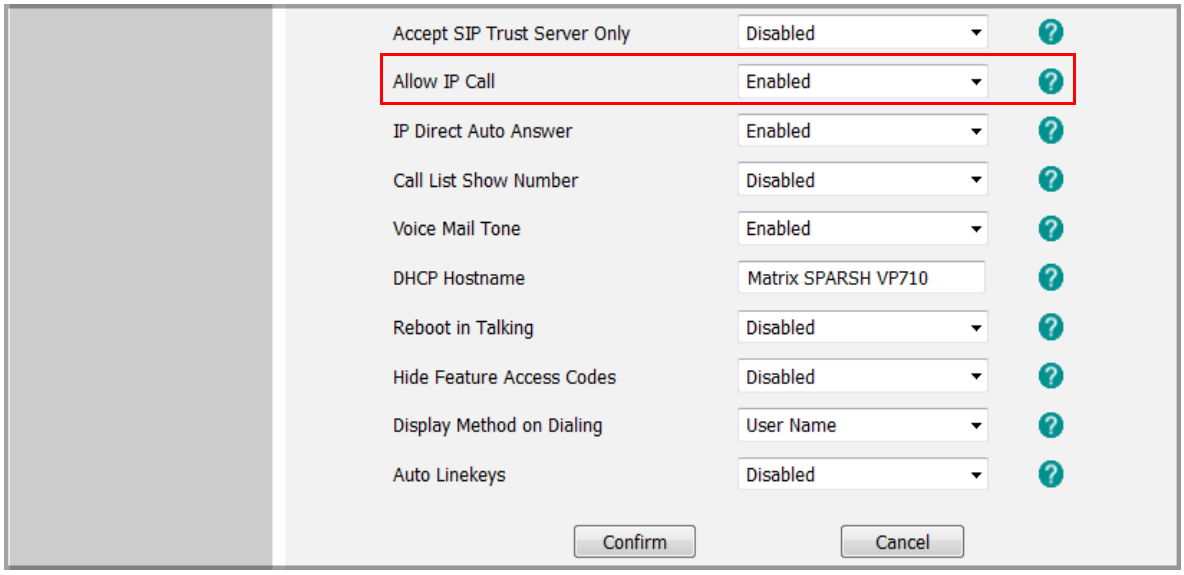
Configuration File	<MAC>.cfg	Configure allow IP call. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure allow IP call. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure allow IP call feature via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Allow IP Call** list.



- Click **Confirm** to save the change.

Anonymous Call

You can use anonymous call to block the identity and phone number from showing up to the called party when you call someone. For example, you call a consulting service for information, but you don't want them to harass you. You can use this feature to hide your identity.

Anonymous call is configurable on a per-line basis. The anonymous call on code and anonymous call off code configured on the IP phone are used to activate/deactivate the server-side anonymous call feature. They may vary on different servers.

Anonymous call is not available on all servers. Contact your system administrator for the anonymous call on code and off code.

Example of anonymous SIP header:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3074920774
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=131654239
To: <sip:1006@10.2.1.48:5060>
Call-ID: 0_288363101@10.3.20.14
CSeq: 1 INVITE
Contact: <sip:1009@10.3.20.14:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Allow-Events: talk,hold,conference,refer,check-sync
P-Preferred-Identity: <sip:1009@10.2.1.48>
Privacy: id
Content-Length: 302
```

Procedure

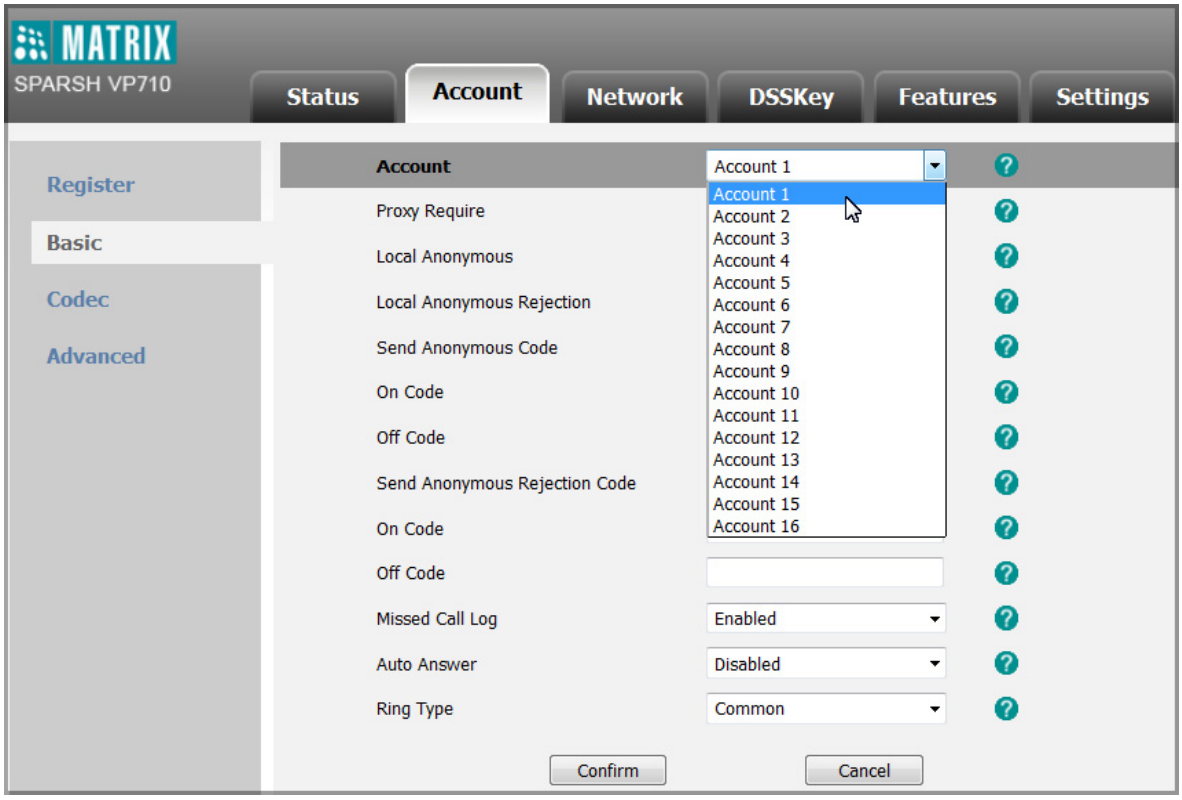
Anonymous call can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure anonymous call. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure anonymous call **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-basic&q=load&acc=0.
	Phone User Interface	Configure anonymous call.

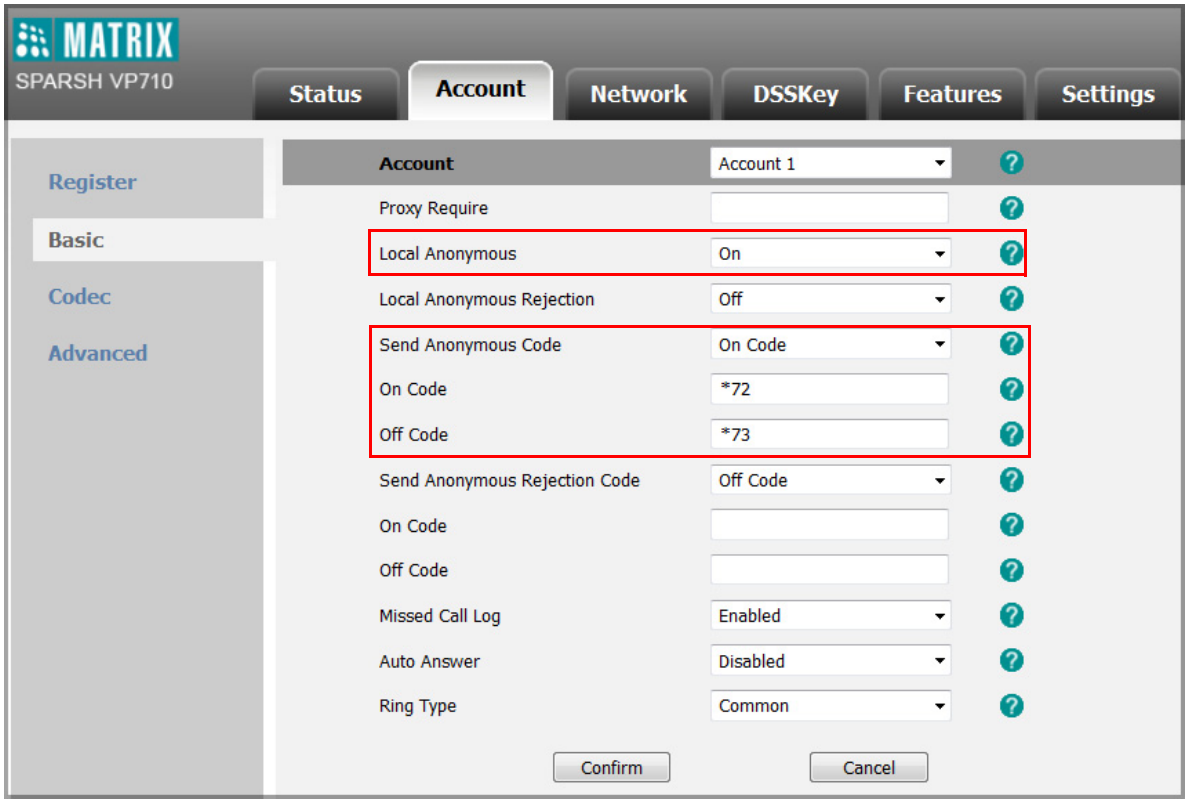
To configure anonymous call via web user interface:

- Click on **Account-> Basic**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Local Anonymous** list.
- Select the desired value from the **Send Anonymous Code** list.
- In the **On Code** field (optional), enter the anonymous call on code.

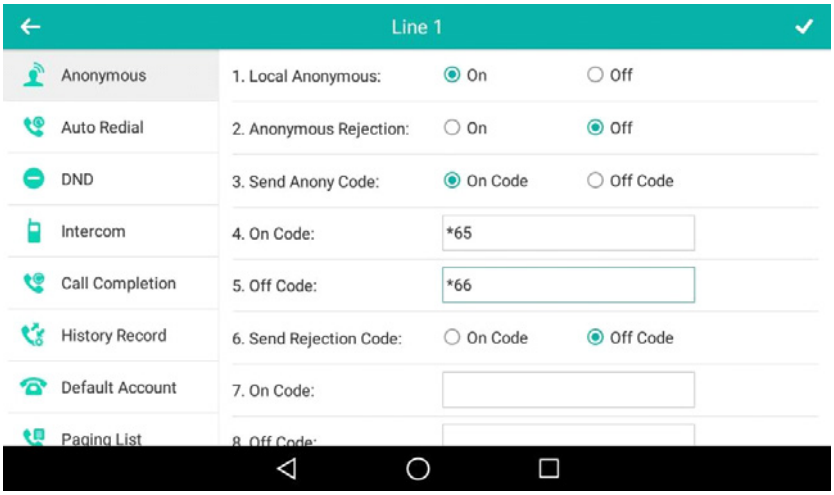
- In the **Off Code** field (optional), enter the anonymous call off code.



- Click **Confirm** to save the change.

To configure anonymous call via phone user interface:

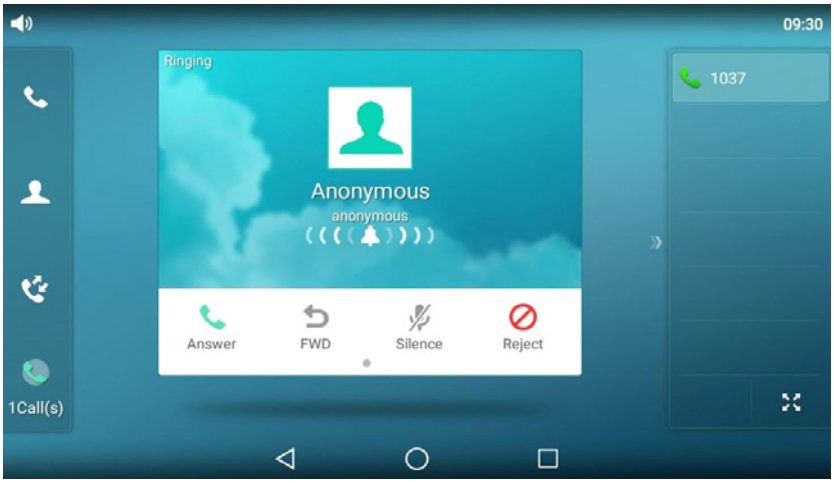
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Anonymous.**
- Tap the desired line.
- Tap the **On** radio box in the **Local Anonymous** field.
- In the **Send Anony Code** field (optional), tap the **On Code** or **Off Code** radio box.
- In the **On Code** and **Off Code** field (optional), enter the anonymous call on code and off code.



- The phone will send the configured on code or off code depending on your selection when you enable or disable anonymous call feature on the phone.
- Tap ✓ to save the change or ◁ to cancel.

To place an anonymous call:

- Using the specific line on the phone to place a call to phone B.
- The touch screen of phone B prompts an incoming call from anonymous.



Anonymous Call Rejection

You can use anonymous call rejection to reject incoming calls from anonymous callers. Anonymous call rejection automatically rejects incoming calls from callers who deliberately block their identities and numbers from being displayed. The anonymous caller's phone touch screen presents "Anonymity Disallowed".

Anonymous call rejection is configurable on a per-line basis.The anonymous call rejection on code and anonymous call rejection off code configured on the IP phone are used to activate/deactivate the server-side anonymous call rejection feature.They may vary on different servers.

Example of anonymous SIP header:

SIP/2.0 **433 Anonymity Disallowed**

Via: SIP/2.0/UDP 10.10.20.32:5060;branch=z9hG4bK2816884590

From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=2625078618

To: <sip:1058@10.2.1.48:5060>;tag=2781829106

Call-ID: 4_510565349@10.10.20.32

CSeq: 1 INVITE

Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH,UPDATE, MESSAGE

User-Agent: MATRIX SPARSH VP710 58.80.196.2 00:1b:09:04:8e:3c

Allow-Events: talk, hold, conference, refer, check-sync

Content-Length: 0

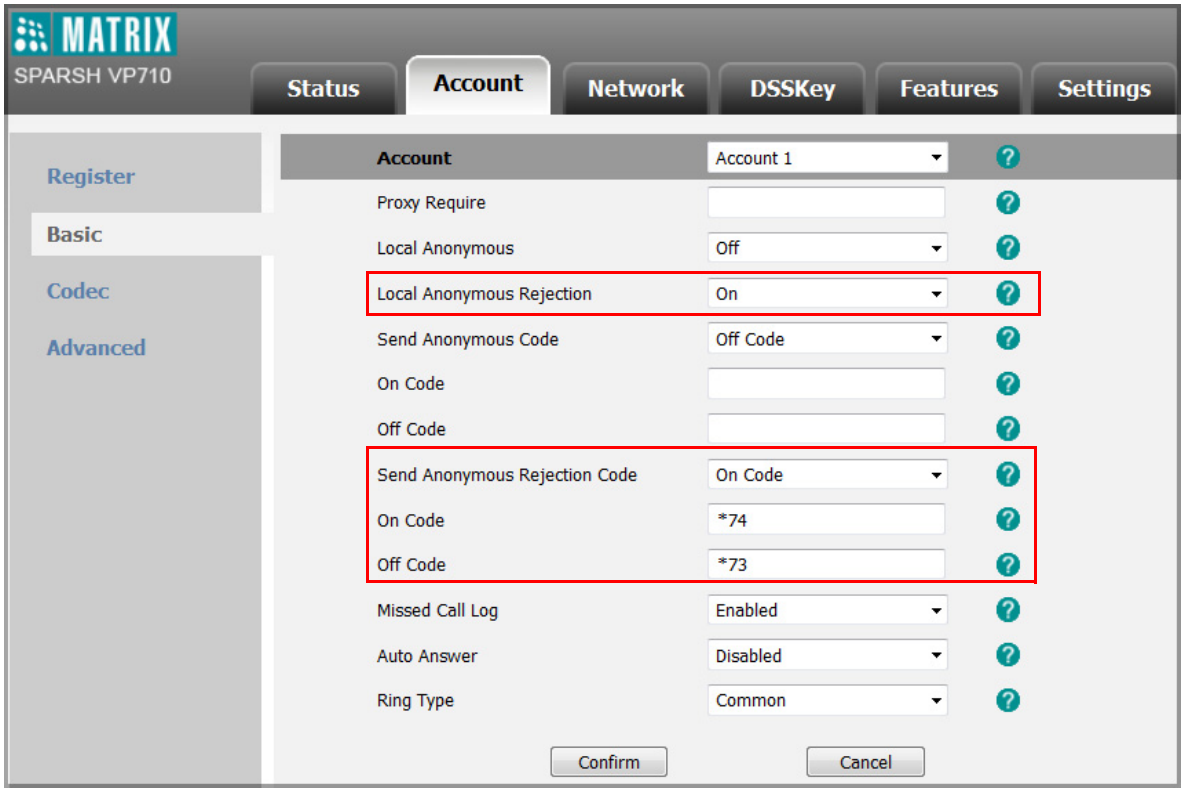
Procedure

Anonymous call rejection can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure anonymous call rejection. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure anonymous call rejection. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-basic&q=load&acc=0
	Phone User Interface	Configure anonymous call rejection.

To configure anonymous call rejection via web user interface:

- Click on **Account->Basic**.
- Select the desired account from the **Account** list.
- Select the desired value from the **Local Anonymous Rejection** list.
- Select the desired value from the **Send Anonymous Rejection Code** list.
- In the **On Code** field (optional), enter the Send Anonymous Rejection on code.
- In the **Off Code** field (optional), enter the Send Anonymous Rejection off code

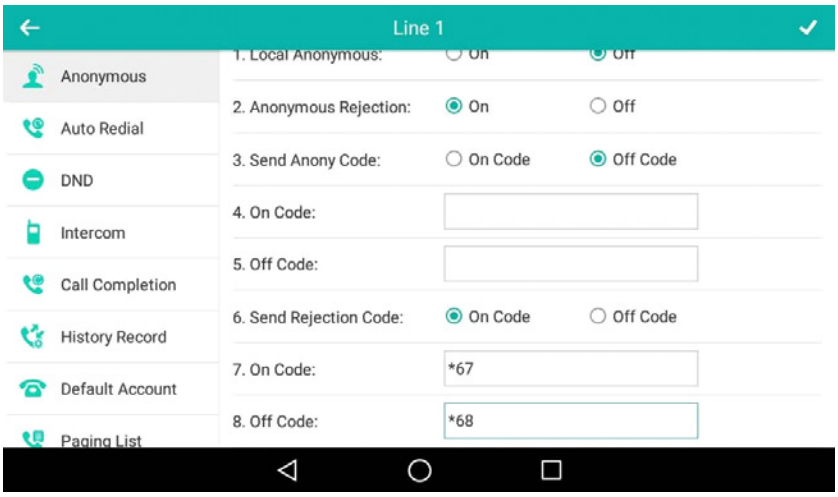


- Click **Confirm** to save the change.

To configure anonymous call rejection via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen
- Tap **Settings->Features->Anonymous Call**.
- Tap the desired line.
- Tap the **On** radio box in the **Anonymous Rejection** field.
- In the **Send Rejection Code** field (optional), tap the **On Code** or **Off Code** radio box.
- In the **On Code** and **Off Code** field (optional), enter the anonymous call rejection on code and off code respectively.

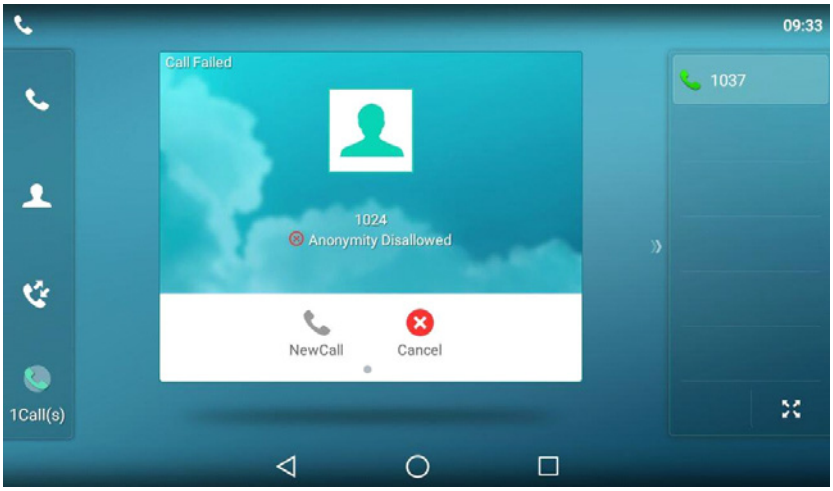
The phone will send the configured on code or off code depending on your selection when you enable or disable anonymous call rejection feature on the phone.



- Tap ✓ to save the change or ◁ to cancel.

If you have enabled anonymous call rejection and you receive an anonymous call:

If phone B has anonymous call feature enabled and places a call to you, the call will be rejected. Your touch screen phone will display the following:



Auto Answer

Auto answer allows you to automatically answer an incoming call.Auto answer will only work when there is no other call in progress or the phone is in idle state. Auto answer is configurable on a per-line basis.

Auto-Answer delay defines a period of delay time before the IP phone automatically answers the incoming call.

Auto answer tone allows the IP phone to play a tone when an incoming call is automatically answered. You can customize the auto answer tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to "Tones".



Auto answer is not applicable to automatically answer an IP address call. Automatically answering an IP address call works only if IP direct auto answer feature is enabled. For more information, refer to "IP Direct Auto Answer"

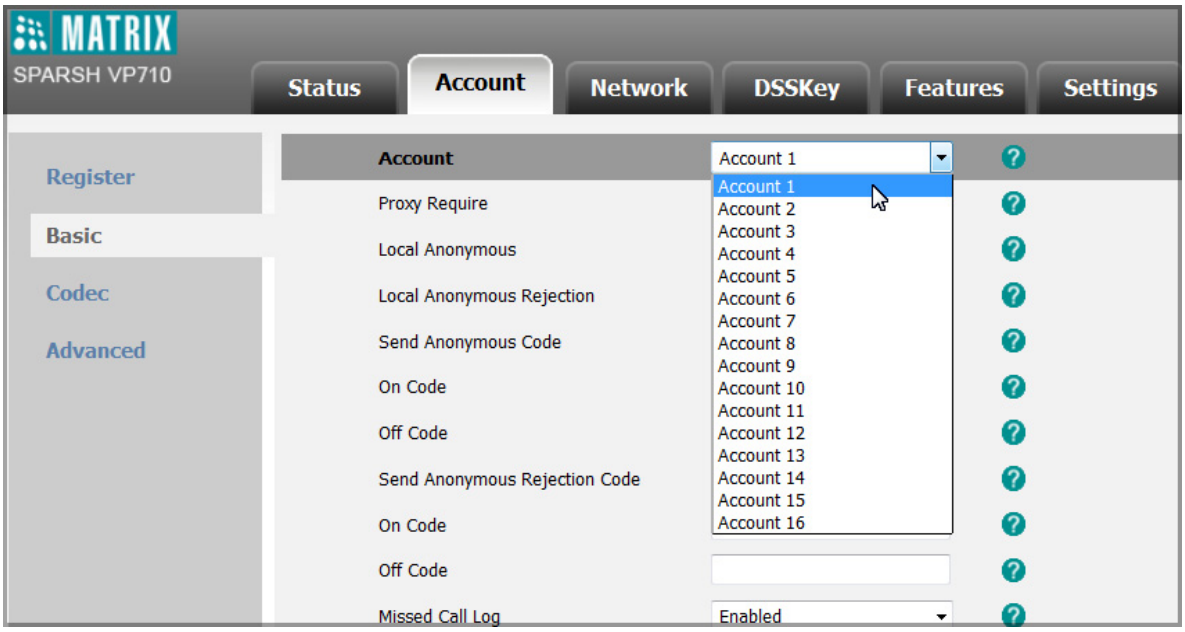
Procedure

Auto answer can be configured using the configuration files or locally.

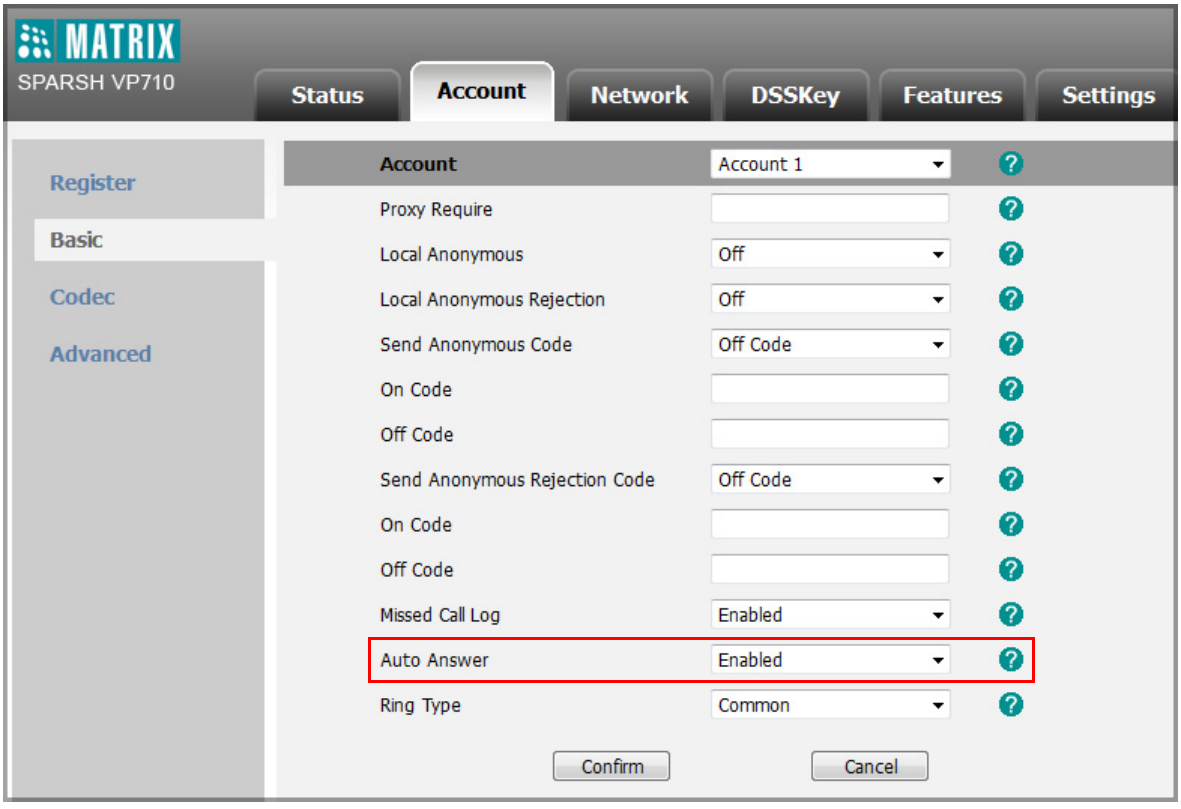
Configuration File	<MAC>.cfg	Configure auto answer. Specify a period of delay time for auto answer. Configure auto answer tone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure auto answer. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-basic&q=load&acc=0 Specify a period of delay time for auto answer. Configure auto answer tone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure auto answer.

To configure auto answer via web user interface:

- Click on **Account-> Basic**.
- Select the desired account from the **Account** list.



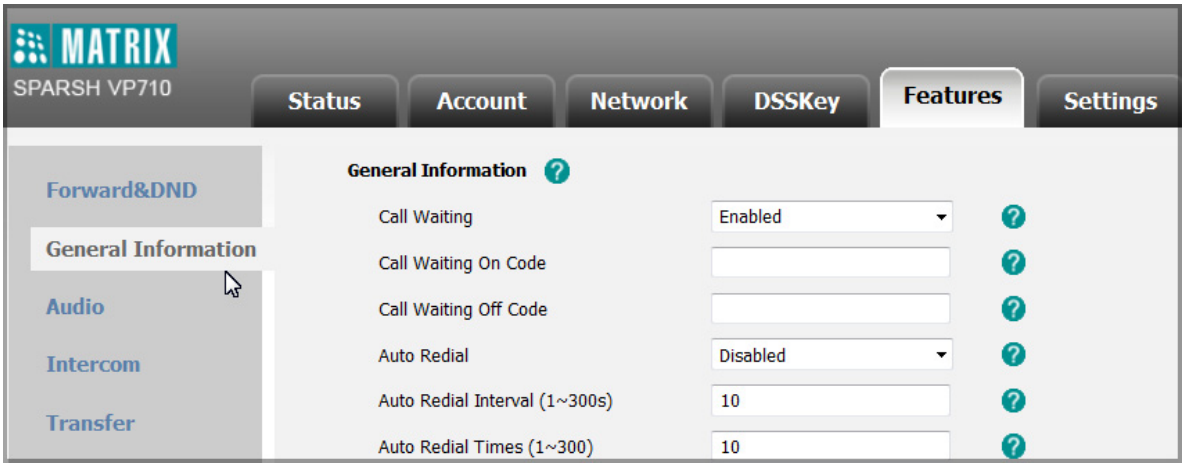
- Select the desired value from the **Auto Answer** list.



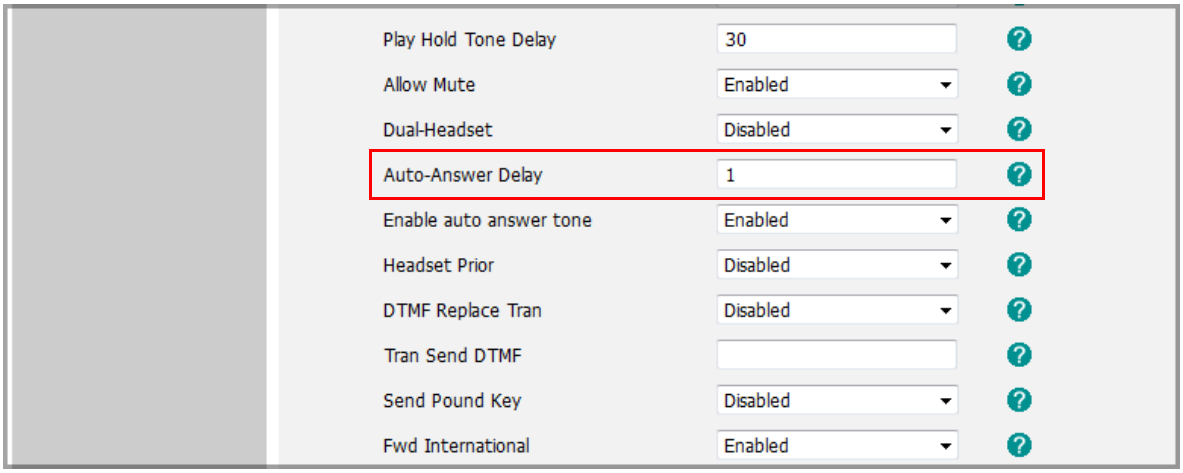
- Click **Confirm** to save the change.

To configure a period of delay time for auto answer via web user interface:

- Click on **Features->General Information**.



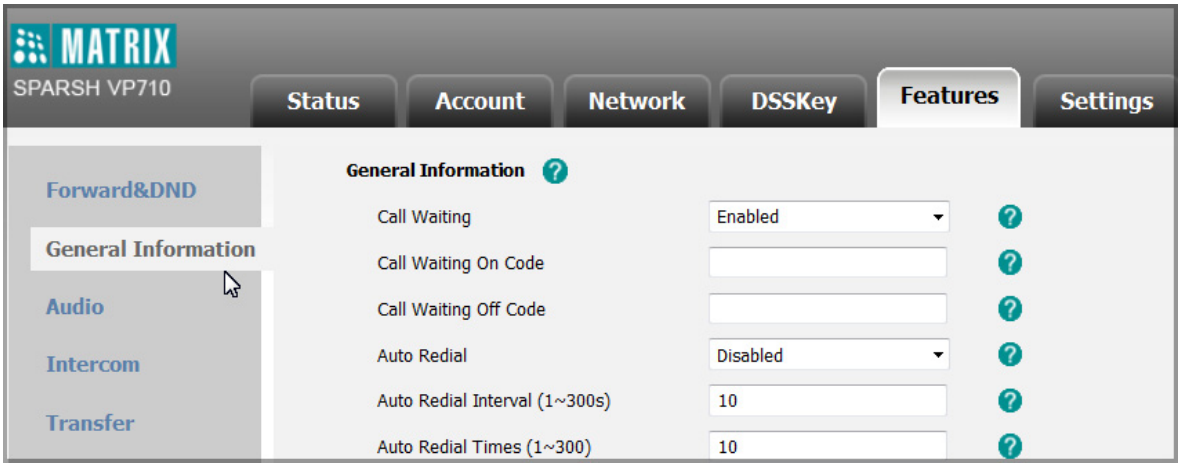
- Enter the desired time in the **Auto-Answer Delay(1~4s)** field.



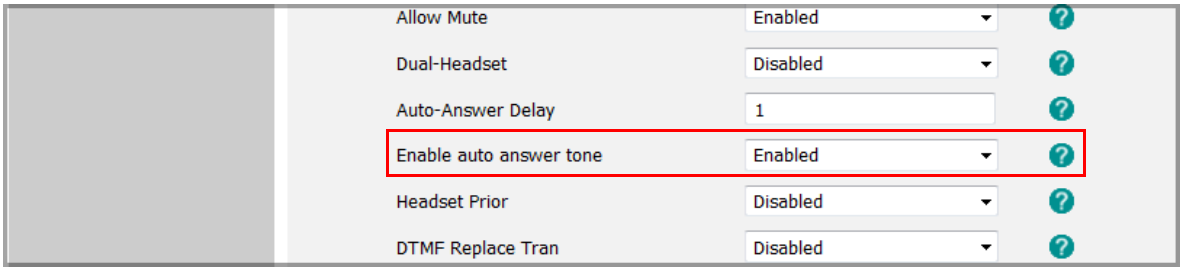
- Click **Confirm** to save the change.

To configure auto answer tone via web user interface:

- Click on **Features->General Information**.



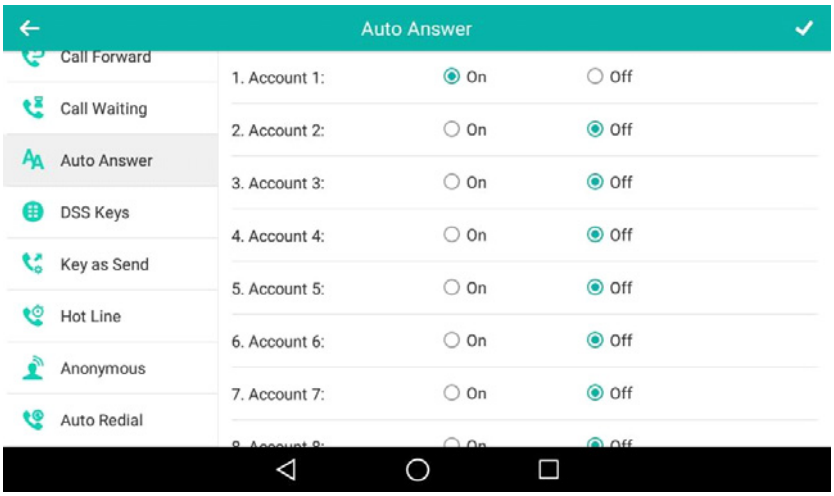
- Select the desired value in the **Enable auto answer tone** list.



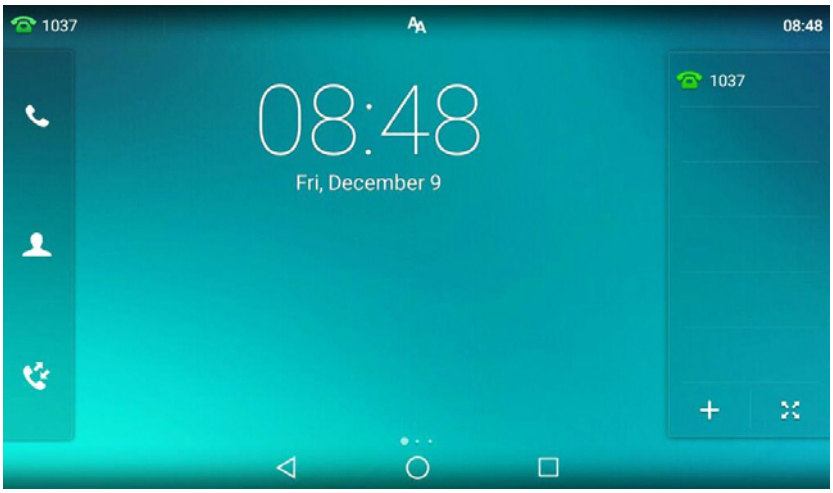
- Click **Confirm** to save the change.

To configure auto answer via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Auto Answer**.
- Tap the **On** radio box in the desired line.



- Tap ✓ to save the change or ◁ to cancel.
- The icon AA will display on the status bar.



You can also swipe down from the top of the screen.Tap AA **Auto Answer** to enter Auto Answer setting screen and enable auto answer feature for the desired line.

Auto Redial

Auto redial feature allows the IP phone to redial the phone number automatically when the called party is busy. Both the number of attempts and waiting time between redials are configurable.

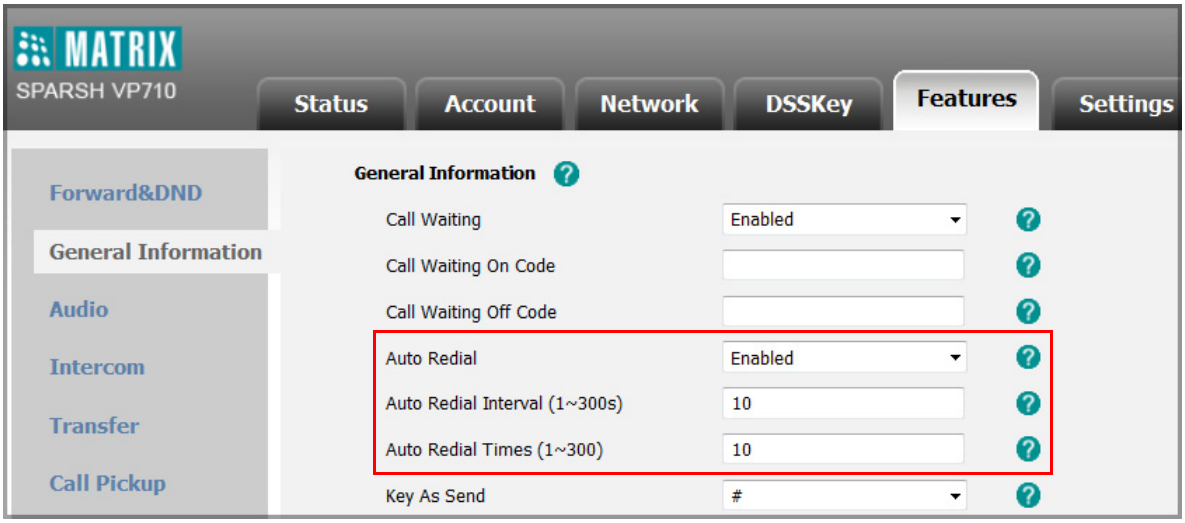
Procedure

Auto redial can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure auto redial feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure auto redial feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure auto redial feature.

To configure auto redial via web user interface:

- Click on **Features->General Information**.
- Select the desired value from the **Auto Redial** list.
- Enter the waiting time in the **Auto Redial Interval (1~300s)** field.
  The default waiting time is 10s.
- Enter the desired times in the **Auto Redial Times (1~300)** field.
  The default value is 10.



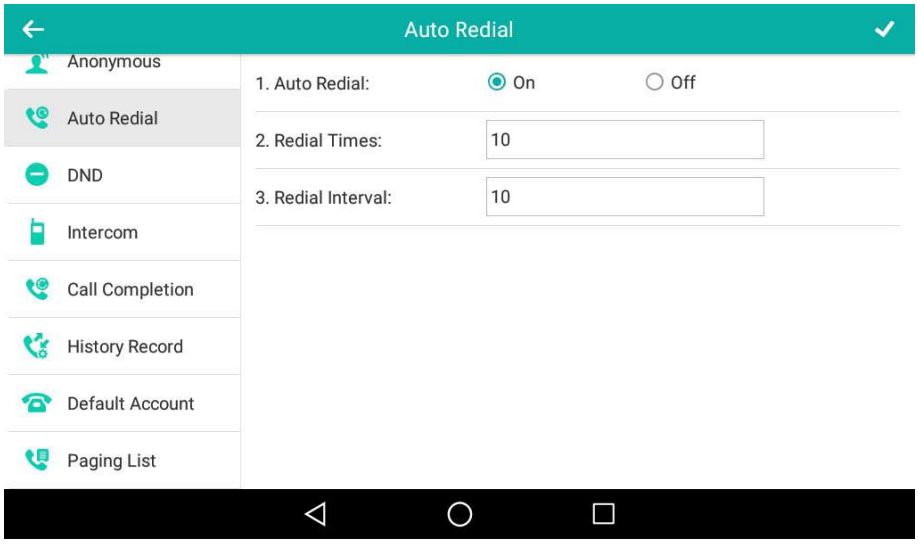
- Click **Confirm** to save the change.

To configure auto redial via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Auto Redial**.
- Tap the **On** radio box in the Auto Redial field.
- Enter the desired time (in seconds) in the **Redial Interval field**.

  The default value is “10”.
- Enter the desired number of redial attempts in the **Redial Times field**.

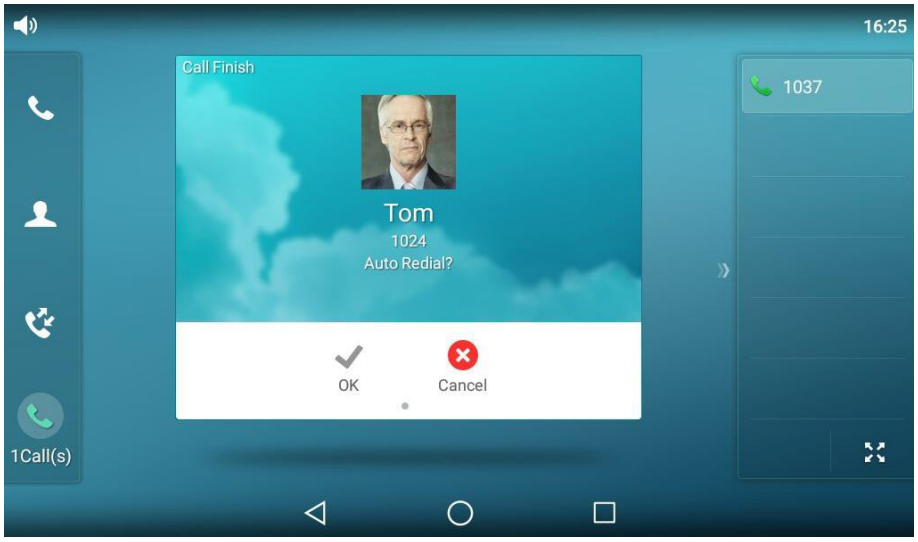
  The default value is “10”.



- Tap ✓ to accept the change or ◁ to cancel.

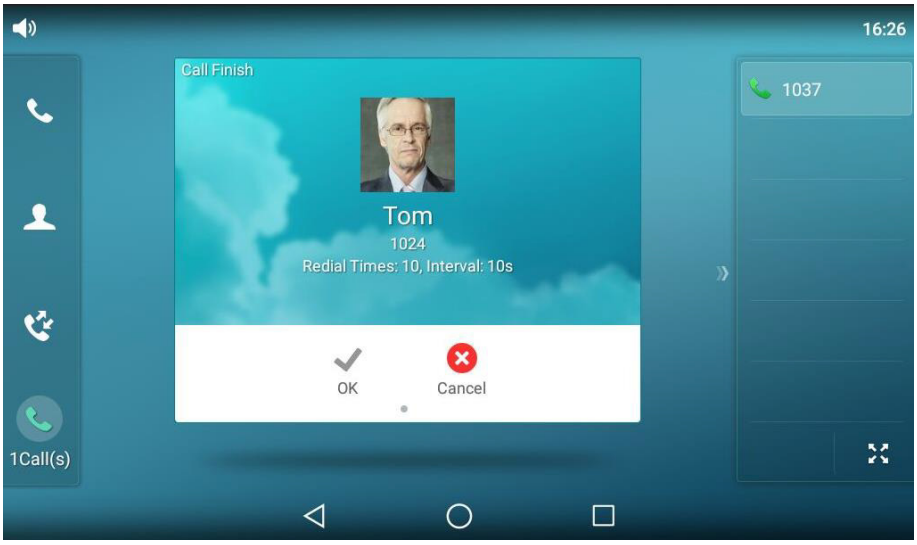
To use auto redial:

When the called party is busy, the following prompt will appear on the touch screen of the phone:



- Tap **OK** to activate auto redial.

  The following prompt will appear on the touch screen of the phone:



- Wait for a period of time or tap **OK** to redial the phone number.

  The phone will retry as many times as configured until the called party is idle.

Call Completion

Call completion allows you to monitor the busy party and establish a call when the busy party becomes available to receive a call. Two factors commonly prevent a call from connecting successfully:

- Callee does not answer
- Callee actively rejects the incoming call before answering

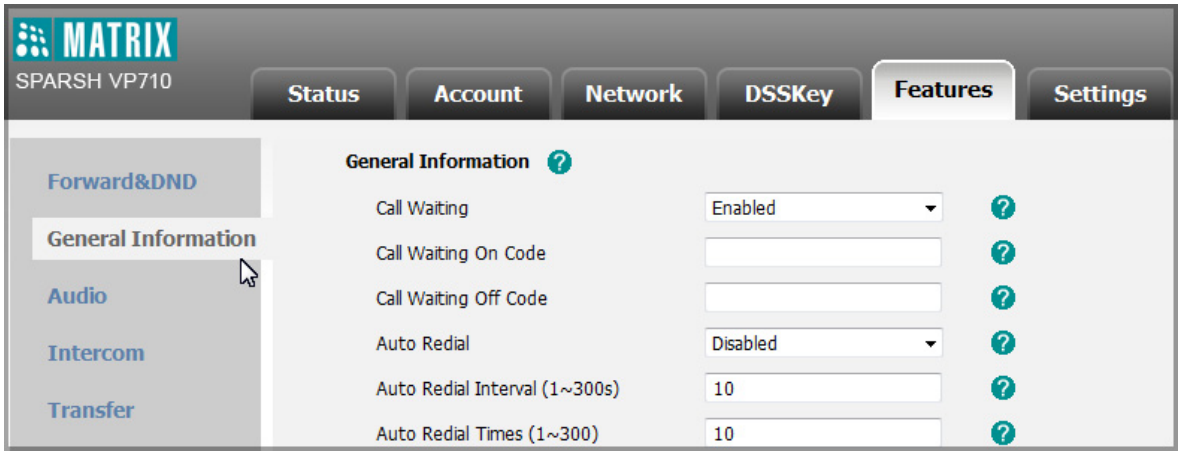
Procedure

Call completion can be configured using the configuration files or locally.

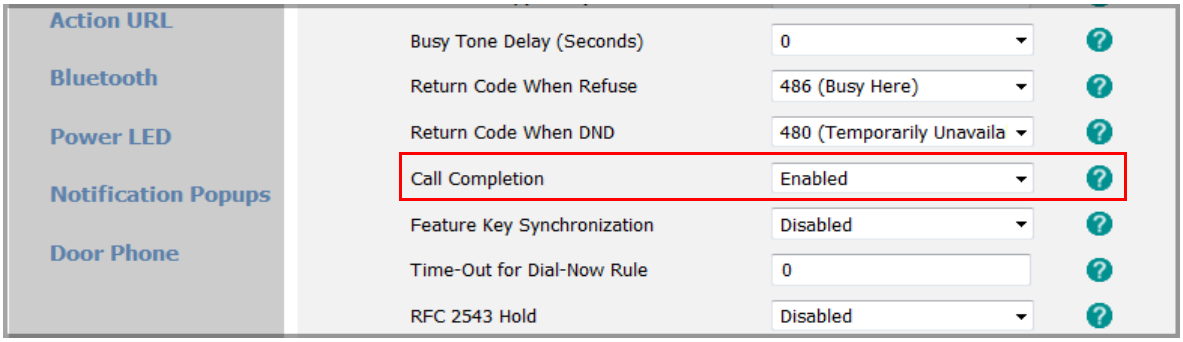
Configuration File	<MAC>.cfg	Configure call completion. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure call completion. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure call completion.

To configure call completion via web user interface:

- Click on **Features->General Information**.



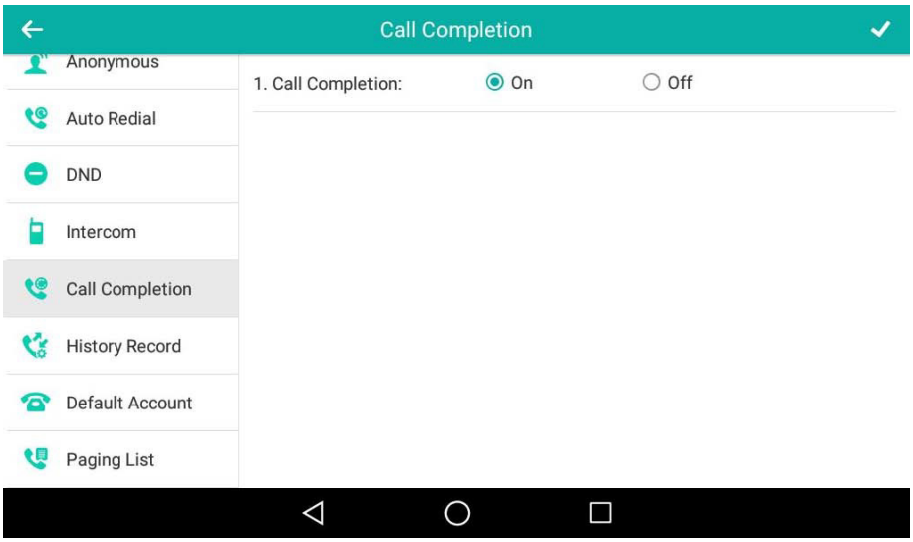
- Select the desired value from the **Call Completion** list.



- Click **Confirm** to save the change.

To configure call completion via phone user interface:

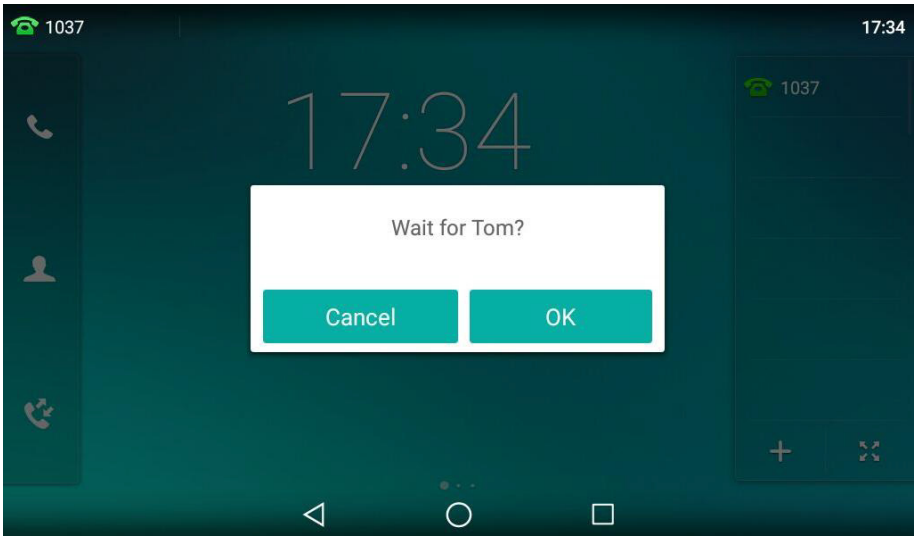
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Call Completion**.
- Tap the **On** radio box in the **Call Completion** field.



- Tap ✓ to accept the change or ◁ to cancel.

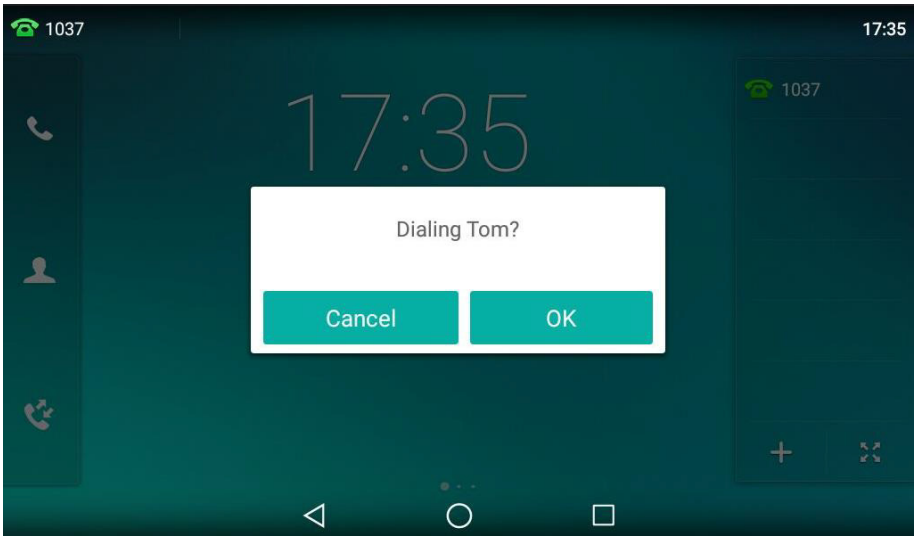
To use call completion:

When the called party is busy, the following prompt will appear on the touch screen of the phone:



- Tap **OK**, the phone returns to the idle screen and call completion is activated.

When the called party becomes idle, the following prompt appears on the touch screen of the phone:



- Tap **OK** to redial the number.

- *Call completion is not available on all servers. For more information, contact your system administrator.*
- *IP phone supports call completion using the SUBSCRIBE/NOTIFY method, which is specified in draft-poetzl-sipping-call-completion-00, to subscribe to the busy party and receive notifications of their status changes.*

Call Forward

Call forward allows you to redirect an incoming call to a third party.

There are two types of forwarding:

- Static Forwarding - This is used to forward incoming calls to another party.
- Dynamic Forwarding - This is used to forward incoming calls to another party while the phone is ringing.

Static Forwarding

Three types of static forwarding:

- **Always Forward:** Incoming calls are immediately forwarded.
- **Busy Forward:** Incoming calls are immediately forwarded if the phone is busy.
- **No Answer Forward:** Incoming calls are forwarded if not answered after a certain period of time.

Call forward can be configured on a phone or a per-line basis depending on the call forward mode. Two types of call forward modes are:

- **Phone:** Call forward is effective for the specific phone. By default, it is enabled.
- **Custom:** Call forward can be configured for each or all accounts.

IP phone redirects an incoming INVITE message by responding with a 302 Moved Temporarily message, which contains a Contact header with a new URI that should be tried.

The server-side call forward settings disable the local call forward settings. If the server-side call forward feature is enabled on any of the phone's registrations, the other registrations are not affected. DND activated on the IP phone disables the local no answer forward settings.

The call forward on code and call forward off code configured on IP phones are used to activate/deactivate the server-side call forward feature. They may vary on different servers.

Diversion/History-Info

IP phone supports the redirected call information sent by the SIP server with Diversion header, per draft-levy-sip-diversion-08, or History-info header, per RFC 4244. The Diversion/History-info header is used to inform the IP phone of a call's history.
For example, when call forward is enabled on the phone, the Diversion/History-info header allows the receiving phone to indicate the caller's name/number and number of the forwarding party.

Forward International

Forward international allows users to forward an incoming call to an international telephone number. By default, this feature is enabled.

Forward Emergency

Forward emergency allows the incoming calls from some authorized numbers not to be forwarded when the call forward feature is enabled. The incoming call will not be logged in the Forwarded Calls list. By default, this feature is disabled.

Procedure

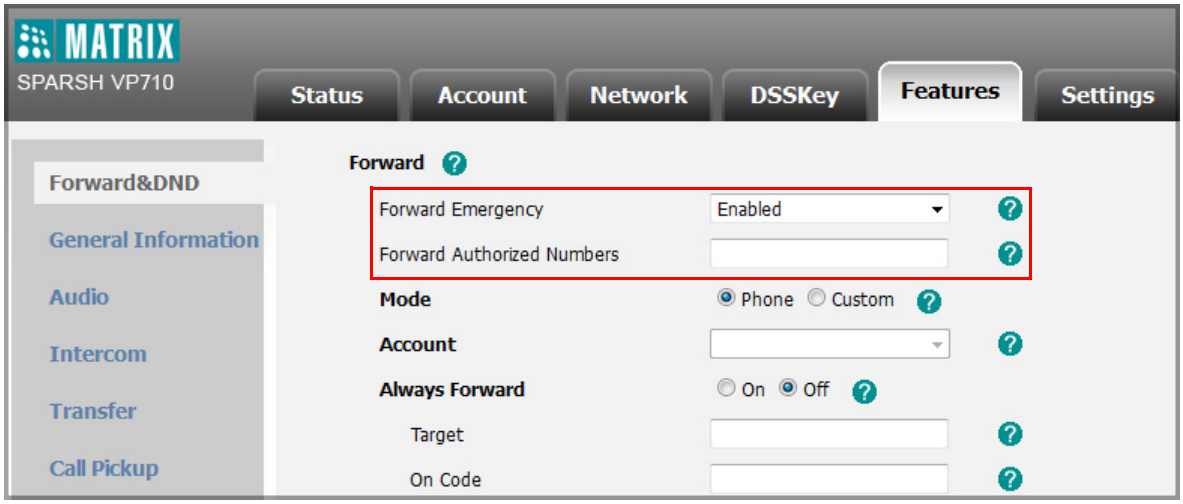
Call forward can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure call forward in custom mode. Specify the authorized numbers when call forward is enabled. Configure the call forward mode. Configure call forward in phone mode. Configure diversion/history-info feature. Configure forward international. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify the authorized numbers when call forward is enabled. Configure call forward. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-forward&q=load Configure diversion/history-info feature. Configure forward international. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load
	Phone User Interface	Configure call forward. Configure forward international.

To specify the authorized numbers when call forward is enabled via web user interface:

When call forward is enabled on the phone, the phone cannot forward incoming calls from the numbers specified in the **Forward Authorized Numbers** field.The incoming call will not be logged in the Forwarded Calls list. By default, this feature is disabled.

- Click on **Features->Forward&DND**.
- Select the desired value in the **Forward Emergency** list.
- Enter the desired value in the **Forward Authorized Numbers** field.
- For multiple numbers, enter a comma between every two numbers.

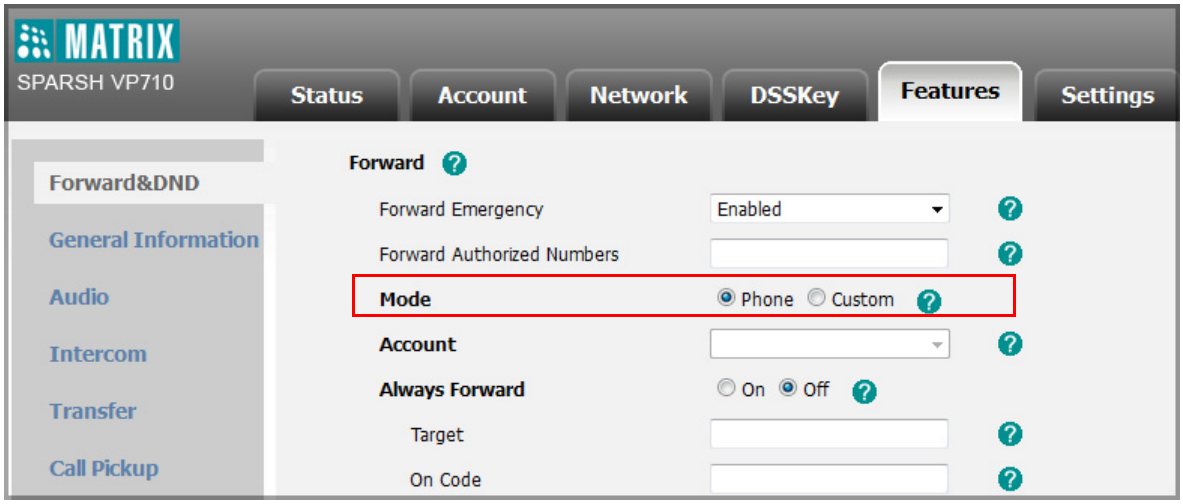


- Click **Confirm** to save the change.

Forward authorized number is configurable via web user interface only.

To configure the call forward mode via web user interface:

- Click on **Features->Forward & DND.**
- In the **Forward** block, select the desired radio box in the **Mode** field.



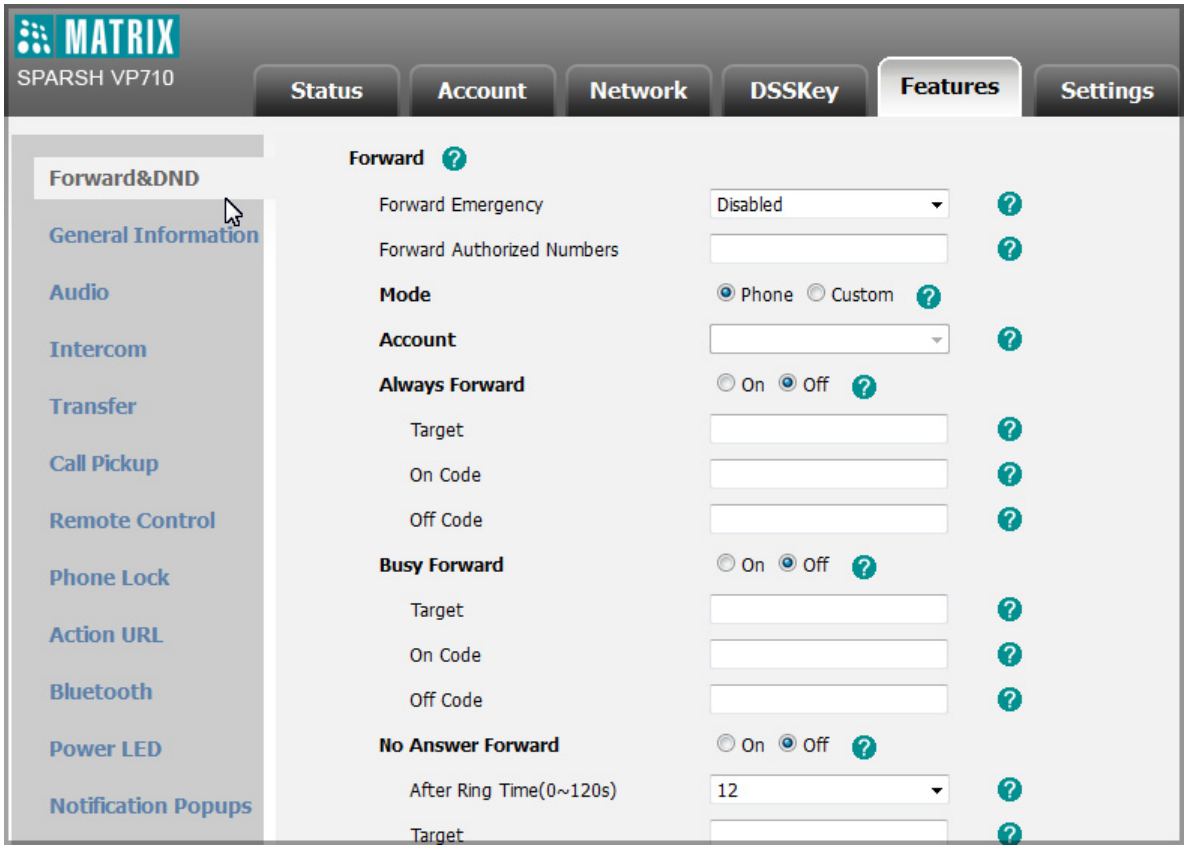
- Click **Confirm** to save the change.

Call forward mode is configurable via web user interface only.

To configure call forward via web user interface:

- Click on **Features->Forward&DND**.
- In the **Forward** block, select the desired radio box in the **Mode** field.
  - If you select the **Phone** radio box:

    - Select the desired radio box in the **Always Forward/Busy Forward/No Answer Forward** field.
    - In the **Target** field, enter the destination number you want to forward the call.
    - In the **On Code** and **Off Code** fields (optional), enter the on code and off code respectively.

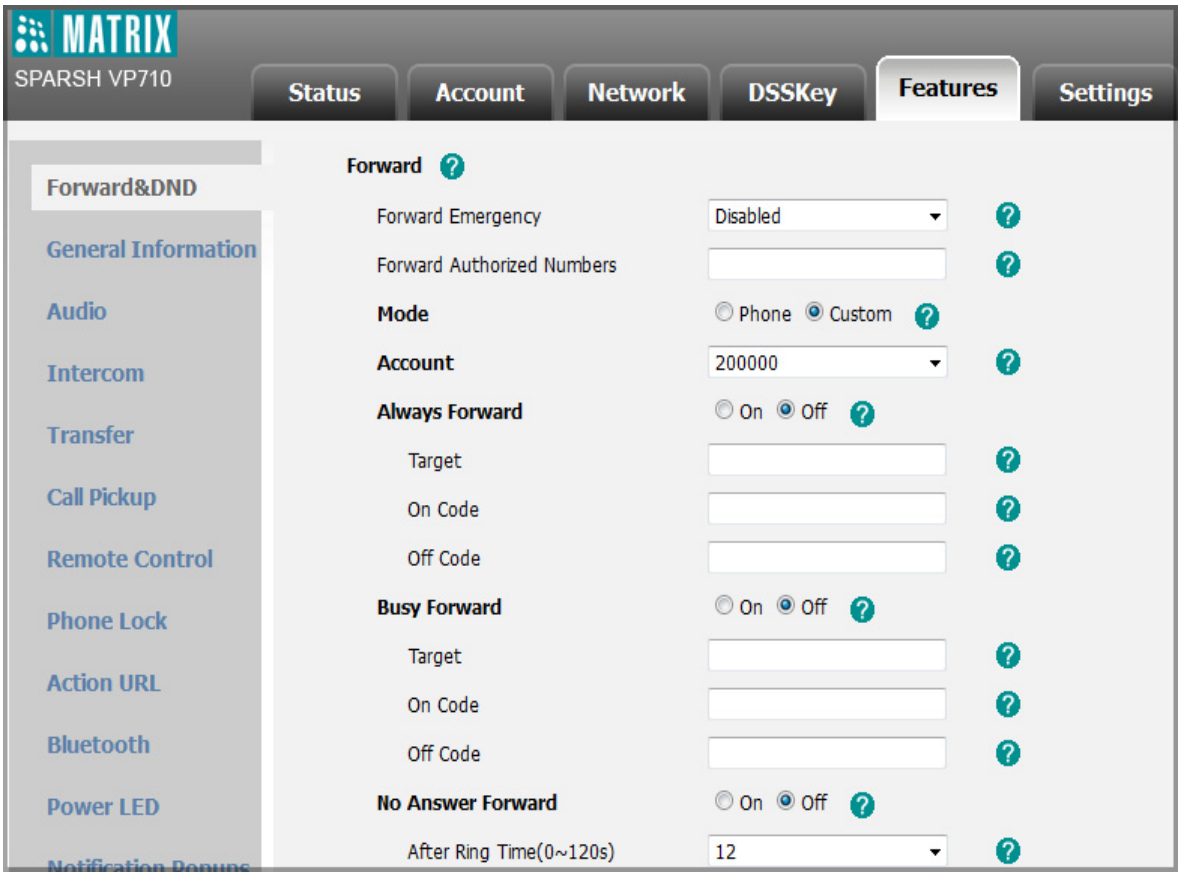


    - In **No Answer Forward** field, select the ring time to wait before forwarding the call from **After RingTime(0~120s)** list.

  - If you select the **Custom** radio box:

    - Select the desired account from the **Account** list.
    - Select the desired radio box in the **Always Forward/Busy Forward/No Answer Forward** field.
    - In the **Target** field, enter the destination number you want to forward the call.
    - In the **On Code** and **Off Code** fields, enter the on code and off code respectively.

- In **No Answer Forward** field, select the ring time to wait before forwarding the call from **After RingTime(0~120s)** list.

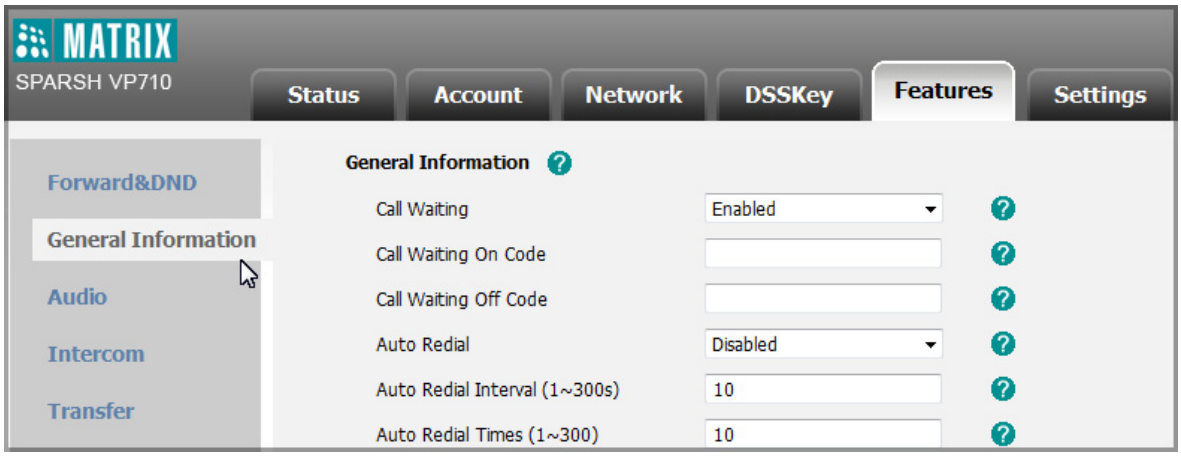


- Click **Confirm** to save the change.

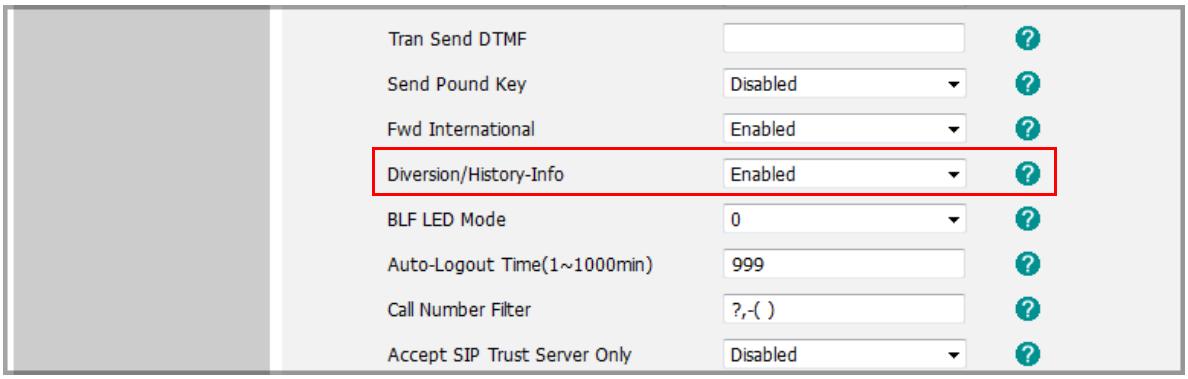
.

To configure Diversion/History-Info feature via web user interface:

- Click on **Features**->**General Information**.



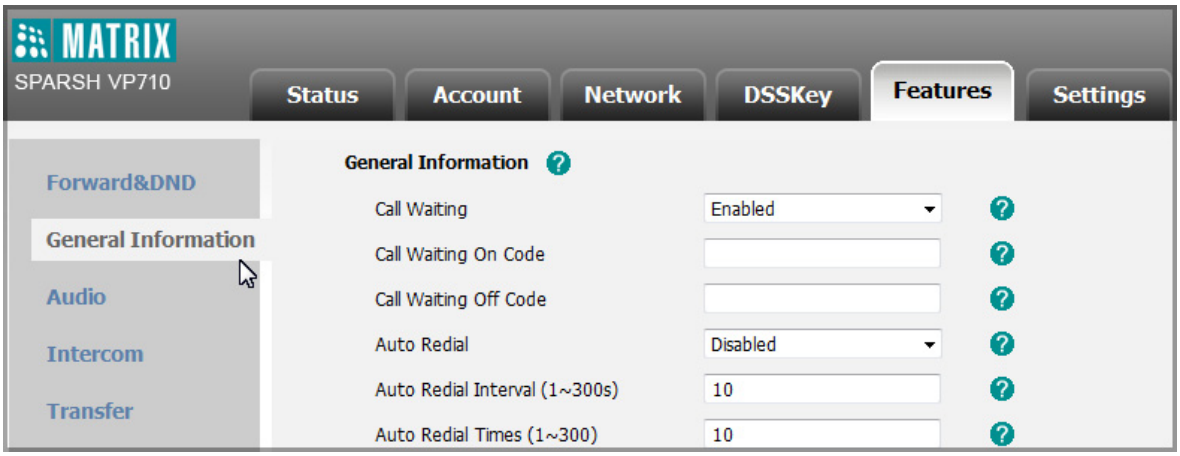
- Select the desired value from the **Diversion/History-Info** list.



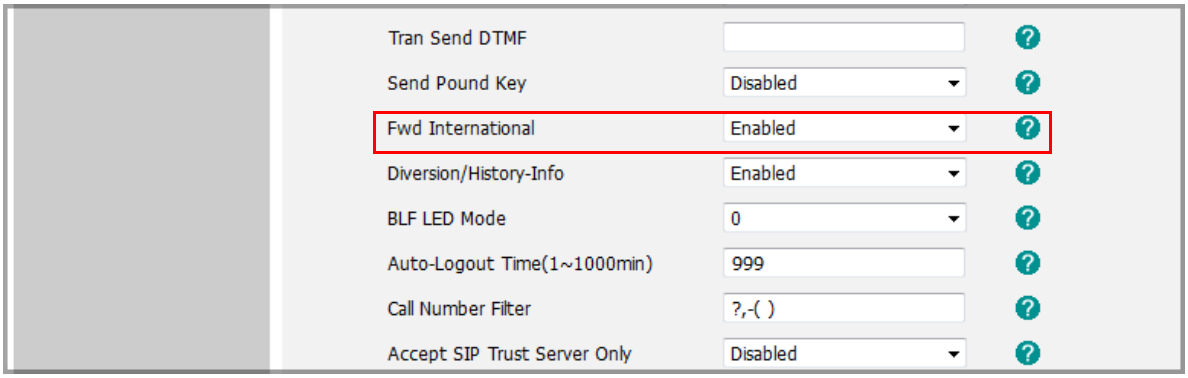
- Click **Confirm** to save the change.

To configure forward international via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Fwd International** list.



- Click **Confirm** to save the change.

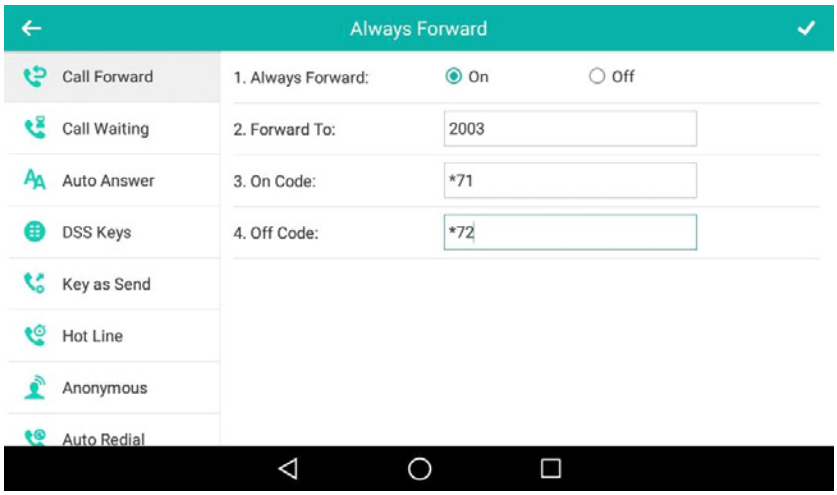
By default, the Fwd International prefix value is "00" and it is not configurable.

To configure call forward via phone user interface:

To enable call forward in phone mode:

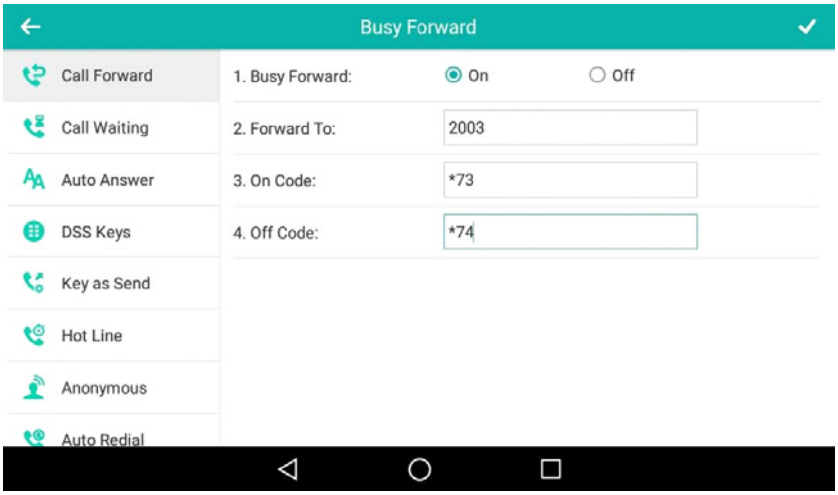
- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings**->**Features**->**Call Forward**.
- Tap the desired forwarding type.
- Depending on your selection:
  - If you tap **Always Forward:**
    - Tap the **On** radio box in the **Always Forward** field.
    - In the **Forward To** field**,** enter the destination number you want to forward all incoming calls to.

- In the **On Code** or **Off Code** field (optional), enter the always forward on code or off code respectively.



- If you tap **Busy Forward**:

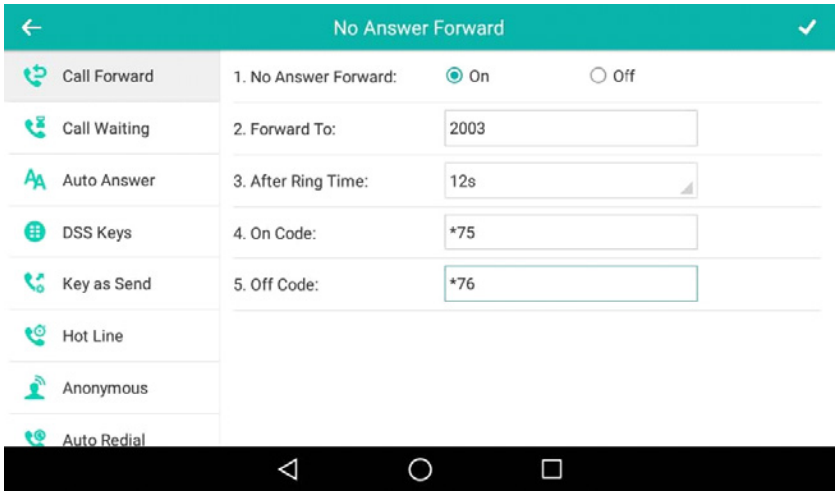
  - Tap the **On** radio box in the **Busy Forward** field.
  - In the **Forward To** field**,** enter the destination number you want to forward incoming calls to when the phone is busy.
  - In the **On Code** or **Off Code** field (optional), enter the busy forward on code or off code respectively



- If you tap **No Answer Forward:**

  - Tap the **On** radio box in the **No Answer Forward** field.
  - In the **Forward To** field**,** enter the destination number you want to forward unanswered incoming calls to.
  - Tap the desired ring time for the call to wait before forwarding the call in the **After Ring Time** list.The default ring time is 12 seconds.

- In the **On Code** or **Off Code** field (optional), enter the no answer forward on code or off code respectively.



- Tap ✓ to accept the change or ◁ to cancel.

The icon ↩ on the status bar indicates that the call forward is enabled.

You can also swipe down from the top of the screen to enter the control center, tap **Forward** to enter the desired Call Forward setting screen according to the current forward type.

If you have not configured the call forward type, tapping **Forward** will turn to the Always Forward setting screen.

To enable call forward in custom mode:

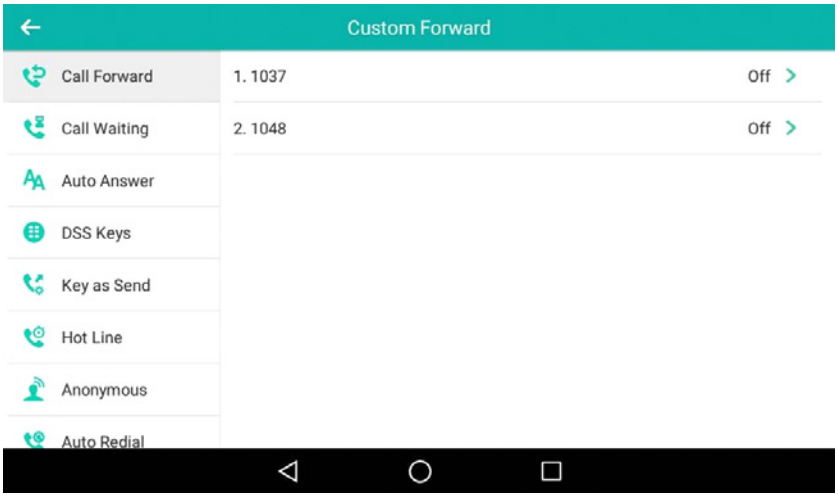
- Do one of the following:
  - Swipe down from the top of the screen or swipe left/right to go to the second idle screen
  - Tap **Settings->Features->Call Forward**.

  OR

  - Press ↪ when the phone is idle.

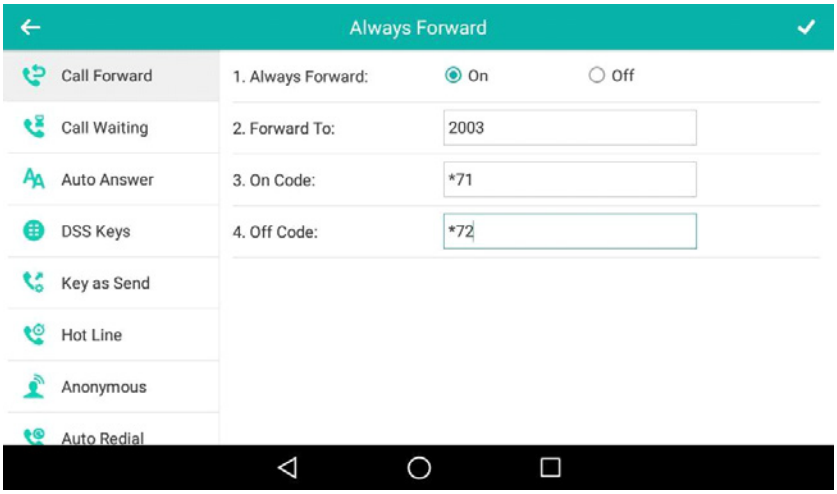
  OR

  - Swipe down from the top of the screen to enter the control center and then tap **Forward**.
- Tap the desired account.



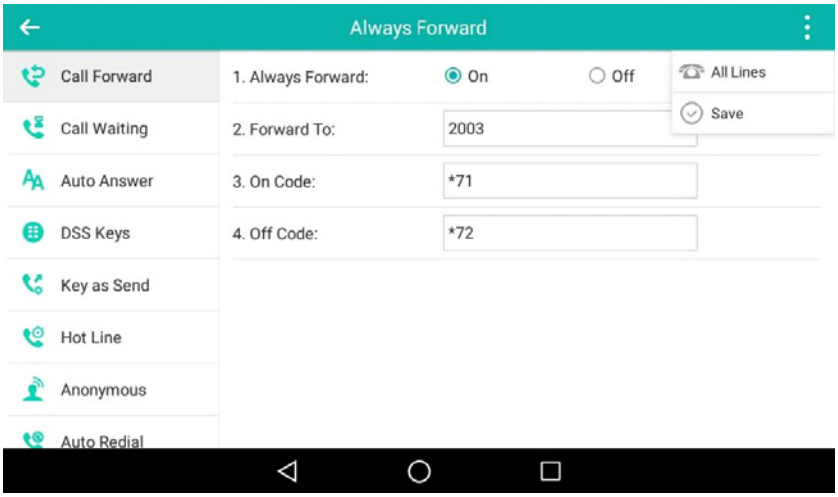
- Tap the desired forwarding type.
- Depending on your selection:
  - If you tap **Always Forward**:

    - Tap the **On** radio box in the **Always Forward** field.
    - In the **Forward To** field, enter the destination number you want to forward all incoming calls to.
    - In the **On Code** or **Off Code** field (optional), enter the always forward on code or off code respectively.

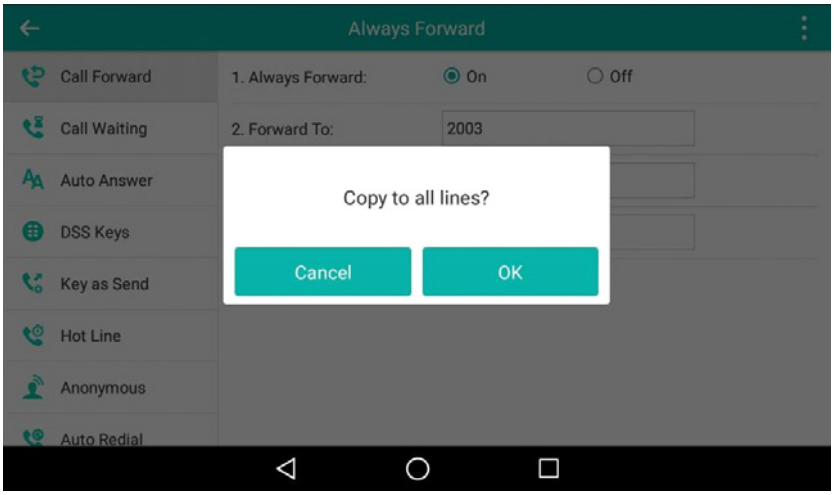


    - You can also enable always forward for all accounts. Do the following:

    - Tap and then tap **All Lines**.



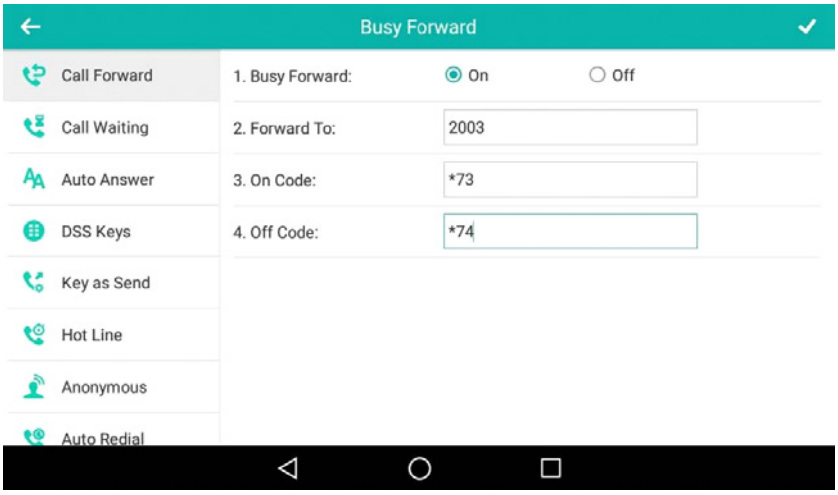
- The touch screen prompts **“Copy to all lines?”**.



- Tap **OK** to accept the change or **Cancel** to cancel.

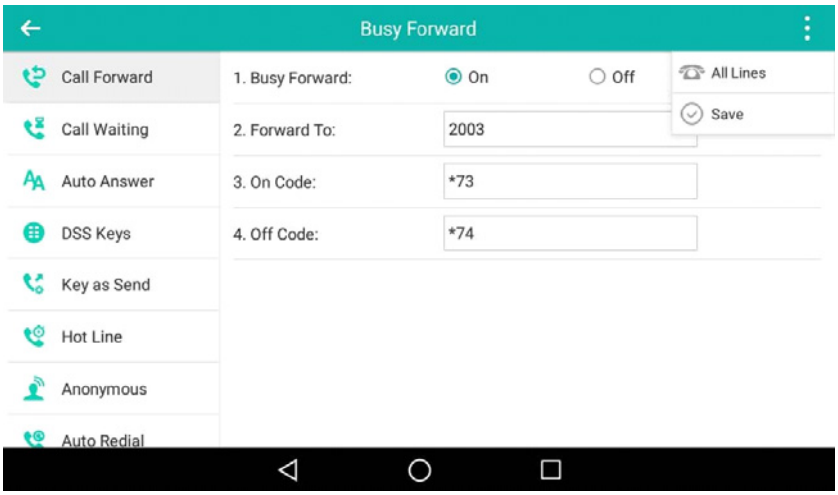
- If you select **Busy Forward**:

  - Tap the **On** radio box in the B**usy Forward field**.
  - In the **Forward To** field**,** enter the destination number you want to forward incoming calls to when the phone is busy
  - In the **On Code** or **Off Code** field (optional), enter the busy forward on code or off code respectively.

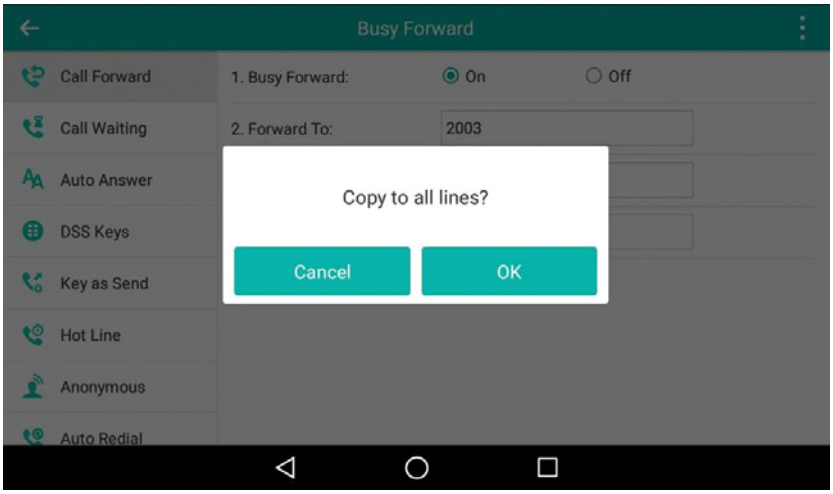


- You can also enable busy forward for all accounts. Do the following:

- Tap and then tap **All Lines.**



- The touch screen prompts **"Copy to all lines?"**.

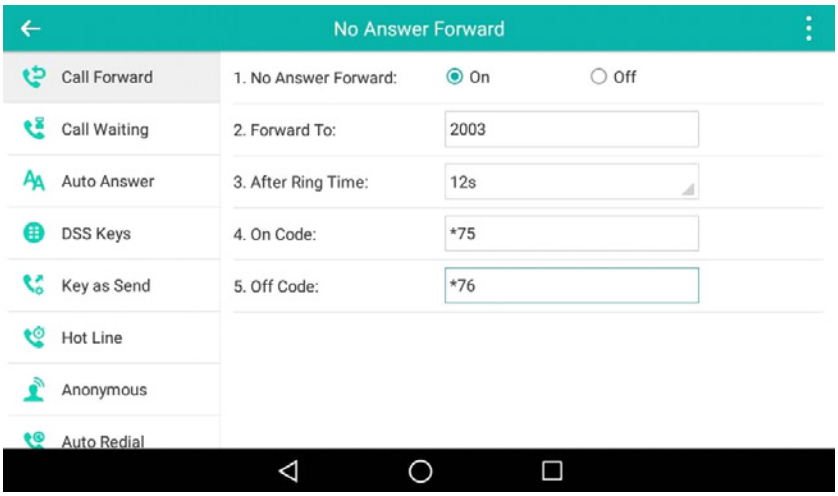


- Tap **OK** to accept the change or **Cancel** to cancel.

- If you select **No Answer Forward**:

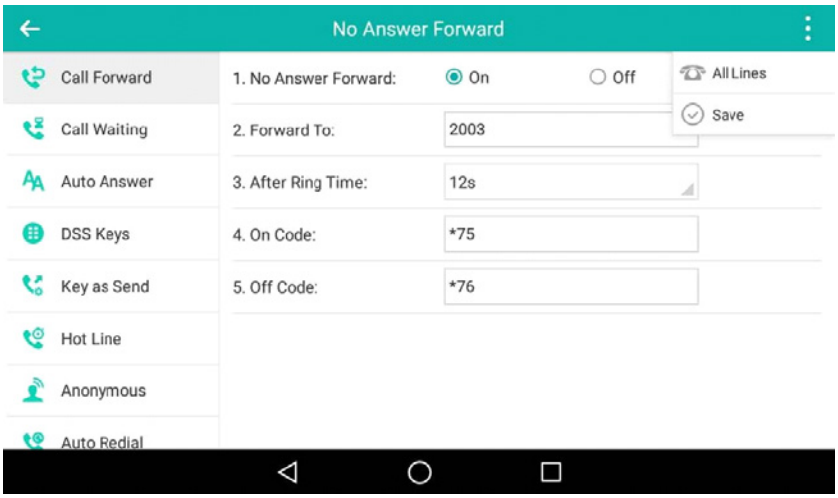
  - Tap the **On** radio box in the **No Answer Forward** field.
  - In the **Forward To** field, enter the destination number you want to forward unanswered incoming calls to.
  - Tap the desired ring time for the call to wait before forwarding the call in the **After Ring Time** list.The default ring time is 12 seconds.

- In the **On Code** or **Off Code** field (optional), enter the no answer forward on code or off code respectively

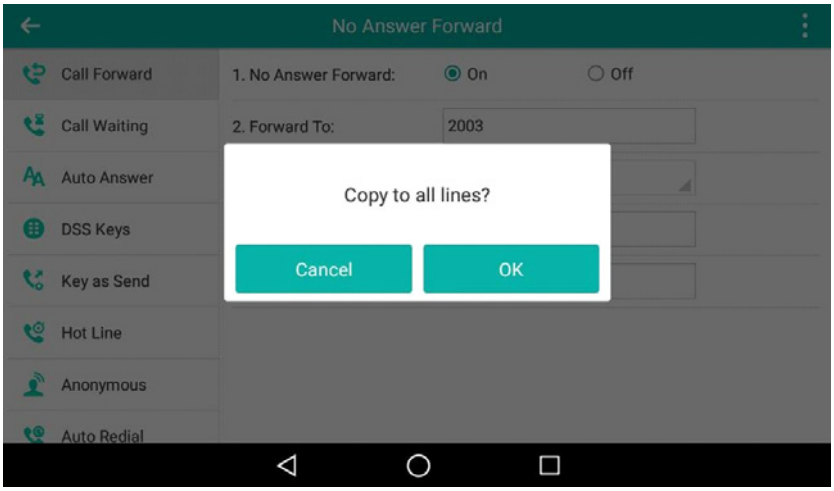


- You can also enable no answer forward for all accounts. Do the following:

- Tap and then tap **All Lines.**



- The touch screen prompts **“Copy to all lines?”**



- Tap **OK** to accept the change or **Cancel** to cancel.

- Tap and then tap **Save** to save the change.

If you enable call forward for the default account, the associated line icon will change to and the icon will appear on the status bar.



If you enable call forward for the non-default account, only the associated line icon will change to , the icon will not appear on the status bar.



If you enable call forward for all accounts, all line icons will change to , and the icon will appear on the status bar.

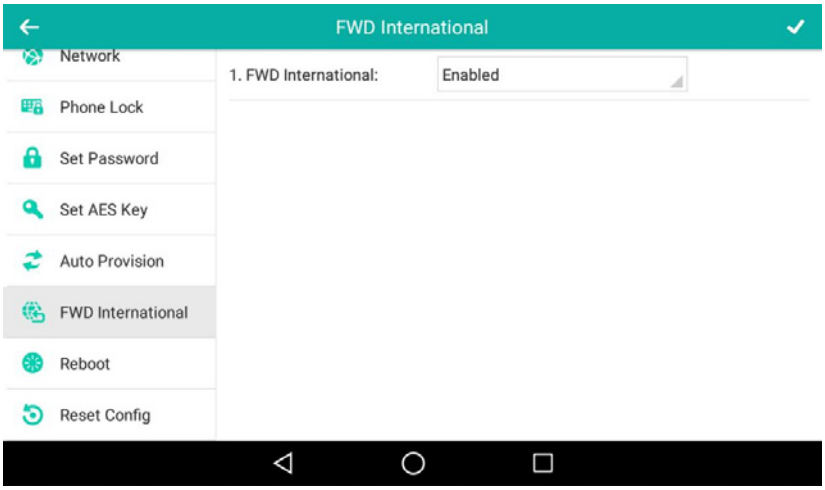


If the forward on code or off code is configured, the IP phone sends the corresponding code to the server when a certain action takes place.
For example, if you set the Always Forward on code to be *71 (may vary on different servers), and the target number to be 1234. When you enable Always Forward on the IP phone, the IP phone sends *711234 to the server, and then the server will enable Always Forward feature on the server side, hence being able to get the right status of the extension.

*You can also enter the SIP URI or IP address in the **Forward To** field. For more information on using the SIP URI or IP address, refer to "Placing Calls".*

To configure forward international via phone user interface:

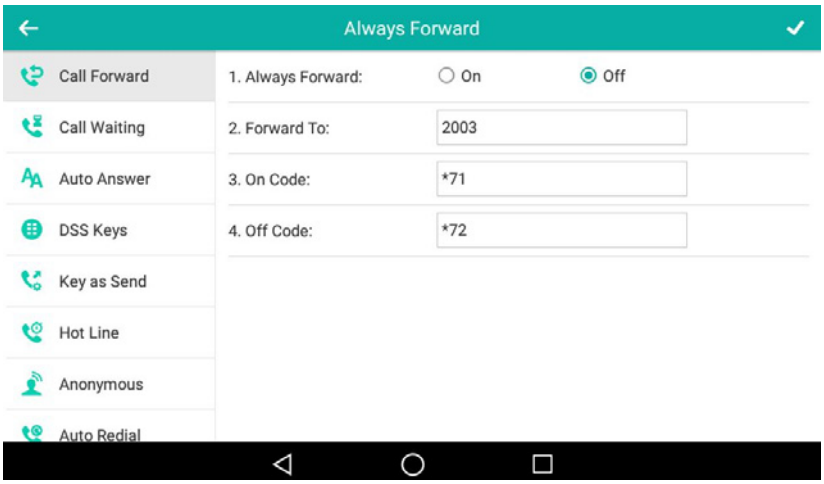
- Tap **Settings->Advanced** (default password: admin) **->FWD International**.
- Tap the **FWD International field**.



- Tap the desired value in the pop-up dialog box.
- Tap to accept the change or to cancel.

To disable call forward in phone mode via phone user interface:

- Do one of the following:
  - Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
  - Tap **Settings->Features->Call Forward.**
  - Tap the desired forwarding type.
  - Tap the **Off** radio box.



- Tap to accept the change or to cancel.

OR

- Press when the phone is idle.

OR

- Swipe down from the top of the screen to enter the control center, and then tap **FWD**

To disable call forward in custom mode for a specific account via phone user interface:

- Do one of the following:
  - Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
  - Tap **Settings->Features->Call Forward**.

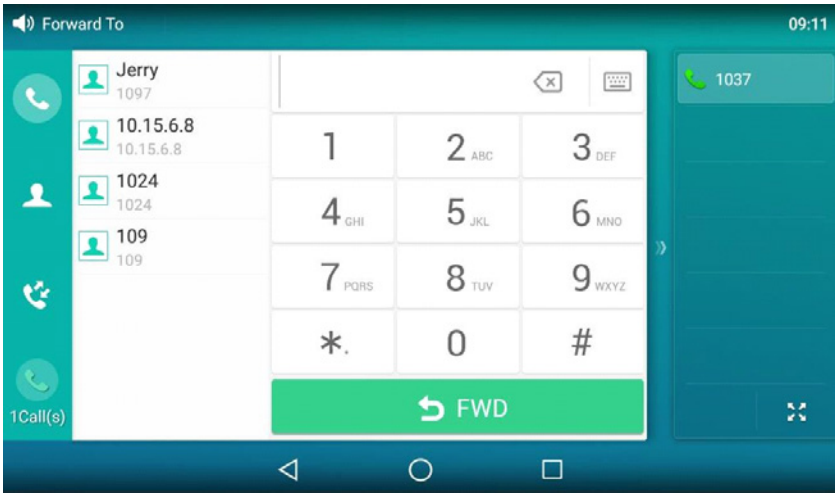
  OR
  - Press  when the phone is idle.

  OR
  - Swipe down from the top of the screen to enter the control center and then tap **Forward**. The touch screen displays a list of accounts registered on the phone.
- Tap the desired account.
- Tap the desired forwarding type.
- Tap the **Off** radio box.
- Tap and then tap **Save** to save the change.

Dynamic Forwarding

To forward an incoming call to another party:

- When the phone is ringing, press the **FWD** soft key.
- Enter the number you want to forward the incoming call to.



- .Press  or tap **FWD**.
- The touch screen prompts a call forward message.

- *Call forward is local to the phone, and may be overridden by the server settings. Call forward on code or off code may be different among servers.For more information, contact your system administrator.*
- *If DND feature is enabled, the incoming calls will not be forwarded even if you configure no answer forward feature.*
- *When the phone forwards a call, a prompt window will pop up by default. If you want to disable the feature, contact your system administrator for more information.*

Call Hold/Resume

Call hold allows you to place an active call on hold. The purpose of call hold is to pause the activity on the existing call so that you can use the phone for other task (e.g. to place or receive another call).
Only one active call can be in progress at a time. Other calls can be made or received while placing the original call on hold. When you place a call on hold, your IP PBX may play Music to the other party while waiting.
Call hold tone allows the IP phone to play a warning tone at regular intervals when a call is on hold. The warning tone is played through the speakerphone.

When a call is placed on hold, the IP phone send an INVITE request with HOLD SDP to request remote parties to stop sending media and to inform them that they are being held.

IP phone supports two call hold methods:

- RFC 3264, which sets the “a” (media attribute) in the SDP to sendonly, recvonly or inactive (e.g., a=sendonly).By default, this method is selected.
- RFC 2543, which sets the “c” (connection addresses for the media streams) in the SDP to zero (e.g., c=0.0.0.0).

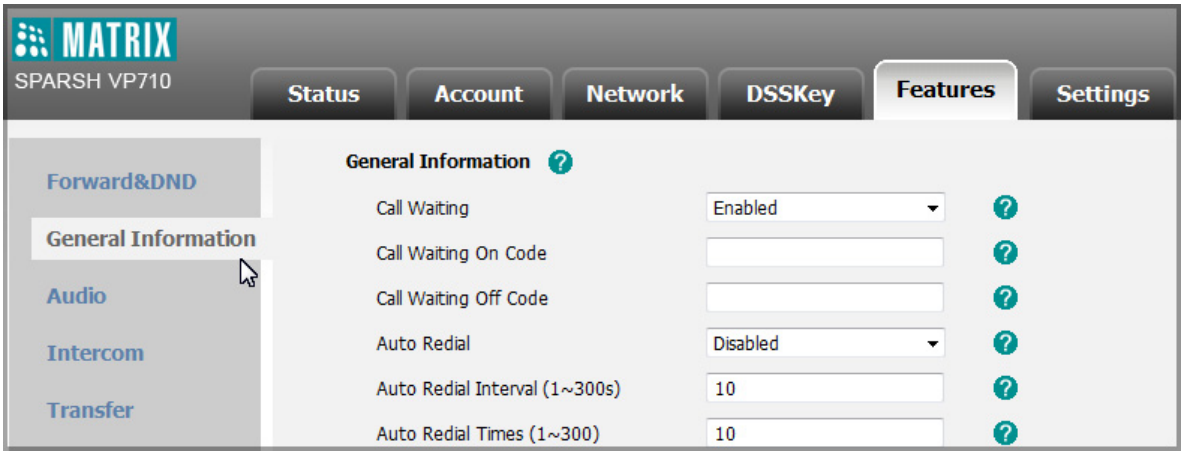
Procedure

Call hold can be configured using the configuration files or locally.

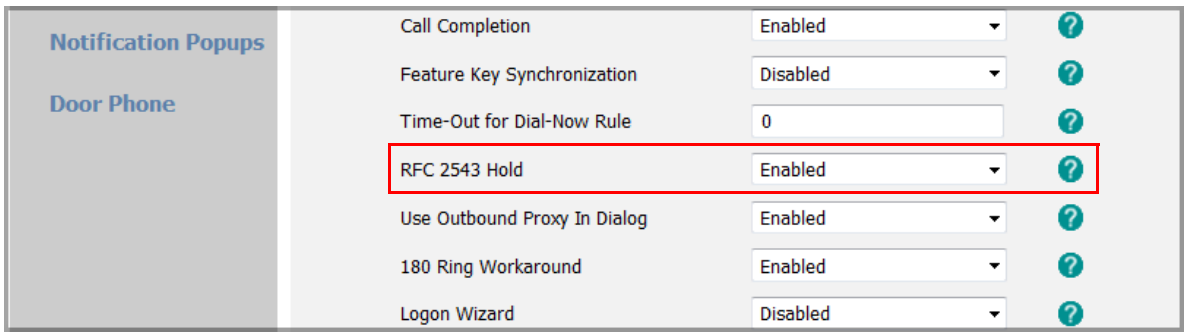
Configuration File	<MAC>.cfg	Configure the call hold tone and call hold tone delay. Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Configure the call hold tone and call hold tone delay. Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure call hold method via web user interface:

- Click on **Features->General Information**.



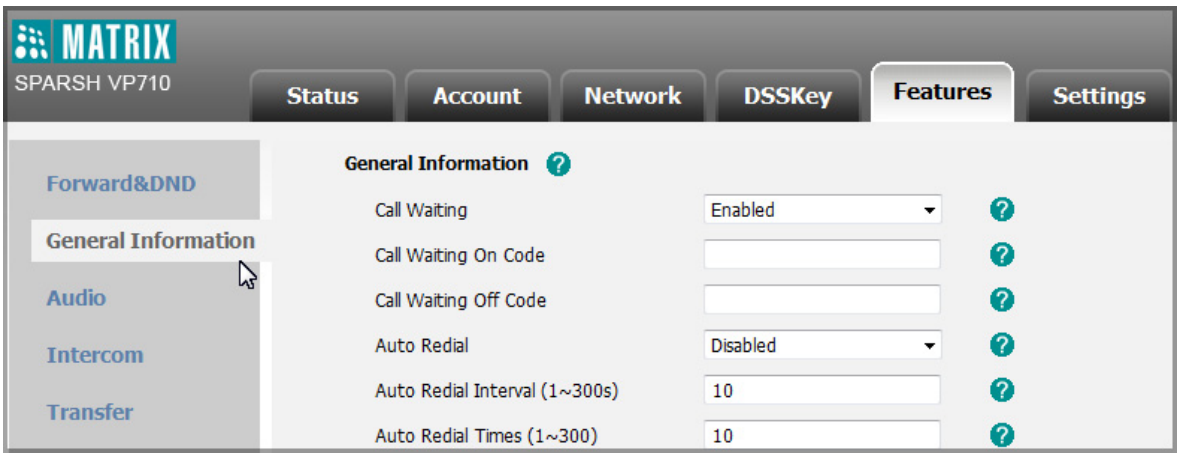
- Select the desired value from the **RFC 2543 Hold** list.



- Click **Confirm** to save the change.

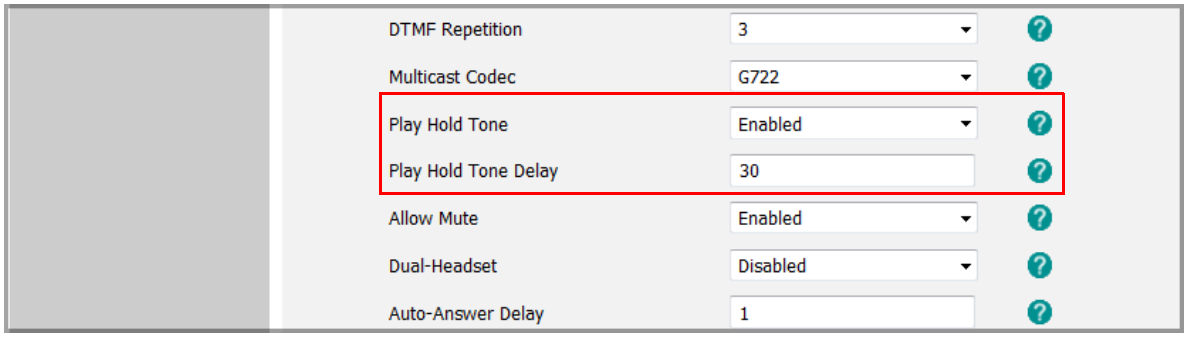
To configure call hold tone and call hold tone delay via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Play Hold Tone** list.

- Enter the desired time in the **Play Hold Tone Delay** field.

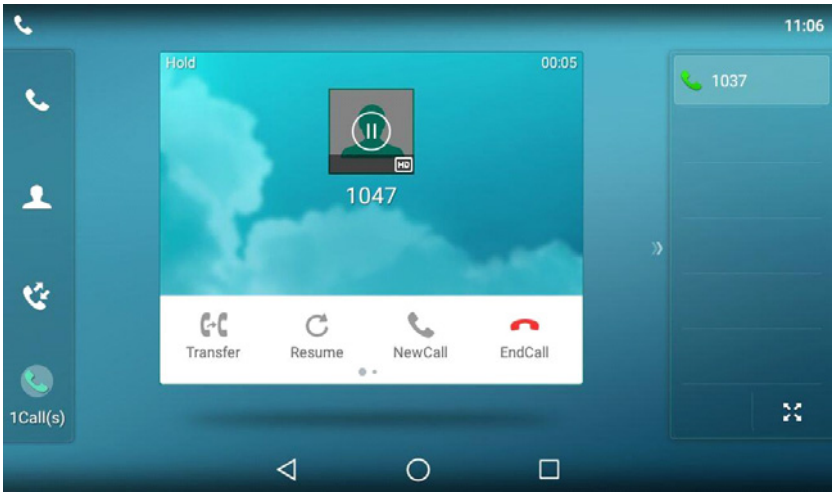


- Click **Confirm** to save the change.

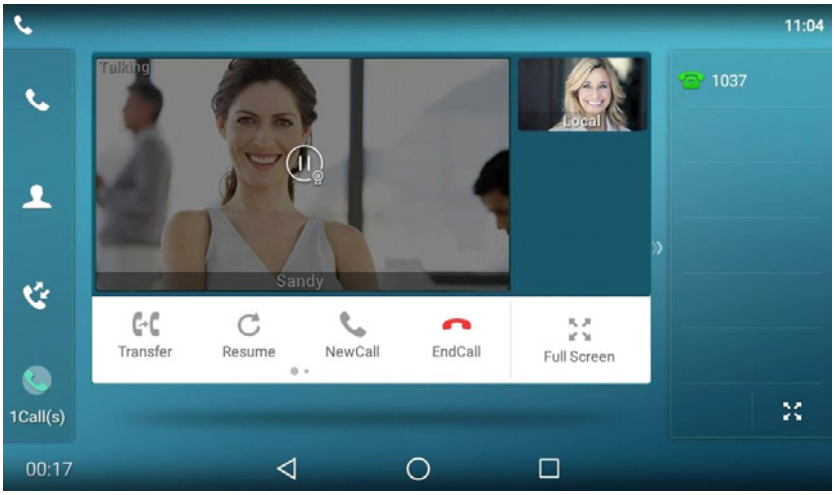
To place a call on hold:

- Press  or tap the **Hold** soft key during a call.

  The following shows an audio-only call has been held:



The following shows a video call has been held:



The phone will beep softly every 30 seconds to remind you that you still have a call on hold**.**

To resume a held call:

- Press  or tap the **Resume** soft key.

Multiple Calls on Hold:

If multiple calls are placed on hold, do one of the following:

- Tap the desired call and then tap the **Resume** soft key to retrieve it.
- Tap the desired call and then tap the corresponding line key to retrieve it.

Call Number Filter

Call number filter allows the IP phone to automatically filter designated characters when dialing.

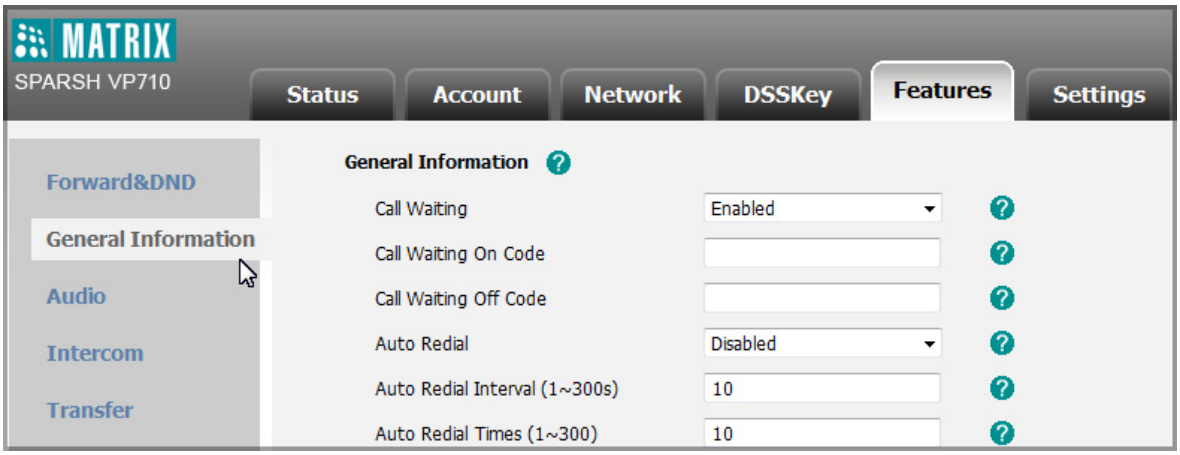
Procedure

Call number filter can be configured using the configuration files or locally.

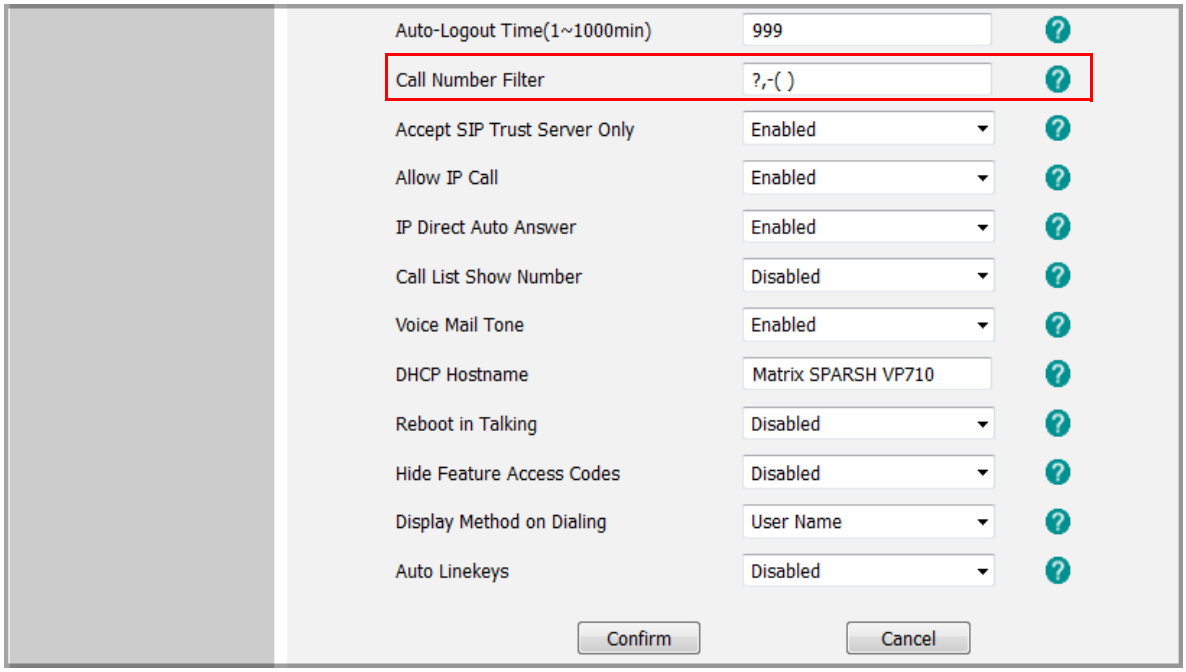
Configuration File	<MAC>.cfg	Configure the characters the IP phone filters when dialing. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the characters the IP phone filters when dialing. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure the characters the IP phone will filter via web user interface:

- Click on **Feature->General Information**.



- Enter the desired characters in the **Call Number Filter** field.



- Click **Confirm** to save the change.

Call Park

Call park allows you to place a call on hold so that it can be retrieved from any other phone. For example: a phone in another office or conference room.

You can park an active call on the extension known as call park orbit by tapping a call park key on the phone. If the call is parked successfully, the response is either a voice prompt confirming that the call was parked or a visible prompt on the touch screen.The current call is placed on hold and can be retrieved from another phone.
The parked call can now be retrieved by dialing call park retrieve code.

If the parked call is not retrieved within a period of time assigned by the system, the phone performing call park will receive a call back. This feature depends on support from a SIP server. For more information on how to configure DSS Key for Call Park, refer "DSS Keys".



Call park is not available on all servers. Contact your system administrator for more information.

Procedure

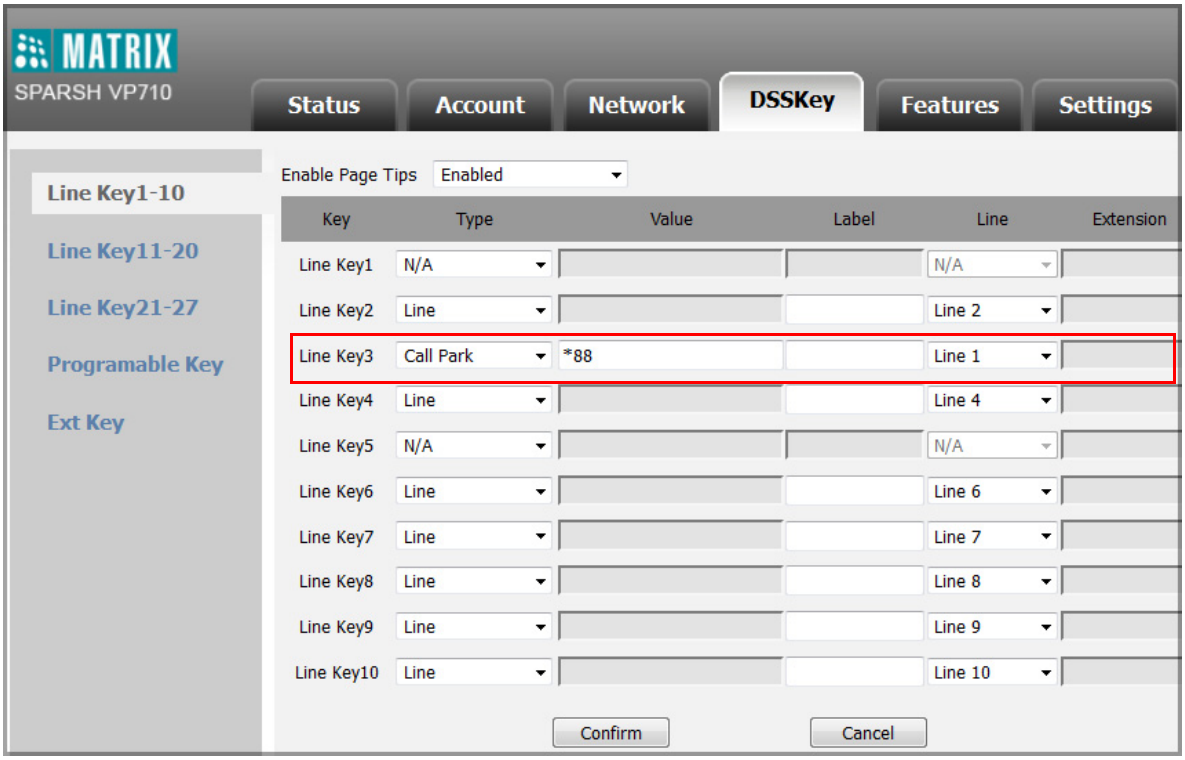
Call park key can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Assign a call park key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Assign a call park key. **Navigate to:** http://<phoneIPAddress>/servlet ?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a call park key.

To configure a call park key via web user interface:

- Click on **DSSKey->Line Key.**
- In the desired DSS key field, select **Call Park** from the **Type** list.
- Enter the desired value (e.g., call park feature code) in the **Value** field.
- In the **Label** field (optional), enter the string that will appear on the touch screen.

- Select the desired line from the **Line** list.

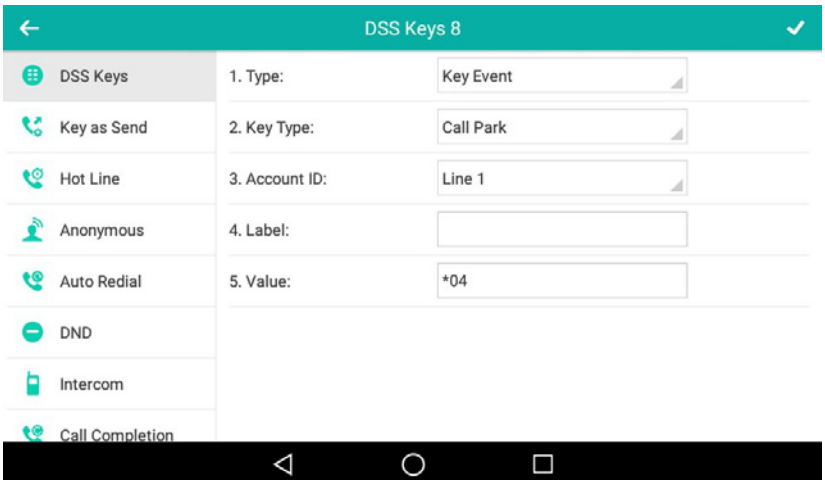


- Click **Confirm** to save the change.

To configure a call park key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Tap the desired line key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **Call Park** in the pop-up dialog box.
- Tap the **Account ID** field.
- Tap the desired line in the pop-up dialog box.
- In the **Label** field (optional), enter the string that will appear on the touch screen.

- Enter the call park code in the **Value** field.



- Tap ✓ to save the change or ◁ to cancel.

To use call park:

- A places a call to B.
- B wants to take the call in a conference room for privacy and so he taps the call park key.
- B retrieves the parked call by dialing the call park retrieve code from the extension present in the conference room.
- The call is established by the system.

- *The call park code and call park retrieve code are predefined on the system server. Contact your system administrator for more information.*
- *If the parked call is not retrieved within a period of time assigned by the system, the phone performing call park will receive a call back.*

Call Pickup

Call pickup allows you to answer someone else's incoming call from your phone, without physically going to the ringing phone.

The IP phone supports two types of pickup:

- Directed Call Pickup: This is used for picking up a call that is ringing at a target phone number.
- Group Call Pickup: This is used for picking up a call that is ringing at any phone number in the group.

To use call pickup, you need to configure the call pickup code before-hand on a global basis or per-line basis via web user interface.The call pickup code configured on a per-line basis takes precedence over that configured on a global basis.

Directed Call Pickup

Directed call pickup is used for picking up an incoming call on a specific extension. You can pick up the incoming call using a directed pickup key or the **DPickup** soft key. For more information on how to configure DSS Key for directed call pickup, refer "DSS Keys".

This feature depends on the support from a SIP server. For many SIP servers, directed call pickup requires a directed pickup code, which can be configured on a phone or a per-line basis.

It is recommended not to configure the directed call pickup key and the Dpickup soft key simultaneously. If you do, the directed call pickup key will not be used correctly.

Procedure

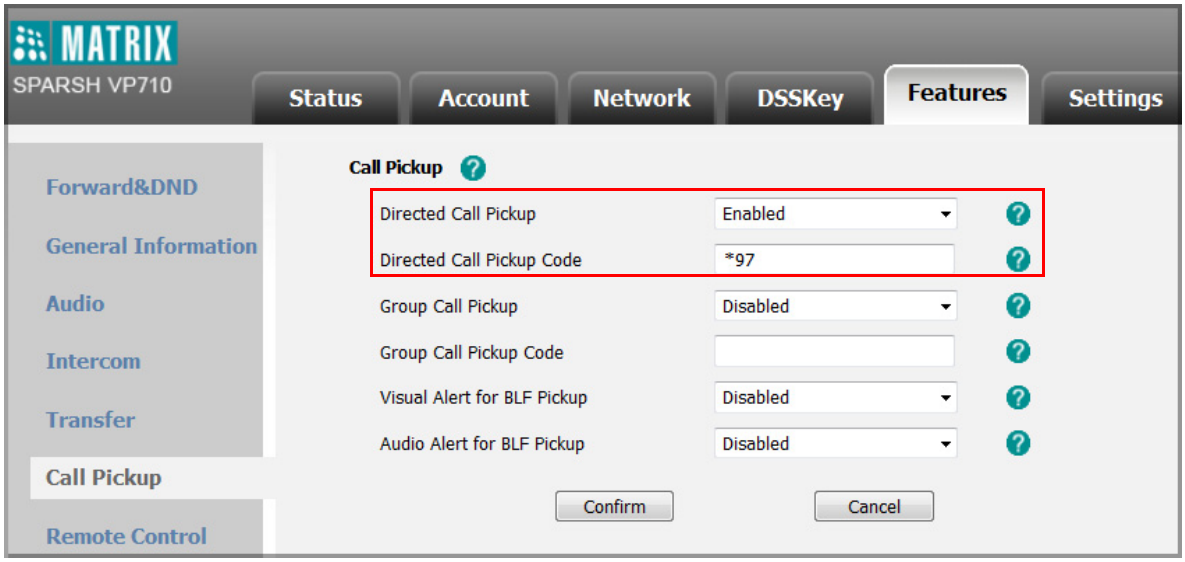
Directed call pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the directed call pickup code on a per-line basis. Configure directed call pickup features on a phone basis. Assign a directed call pickup key. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Assign a directed call pickup key. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load Configure directed call pickup code on a per-line basis. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0 Configure directed call pickup feature on a phone basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-callpickup&q=load
	Phone User Interface	Assign a directed call pickup key.

To enable directed call pickup and configure the directed call pickup code on a global basis via web user interface:

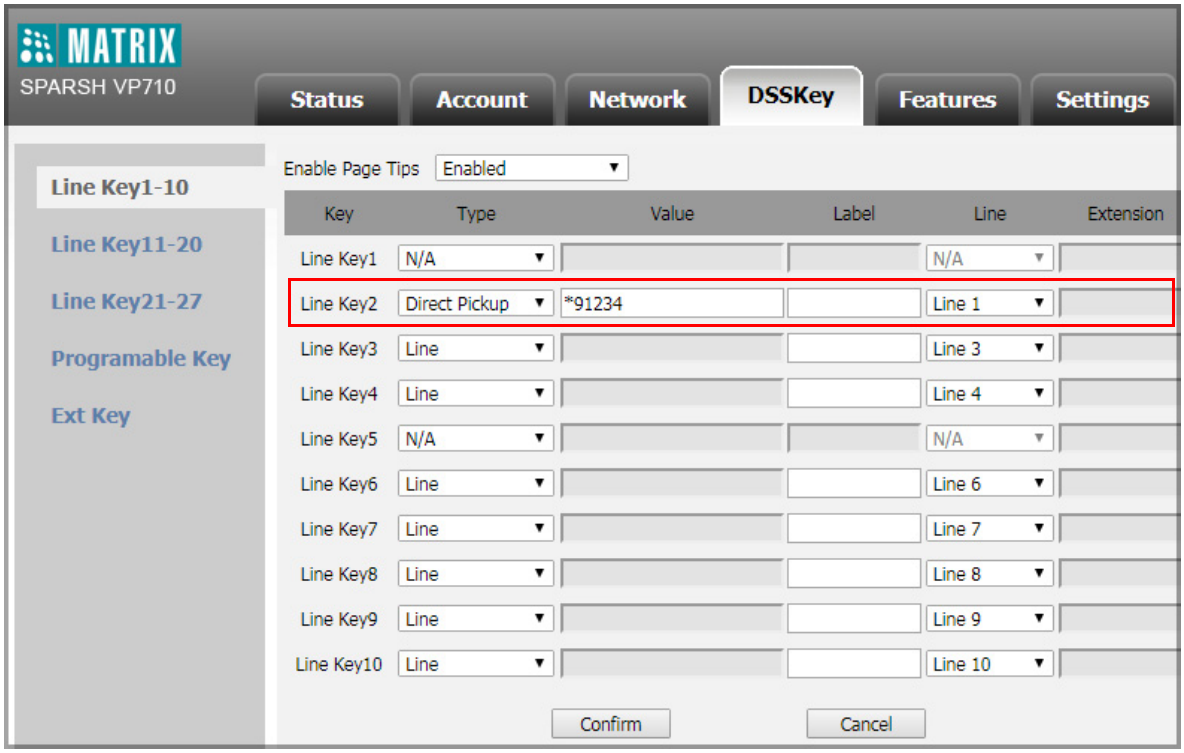
- Click on **Features->Call Pickup**.
- Select **Enabled** from the **Directed Call Pickup** list.
- In the **Directed Call Pickup Code** field, enter the directed call pickup code.



- Click **Confirm** to save the change.

To configure a directed call pickup key via web user interface:

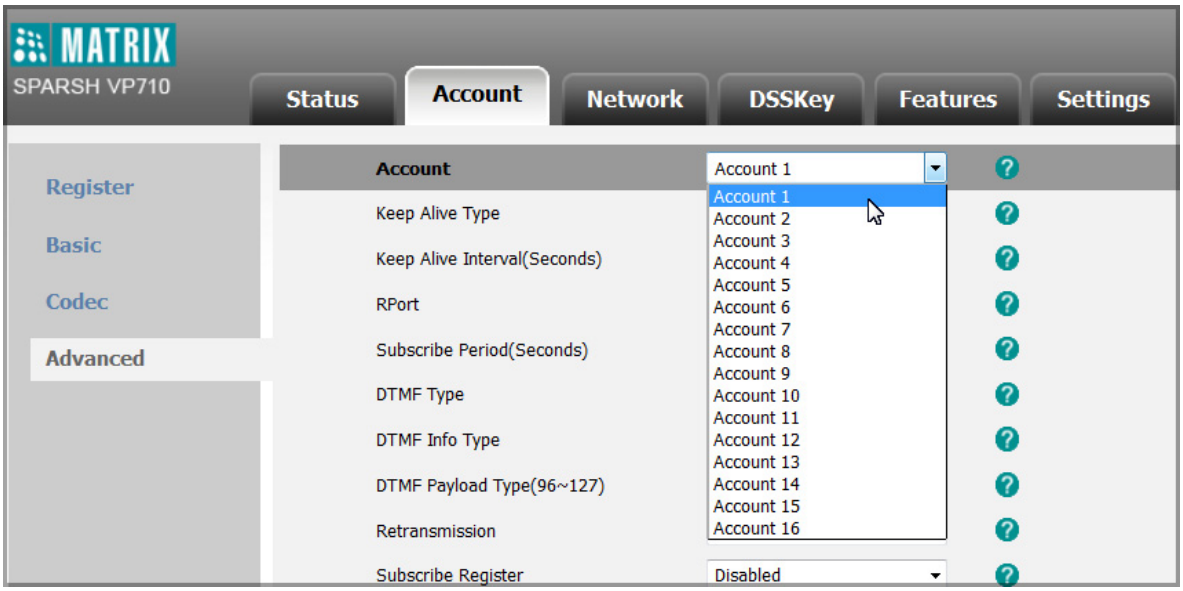
- Click on **DSSKey->Line Key (**or **Programable Key)**.
- In the desired DSS key field, select **Direct Pickup** from the **Type** list.
- In the **Value** field, enter the directed call pickup code followed by the specific extension.
- In the **Label** field (optional), enter the string that will appear on the touch screen.
- Select the desired line from the **Line** list.



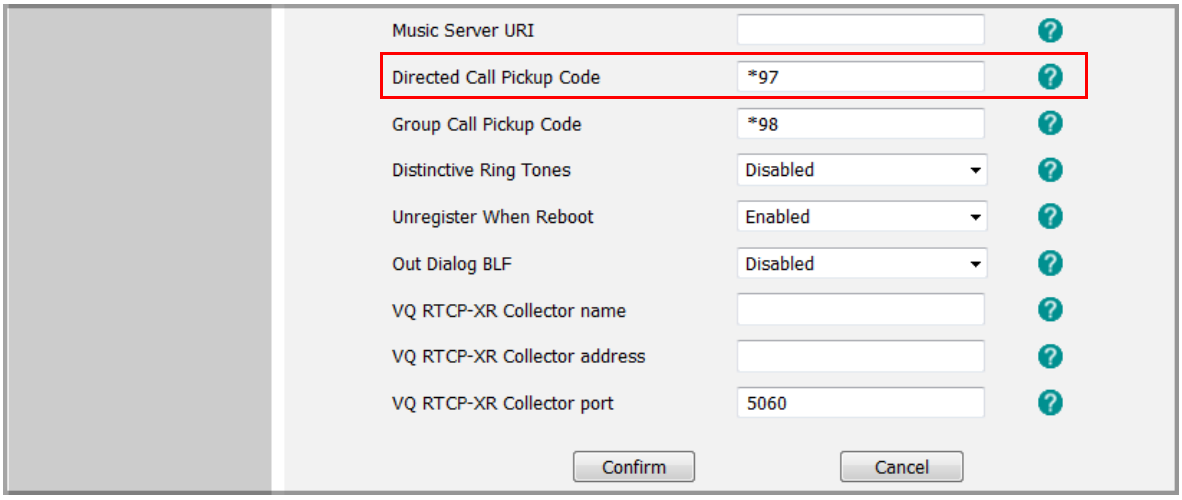
- Click **Confirm** to save the change.

To configure the directed call pickup code on a per-line basis via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- In the **Directed Call Pickup Code** field, enter the directed call pickup code.

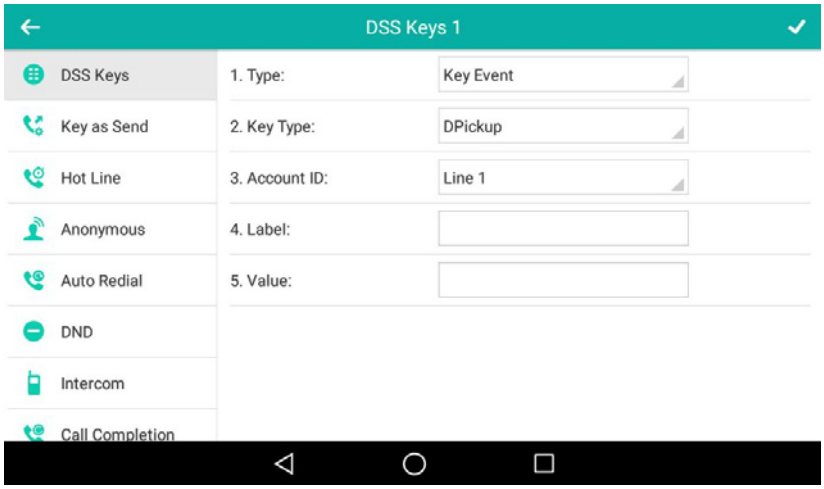


- Click **Confirm** to save the change.

To configure a directed pickup key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Select the desired DSS key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **DPickup** in the pop-up dialog box.

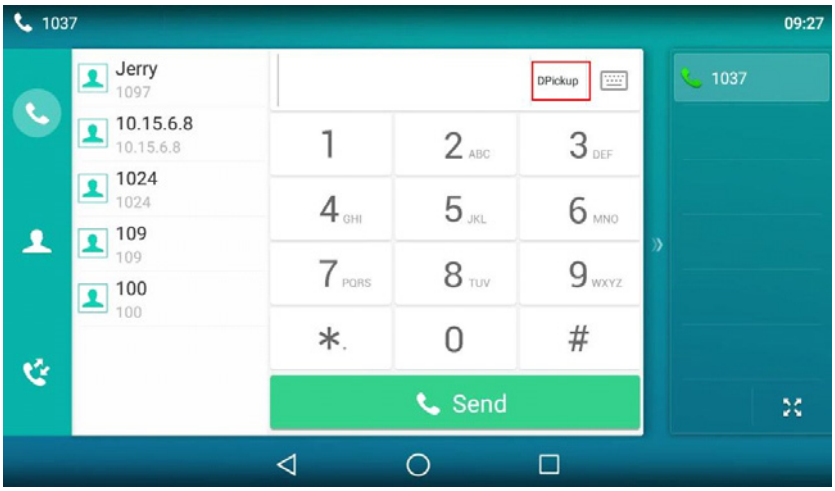
- Tap the **Account ID** field.
- Tap the desired line in the pop-up dialog box.
- In the **Label** field (optional), enter the string that will appear on the touch screen.
- In the **Value** field, enter the directed call pickup code followed by the specific extension.



- Tap to save the change or to cancel.

To pick up a call directly:

- Pick up the handset, press the Speakerphone key or tap the line key.**DPickup** appears on the touch screen



- Tap **DPickup** on your phone when the target phone receives an incoming call.
- Enter the phone number which is receiving an incoming call.
- Tap **DPickup** again.

  The call is answered on your phone.

You can also configure a line key as a direct pickup key via phone user interface or web user interface. Once configured, you can pick up a call by tapping the direct pickup key directly. For more information, refer to “Line Keys”.

You can configure a programmable key as directed pickup key via web user interface only. Relevant instructions to do that have been mentioned above. You can pick up a call by pressing the directed pickup key directly.

Group Call Pickup

Group call pickup is used for picking up a call that is ringing at any phone in the group. The pickup group should be predefined.If the group receives many incoming calls at once, you can pick up the first incoming call, using a group pickup key or the **GPickup** soft key For more information on how to configure DSS Key for group call pickup, refer "DSS Keys".

This feature depends on support from a SIP server. For many SIP servers, group call pickup requires a group pickup code, which can be configured on a phone.

- *If there are many incoming calls at the same time, tapping the GPickup soft key on the phone will pick up the call that rings first.*
- *The pickup group should be predefined, contact your system administrator for more information.*

Procedure

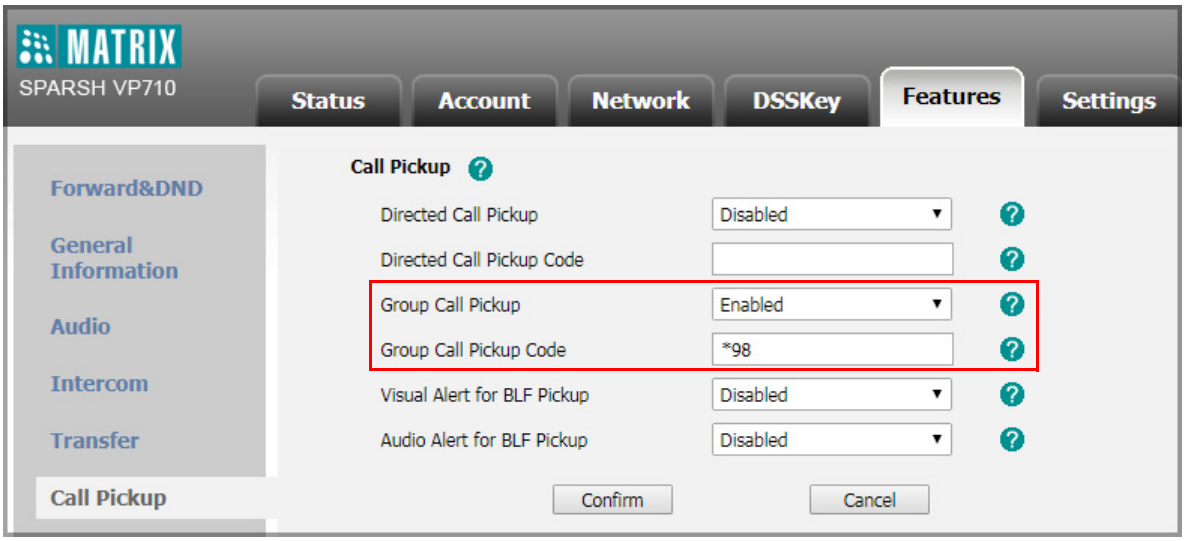
Group call pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the group call pickup code on a per-line basis. Configure group call pickup feature on a phone basis. Assign a group call pickup key. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Assign a group call pickup key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load Configure the group call pickup code on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0 Configure group call pickup feature on a phone basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-callpickup&q=load
	Phone User Interface	Assign a group call pickup key.

To enable group call pickup and configure the group call pickup code on a global basis via web user interface:

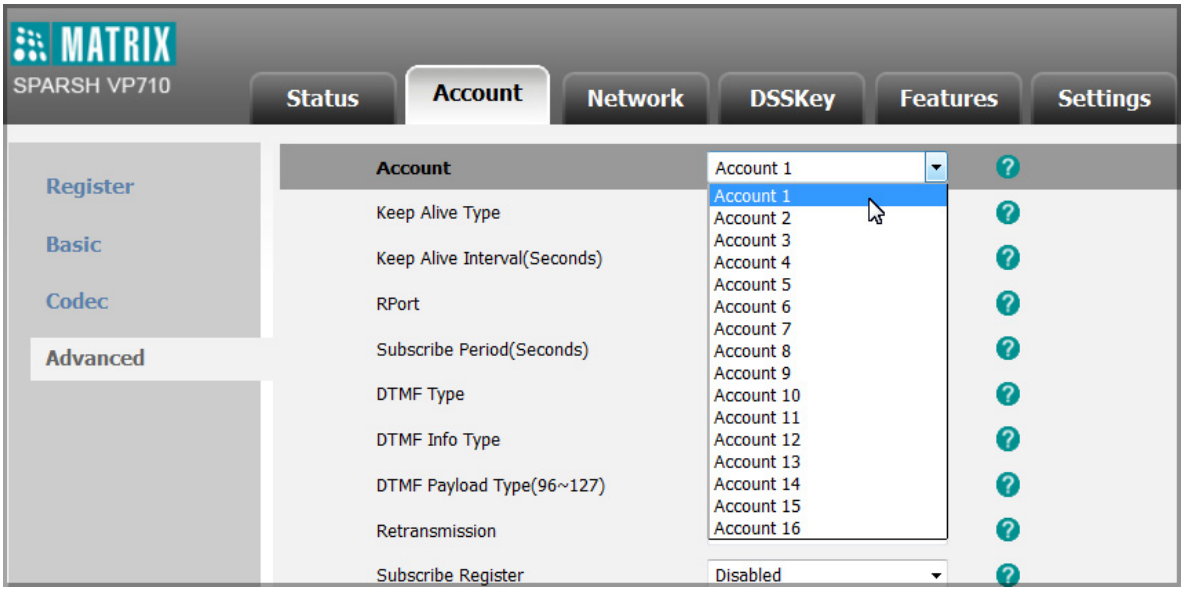
- Click on **Features->Call Pickup**.
- Select the desired value from the **Group Call Pickup** list.
- In the **Group Call Pickup Code** field, enter the group call pickup code.



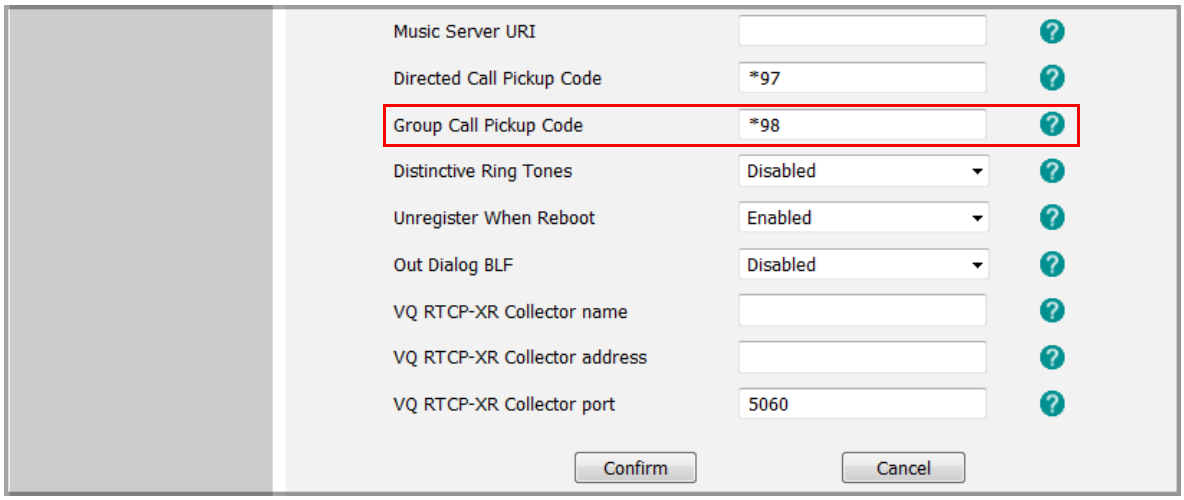
- Click Confirm to save the change.

To configure the group call pickup code on a per-line basis via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- In the **Group Call Pickup Code** field., enter the group call pickup code.

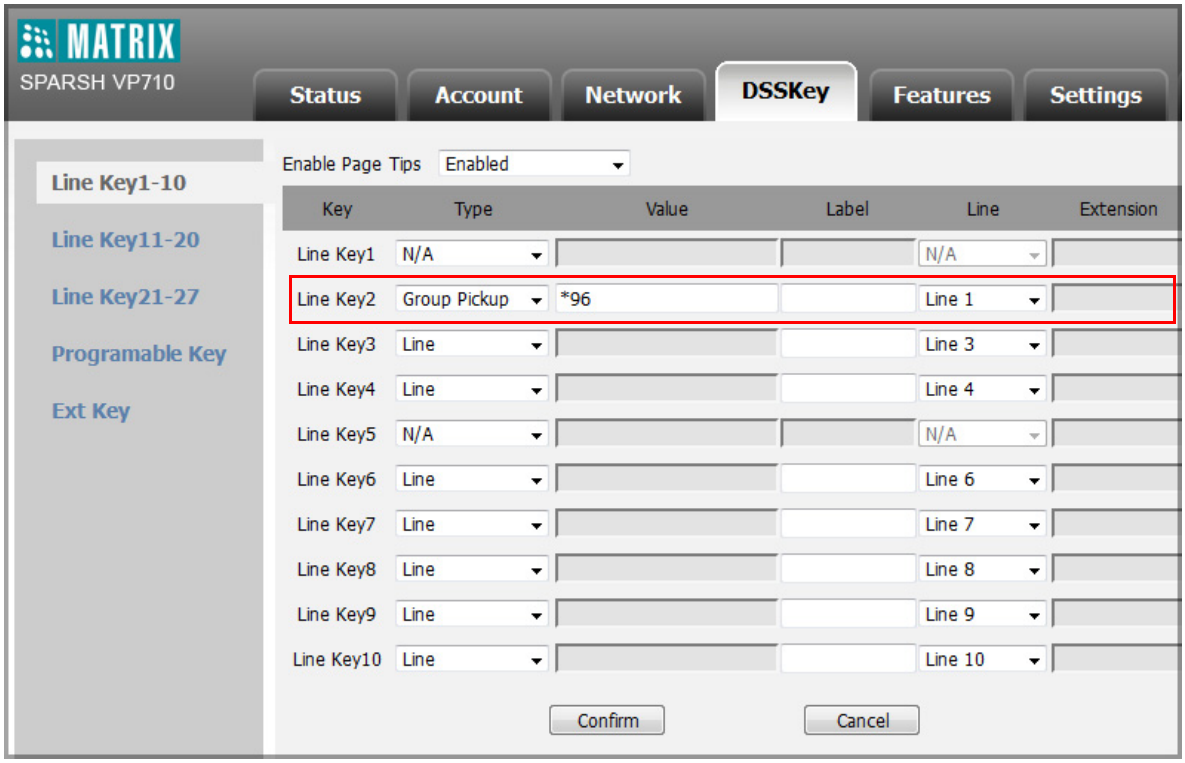


- Click **Confirm** to save the change.

To configure a group call pickup key via web user interface:

- Click on **DSS Key-> Line Key** (or **Programable Key).**
- In the desired DSS key field, select **Group Pickup** from the **Type** list.
- In the **Value** field, enter the group call pickup code.
- In the **Label** field (optional), enter the string that will appear on the touch screen.

- Select the desired line from the **Line** list.

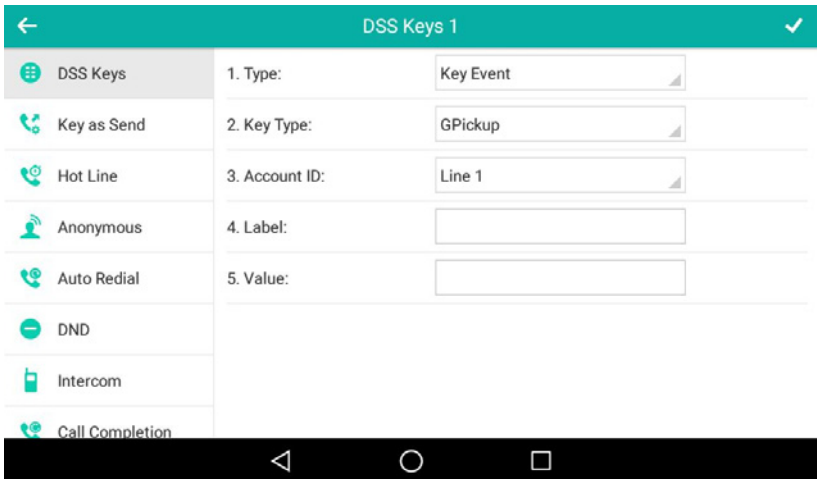


- Click **Confirm** to save the change.

To configure a group pickup key via phone user interface:

- Tap **Settings->Features->DSS Keys**.
- Select the desired DSS key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **GPickup** in the pop-up dialog box.
- Tap the **Account ID** field.
- Tap the desired line in the pop-up dialog box.
- In the **Label** field (optional), enter the string that will appear on the touch screen.

- In the **Value** field, enter the group call pickup code.

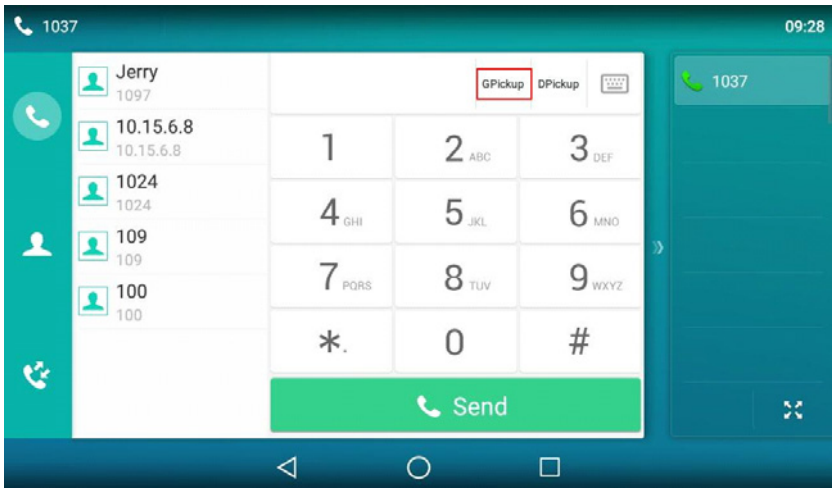


- Tap ✓ to save the change or ◁ to cancel.

To pick up a call in the group:

- Pick up the handset, press the Speakerphone key or tap the line key.

  GPickup appears on the touch screen



- Tap **GPickup** on your phone when a phone in the group receives an incoming call.
- The call is answered on your phone.

You can also configure a line key as a group pickup key via phone user interface or web user interface. Once configured, you can pick up a call by tapping the group pickup key directly.

The directed call pickup code and group call pickup code are predefined on the system server. Contact your system administrator for more information.

Dialog Info Call Pickup

Call pickup is implemented through SIP signals on some specific servers. When this feature is enabled, IP phones support picking up incoming calls via the INVITE message which includes a Replaces header in the message body. The value of Replaces is derived from a NOTIFY message with dialog-info event. You can pick up an incoming call

by tapping the DSS key used to monitor a specific extension (such as the BLF key). For more information on BLF, refer to "Busy Lamp Field (BLF)".

If the visual alert for BLF pickup feature is enabled, you can also pick up an incoming call by tapping the DPickup key. For more information on visual alert for BLF pickup, refer to "Visual Alert and Audio Alert for BLF Pickup".
.

In this way, you do not need to configure the directed call pickup code.

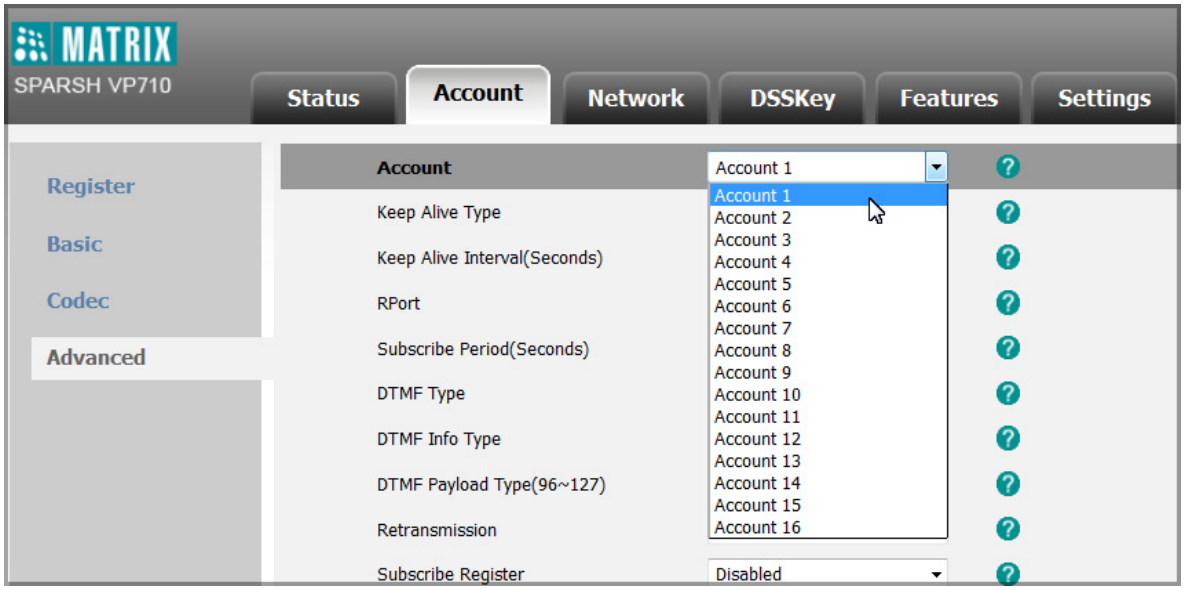
Procedure

Dialog info call pickup can be configured using the configuration files or locally.

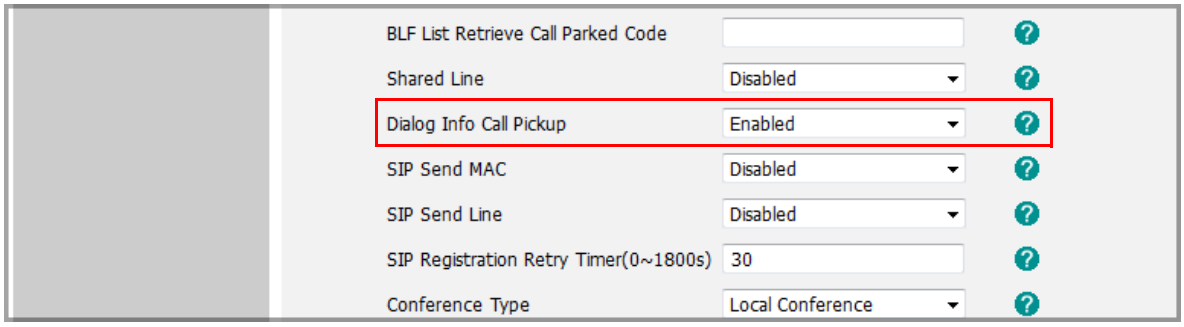
Configuration File	<MAC>.cfg	Configure dialog info call pickup. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure dialog info call pickup. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure dialog info call pickup via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Dialog Info Call Pickup** list.



- Click **Confirm** to save the change.

Call Recording

Call and Conference Recording Using a Softkey

The IP phone supports recording calls (audio-only calls or video calls) or conferences by tapping the **Record** soft key during a call.There is a time limit - 12 hours

You can also set a mark while recording.This feature is useful during important meetings or discussions, allowing you to mark some pivotal point in the discussion that you may need as a reference later.

The recorded video calls are saved in *.mkv format and include a date & time stamp and the other party's number/IP address/name (or the first person's number/IP address/name you called).
For example, 20160302-1452-Tom was created on March 2, 2016, at 14:52 and you have a call with Tom.
The recorded audio-only calls are saved in *.aac format. ou can view the recorded files by **FileManager** application

By default, the recorded files are saved in the internal SD card. But if there is a USB flash drive connected to your phone, the recorded files will be saved in the storage device according to the priority: USB flash drive>internal SD card.

You can also record audio-only calls by tapping record/URL record key. For more information, refer to "Call Recording Using DSS Keys (Record and URL Record)".

In addition, the IP phone allows you to record audio and access audio recording files by **Recorder** application. For more information, refer to "Recorder".



- *If you use a record key to record a call, the recorded file will be stored in servers. For more information, refer to Call Recording Using a Record Key.*
- *Before recording any call, especially those involving PSTN, it is necessary to know about the rules and restrictions of any governing call-recording in the place you are in. It is also very important to have the consent of the person you are calling before recording the conversation.*

The Record soft key controls the recording function, and is available (you need to swipe the soft key area left to see it):

- When there are one or more calls connected to your phone
- During an active call
- When calls are muted
- During an attended transfer
- During a conference call

The Record soft key is not available:

- When there are no connected calls on your phone
- When you place a new call (to transfer or to conference)
- When the phone prompts you to answer an incoming call

The recording is not be paused when the following occurs:

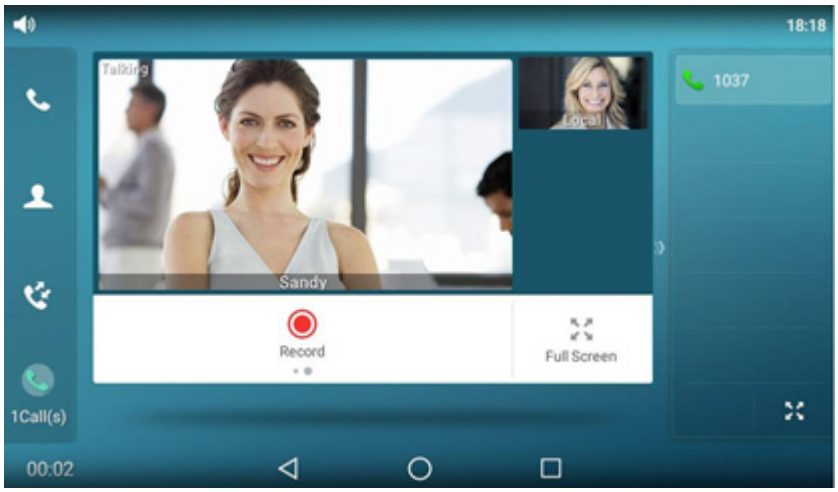
- You place a call on hold.

- You mute a call.
- You set up a conference call
- You perform the semi-attended or attended transfer
- An incoming calls arrives on your phone

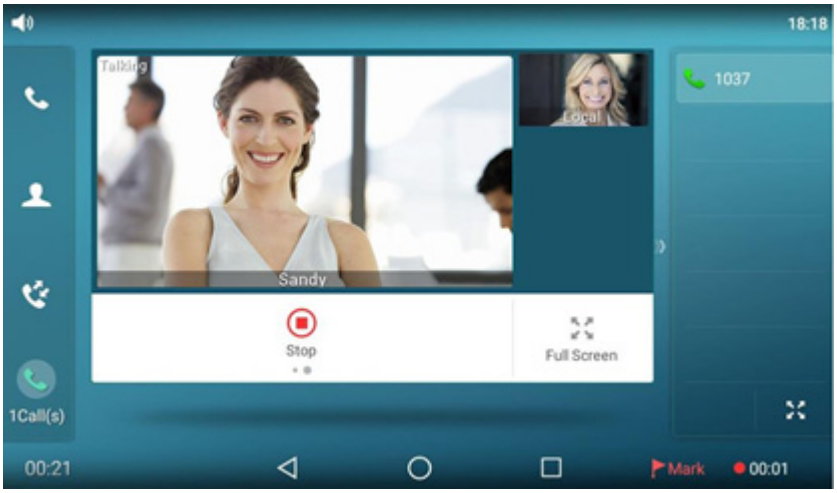
Call Recording

To record a call via phone user interface:

- Swipe the soft key area left and then tap the Record soft key during a call.



If the recording starts successfully, the recording icon and recording duration will appear on the bottom-right of the touch screen.



- Tap Mark on the bottom-right of the screen to set a mark (Optional).

  The icon changes to +1, the number will increase if you set more marks.There is a red flag on this moment when playing the recorded file.

  If you tap the Hold soft key or press the key while recording, only you are recorded.

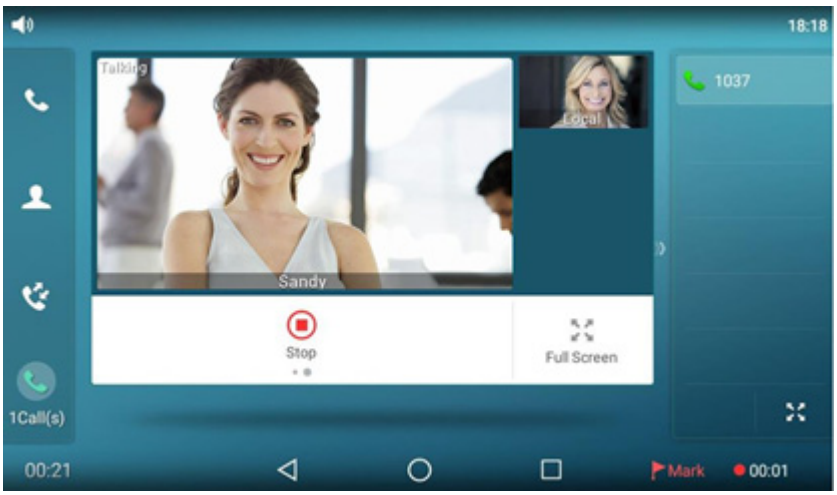
  If you press while recording, only the callee is recorded.

You can hold, transfer or set up a conference call while recording.

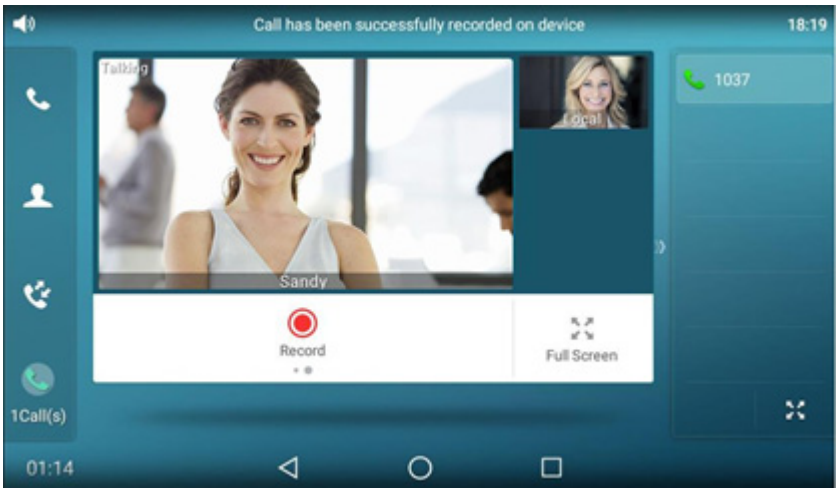
When you end a call while recording, the recording will be stopped and saved automatically.

To stop recording while the phone records, do one of the following:

- Swipe the soft key area left and then tap the **Stop** soft key during a call.

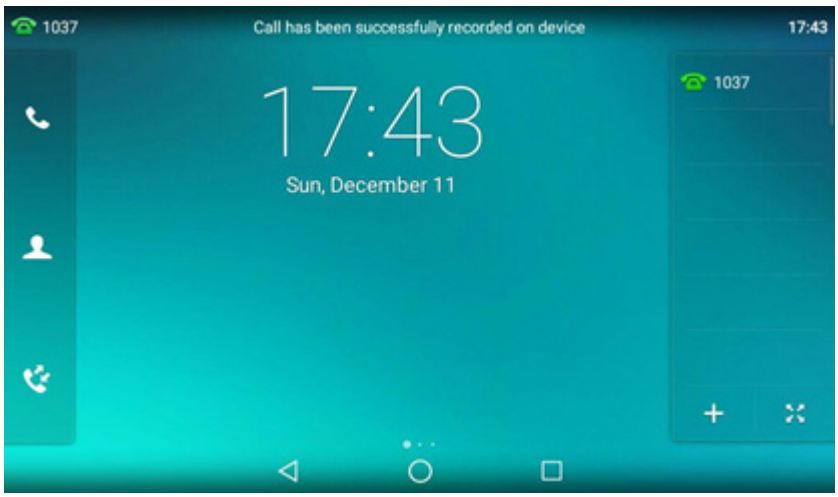


The status bar prompts “Call has been successfully recorded on device”.



- When there is only a call on the phone, tap the **EndCall** soft key.

  The status bar prompts “Call has been successfully recorded on device”, the recording icon and recording duration disappear, and the phone returns back to the idle screen.



The recording will be stored as a new file when the phone starts recording again.

Conference Recording

You can record conference calls in the same way as other calls with the following exceptions:

- All conference participants are recorded while recording. If one of the participants taps the **Hold** soft key or press the key only that participant is recorded. When a conference call is placed on hold, recording of the conference is paused. You can place or answer other calls, which will be recorded in the same file. When the conference call is resumed, recording of the conference resumes.
- All conference participants are recorded while recording. If one of the participants presses , only that participant is not recorded.

Call Recording Using a Record Key

You can record calls by tapping a record key on the phone. The IP phone supports record and URL record.

Two ways of call recording:

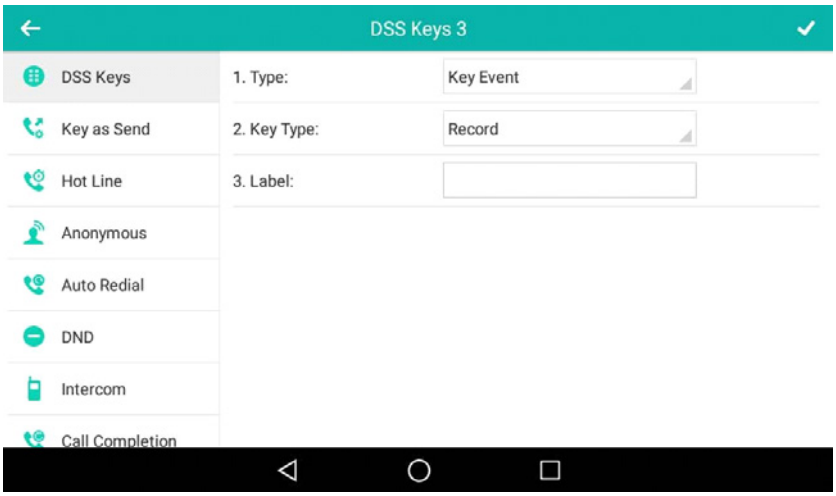
- **Record:** The phone sends a SIP INFO message containing a specific header ¡§Record: on/off¡¨ to trigger a recording.
- **URL Record:** The phone sends an HTTP URL request to trigger a recording. Contact your system administrator for the predefined URL.

- *If you record a call using a softkey, the recorded file will be stored in the internal SD card or USB flash drive.For more information, refer to Call and Conference Recording Using a Softkey*
- *Call recording is not available on all servers. Contact your system administrator for more information.*

.

To configure a record key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired line key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **Record** in the pop-up dialog box.
- In the **Label** field (Optional), enter the string that will appear on the touch screen.

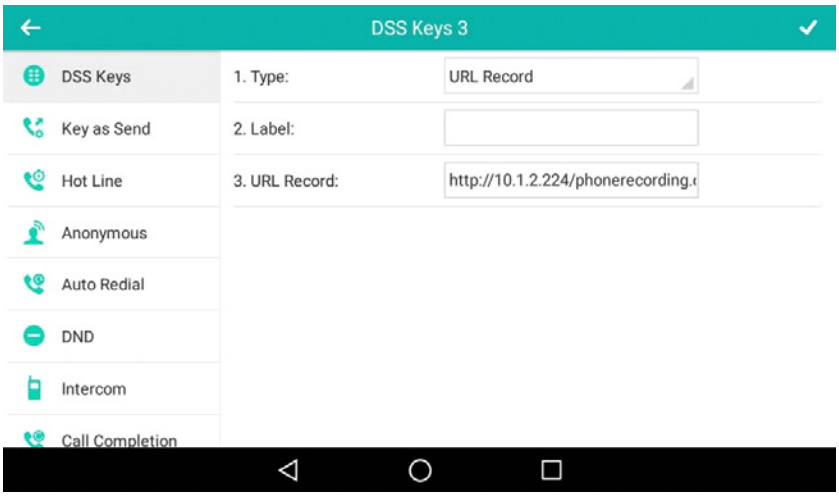


- Tap ✓ to save the change or ◁ to cancel.

To configure a URL record key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Tap the desired line key.
- Tap the **Type** field.
- Tap **URL Record** in the pop-up dialog box.
- In the **Label** field (Optional), enter the string that will appear on the touch screen.

- Enter the URL (e.g., http://10.1.2.224/phonerecording.cgi) in the **URL Record** field.



- Tap ✓ to save the change or ◁ to cancel.

The Record and URL Record keys control the recording function, and are available:

- During an active call
- When calls are on hold or muted
- During a conference call

The Record and URL Record keys are not available when:

- There are no connected calls on your phone
- You place a new call

To record a call:

- Tap the Record key or URL Record key during a call.
  If the recording starts successfully, the recording icon will appear at the bottom-right corner of the contact avatar and the icon indicator of the Record key or URL Record key will change to .
- Tap the Record key or URL Record key again to stop recording.
  The recording icon disappears from the bottom-right corner of the contact avatar and the icon indicator of the Record key or URL Record key changes to .

To know the Recording status indications, refer “Icon Instructions” for details.

You can listen to the recordings stored on your server system. For example, you can dial an access code to listen to the recordings.

The way in which you listen to the recordings may be different depending on the server. Contact your system administrator for more information.

Call Recording Using DSS Keys (Record and URL Record)

The IP phone supports recording calls by tapping the call record key. It depends on support from a SIP server. When you tap the call record key, the IP phone sends a record request to the server.

IP phone itself do not have memory to store the recording, what they do is to trigger the recording and indicate the recording status.
Normally, there are 2 main methods to trigger a recording on a certain server. We call them record and URL record. Record is for the IP phone to send the server a SIP INFO message containing a specific header. URL record is for the IP phone to send the server an HTTP GET message containing a specific URL. The server processes these messages and decides to start or stop a recording.

If it is a video call, you can only record the audio but not video by tapping record/URL record key.For more information on recording video calls, refer to Call Recording Using Soft Key.

Record

When a user taps a record key for the first time during a call, the IP phone sends a SIP INFO message to the server with the specific header “Record: on”, and then the recording starts.

Example of a SIP INFO message:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK1870385345
From: "1009" <sip:1009@10.2.1.48:5060>;tag=1385842459
To: <sip:1006@10.2.1.48:5060>;tag=2383911905
Call-ID: 0_1289812066@10.3.20.14
CSeq: 2 INFO
Contact: <sip:1009@10.3.20.14:5060>
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Record: on
Content-Length: 0
```

When the user taps the record key for the second time, the IP phone sends a SIP INFO message to the server with the specific header “Record: off”, and then the recording stops.

Example of a SIP INFO message:

```
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK175716007
From: "1009" <sip:1009@10.2.1.48:5060>;tag=1385842459
To: <sip:1006@10.2.1.48:5060>;tag=2383911905
Call-ID: 0_1289812066@10.3.20.14
CSeq: 3 INFO
Contact: <sip:1009@10.3.20.14:5060>
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Record: off
Content-Length: 0
```

URL Record

When a user taps a URL record key for the first time during a call, the IP phone sends an HTTP GET message to the server.If the recording is successfully started, the server will respond with a 200 OK message.If the recording fails for some reasons, for example, the recording box is full, the server will respond with a 200 OK message When the user taps the URL record key for the second time, the IP phone sends an HTTP GET message to the server, and then the server will respond with a 200 OK message

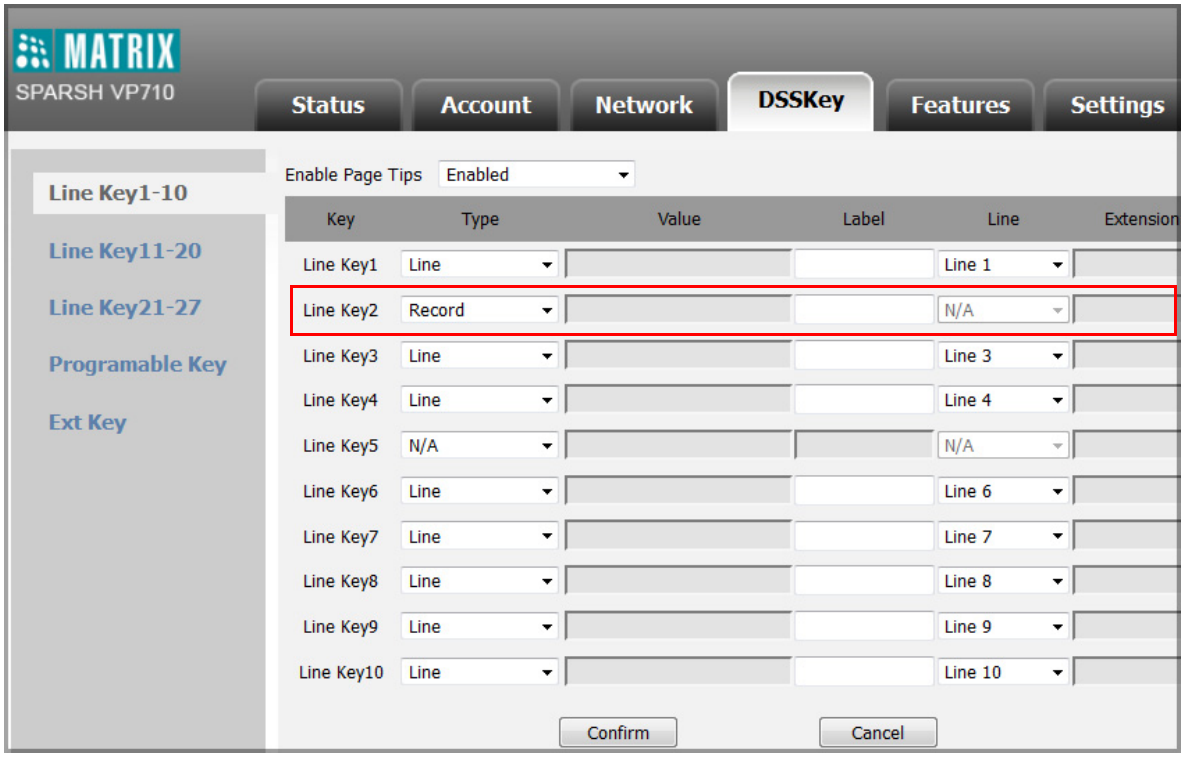
Procedure

Record or URL record key can be configured using the configuration files.

Configuration File	<MAC>.cfg	Assign a record key. Assign a URL record key. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Assign a record key and URL record key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a record key and URL record key.

To configure a record key via web user interface:

- Click on **DSSKey->Line Key**.
- In the desired DSS key field, select **Record** from the **Type** list.
- In the Label field (Optional), enter the string that will appear on the touch screen.

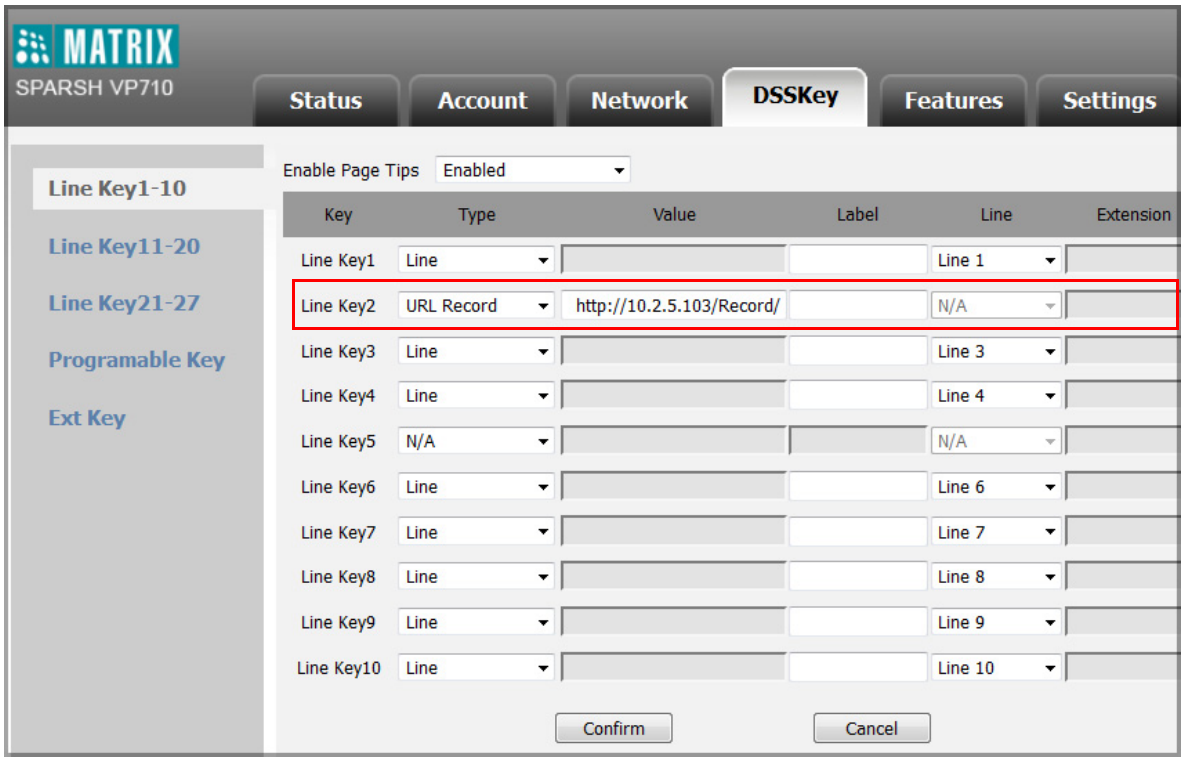


- Click **Confirm** to save the change.

To configure a URL record key via web user interface:

- Click on **DSSKey->Line Key**.
- In the desired DSS key field, select **URL Record** from the **Type** list.
- Enter the URL in the **Value** field.

- In the Label field (optional), enter the string that will appear on the touch screen.



- Click **Confirm** to save the change.

Call Timeout

Call timeout defines a specific period of time within which the IP phone will cancel the dialing if the call is not answered.

Procedure

Call timeout can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the duration time in the ringback state. For more information, refer "Appendix D - Configuration Parameters".

Call Transfer

Call transfer enables IP phones to transfer an existing call to another party. IP phone supports call transfer using the REFER method specified in RFC 3515 and offers three types of transfer:

- **Blind Transfer**—Transfer a call directly to another party without consulting. Blind transfer is implemented by a simple REFER method without Replaces in the Refer-To header.

- **Semi-attended Transfer**—Transfer a call after hearing the ringback tone. Semi-attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

- **Attended Transfer**—Transfer a call with prior consulting. Attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

Normally, call transfer is completed by pressing the transfer key. Blind transfer on-hook and Attended transfer on-hook features allow the IP phone to complete the transfer through on-hook.

When a user performs a semi-attended transfer, semi-attended transfer feature determines whether to display the prompt "n New Missed Call(s)" ("n" indicates the number of the missed calls) on the destination party's phone touch screen.

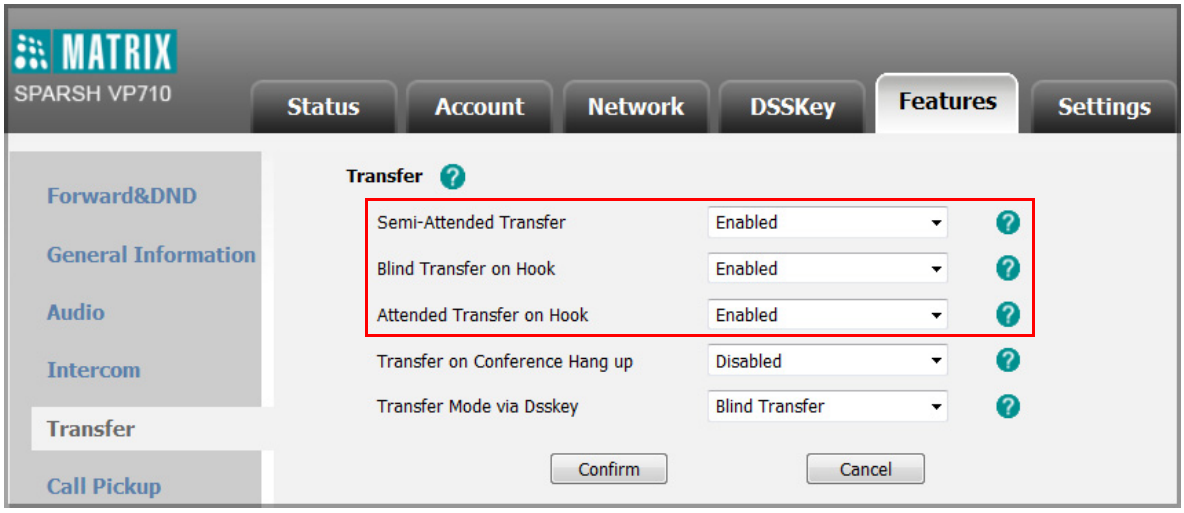
Procedure

Call transfer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify whether to complete the transfer through on-hook. Configure semi-attended transfer feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify whether to complete the transfer through on-hook. Configure semi-attended transfer feature. **Navigate to**: http://<phoneIPAddress>/servlet?p=features-transfer&q=load

To configure call transfer via web user interface:

- Click on **Features->Transfer**.



- Select the desired values from **Semi-Attended Transfer, Blind Transfer On Hook** and **Attended Transfer On Hook** lists.
- Click **Confirm** to save the change.

Blind Transfer

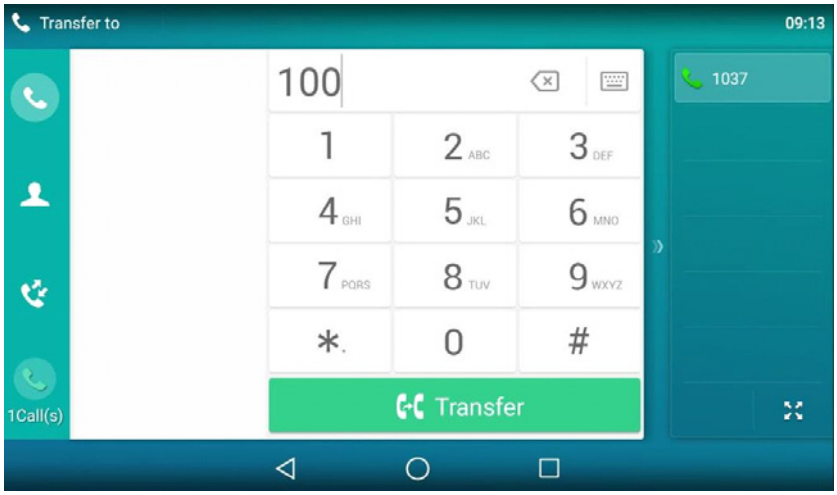
Before performing a blind transfer (except you use dial number method), make sure that the transfer mode via Dsskey is configured as **Blind Transfer**. For more information on how to configure it, refer "Transfer Mode via Dsskey"

To perform a blind transfer:

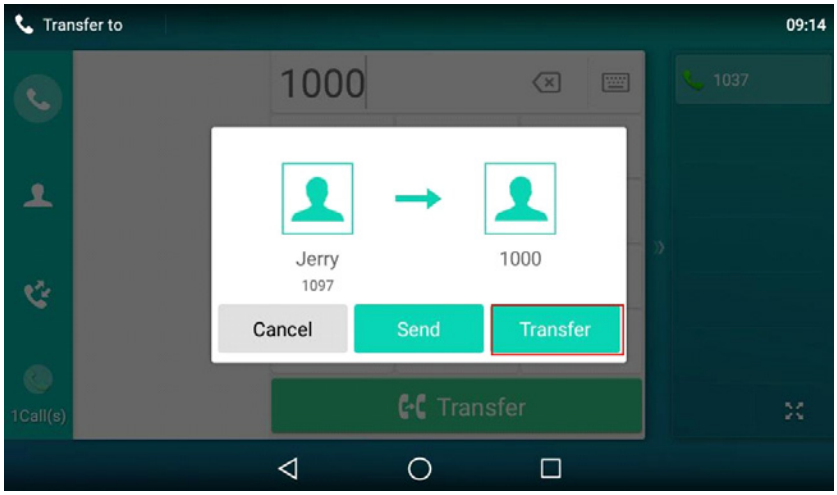
- Press or tap the **Transfer** soft key during a call.
- Do one of the following:
  - Enter the number you want to transfer the call to.
  - Press .

  OR
  - Enter the number you want to transfer the call to.

- Tap the **Transfer** soft key.



The phone will prompt a dialog box as the following picture shows:



- Tap **Transfer** to complete the transfer. (The call will automatically dial out in about 5seconds if you do not tap Transfer.)

OR

- Tap  and then tap **Local Directory or Remote Phone Book** to search the desired contact.
- Tap the desired contact to complete the transfer.For more information refer to “Remote Phone Book”

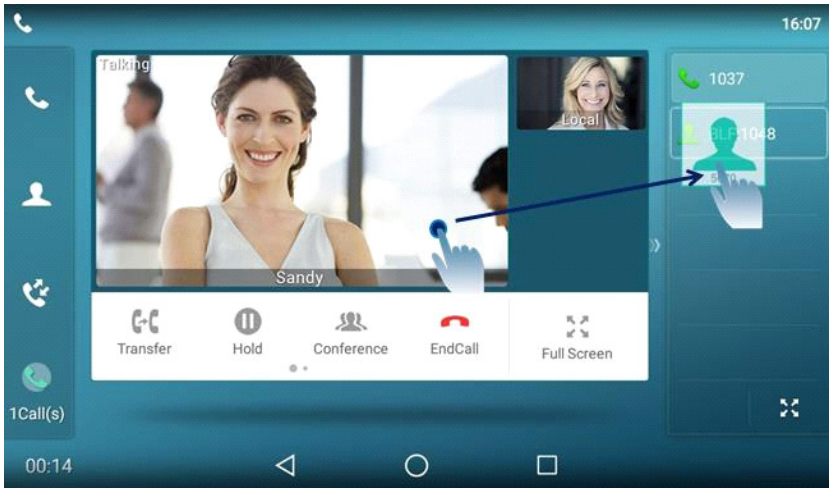
OR

- Tap  and then tap the desired list.
- Tap the desired entry to complete the transfer.

The call is then connected to the number to which you are transferring.

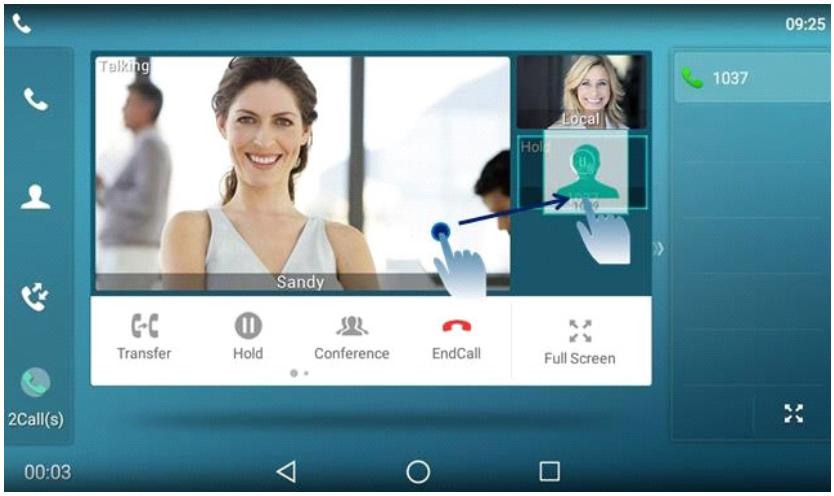
To perform a blind transfer, do one of the following:

- When there is an active call for the account selected from the **Account ID** field on the speed dial or BLF key. Drag the far-site window to that speed dial or BLF key

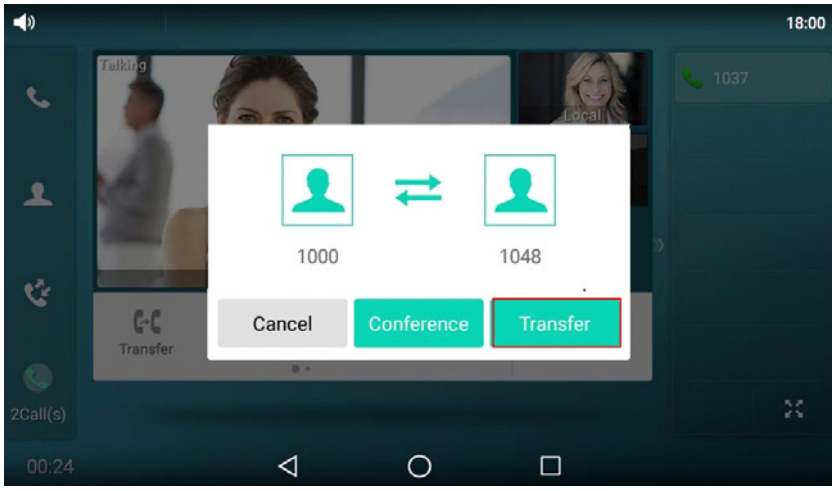


Then the call is connected to the number specified in the **Value** field of the speed dial or BLF key. For more information on how to configure a speed dial or BLF key, refer to “DSS Keys”

- Make sure you have an active call and one or more calls on hold, drag one far-site window to another far-site window.



The touch screen prompts the following:



- Tap **Transfer**.

The active call is connected to the number of the held party.

Semi-Attended Transfer

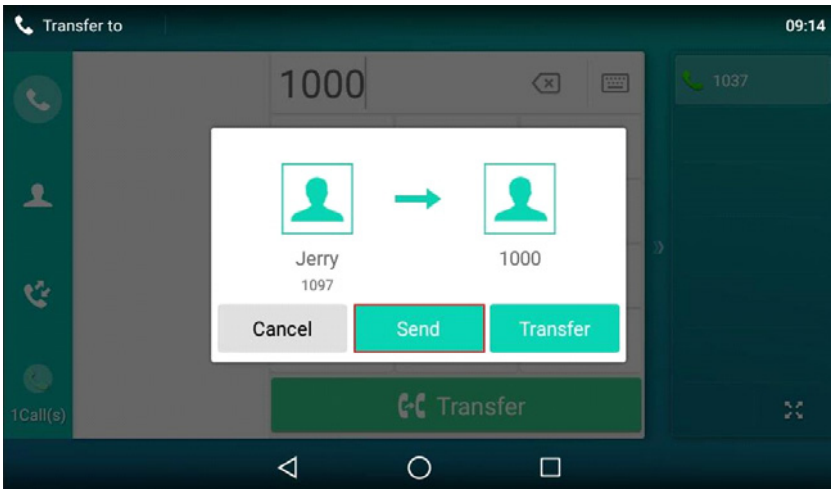
Before performing a semi-attended transfer (except you use dial number method), make sure that the transfer mode via Dsskey is configured as New Call or Attended Transfer. For more information on how to configure it, refer “Transfer Mode via Dsskey”.

To perform a semi-attended transfer:

- Press or tap the **Transfer** soft key during a call.
- Do one of the following:
  - Enter the number you want to transfer the call to.
  - Press # SEND to dial out.

  OR
  - Enter the number you want to transfer the call to.
  - Tap the **Transfer** soft key.

The phone will prompt a dialog box.



- Tap **Send** to dial out. (The call will automatically dial out in about 5 seconds if you do not tap Send.)

OR

- Tap and then tap **Local Directory or Remote Phone Book** to search the desired contact.
- Tap the desired contact to dial out.For more information refer to "Remote Phone Book"

OR

- Tap and then tap the desired list.
- Tap the desired entry to dial out.

- Press or tap the **Transfer** soft key to complete the transfer when receiving ringback.

  If you are using a handset, the transfer can be completed by hanging up the handset.

Attended Transfer

Before performing an attended transfer (except you use dial number method), make sure that the transfer mode via Dsskey is configured as New Call or Attended Transfer. For more information on how to configure it, refer to "Transfer Mode via Dsskey".

To perform an attended transfer:

- Press or tap the **Transfer** soft key during a call.
- **Do one of the following:**
  - Enter the number you want to transfer the call to.
  - Press # SEND to dial out.

  OR
  - Enter the number you want to transfer the call to.
  - Tap **Transfer** soft key.
    The phone will prompt a dialog box.
  - Tap **Send** to dial out. (The call will automatically dial out in about 5 seconds if you do not tap Send.)

  OR
  - Tap and then tap **Local Directory** or **Remote Phone Book** to search the desired contact.
  - Tap the desired contact to dial out.For more information refer to "Remote Phone Book"

OR

- Tap  and then tap the desired list.
- Tap the desired entry to dial out.

- After the party answers the call, press  or tap **Transfer** soft key to complete the transfer.

  If you are using a handset, the transfer can be completed by hanging up the handset.

Call Waiting

Call waiting allows the IP phone to receive a new incoming call when there is already an active call. If call waiting is disabled, another incoming call is automatically rejected by the phone with a busy message.

You can also enable or disable the phone to play a warning tone when receiving another call.You can customize call waiting tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to "Tones".

The call waiting on code and call waiting off code configured on the IP phone are used to activate/deactivate the server-side call waiting feature. They may vary on different servers.

Procedure

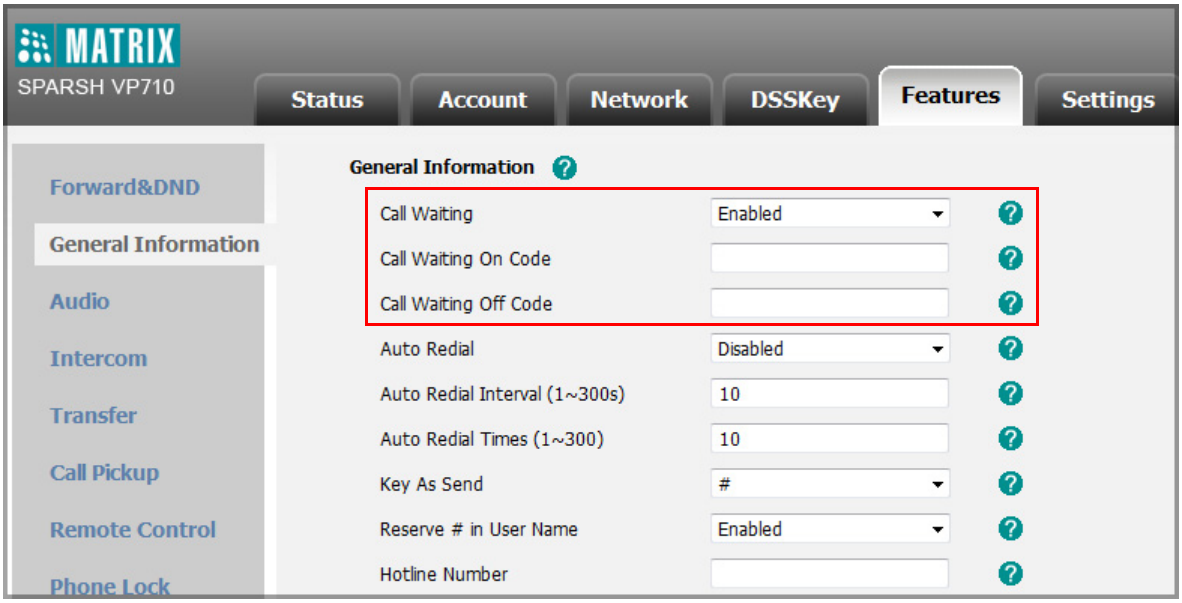
Call waiting and call waiting tone can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure call waiting and call waiting tone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure call waiting. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load Configure call waiting tone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-audio&q=load
	Phone User Interface	Configure call waiting and call waiting tone.

To configure call waiting via web user interface:

- Click on **Features->General Information**.
- Select the desired value from the **Call Waiting** list.
- In the **Call Waiting On Code** field (optional), enter the call waiting on code.

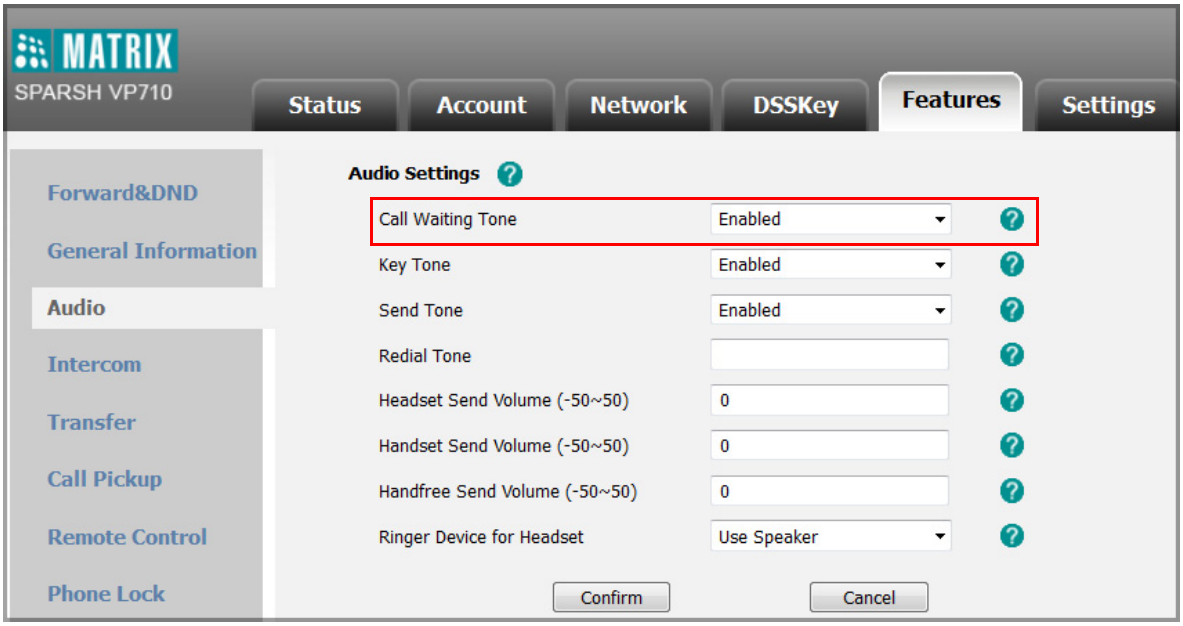
- In the **Call Waiting Off Code** field (optional), enter the call waiting off code.



- Click **Confirm** to save the change.

To configure call waiting tone via web user interface:

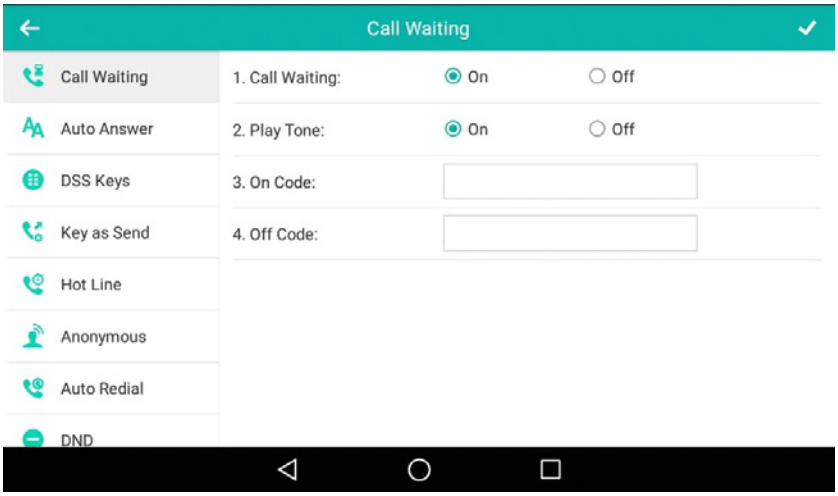
- Click on **Features->Audio**.
- Select the desired value from the **Call Waiting Tone** list.



- Click **Confirm** to save the change.

To configure call waiting and call waiting tone via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Call Waiting.**
- Tap the **On** radio box in the **Call Waiting** field.
- Tap the **On** radio box in the **Play Tone** field.
- In the **On Code** or **Off Code** field (optional), enter the call waiting on code or off code respectively.



- .Tap  to save the change or  to cancel.

Calling Line Identification Presentation (CLIP)

Calling Line Identification Presentation (CLIP) allows the IP phone to display the caller identity, derived from a SIP header contained in the INVITE message when receiving an incoming call.

The IP phone supports deriving caller identity from three types of SIP header:

- From
- P-Asserted-Identity (PAI)
- Remote-Party-ID (RPID)

Identity presentation is based on the identity in the relevant SIP header.
If the caller is present in the local directory, the local contact name assigned to the caller should be preferentially displayed and stored in the call log.

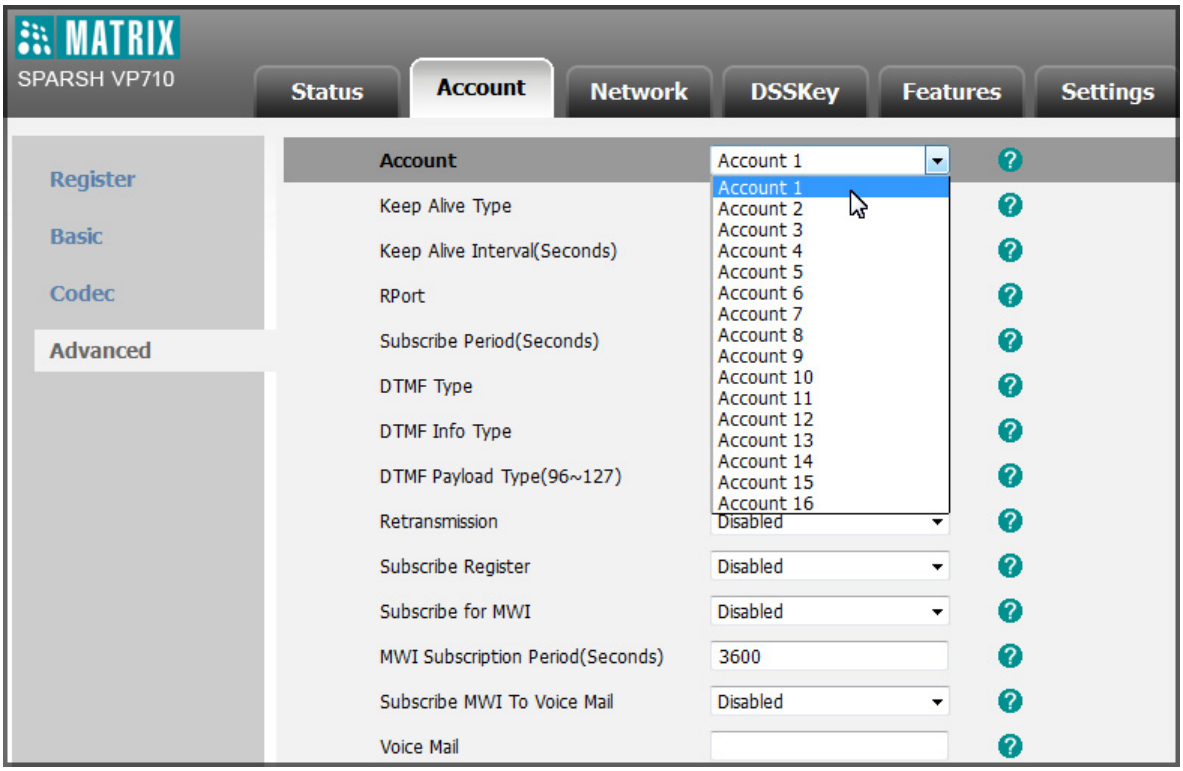
Procedure

CLIP can be configured using the configuration files or locally.

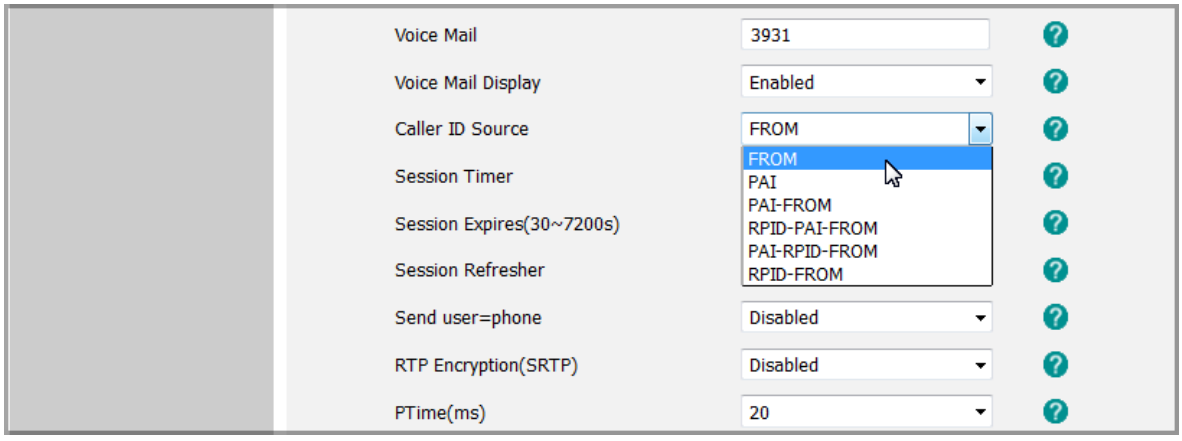
Configuration File	<MAC>.cfg	Configure the presentation of the caller identity. Specify whether to process Privacy header field. Specify whether to process the P-Preferred-Identity (PPI) header for caller identity presentation. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the presentation of the caller identity. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure the presentation of the caller identity via web user interface:

- Click on **Account-> Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Caller ID Source** list.



- Click **Confirm** to save the change.

Connected Line Identification Presentation (COLP)

Connected Line Identification Presentation (COLP) also known as Called line identification presentation allows the IP phone to display the identity of the connected party specified for outgoing calls.

IP phones can display the Dialed Digits, or the identity in a SIP header (Remote-Party-ID or P-Asserted-Identity) received, or the identity in the “From” header carried in the UPDATE message sent by the Callee as described in RFC 4916
In some cases, the remote party will be different from the called line identification presentation due to call diversion.

If the Callee is present in the local directory, the local contact name assigned to the Callee should be preferentially displayed.

Procedure

COLP can be configured only using the configuration files.

Configuration File	<MAC>.cfg	Configure the presentation of the callee’s identity. Specify whether to process Privacy header field. For more information, refer “Appendix D - Configuration Parameters”.

Conference

Conference is a call in which you can talk to more than two persons simultaneously. You can merge two separate calls to create a conference.

During a conference, you can manage each participant.You can mute, hold and remove individual participants in the video call screen. For details refer to "Video Call Screen Display".

You can create a conference with other parties using the phone's local conference or network conference.

Local Conference

In Local conference, you can create up to three-way video conference call and five-way audio-only and video mixed conference. The audio-only and video mixed conference supports five parties (including you), in which you can include a maximum of three video calls.

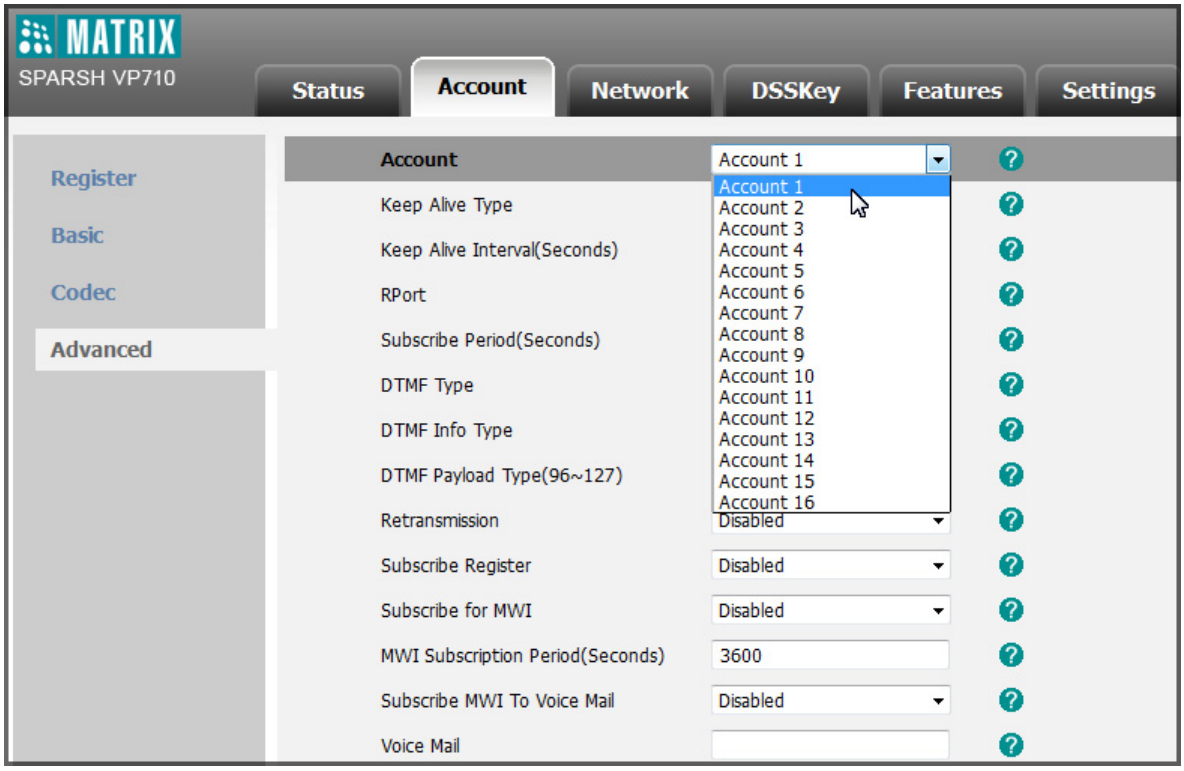
Procedure

Local conference can be configured using the configuration files or locally.

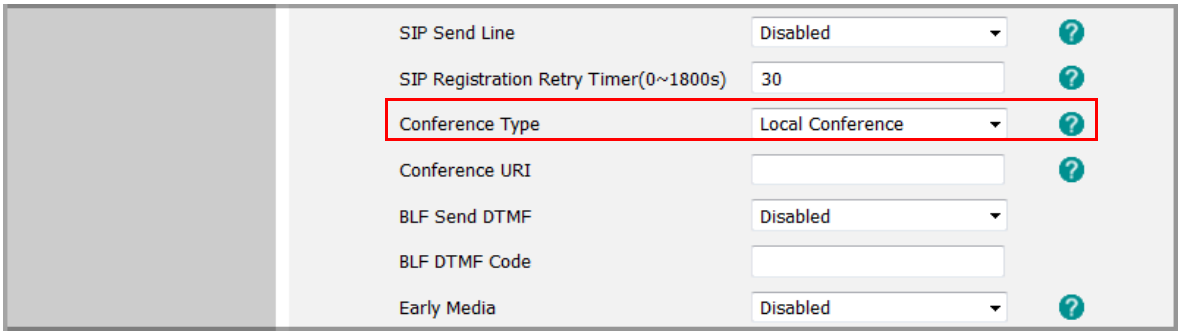
Configuration File	<MAC>.cfg	Configure local conference. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure local conference. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure the local conference via web user interface:

- Click on **Account->Advanced.**
- Select the desired account from the **Account** list.



- Select **Local Conference** from the **Conference Type** list.

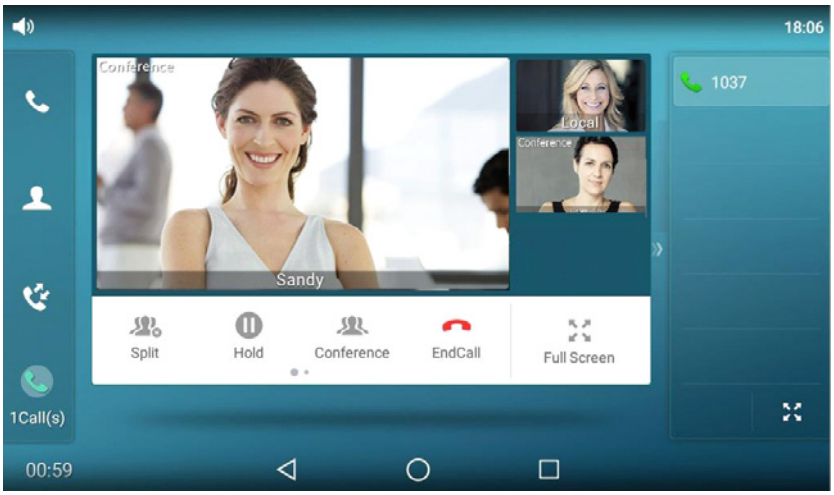


- Click **Confirm** to save the change.

To set up a local conference call:

- Place a call to the first party.
- When the first party answers the call, tap the **Conference** soft key to place a new call.The active call is placed on hold.
- Enter the number of the second party and then press #SEND or tap **Conference**.

- When the second party answers the call, tap the **Conference** soft key on the second party's call screen to join all parties in the conference.

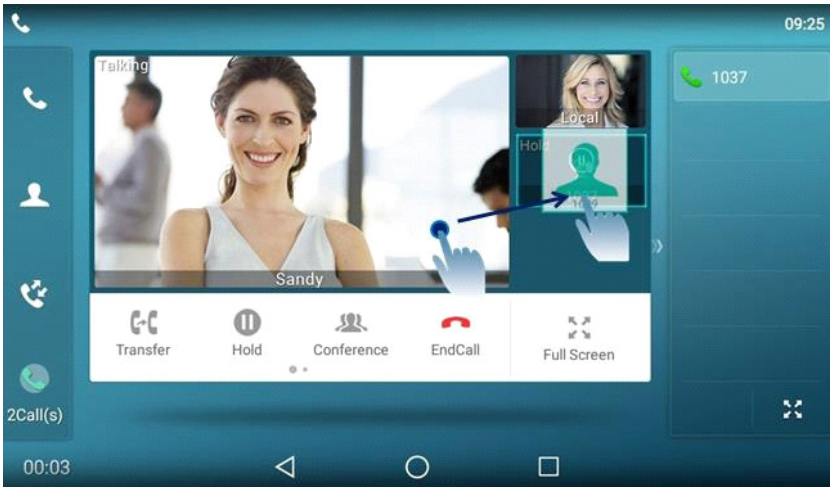


- Repeat the preceding three steps until you have added all intended parties.

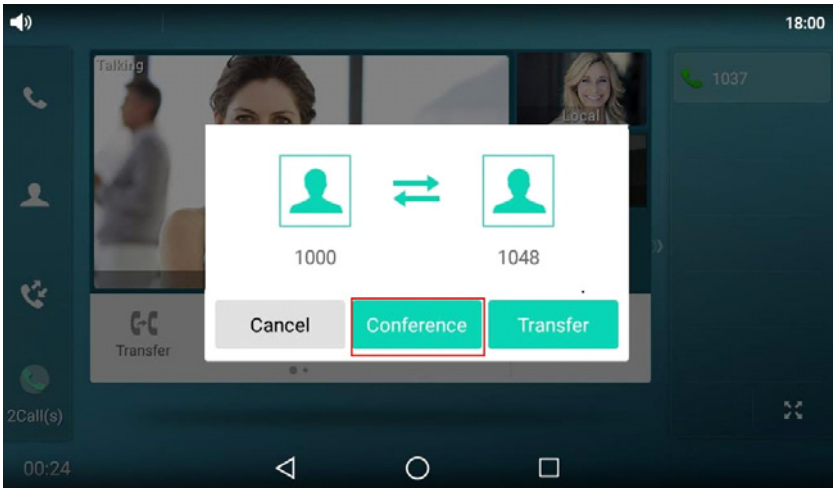
The phone will enter the video call full-screen within 5 seconds during a call, you may tap ◁ or blank area of the touch screen to see the soft keys.

To join calls in a conference:

- Make sure you have an active call and one or more calls are on hold (either on the same line key or a different line key).
- Do one of the following:
  - Drag one far-site window to another far-site window.



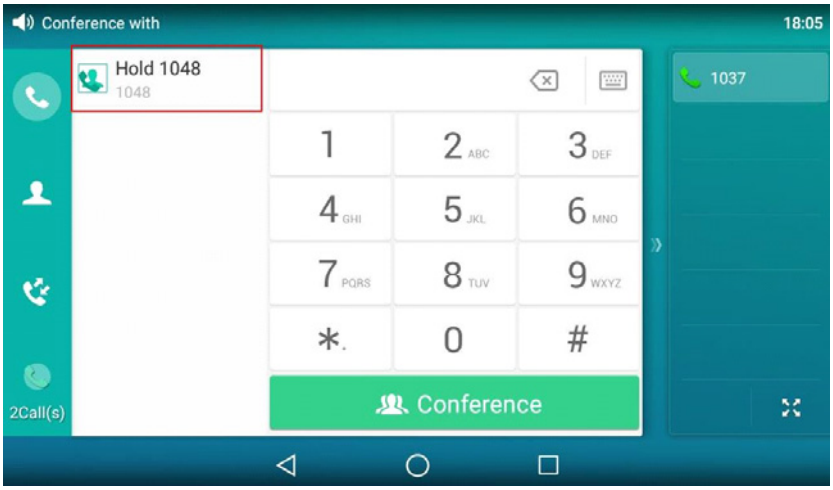
- The touch screen prompts the following:



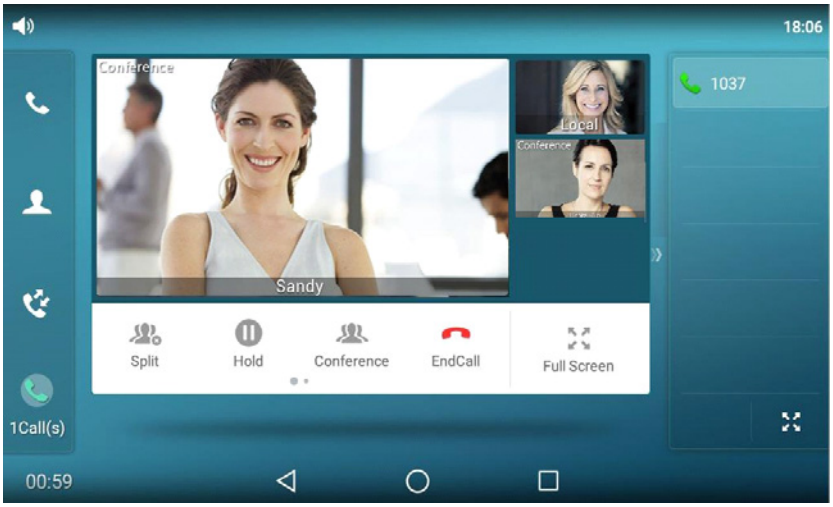
- Tap **Conference**.

OR

- Tap the desired call for conference and ensure that the call is active.
- Tap the **Conference** soft key.



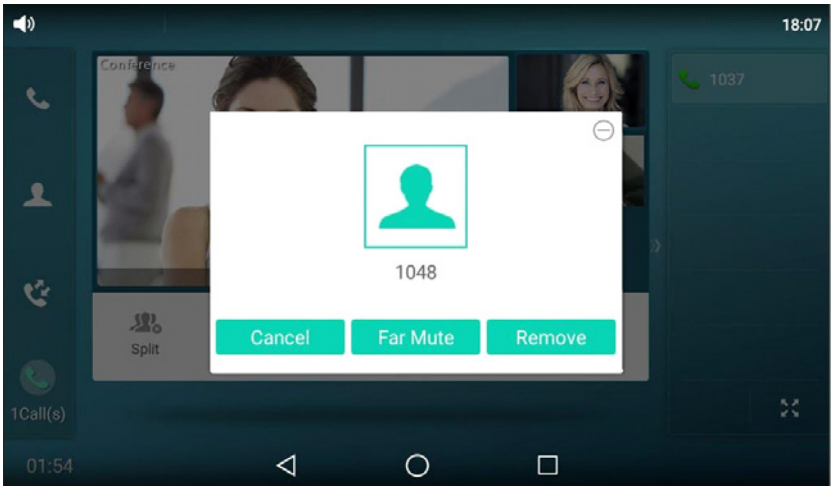
- Tap the Hold call to join the calls in the conference.



- Repeat the step 2 to join more calls in the conference.

During the conference call, you can do the following:

- Press or tap the **Hold** soft key to place the conference call on hold.
- Tap the **Conference** soft key to place a new call.
- Tap the **Split** soft key to split the conference call into individual calls on hold.
- Press to mute the conference call, all other participants can hear each other, but they cannot hear you.
- Tap the **EndCall** soft key to drop the conference call.
- Long tap the desired party (if the far site supports audio calls only, you can just tap it), the phone will prompt a dialog box as the following picture shows:



You can do the following:

- Tap **Far Mute** to mute the party. The muted party can hear everyone, but no one can hear the muted party.

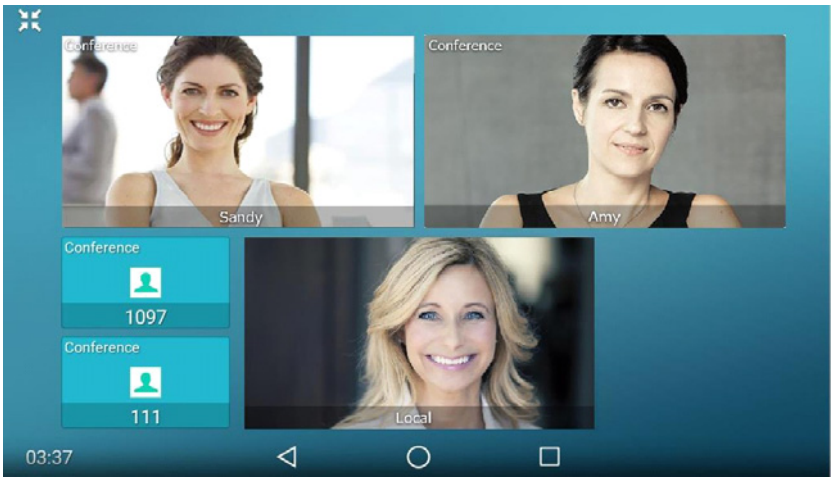
OR

- Tap **Remove** to remove the party.

OR

- Tap **Cancel** or to close the dialog box.

The following screen shows a mixed conference call with three-way video calls and two-way audio-only calls.



Network Conference

Network conference, also known as centralized conference, provides you the flexibility of calling multiple participants (more than three).You can use network conference on the IP phone to conduct an audio-only/video/ audio-only and video mixed conference with multiple participants.

This feature allows you to perform the following:

- Join two calls together into a conference call.
- Invite another party into an active conference call.

- *Network conference is not available on all servers. For more information, contact your system administrator.*
- *IP phone implement network conference using the REFER method specified in RFC 4579.*
- *To use this feature, contact your system administrator for the network conference URI in advance*.

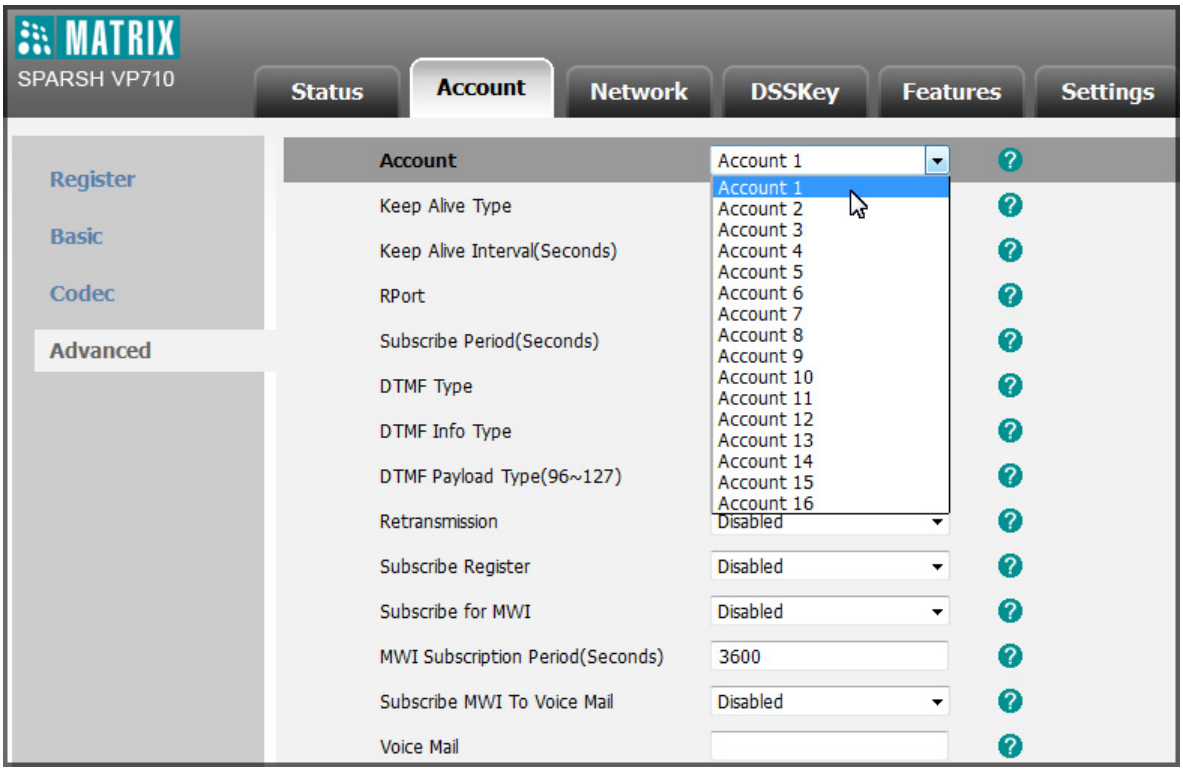
Procedure

Network conference can be configured using the configuration files or locally.

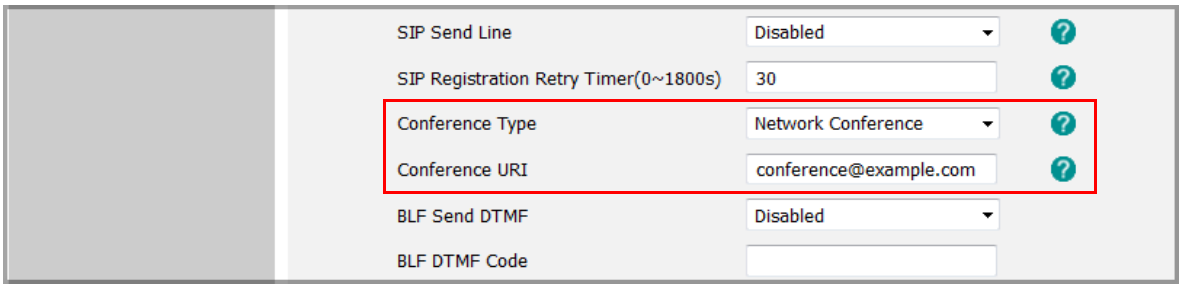
Configuration File	<MAC>.cfg	Configure network conference. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure network conference. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure network conference via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select **Network Conference** from the **Conference Type** list.
- Enter the conference URI (e.g., conference@example.com) in the **Conference URI** field.



- Click **Confirm** to save the change.

Network conference is configurable via web user interface only.

To set up a network conference call:

- Place a call to the first party.
- Tap the **Conference** soft key to place a new call.The active call is placed on hold.
- Enter the number of the second party and press #SEND or tap **Conference**.

- When the second party answers the call, tap the **Conference** soft key to add the second party to the conference.
- Tap the **Conference** soft key to place a new call.The conference call is placed on hold.
- Repeat the preceding three steps until you have added all intended parties.

  The procedures to set up a network conference call for specific servers may be different from introduced above.

Transfer on Conference Hang Up

Transfer on conference hang up feature allows the other two parties to remain connected when the conference initiator drops the conference call

In a local conference, the calls of all the parties are dropped, when the conference initiator drops the call.

Network conference does not have a conference initiator, so if any party exits the network conference, the remaining parties are still connected.For more information on network conference, refer to "Network Conference".

Procedure

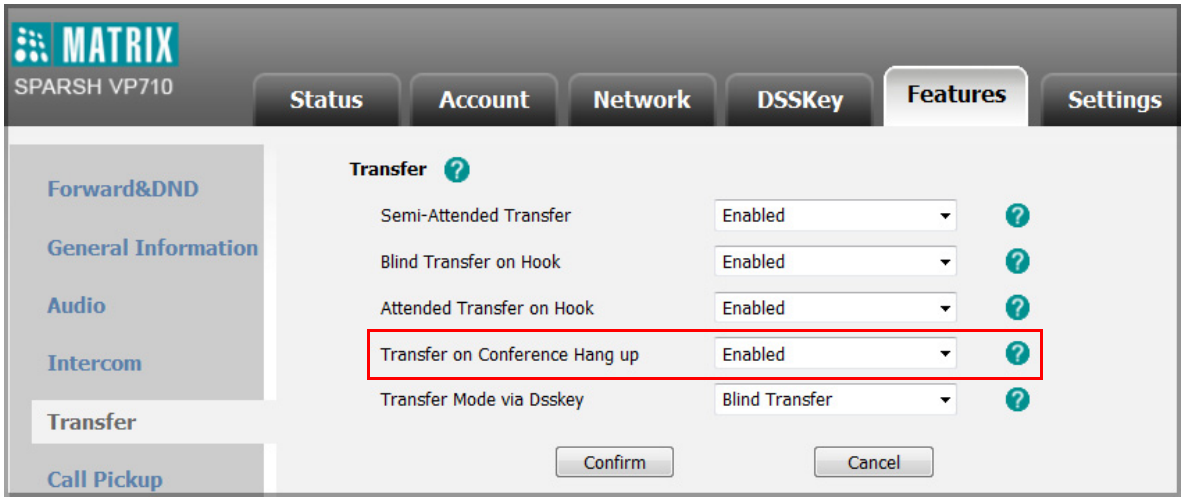
Transfer on conference hang up can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transfer on conference hang up. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the transfer on conference hang up. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-transfer&q=load

To configure Transfer on conference hang up via web user interface:

- Click on **Features->Transfer**.

- Select the desired value from the **Transfer on Conference Hang up** list.



- Click **Confirm** to save the change.

Door Phone

Door phone is used to control the opening of the door giving access to any kind of buildings, offices, or apartment blocks. You can connect the door phone to one or multiple IP phones and control the opening of the door on the IP phone.The IP phone is only compatible with the door phone of 2N. For more information on how to connect the door phone and IP phone, refer to the documentation from the 2N.

You can configure up to two door phones on the IP phone. You can also configure call settings (transmit audio/ video or not, display the call in full-screen mode) for the door phone call.

Procedure

Door phone settings can be configured using the configuration files or locally.

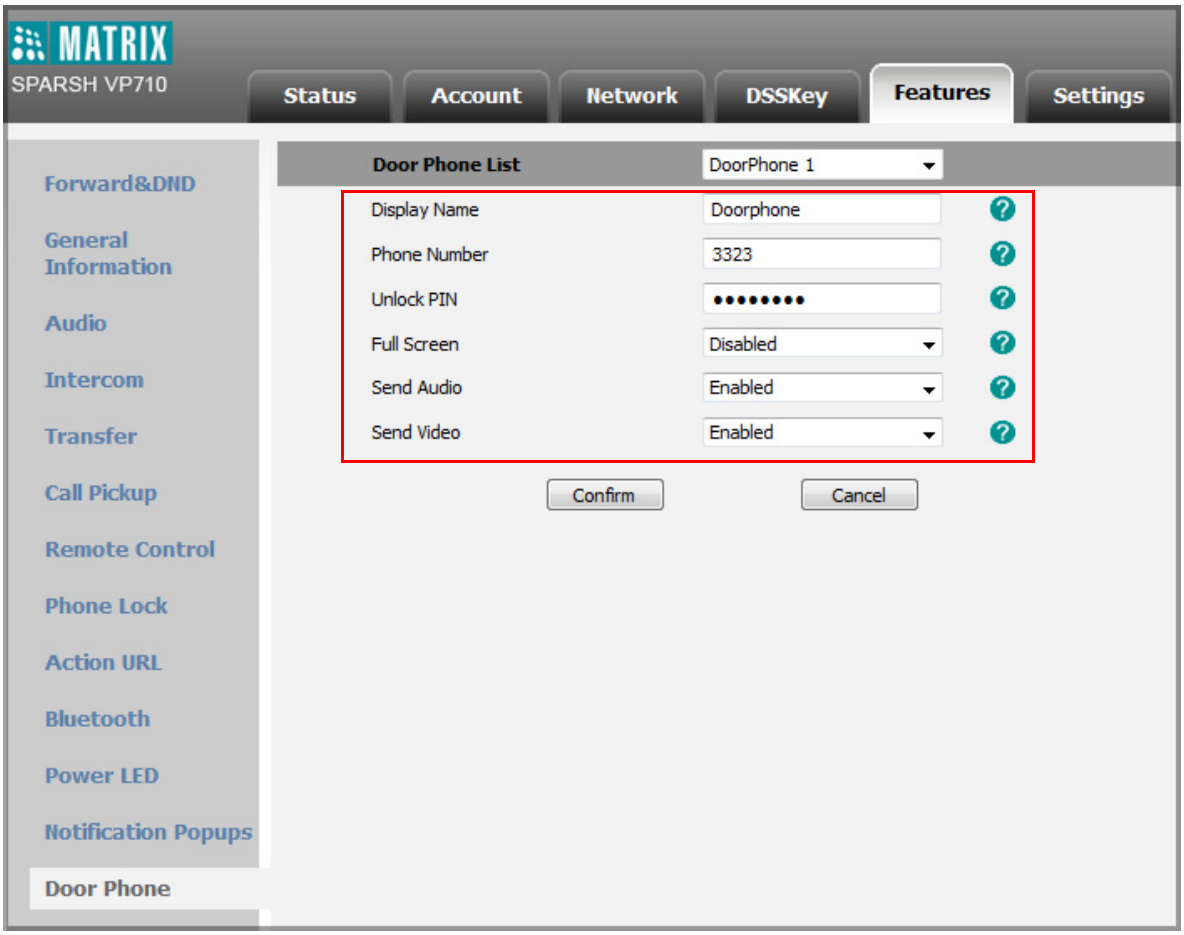
Configuration File	<MAC>.cfg	Configure the door phone information. Configure the unlock PIN of the door phone. Configure the call settings for door phone call. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the door phone information. Configure the unlock PIN of the door phone. Configure the call settings between the door phone and IP phone. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-doorphone&q=load

To configure door phone feature via web user interface:

- Click on **Feature->Door Phone**.
- Select the desired door phone from the **Door Phone** List.
- Enter the display name of the door phone in the **Display Name** field.
- Enter the number of the door phone in the **Phone Number** field.

  If you leave it blank or enter the wrong number, the IP phone will take the incoming door phone call as a normal call.
- Enter the unlock PIN in the **Unlock PIN** field.
- Select the desired value from the **Full Screen** list.
- Select the desired value from the **Send Audio** list.

- Select the desired value from the **Send Video** list.



- Click **Confirm** to save the change.

Do Not Disturb (DND)

You can use DND to reject incoming calls automatically on the phone. Callers will receive a busy message.All calls you receive when DND is enabled are logged in the missed calls list. You can receive incoming calls from authorized numbers when DND is enabled.

DND feature can be configured on a phone or a per-line basis depending on the DND mode.

Two DND modes:

- Phone (default): DND feature is effective for the IP phone.
- Custom: DND feature can be configured for each or all accounts.

A user can activate or deactivate DND using the DND key on the phone. The server-side DND feature disables the local DND and call forward settings. If the server-side DND feature is enabled on any of the IP phone's registrations, the other registrations are not affected.

The DND on code and DND off code configured on IP phones are used to activate/deactivate the server-side DND feature. They may vary on different servers.

Return Message When DND

This feature defines the return code and the reason of the SIP response message for the rejected incoming call when DND is enabled on the IP phone. The caller's phone touch screen displays the received return code.

DND Emergency

This feature allows users to receive the incoming calls from some authorized numbers even if the DND feature is enabled. This feature is disabled by default.

Procedure

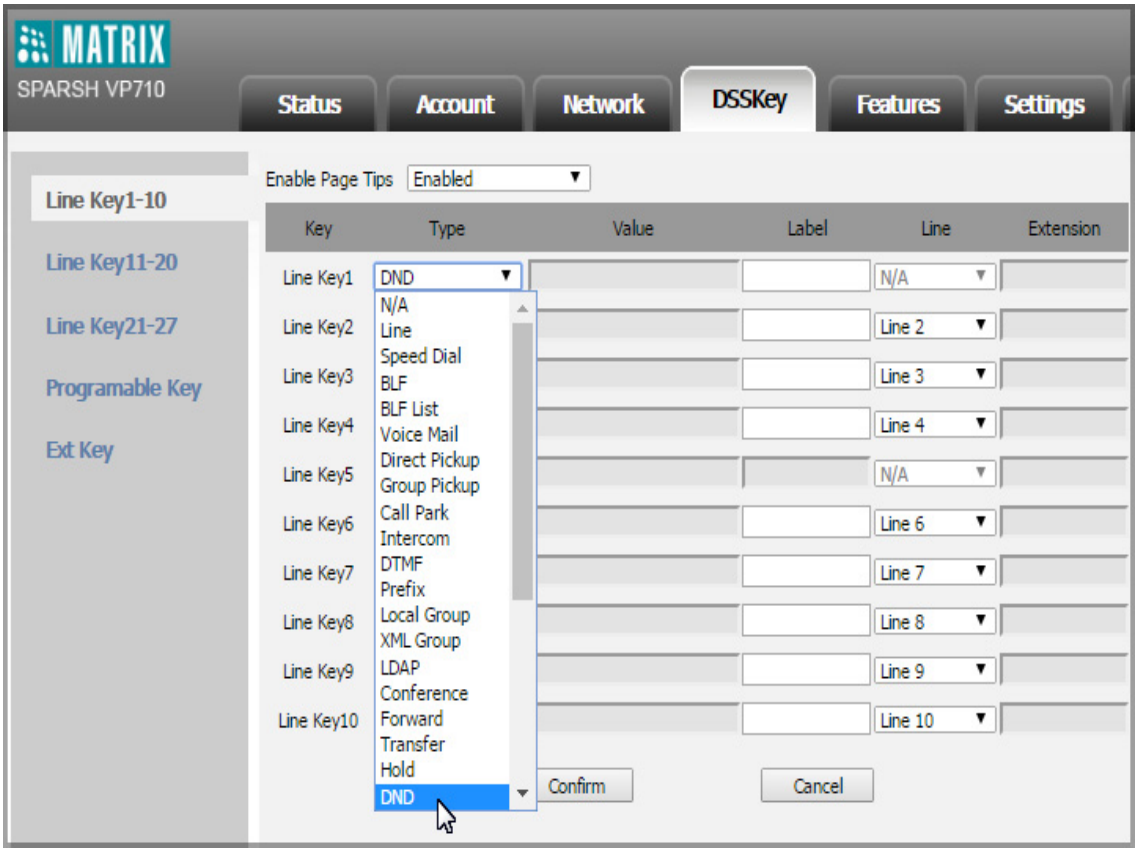
DND can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Assign a DND key. Configure the DND mode. Configure DND in the phone mode. Configure DND in the custom mode. Specify the authorized numbers when DND is enabled. Specify the return code and the reason of the SIP response message when DND is enabled. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Assign a DND key. **Navigate to**: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Configure DND. **Navigate to**: http://<phoneIPAddress>/servlet?p=features-forward&q=load Specify the return code and the reason of the SIP response message when DND is enabled. **Navigate to**: http://<phoneIPAddress>/servlet?p=features-general&q=load
	Phone User Interface	Configure DND.

To configure a DND key via web user interface:

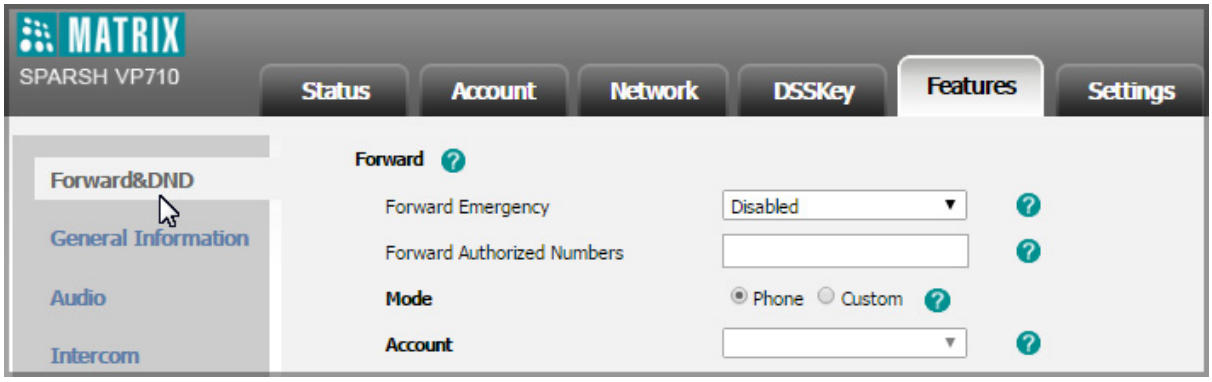
- Click on **DSSKey->Line Key (or Programable Key)**.
- In the desired DSS key field, select **DND** from the **Type** list.
- Enter the string that will appear on the touch screen in the **Label** field.



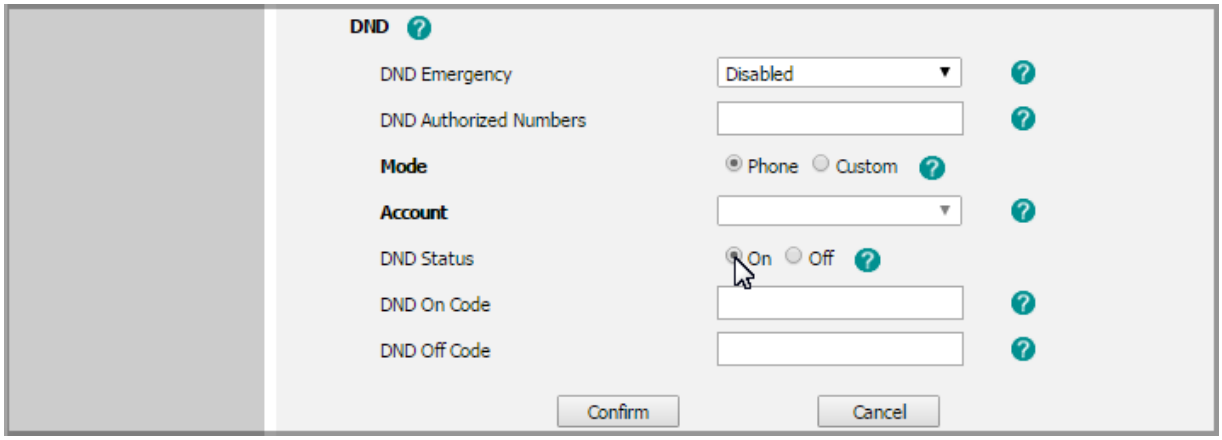
- Click **Confirm** to save the change.

To configure DND feature via web user interface:

- Click on **Features->Forward&DND**.



- Click the desired option in **DND Status**.

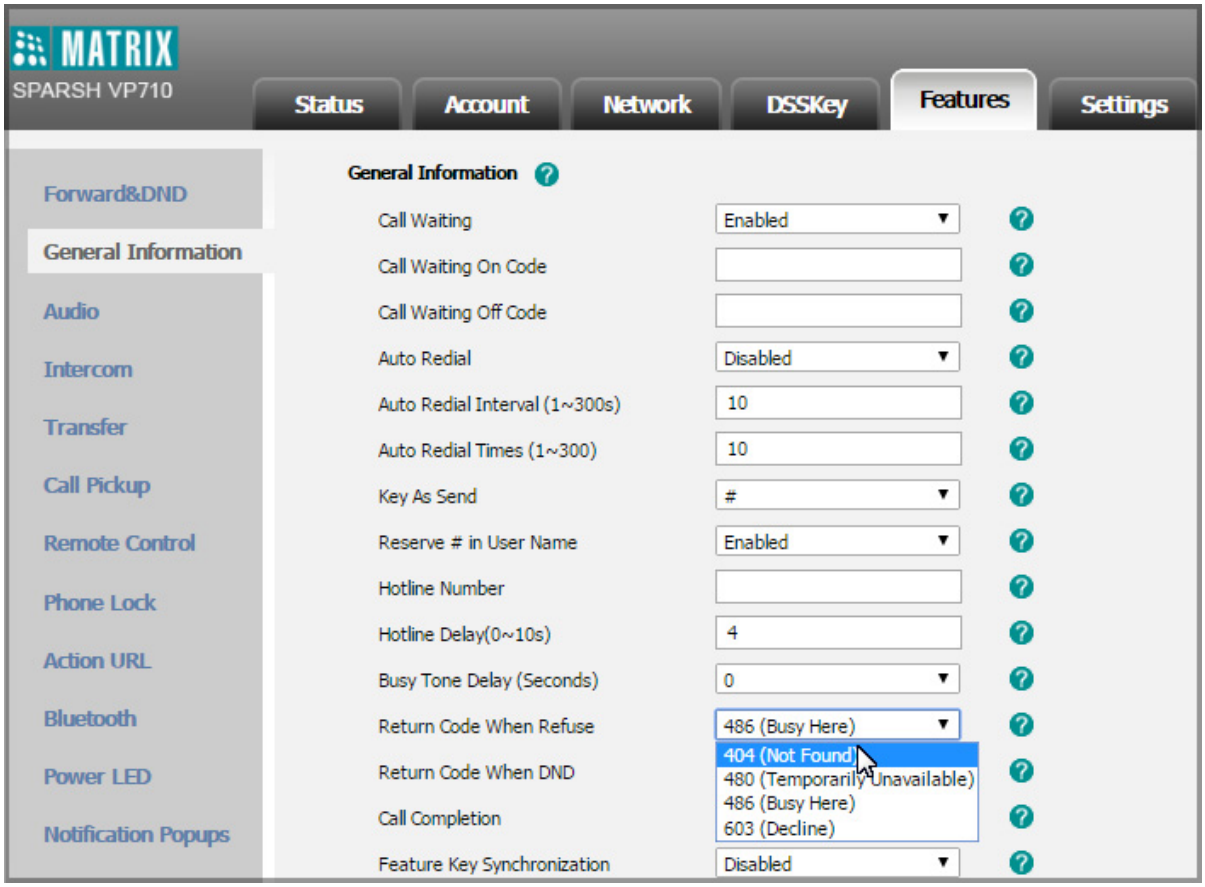


- Enter the DND on code in the **DND On Code** field (optional).
- Enter the DND off code in the **DND Off Code** field (optional).
- Click **Confirm** to save the change.

To specify the return code and the reason when DND is enabled via web user interface:

- Click on **Features->General Information**.

- Select the desired type from the **Return Code When DND** list.

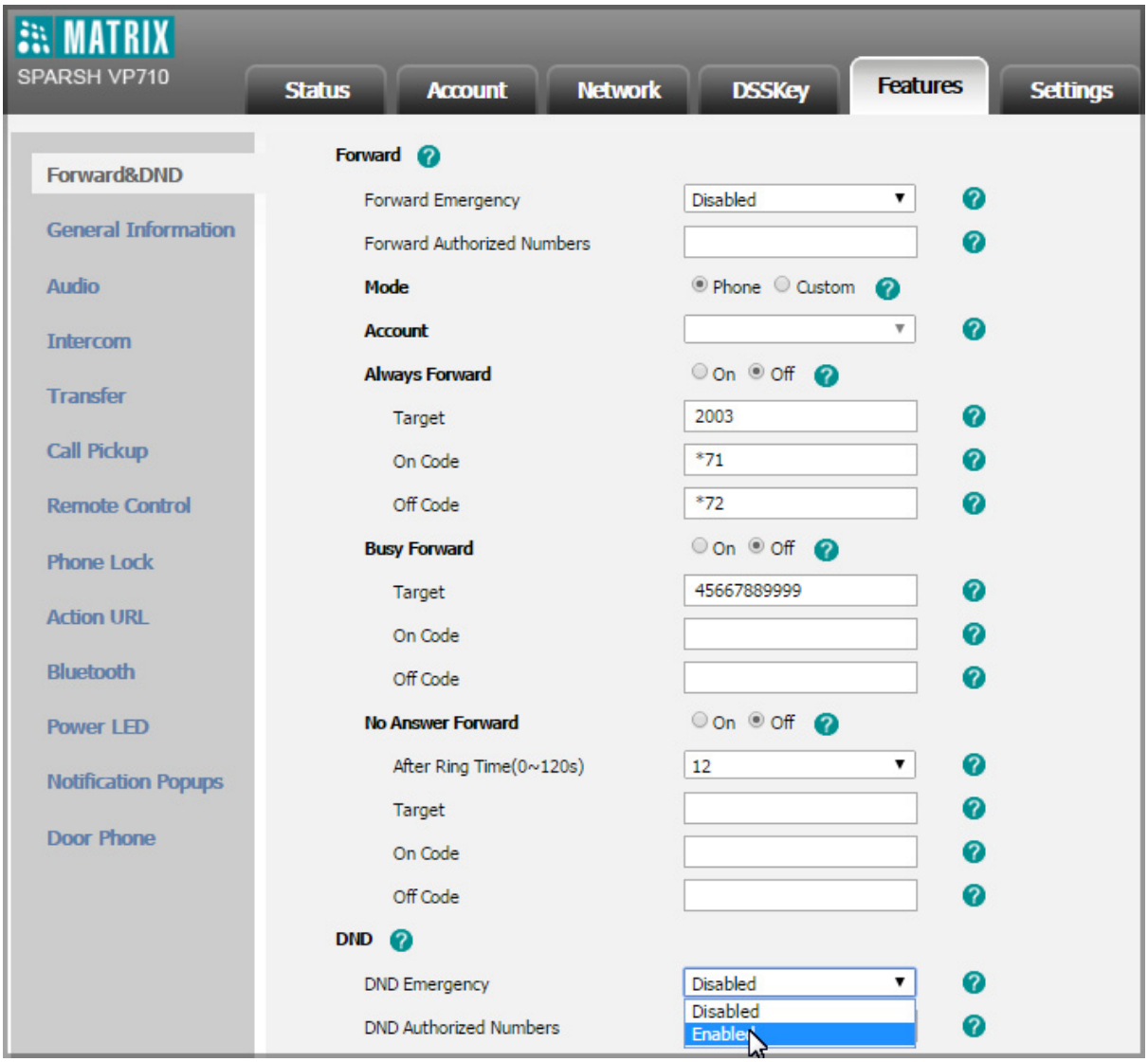


- Click **Confirm** to save the change.

To specify the authorized numbers when DND is enabled via web user interface:

- Click on **Features->General Information.**
- Select the desired value from the DND Emergency list.

- Enter the desired value in the DND Authorized Numbers field. Multiple numbers are separated by commas.



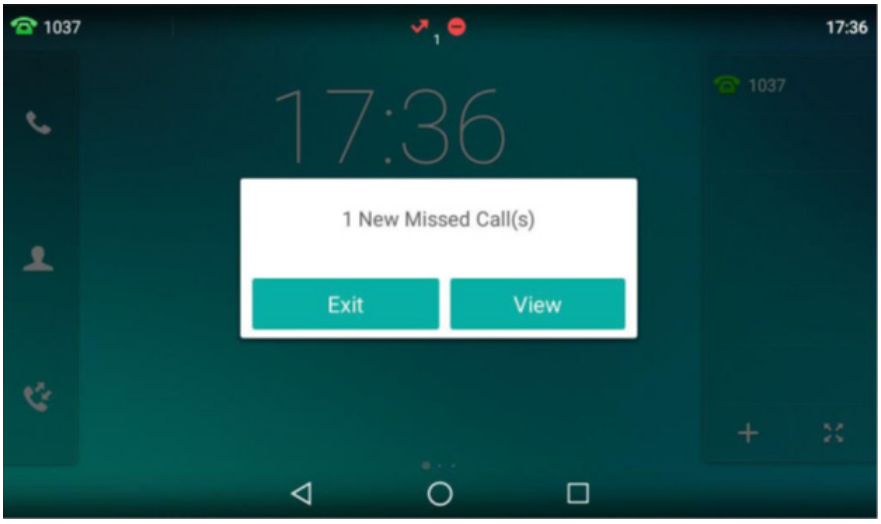
- Click **Confirm** to save the change.

To enable DND in phone mode:

- Swipe down from the top of the screen to enter the control center and toggle DND on.

  The icon on the status bar indicates that DND is enabled.

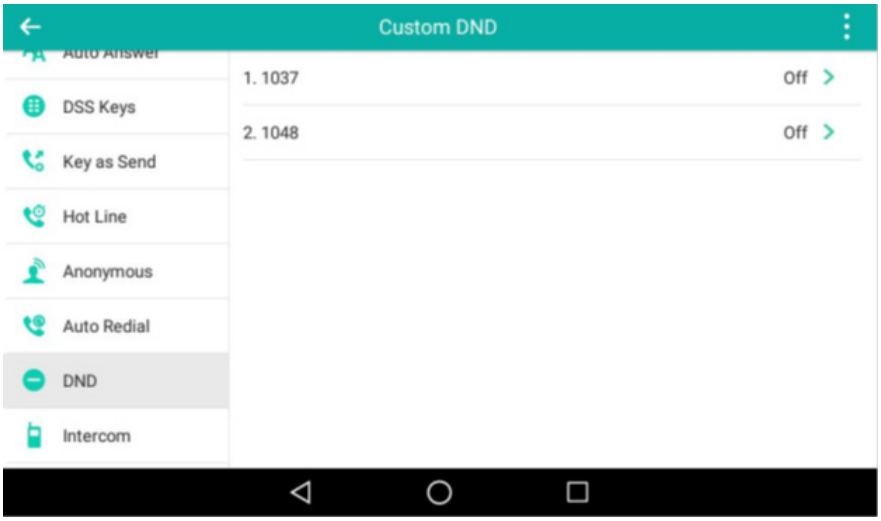
Incoming calls will be rejected automatically and "n New Missed Call(s)" ("n" indicates the number of missed calls. e.g., 1 New Missed Call(s)) will appear on the touch screen.



To enable DND in custom mode for a specific account:

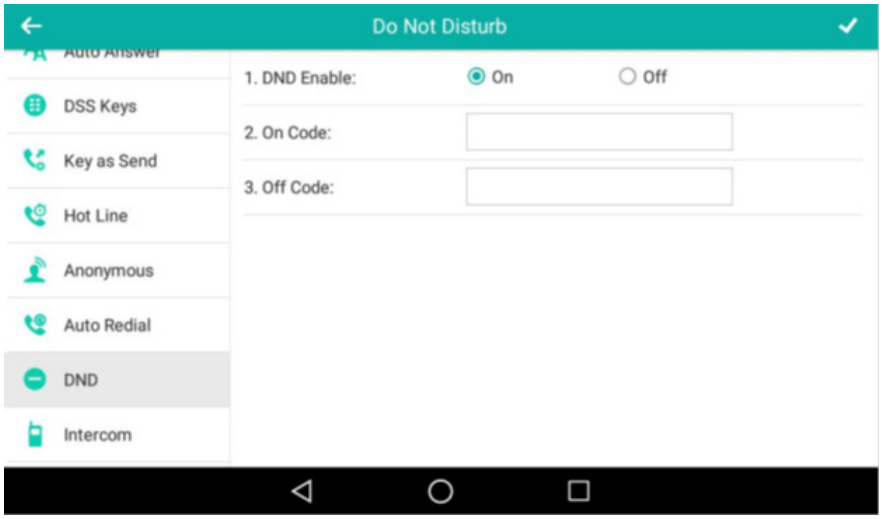
- Swipe down from the top of the screen to enter the control center and then tap DND.

  The touch screen displays a list of accounts registered on the phone.



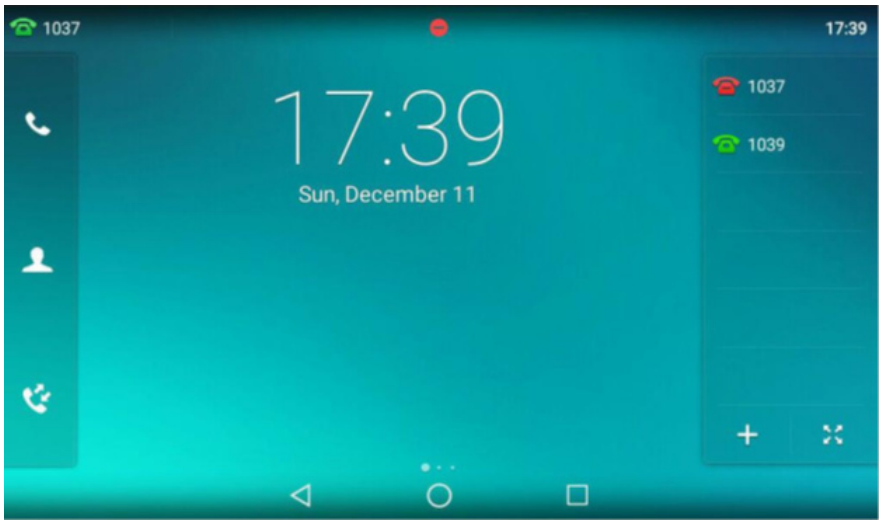
- Tap the desired account.
- Tap the **On** radio box in the **Status** field.

- Enter the DND on code or off code respectively in the **On Code** or **Off Code** field.



- Tap ✓ to save the change or ◁ to cancel.

If you enable DND for the default account, the associated line icon will change to ☎, and the icon ⊖ will appear on the status bar.



- If you enable DND for the non-default account, only the associated line icon will change to the icon will not appear on the status bar.



- If you enable DND for all accounts, all line icons will change to and the icon will appear on the status bar.



To disable DND in phone mode:

- Swipe down from the top of the screen to enter the control center and then tap **DND**.

To disable DND in custom mode:

- Swipe down from the top of the screen to enter the control center and then tap **DND**.
- Tap the desired account.
- Tap the **Off** radio box of the **DND Enable** field.

- Tap ✓ to save the change or ◁ to cancel.

DND authorized number is configurable via web user interface only.

Feature Key Synchronization

Feature key synchronization provides the capability to synchronize the status of the following features between the IP phone and the server:

- Do Not Disturb (DND)
- Call Forwarding Always (CFA)
- Call Forwarding Busy (CFB)
- Call Forwarding No Answer (CFNA)

If feature key synchronization is enabled and you change the status of one of these features on the server then the server notifies the IP phone for synchronizing the status. Conversely, if you change the status of one of these features on the IP phone then the IP phone notifies the server for synchronizing the status.

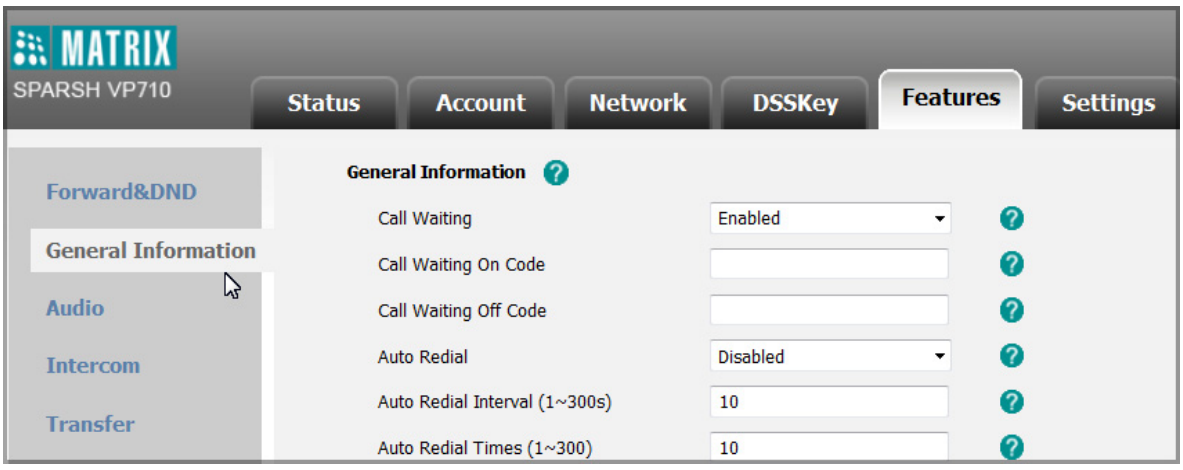
Procedure

Feature key synchronization can be configured using the configuration files or locally.

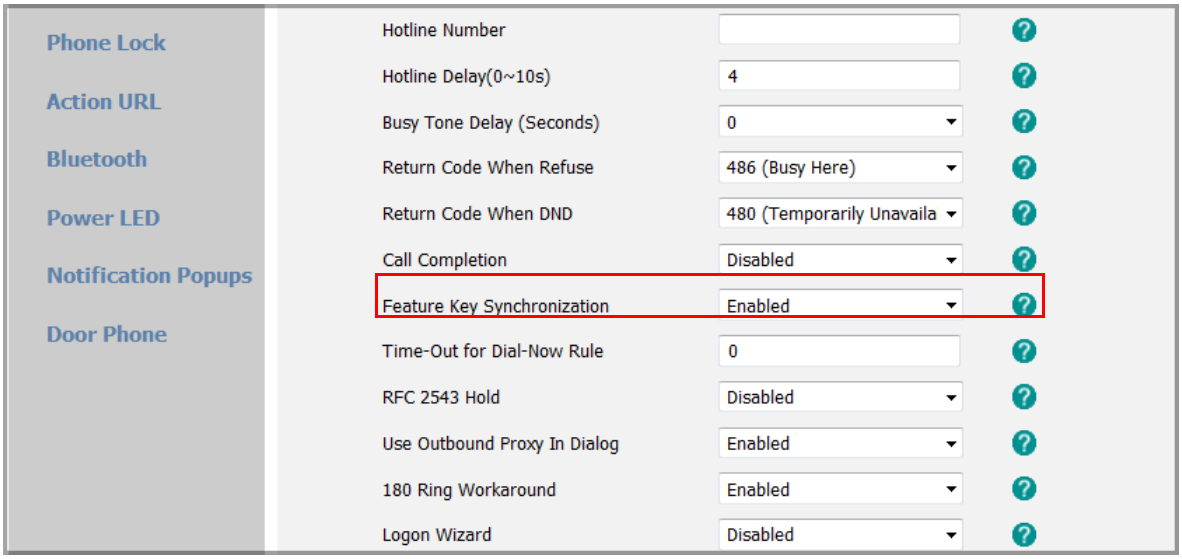
Configuration File	<MAC>.cfg	Configure feature key synchronization. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure feature key synchronization. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure key synchronization via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Feature Key Synchronization** list.



- Click **Confirm** to save the change.

Hide Feature Access Codes

This feature enables the IP phone to display the feature name instead of the dialed feature access code automatically.For example, the dialed call park code will be replaced by the identifier “Call Park” when you park an active call.

The feature is applicable to the following features:

- Voice Mail
- Pick up
- Group Pick up
- Barge In
- Retrieve
- Call Park
- Call Pull

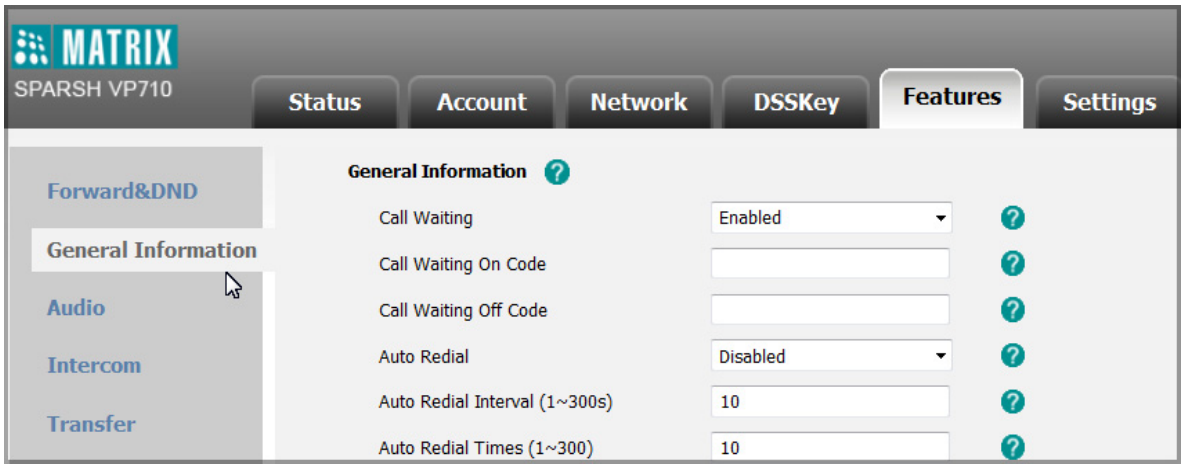
Procedure

Hide feature access codes feature can be configured using the configuration files or locally.

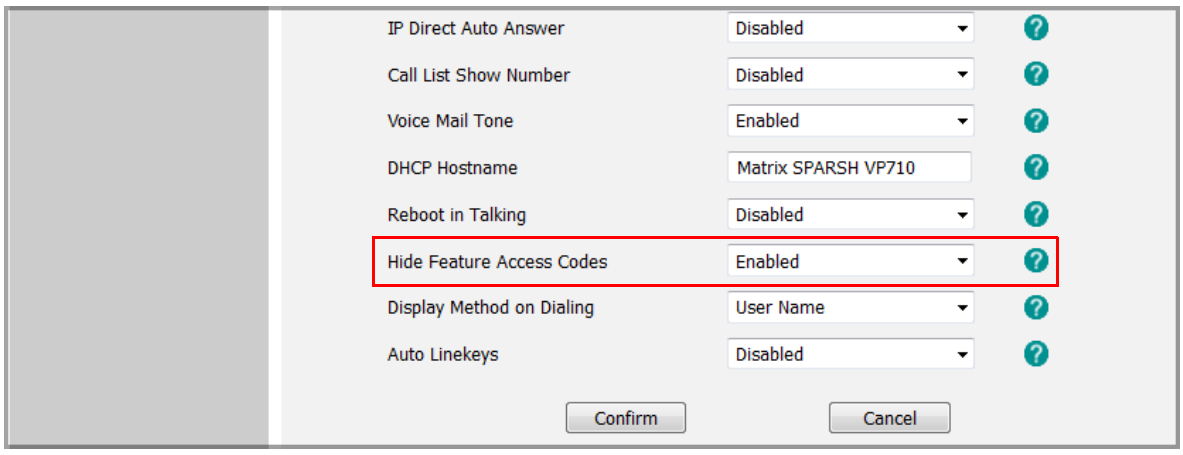
Configuration File	<MAC>.cfg	Configure the hide feature access codes feature. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Configure the hide feature access codes feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure hide feature access codes feature via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Hide Feature Access Codes** list.



- Click **Confirm** to save the change.

Intercom

Intercom allows establishing an audio conversation directly.The IP phone can answer intercom calls automatically. You can tap the intercom key to automatically connect with a preconfigured target extension for outgoing intercom calls, and the target extension will automatically answer incoming intercom calls by default.

You can also use intercom key to monitor the specific extension for status changes on the IP phone.
IP phone supports this feature using a SUBSCRIBE/NOTIFY mechanism as specified in RFC 3265. This feature depends on support from a SIP server

For example:

An intercom key is configured on a supervisor's phone to monitor the specific IP phone user status (busy or idle). When the monitored user places a call, a busy indicator on the supervisor's phone indicates that the user's phone is in use.
When the monitored user is idle, the supervisor can tap the intercom key to automatically connect with a pre configured target extension for outgoing intercom calls.
When the monitored user receives an incoming call, the supervisor can tap the intercom key to pick up the call directly.

Icon Indicator (configured as an intercom key)

Icons	Description
	Intercom idle state
	Intercom ringing state
	Intercom callout state
	Intercom talking state
	Intercom failed state

.

Intercom is not available on all servers. Contact your system administrator for more information.

Outgoing Intercom Calls

Intercom is a useful feature in office environments to quickly connect with an operator or secretary. You can tap an intercom key to automatically initiate an outgoing intercom call with a remote extension.

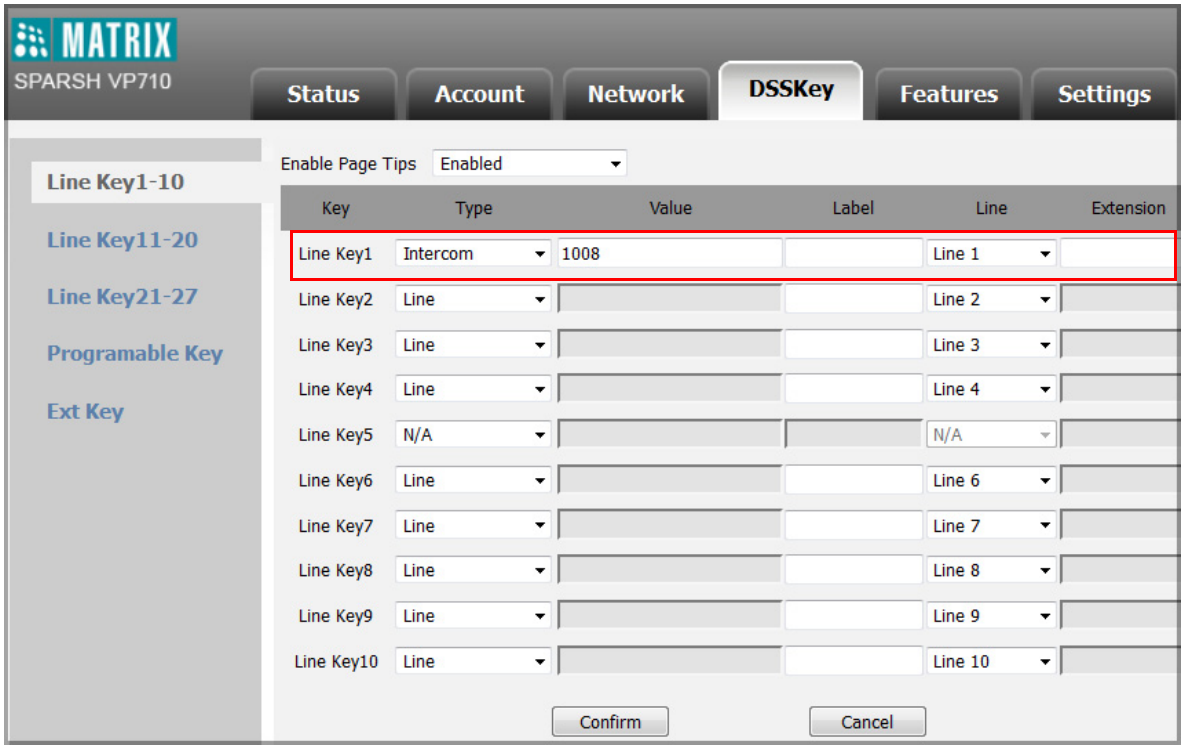
Procedure

Intercom can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the intercom subscription. Assign an intercom key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Assign an intercom key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign an intercom key.

To configure an intercom key via web user interface:

- Click on **DSS Key-> Line Key**.
- In the desired DSS key field, select **Intercom** from the **Type** list.
- Enter the remote extension number in the **Value** field.



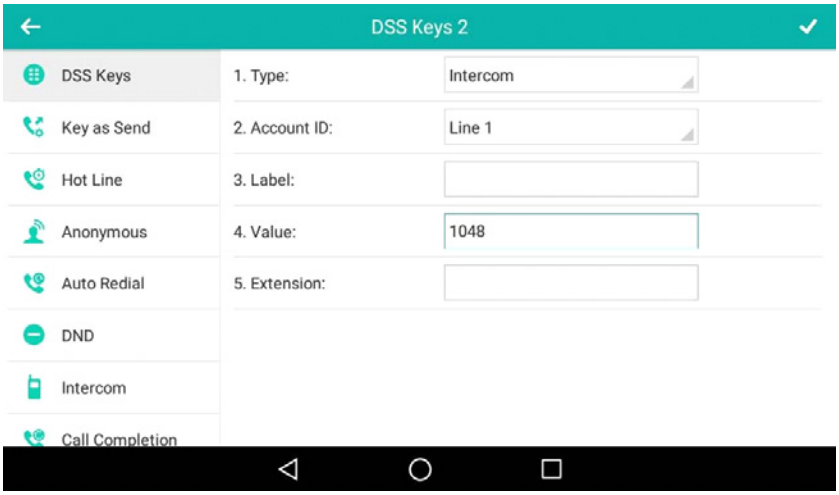
- In the **Label** field (optional), enter the string that will appear on the touch screen.
- In the **Extension** field (optional), enter the directed call pickup code.

- Select the desired line from the **Line** list**.**
- Click **Confirm** to save the change.

To configure an intercom key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired line key.
- Tap the **Type** field.
- Tap **Intercom** in the pop-up dialog box.
- Tap the **Account ID** field.
- Tap the desired line in the pop-up dialog box.
- In the Label field (optional), enter the string that will appear on the touch screen.
- Enter the target extension number in the **Value** field.
- In the **Extension** field (optional), enter the directed call pickup code.

  If it is configured, when the target extension receives an incoming call, you can tap the intercom key to pick up the call directly. Contact your system administrator for more information.



- .Tap  to save the change or  to cancel.

Incoming Intercom Calls

The IP phone supports automatically answering an incoming intercom call by default. The phone automatically plays a warning tone when it receives an incoming intercom call. In addition, you can enable the phone to mute the microphone when it automatically answers an incoming intercom call. You can also enable the phone to automatically answer an incoming intercom call while there is already an active call on the phone, the active call is placed on hold.

The IP phone can process incoming calls differently depending on settings. There are four configuration options for incoming intercom calls:

Intercom Feature	Description
Accept Intercom	Enable or disable the IP phone to automatically answer an incoming intercom call.
Intercom Mute	Enable or disable the microphone on the IP phone for intercom calls.
Intercom Tone	Enable or disable the IP phone to play a warning tone when it receives an incoming intercom call.
Intercom Barge	Enable or disable the IP phone to automatically answer an incoming intercom call while there is already an active call on the phone.

Accept Intercom

You can enable or disable the phone to automatically answer an incoming intercom call. If Accept Intercom is enabled, the phone automatically answers an incoming intercom call. If Accept Intercom is disabled, the phone rejects incoming intercom calls and sends a busy message to the caller.By default, it is enabled.

You can set a delay time before the phone automatically answers intercom calls.Contact your system administrator for more information.

Intercom Mute

You can mute or un-mute the microphone on the phone for intercom calls automatically. If Intercom Mute is enabled, the microphone is muted for intercom calls. If Intercom Mute is disabled, the microphone works for intercom calls.By default, it is disabled.

Intercom Tone

You can enable or disable the phone to play a warning tone when receiving an intercom call. If Intercom Tone is enabled, the phone plays a warning tone before answering the intercom call. If Intercom Tone is disabled, the phone automatically answers the intercom call without warning. By default, it is enabled.

Intercom Barge

You can enable or disable the phone to automatically answer an incoming intercom call while there is already an active call on the phone. If Intercom Barge is enabled, the phone automatically answers the intercom call and places the active call on hold. If Intercom Barge is disabled, the phone handles an incoming intercom call like a waiting call.By default, it is disabled.

To enable the phone to receive a new incoming call when it has an active call, make sure that call waiting feature is enabled on the phone in advance. For more information, refer to “Call Waiting”.

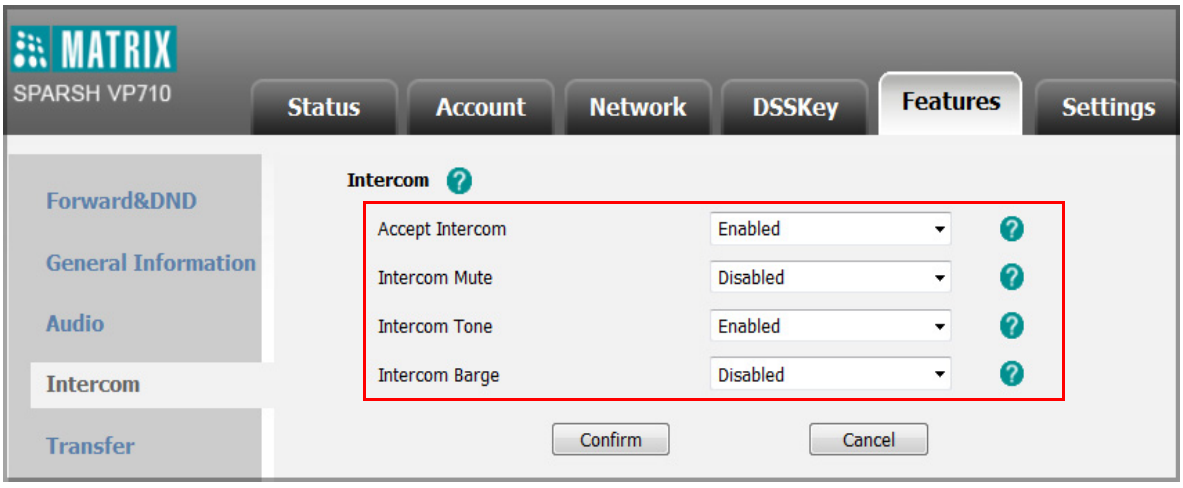
Procedure

Incoming intercom calls can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure incoming intercom call feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure incoming intercom call feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-intercom&q=load
	Phone User Interface	Configure incoming intercom call feature.

To configure intercom via web user interface:

- Click on **Features->Intercom**.
- Select the desired values from the **Accept Intercom**, **Intercom Mute**, **Intercom Tone** and **Intercom Barge** lists.

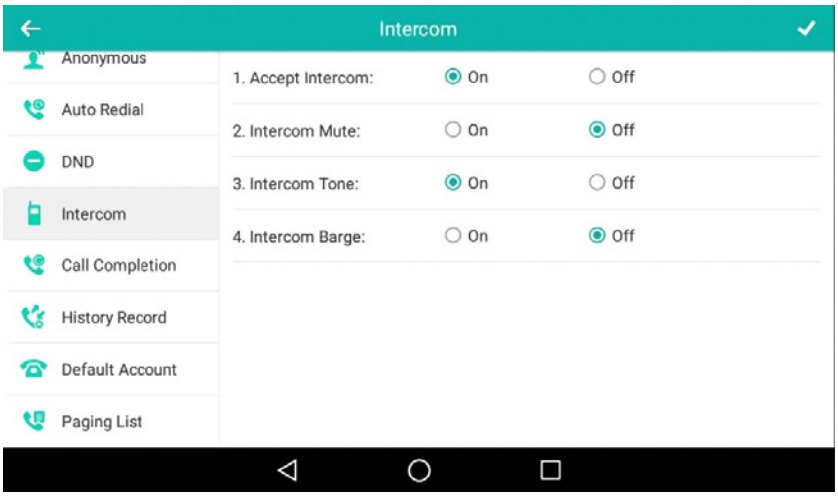


- Click **Confirm** to save the change.

To configure intercom features via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->Intercom**.

- Make the desired changes.



- Tap to save the change or to cancel.

Using Intercom Feature on the Phone

You can use the intercom key in the following ways:

- Place a call to the target extension.
- Pick up an incoming call of the target extension.

Placing an Intercom Call to the Target Extension

To place an intercom call when the target phone is idle:

- Tap the intercom key.

  The target extension plays a warning tone and automatically answers the call in the hands-free (speakerphone) mode by default.

Picking up an Incoming Call of the Target Extension

When the target extension receives an incoming call, the intercom key icon of the target extension will become and solid. If you configure the directed call pickup code in advance, you can pick up the target extension's incoming call by tapping the intercom key. You can configure the directed call pickup code when configuring an intercom key. For more information, refer to "To configure an intercom key via phone user interface:"

To pick up an incoming call when the target phone is ringing:

- Tap the intercom key.

  The incoming call of the target extension is answered on the IP phone.

If the directed call pickup code is not configured, the phone will place a call to the target extension instead of picking up an incoming call of the target extension when you tap the intercom key.

IP Direct Auto Answer

IP direct auto answer allows IP phone to automatically answer an IP address call. IP direct auto answer works only if allow IP call is enabled. For more information on allow IP call, refer to"Allow IP Call".

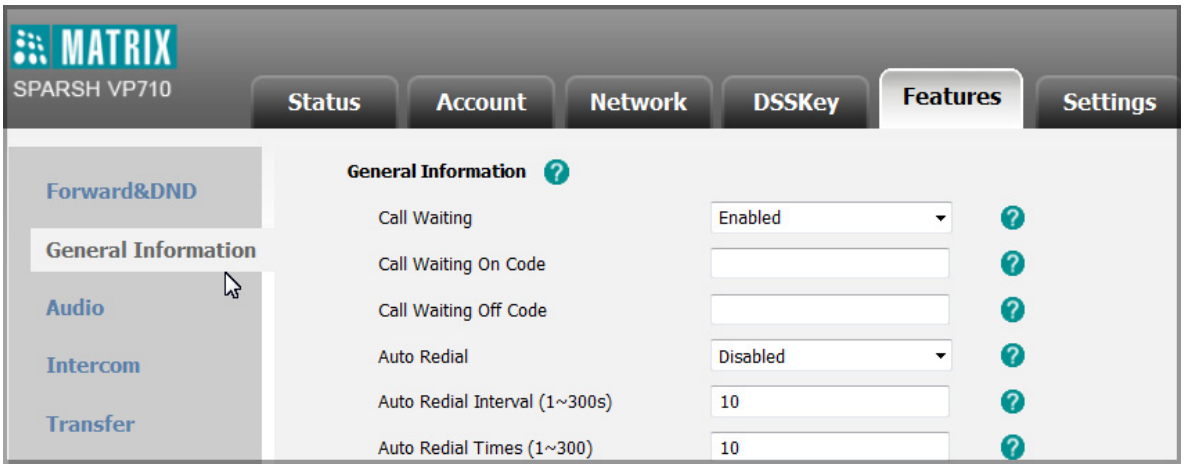
Procedure

IP direct auto answer can be configured using the configuration files or locally.

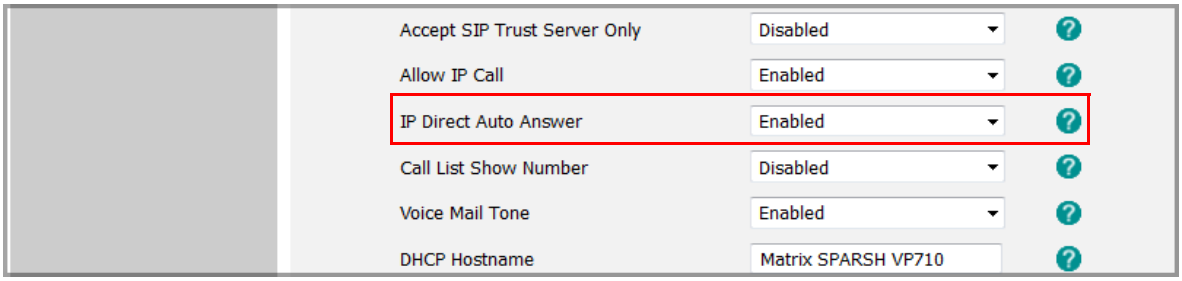
Configuration File	<MAC>.cfg	Configure IP direct auto answer. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure IP direct auto answer. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure IP direct auto answer via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **IP Direct Auto Answer** list.



- Click **Confirm** to save the change.

IPv6 Support

IPv6 is the next generation network layer protocol, designed as a replacement for the current IPv4 protocol. IPv6 uses a 128-bit address, consisting of eight groups of four hexadecimal digits separated by colons, for example, 2026:1234:1:1:215:65ff:fe1f:caa.VoIP network based on IPv6 can provide end-to-end security capabilities, enhanced Quality of Service (QoS), a set of service requirements to deliver performance guarantee while transporting traffic over the network.

If you configure the network settings on the IP phone for an IPv6 network, you can set up an IP address for the phone either by using SLAAC (ICMPv6), DHCPv6 or by manually entering an IP address.



Make sure that your network environment supports IPv6, for more information contact your ISP.

IPv6 Address Assignment Method

Supported IPv6 address assignment methods:

- **Manual Assignment:** An IPv6 address and other configuration parameters (e.g., DNS server) for the IP phone can be statically configured by an administrator.

- **Stateless Address Auto-configuration (SLAAC)/ ICMPv6:** SLAAC is one of the most convenient methods to assign IP addresses to IPv6 nodes. SLAAC requires no manual configuration of the IP phone, minimal (if any) configuration of routers, and no additional servers. To use IPv6 SLAAC, the IP phone must be connected to a network with at least one IPv6 router. This router is configured by the network administrator and sends out Router Advertisement announcements onto the link. These announcements can allow the on-link connected IP phone to configure itself with IPv6 address, as specified in RFC 4862.

- **Stateful DHCPv6:** The Dynamic Host Configuration Protocol for IPv6 (DHCPv6) has been standardized by the IETF through RFC 3315. DHCPv6 enables DHCP servers to pass configuration parameters such as IPv6 network addresses to IPv6 nodes. It offers the capability of automatic allocation of reusable network addresses and additional configuration flexibility. This protocol is a stateful counterpart to “IPv6 Stateless Address Autoconfiguration” (RFC 2462), and can be used separately or concurrently with the latter to obtain configuration parameters.

The following table lists where the IP phone obtains the IPv6 address and other network settings:

DHCPv6	SLAAC (ICMPv6)	How the IP phone obtains the IPv6 address and network settings?
Disabled	Disabled	You have to manually configure the static IPv6 address and other network settings.
Disabled	Enabled	The IP phone can obtain the IPv6 address via SLAAC, but the other network settings must be configured manually.
Enabled	Disabled	The IP phone can obtain the IPv6 address and the other network settings via DHCPv6.
Enabled	Enabled	The IP phone can obtain the IPv6 address via SLAAC and obtain other network settings via DHCPv6.

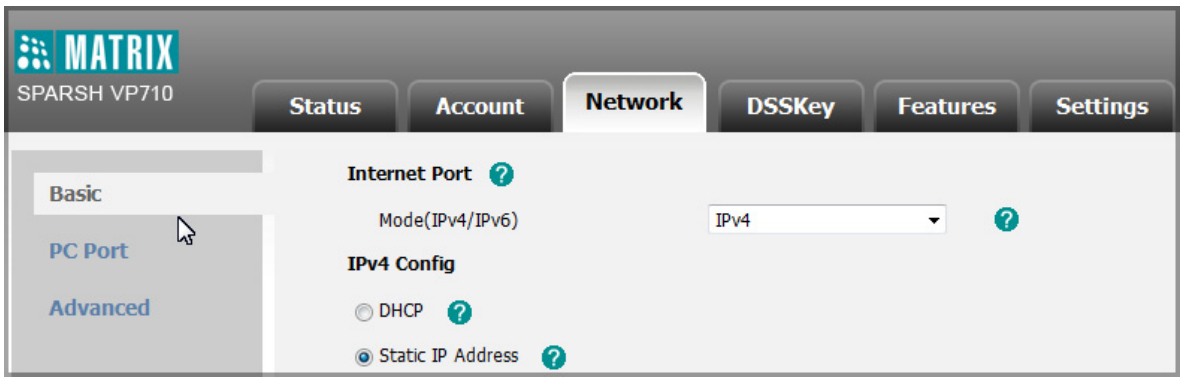
Procedure

IPv6 can be configured using the configuration files or locally.

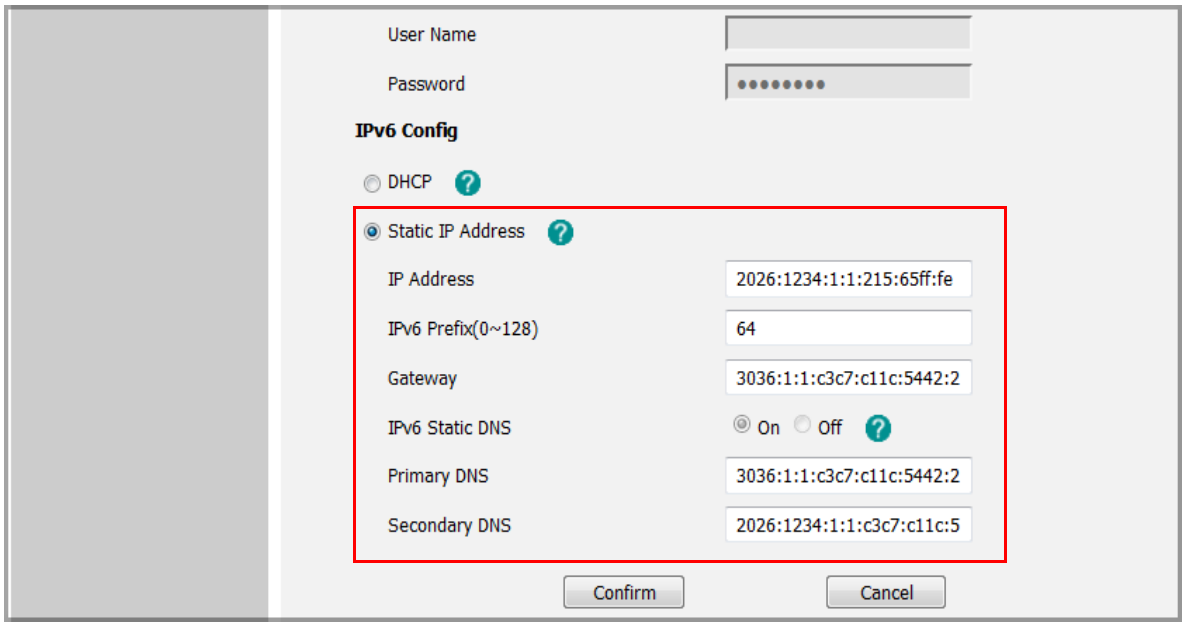
Configuration File	<MAC>.cfg	Configure the IPv6 address assignment method. Configure the IPv6 static DNS address. Configure the IPv6 static DNS. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the IPv6 address assignment method. Configure the IPv6 static DNS. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network&q=load
	Phone User Interface	Configure the IPv6 address assignment method. Configure the IPv6 static DNS. Configure the IPv6 static DNS address

To configure IPv6 address assignment method via web user interface:

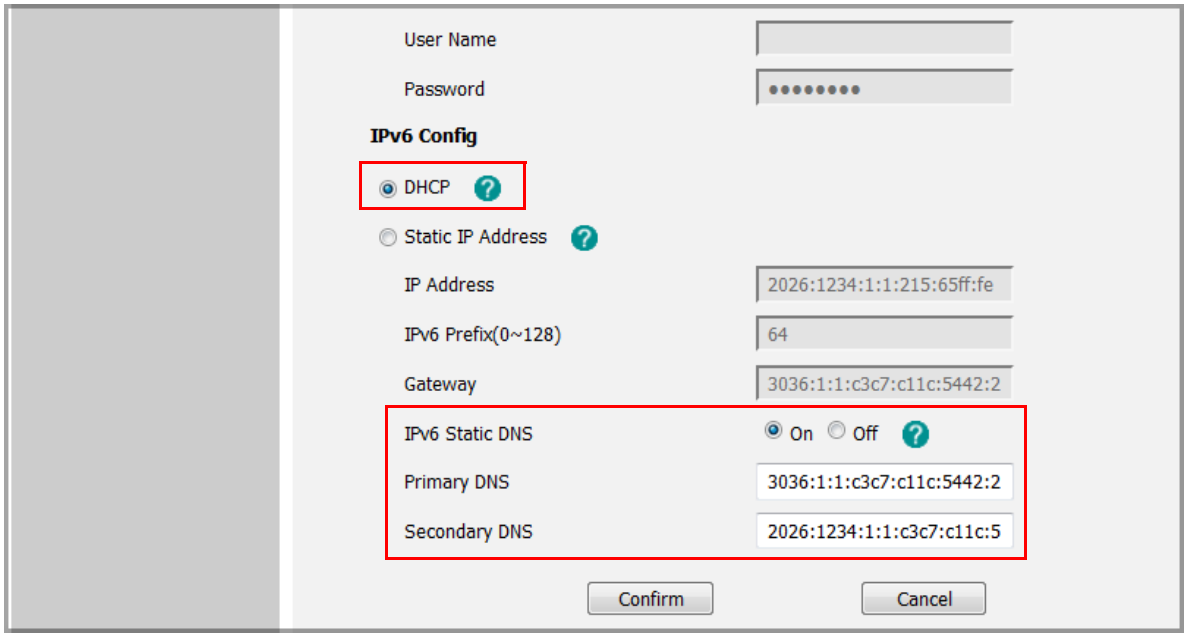
- Click on **Network->Basic**.



- Select the desired address mode (**IPv6** or **IPv4&IPv6**) from the **Mode (IPv4/IPv6)** list.
- In the **IPv6 Config** block, select the **DHCP** or the **Static IP Address** radio box.
  - If you select the **Static IP Address** radio box, configure the IPv6 address and other configuration parameters in the corresponding fields.



- If you select the **DHCP** radio box (Optional), you can configure the static DNS address in the corresponding fields.



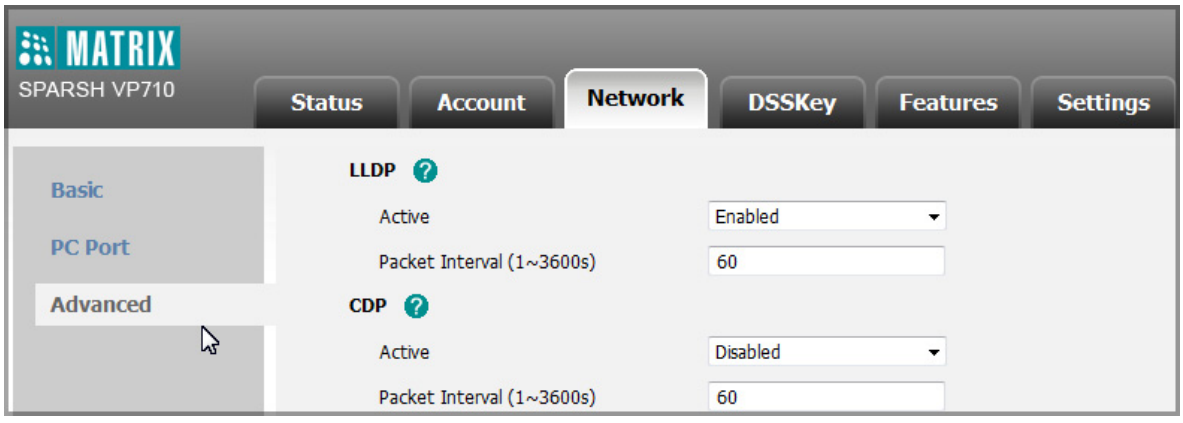
- Click **Confirm** to save the change.

  A dialog box pops up to prompt that the settings will take effect after a reboot.

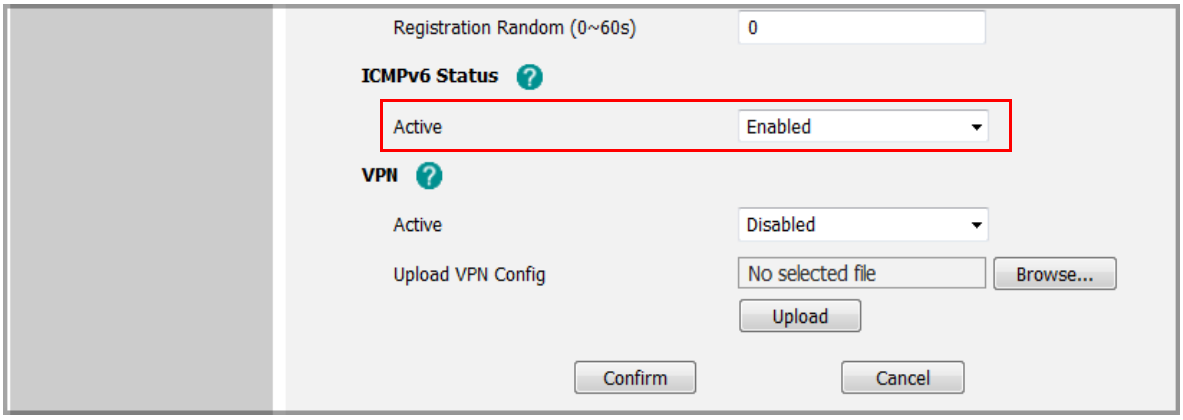
- Click **OK** to reboot the phone.

To configure SLAAC feature via web user interface:

- Click on **Network->Advanced.**



- In the **ICMPv6 Status** block, select the desired value from the **Active** list.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that the settings will take effect after a reboot.

- Click **OK** to reboot the phone.

To configure IPv6 address assignment method via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port**.
- Tap the **IP Mode** field.
- Tap **IPv6** or **IPv4 and IPv6** in the pop-up dialog box.
- Tap **IPv6**.
- Tap the **Type** field.
- Tap the desired IPv6 address assignment method in the pop-up dialog box.
- If you select the **Static IP**, configure the IPv6 address and other network parameters in the corresponding fields.



- Tap ✓ to save the change or ◁ to cancel.

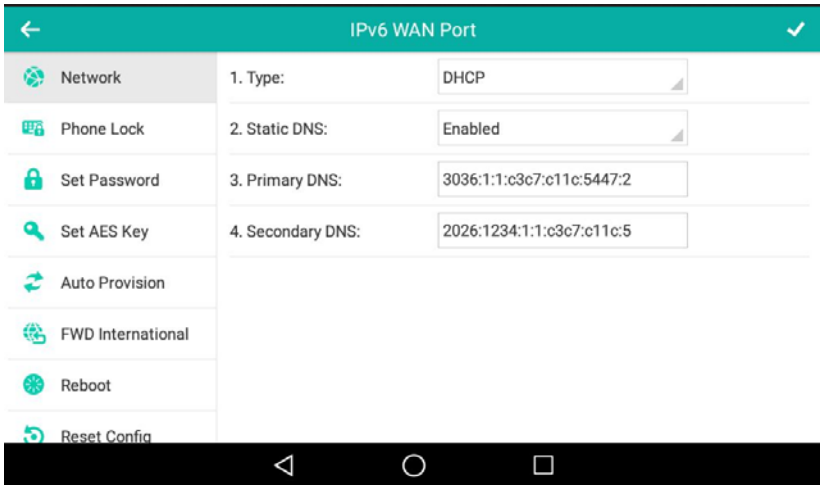
  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

To configure static DNS when DHCP is used via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->WAN Port->IPv6.**
- Tap the **Type** field.
- Tap **DHCP** in the pop-up dialog box.
- Tap the **Static DNS** field.
- Tap **Enabled** in the pop-up dialog box.
- Enter the desired value in the **Primary DNS** and **Secondary DNS** field respectively.



- Tap  to save the change or  to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot

Logon Wizard

Logon wizard allows IP phones to provide the logon wizard during the first startup.



Logon wizard feature works only if there is no registered account on the IP phone.

Procedure

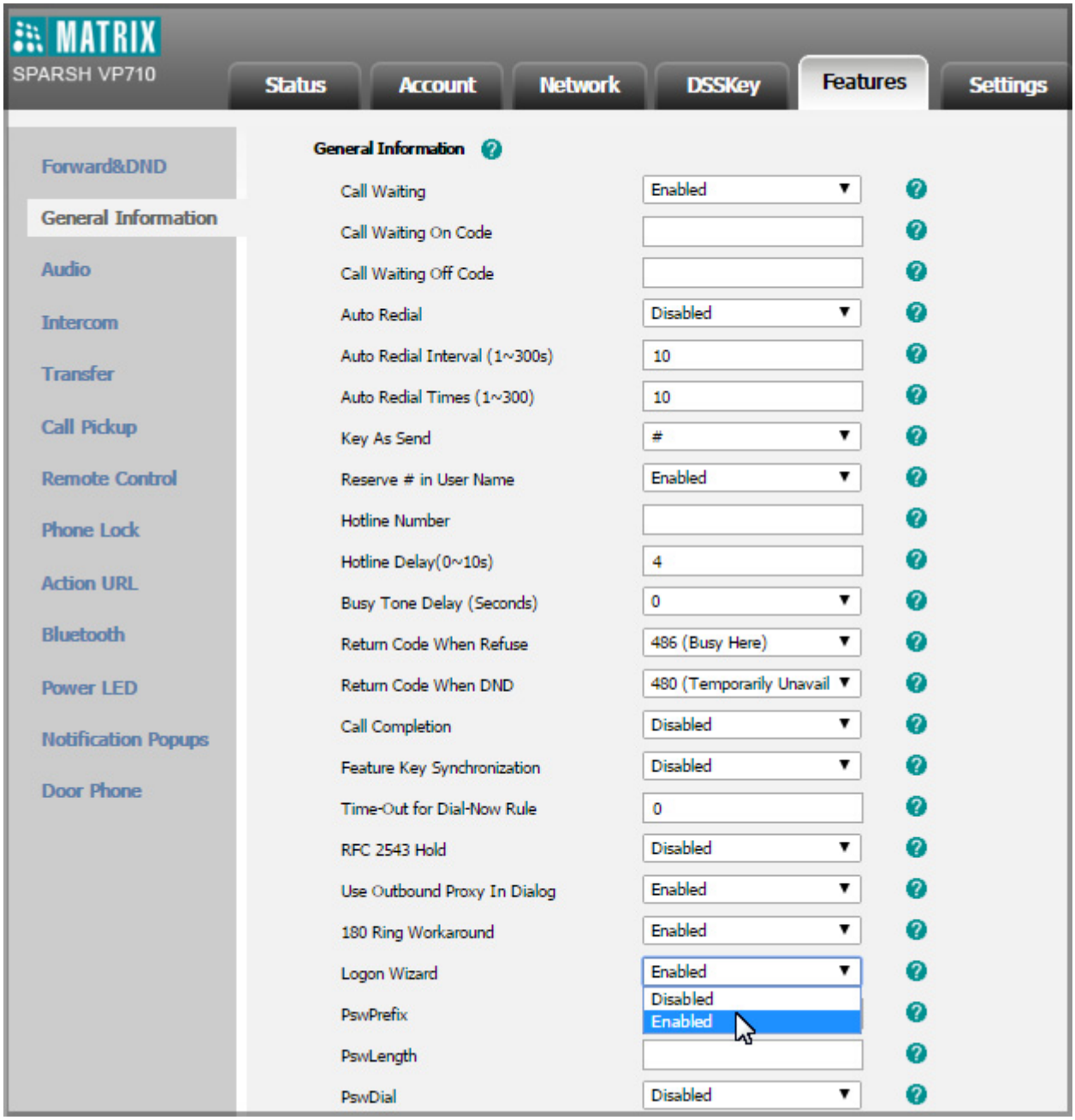
Logon wizard can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the logon wizard. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the logon wizard. **Navigate to**: ttp://<phoneIPAddress>/servlet?m=mod_data&p=features-actionurl&q=load

To configure logon wizard feature via web user interface:

- Click on **Features->General Information**.

- Select the desired value from the **Logon Wizard** list.



- Click **Confirm** to save changes.

Message

Message service in the IP phone can be categorized as follows:

- Video/Voice Mail
- Message Waiting Indicator (MWI)

Video/Voice Mail

You can leave video/ voice mails using SPARSH VP710. You can also access video/voice mails that are stored in the voice mailbox. The usage of video mail is the same as that of voice mail.

Video/voice mail is not available on all servers and differs from different servers. Contact your system administrator for more information.

Leaving Voice Mails

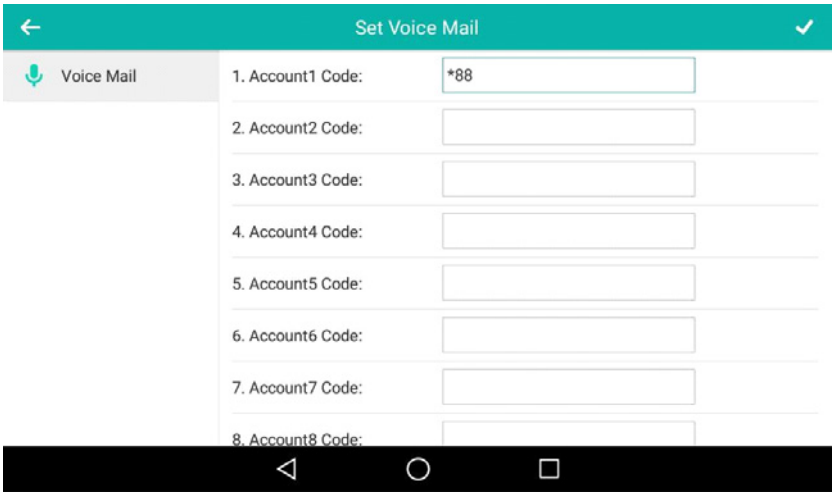
To leave a voice mail:

- You can leave a voice mail when the callee is busy or inconvenient to answer the call.
- Follow the voice prompt from the system server to leave a voice mail and then hang up.

Configuring the Voice Mail Access Codes

To configure voice mail access codes via phone user interface:

- Swipe left or right to go to the second idle screen.
- Tap to launch **Message** application.
- Tap **Set Voice Mail**.
- Enter the voice mail access code (e.g., *88) in the desired account field.



- Tap to save the change or to cancel.

Voice mail access codes must be predefined on the system server. Contact your system administrator for the more information.

Listening to Voice Mails

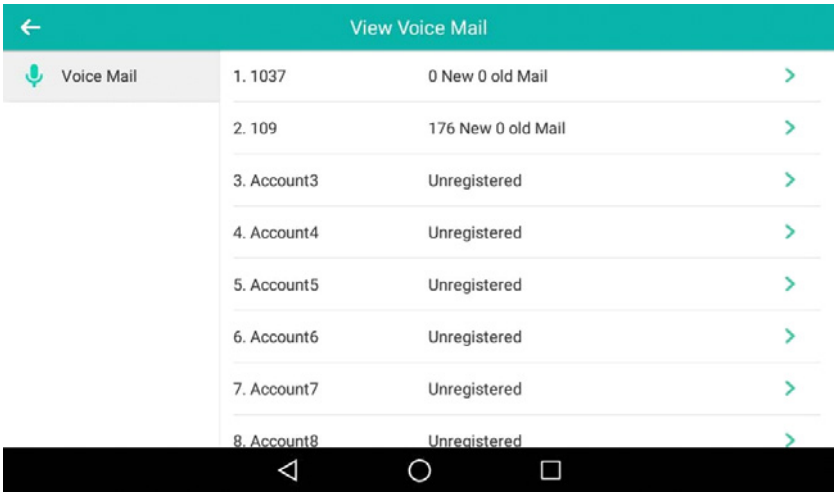
To listen to voice mails:

- The touch screen prompts when the phone receives a new voice mail, press or tap **Connect** to dial out the voice mail access code.
- Follow the voice prompt to listen to your voice mail.

Before listening to voice mails, ensure that the voice mail access code has been configured. For more information, refer to "Configuring the Voice Mail Access Codes".

To view voice mails via phone user interface:

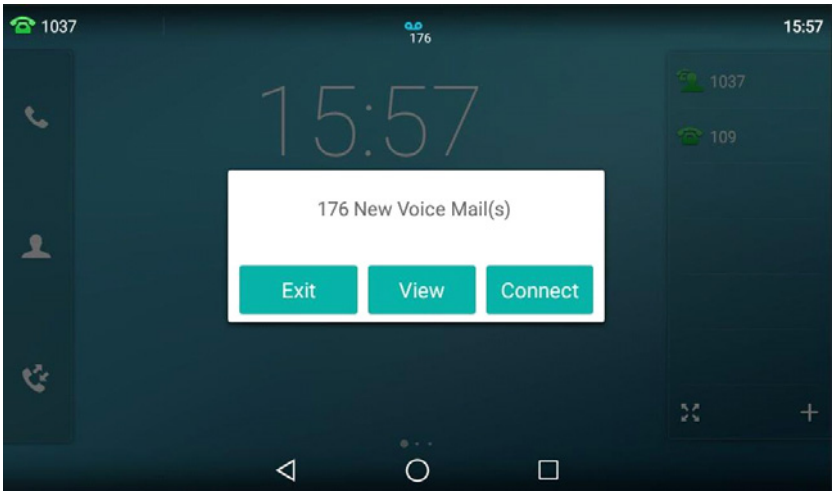
- Swipe left or right to go to the second idle screen.
- Tap to launch **Message** application.
- Tap **View Voice Mail**.
  The touch screen displays the amount of new and old voice mails.



You can tap the account to listen to voice mails.

The usage of video mail is the same as that of voice mail, the following takes voice mail as an example:

When receiving a new voice mail, the phone will play a warning tone.The power indicator LED will slowly flash red and the touch screen will prompt “n New Voice Mail(s)” where (“n” indicates the number of unread voice mails, e.g., 176 New Voice Mail(s)) and display .



If the voice mail pop-up message box disappears, it won't pop up again unless the you receives a new voice mail or re-register the account that has unread voice mail(s).

- *You can configure the phone not to display the pop-up prompt, contact your system administrator for more information.*
- *The icon on the status bar has a counter indicating the number of messages you haven’t heard yet.*

Voice Mail Tone

Voice mail tone feature allows the IP phone to play a warning tone when receiving a new voice mail. You can customize the warning tone or select specialized tone sets (vary from country to country) for your IP phone. For more information, refer to “Tones”.

Procedure

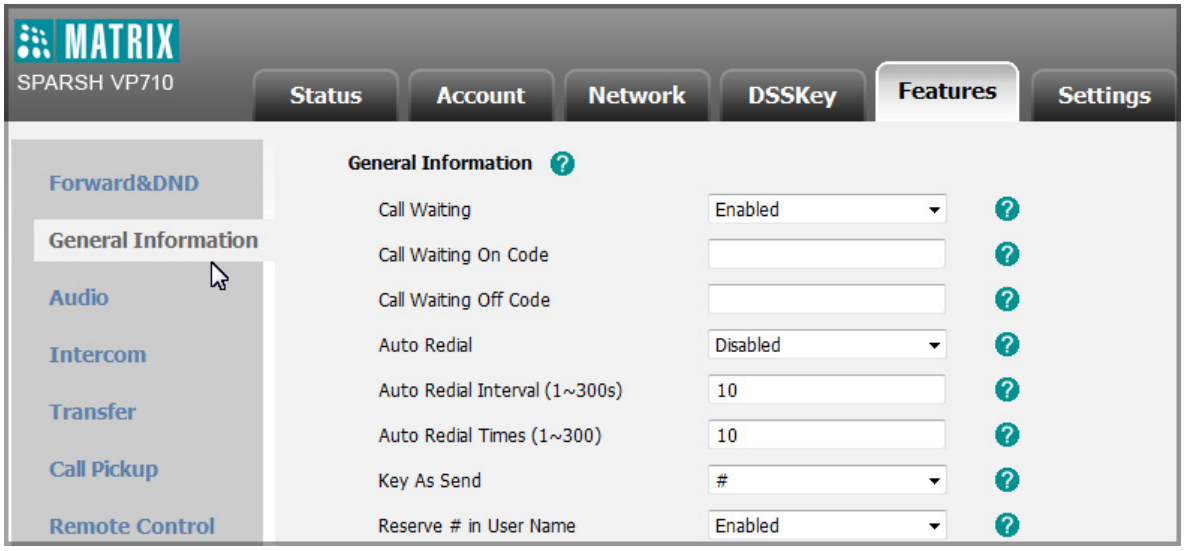
Voice mail tone can be configured using the using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure whether to play a warning tone when the IP phone receives a new voice mail. For more information, refer “Appendix D - Configuration Parameters”.

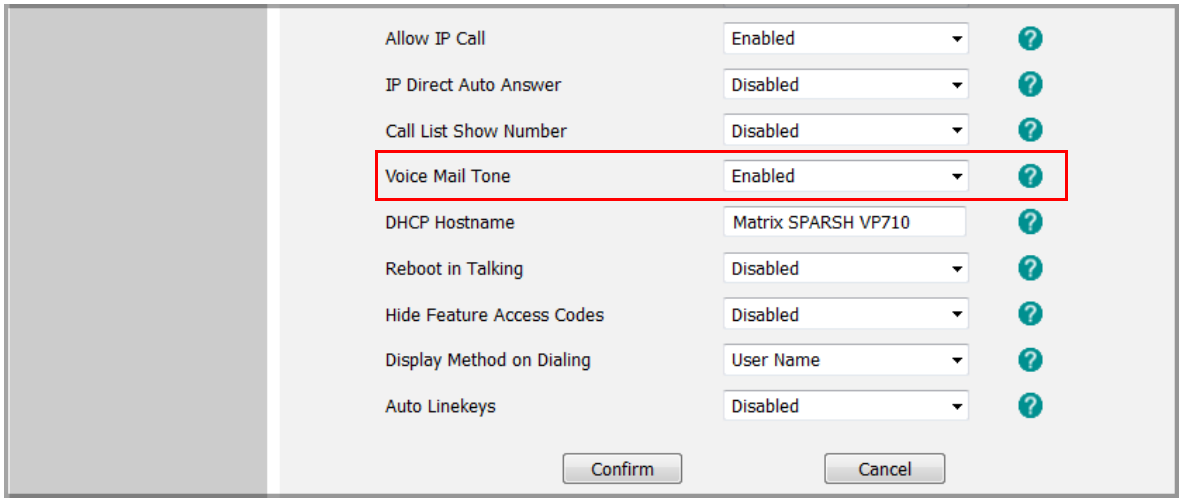
Local	Web User Interface	Configure whether to play a warning tone when the IP phone receives a new voice mail. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure voice mail tone via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Voice Mail Tone** list.



- Click **Confirm** to save the change.

Message Waiting Indicator (MWI)

Message Waiting Indicator (MWI) informs you with the number of unread messages in your mailbox without calling the mailbox.The IP phone supports both audio and visual MWI when receiving new voice messages. This will be cleared when you retrieve all voice mails or delete them

MWI is indicated in three ways:

- A warning tone
- An indicator message (including a voice mail icon) on the touch screen
- Power indicator LED slowly flashing red



MWI service is not available on all servers. Contact your system administrator for more information.

The IP phone supports both solicited and unsolicited MWI.

Unsolicited MWI: It is a server related feature. The IP phone sends a SUBSCRIBE message to the server for the message-summary updates. The server sends a message-summary NOTIFY within the subscription dialog each time the MWI status changes.

Solicited MWI: For this you must enable MWI subscription feature on the IP phone. The IP phone supports subscribing the MWI messages to the account or the voice mail number.

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The MWI Subscription parameters you need to know:

Option	Description
Subscribe for MWI	Enable or disable a subscription for MWI service.
MWI Subscription Period	Period of MWI subscription. The IP phone sends a refresh SUBSCRIBE request before initial subscription expires.
Subscribe MWI to Voice Mail	Enable or disable a subscription to the voice mail number for MWI service.To use this feature, you should also configure the voice mail number.



- *The IP phone will send SUBSCRIBE messages for MWI service to the account or the voice number MWI service depending on the server. Contact your system administrator for more information.*
- *MWI subscription is configurable via web user interface only.*

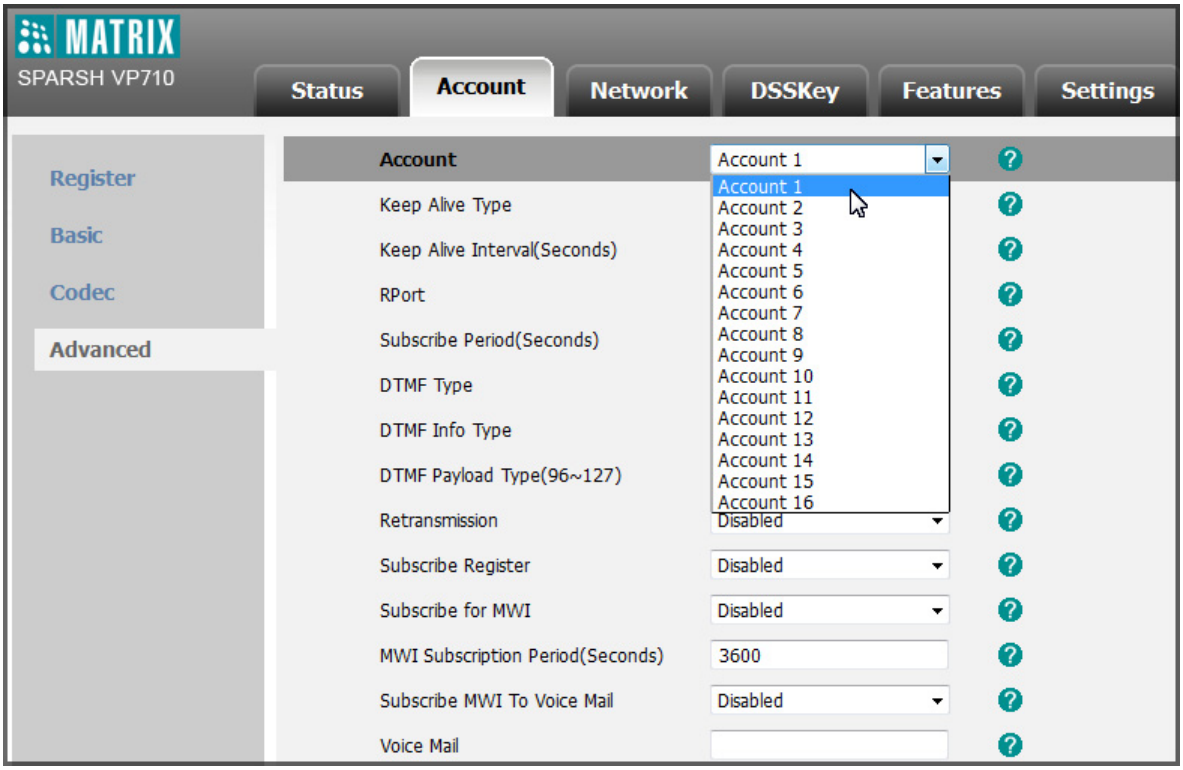
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure subscribe for MWI. Configure subscribe MWI to voice mail. Configure the voice mail number on a per-line basis. Configure the presentation of audio and visual MWI. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure subscribe for MWI. Configure subscribe MWI to voice mail. Configure the voice mail number on a per-line basis. Configure the presentation of audio and visual MWI. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0
	Phone User Interface	Configure the voice mail number on a per-line basis.

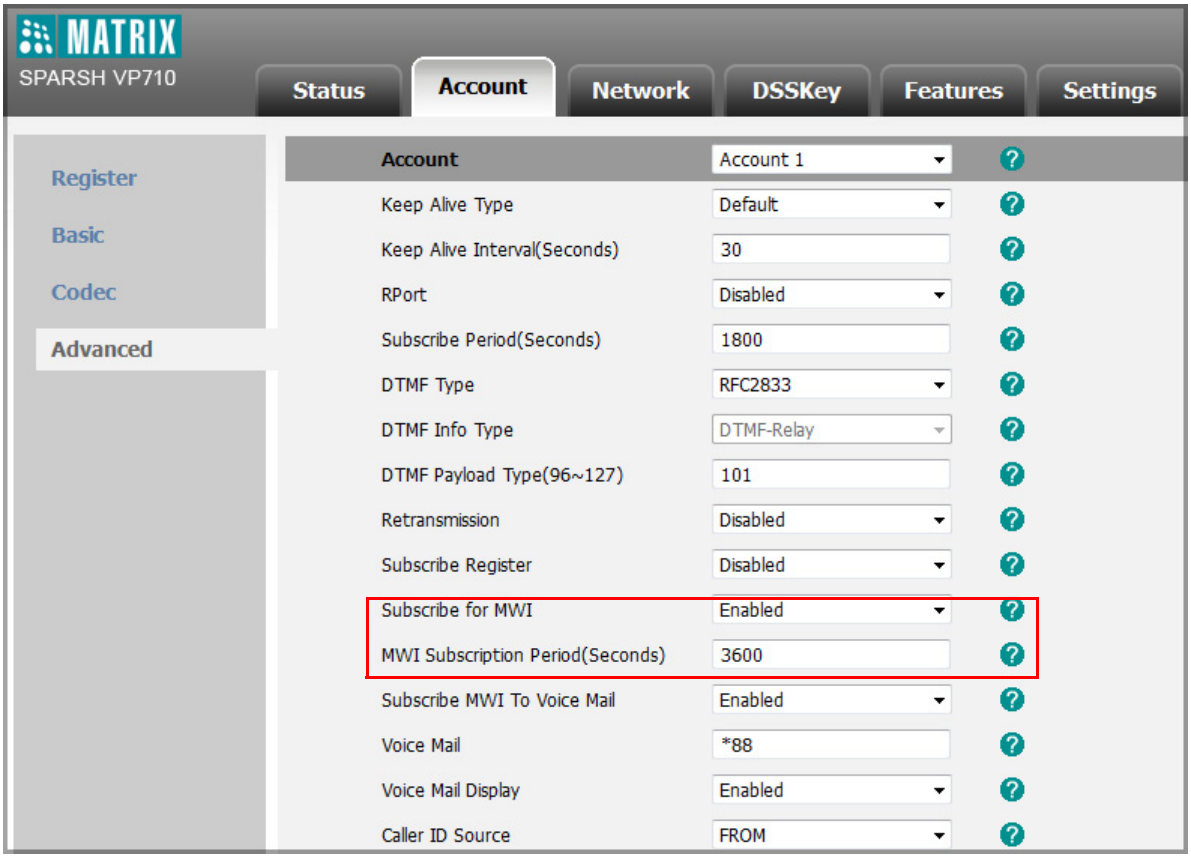
To configure subscribe for MWI via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select **Enabled** from the **Subscribe for MWI** list.

- Enter the period time in the **MWI Subscription Period (Seconds)** field.

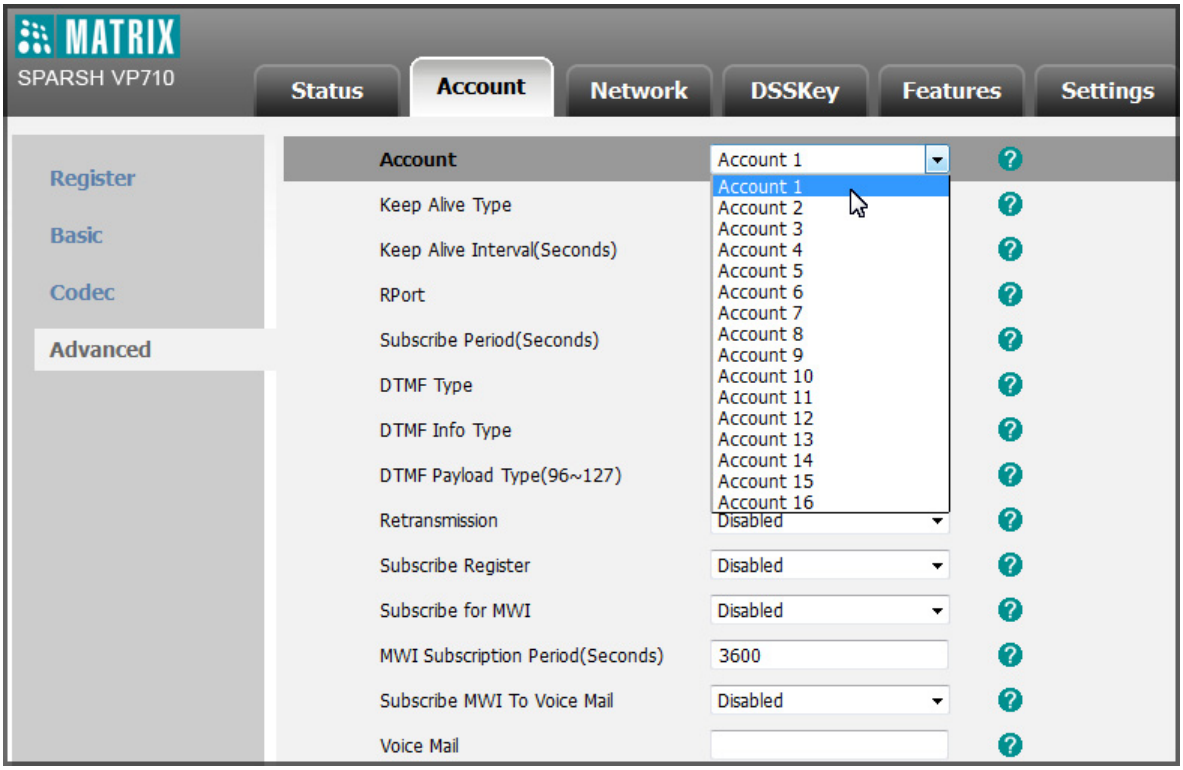


- Click **Confirm** to save the change.

  The IP phone will subscribe to the account number for MWI service by default.

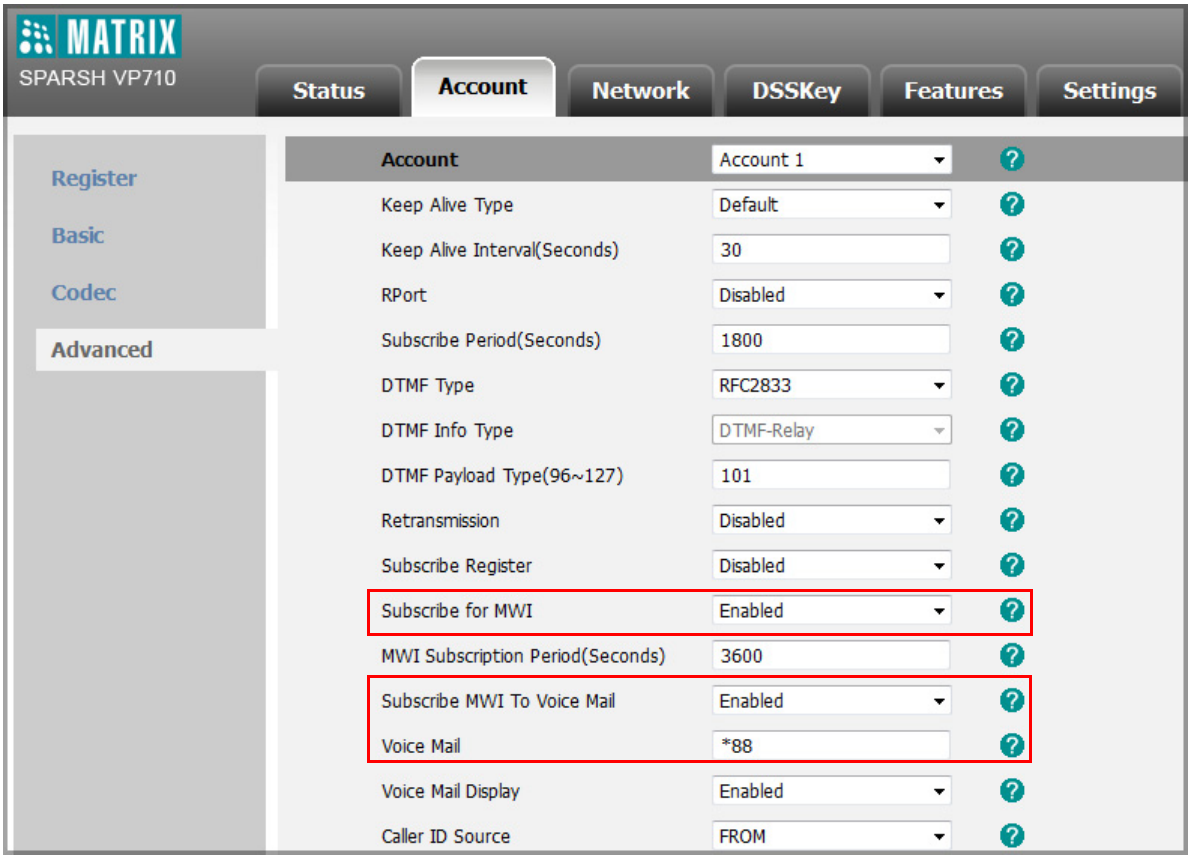
To configure Subscribe MWI to voice mail via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select **Enabled** from the **Subscribe for MWI** list.
- Select **Enabled** from the **Subscribe MWI To Voice Mail** list.

- Enter the desired voice mail number in the **Voice Mail** field.

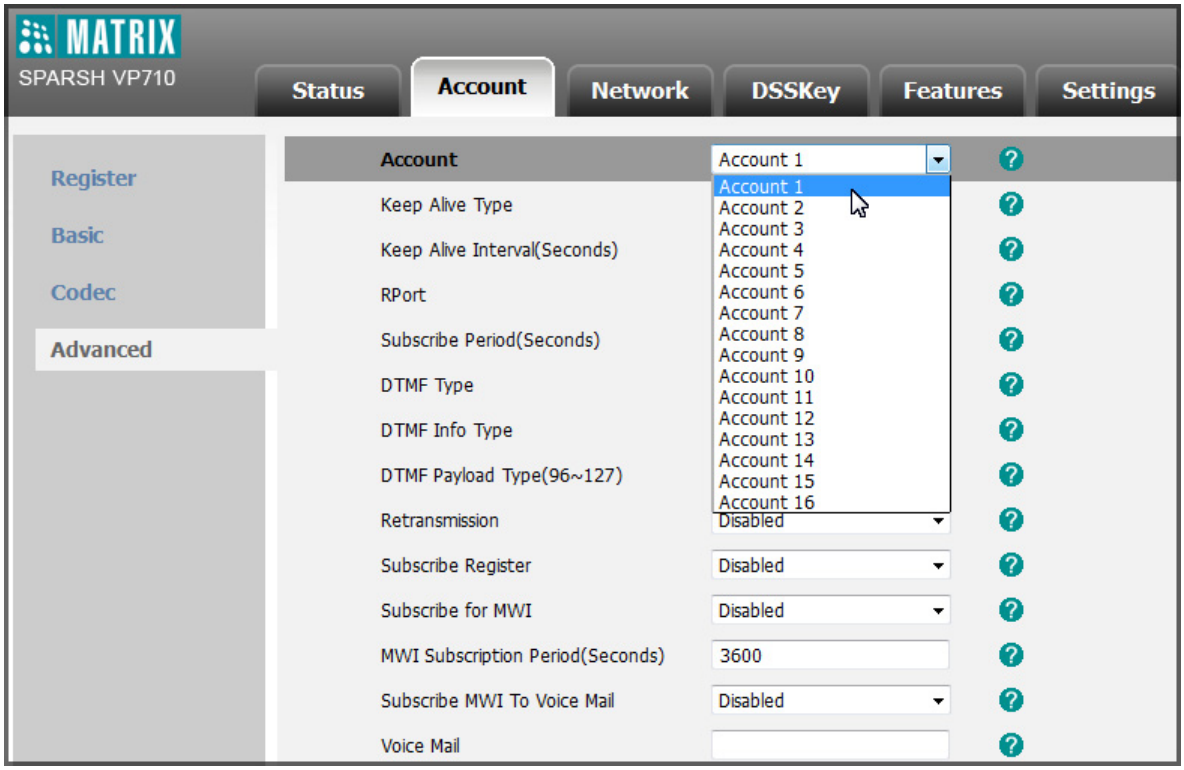


- Click **Confirm** to save the change.

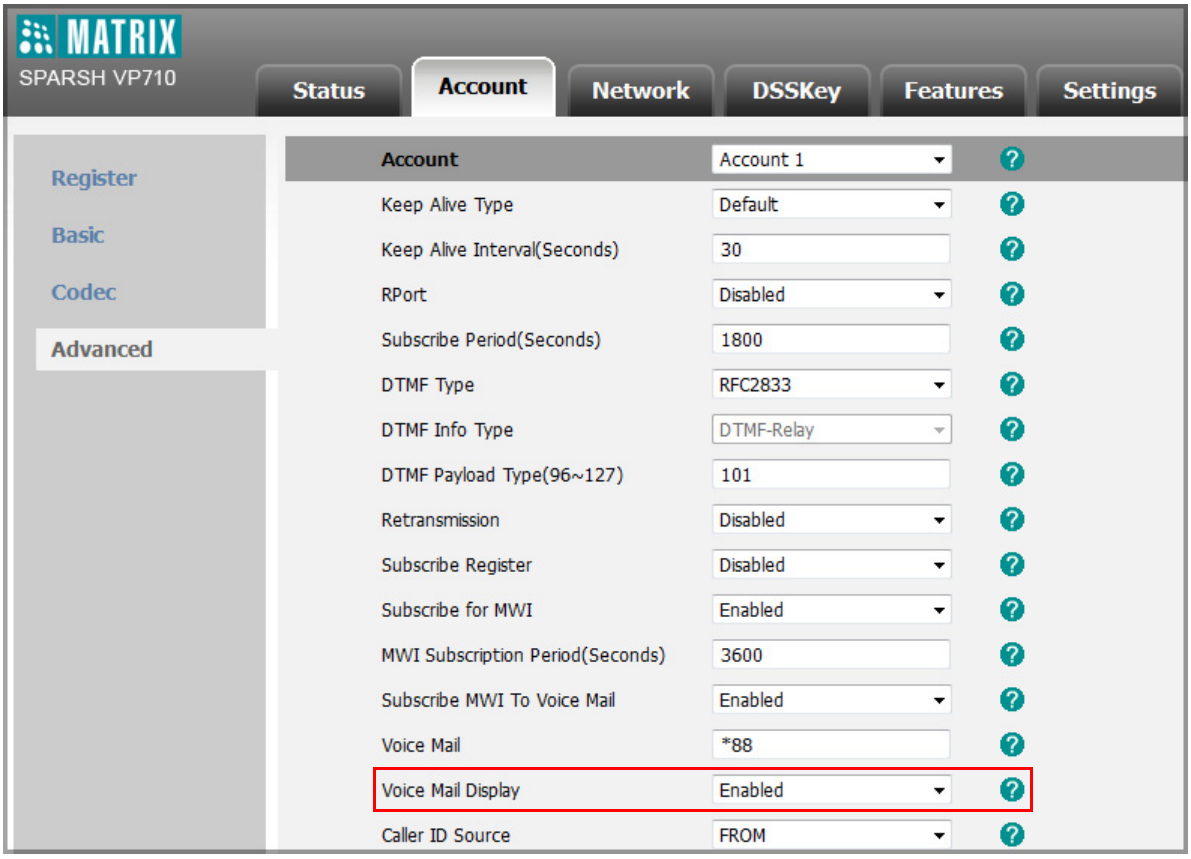
  The IP phone will subscribe to the voice mail number for MWI service using Subscribe MWI to Voice Mail.

To configure the presentation of audio and visual MWI via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Voice Mail Display** list.



- Click **Confirm** to save the change.

- *For proper functioning of the Voice Mail & MWI with the Matrix PBX server, Subscribe for MWI parameter must be Enabled and Subscribe MWI To Voice Mail parameter must be Disabled.*
- *Also the feature access code (Voice Mail Retrieval Number) must be programmed in the Voice Mail box (here it is shown as 9999).*

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Music on Hold

Music on hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by the party placed on hold.

To use this feature, you should specify a SIP URI pointing to a MoH server account.

When a call is placed on hold, the phone will send a SIP INVITE message to the MoH server account. The MoH server account automatically answers to the SIP INVITE messages and immediately plays audio from the source located (LAN, Internet) to the held party.

- *MoH server is configurable via web user interface only.*
- *Contact your system administrator for the SIP URI.*

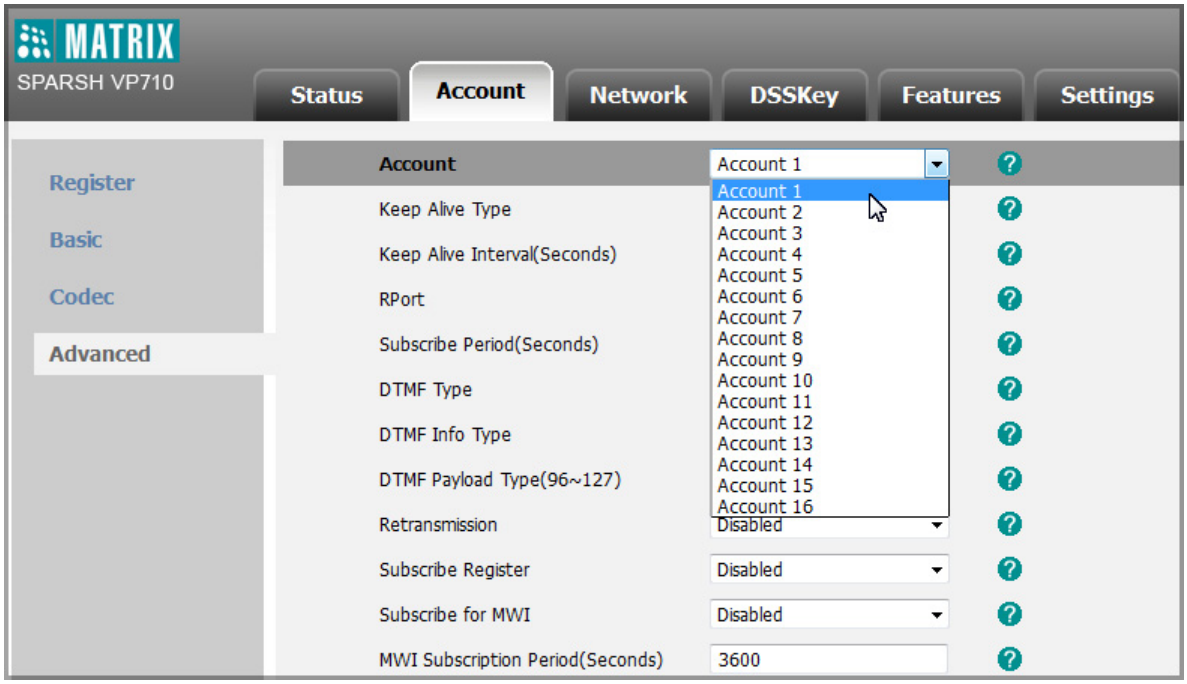
Procedure

Music on hold (MoH) can be configured using the configuration files or locally.

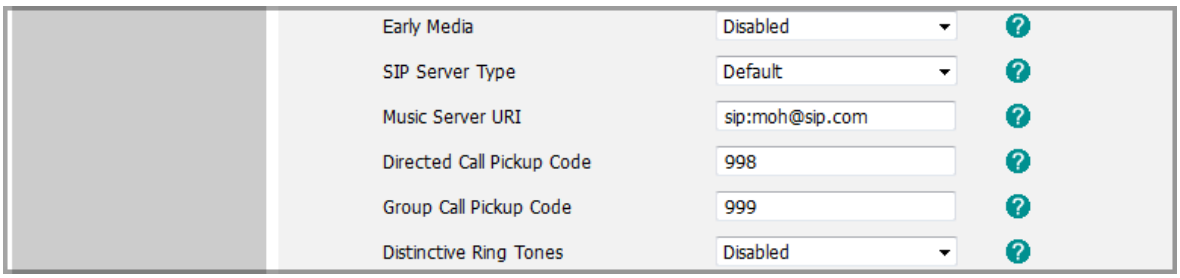
Configuration File	<MAC>.cfg	Configure music on hold on a per-line basis. Configure the way on how the IP phone processes music on hold when placing an active call on hold. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure music on hold on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure MoH server via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Enter the SIP URI (for example, sip:moh@sip.com) in the **Music Server URI** field.



- Click **Confirm** to save the change.

When you have placed a call on hold, the held party can hear the music.

- *Music on Hold is not available on all servers. There is no need to specify the SIP URI if the MoH feature is enabled by default on the server and the server can play audio to the held party. For more information, contact your server administrator.*
- *For this feature to function, all involved parties cannot use encrypted RTP (SRTP).*

Mute

This feature allows you to mute the microphone of the active audio device (handset, headset and speakerphone) during an active call or while dialing so that the other party cannot hear you.You can activate the mute feature by pressing the MUTE key. The mute feature is automatically deactivated when an active call ends.

Allow Mute

You can mute the microphone of the active audio device during an active call so that the other party cannot hear you. If allow mute feature is disabled, you cannot mute an active call.

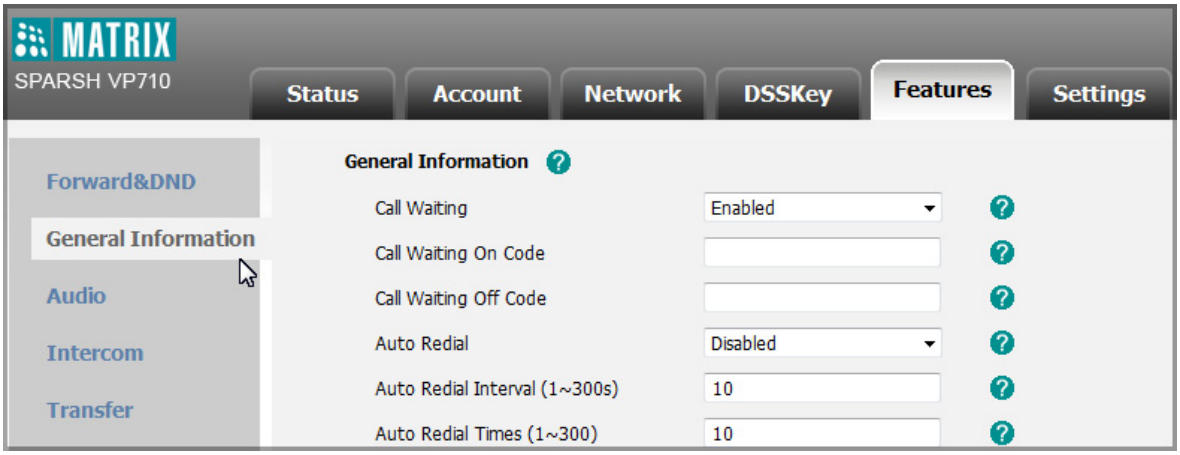
Procedure

Allow mute can be configured using the configuration files or locally.

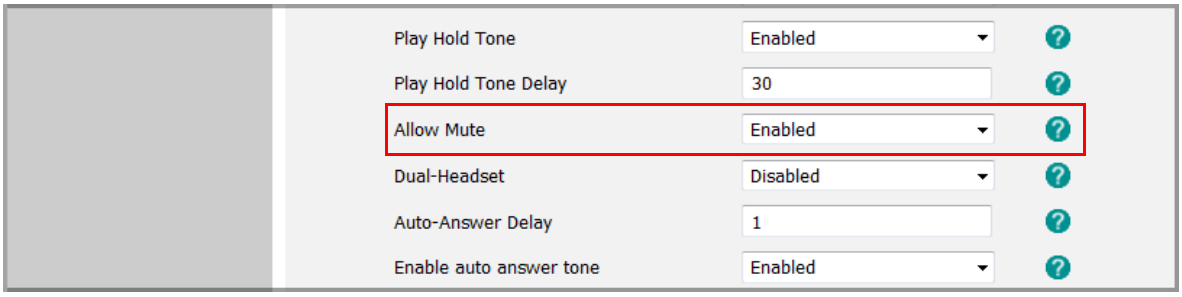
Configuration File	<MAC>.cfg	Configure allow mute feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure allow mute feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure allow mute via web user interface:

- Click on **Feature->General Information**.



- Select the desired value from the **Allow Mute** list.



- Click **Confirm** to save the change.

Keep Mute

Normally, Mute feature automatically deactivates when an active call ends.Keep mute also known as persistent mute allows you to keep the mute state of your phone persist across the calls.

Once keep mute feature is enabled, you can activate the mute feature by pressing the MUTE key in an idle state or any other state.The phone stays in the mute state until you un-mute the microphone or restart the phone.It prevents the other party from hearing the noise coming from your room when auto answer is enabled on your phone.
When you mute the phone in an idle state or any other states, the mute key LED illuminates solid red and the icon  appears on the status bar.

Make sure that the Keep mute feature should be pre-configured by your system administrator.

Procedure

Keep mute can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure keep mute feature. For more information, refer "Appendix D - Configuration Parameters".

To mute a call:

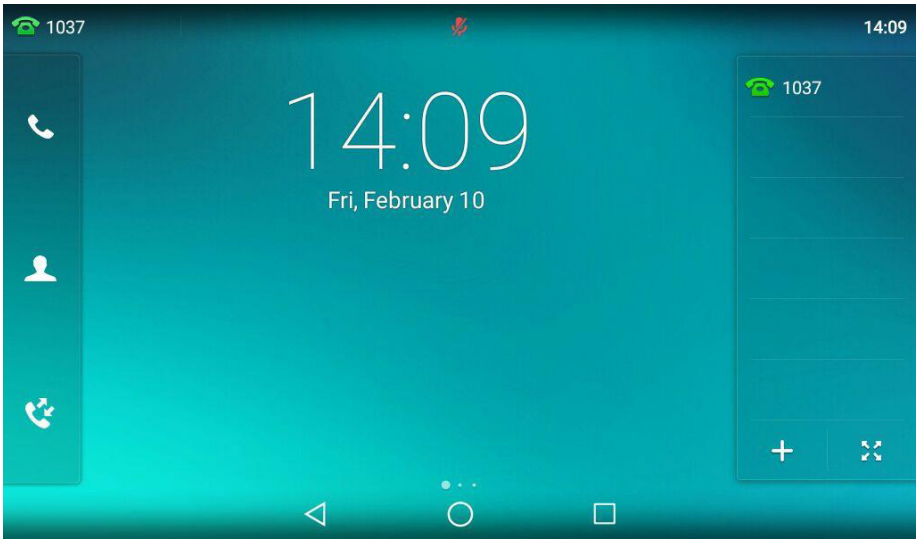
- Press  during an active call.

  The mute key LED illuminates solid red.

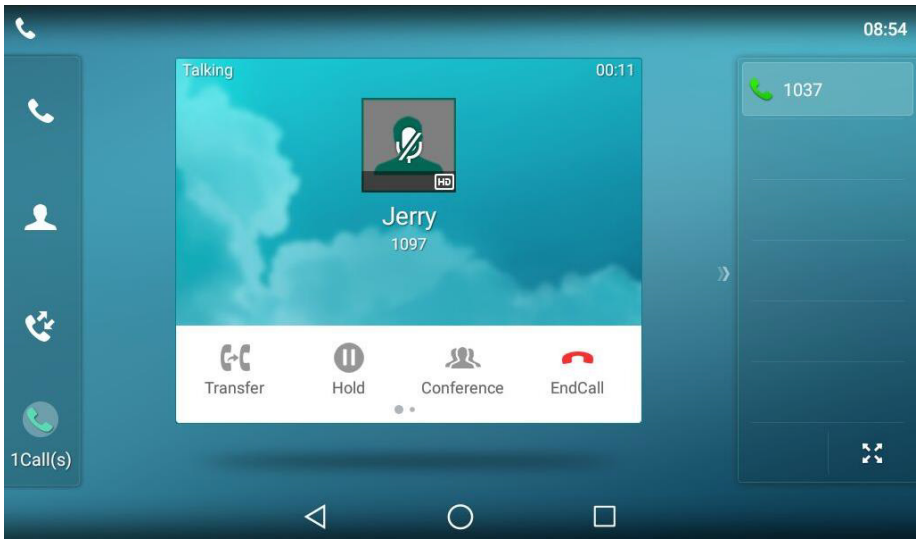
To un-mute a call:

- Press  again to un-mute the call.
- The mute key LED is turned off.

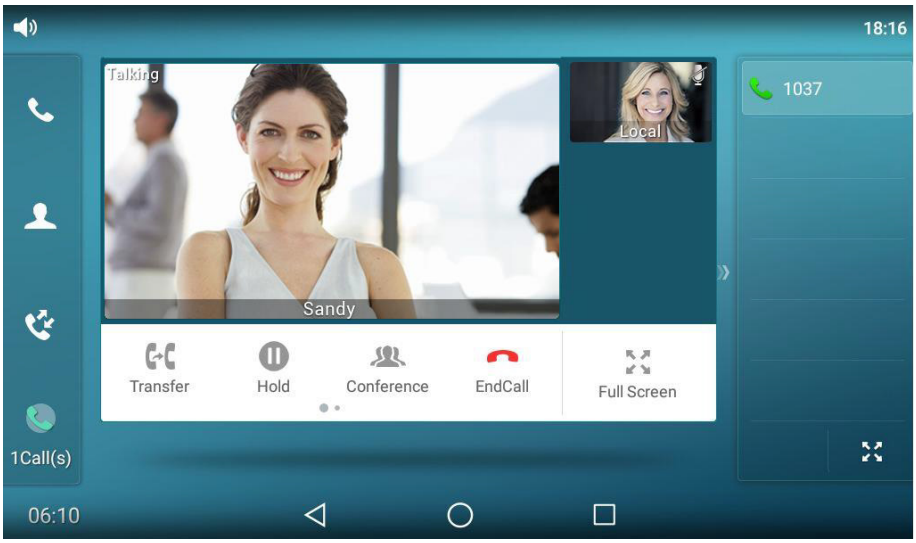
The following shows the keep mute icon on the status bar when mute is enabled:



The following shows an audio-only call has been muted:



The following shows a video call has been muted:



Notification Popups

Notification popups feature allows the IP phone to display the pop-up message box when it misses a call, forwards an incoming call to other party or receives a new voice mail.

Procedure

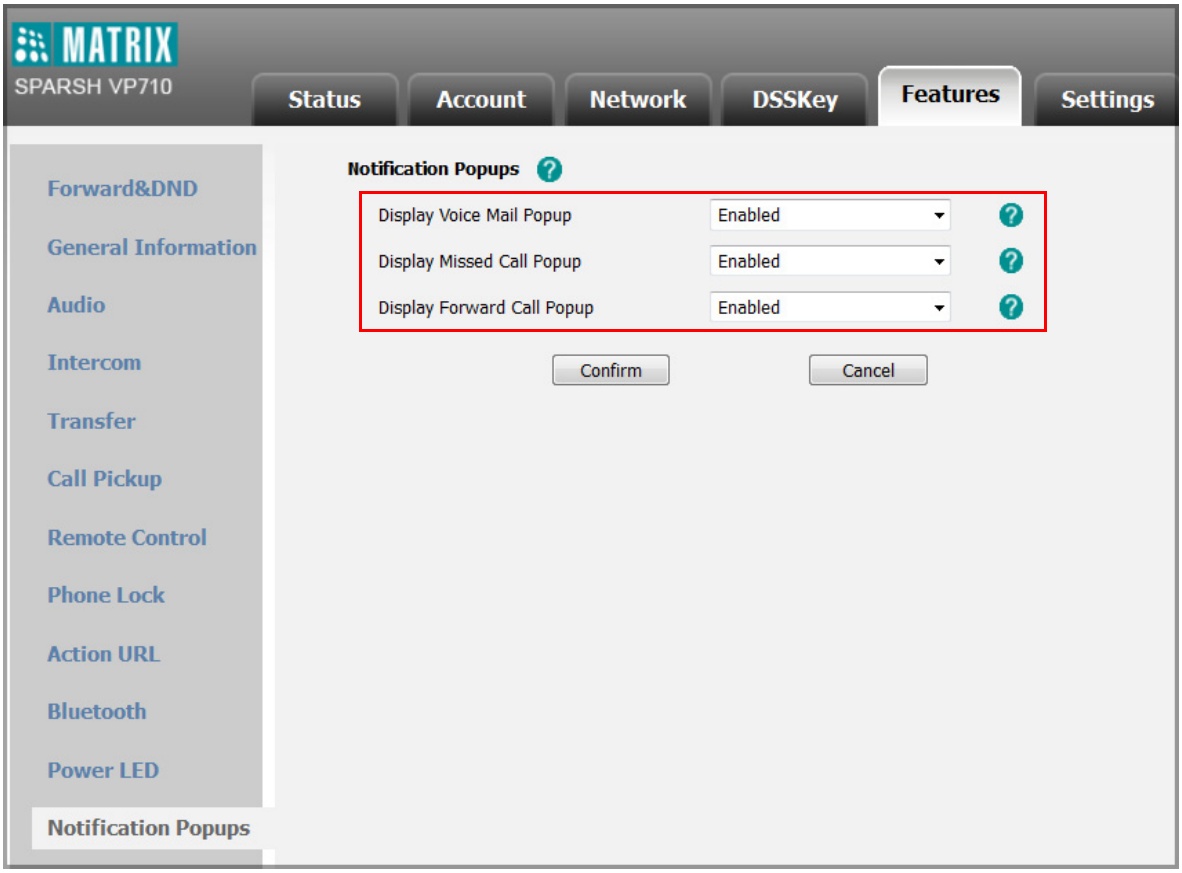
Notification Popups can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure notification popups. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure notification popups. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_ data&p=features-notifypop&q=load

To configure the notification popups via web user interface:

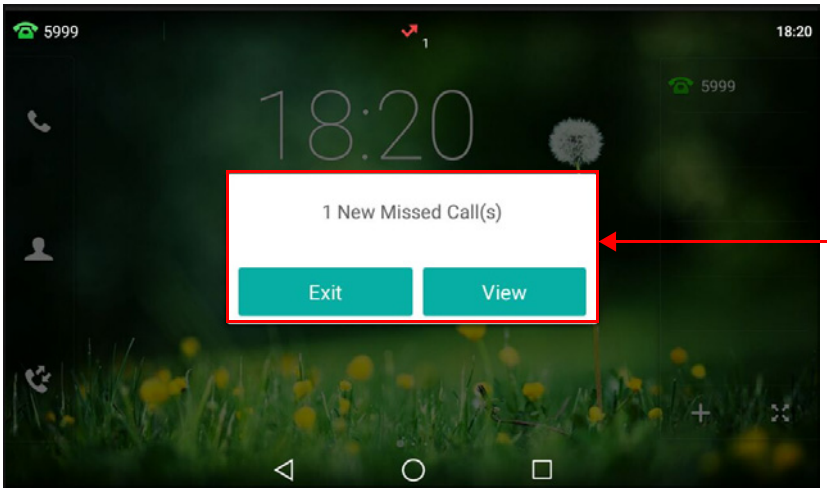
- Click on **Features->Notification Popups**.
- Select the desired value from the **Display Voice Mail Popup** list.
- Select the desired value from the **Display Missed Call Popup** list.

- Select the desired value from the **Display Forward Call Popup** list.



- Click **Confirm** to save the change.

The following shows an example of missing a call:



Pop-up message box

Password Dial

Password dial allows the callee's number to be partly displayed on the IP phone when placing a call. The hidden digits are displayed as asterisks on the touch screen. This feature is especially useful when placing important and confidential calls.

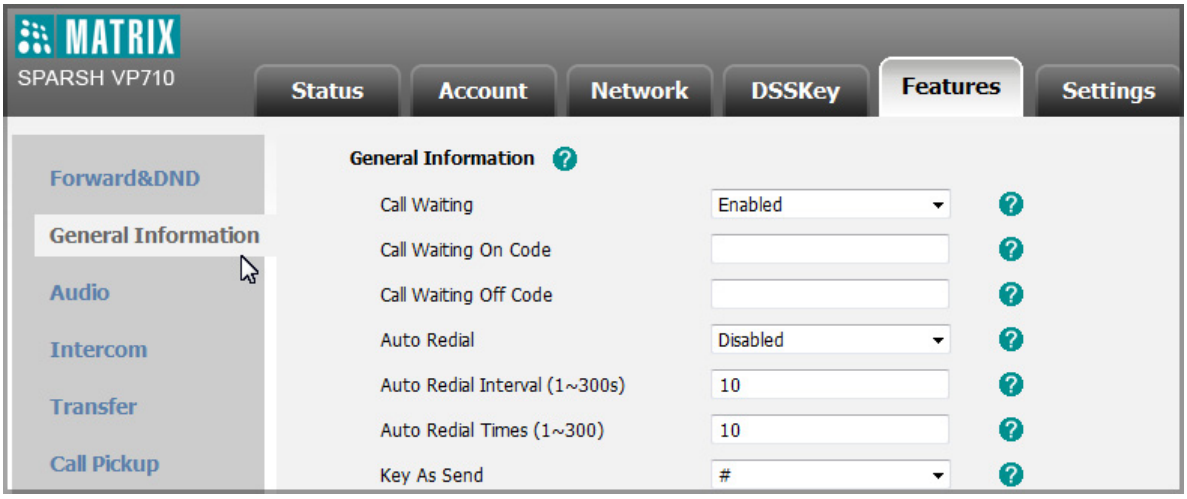
Procedure

Password dial can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure password dial feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure password dial feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

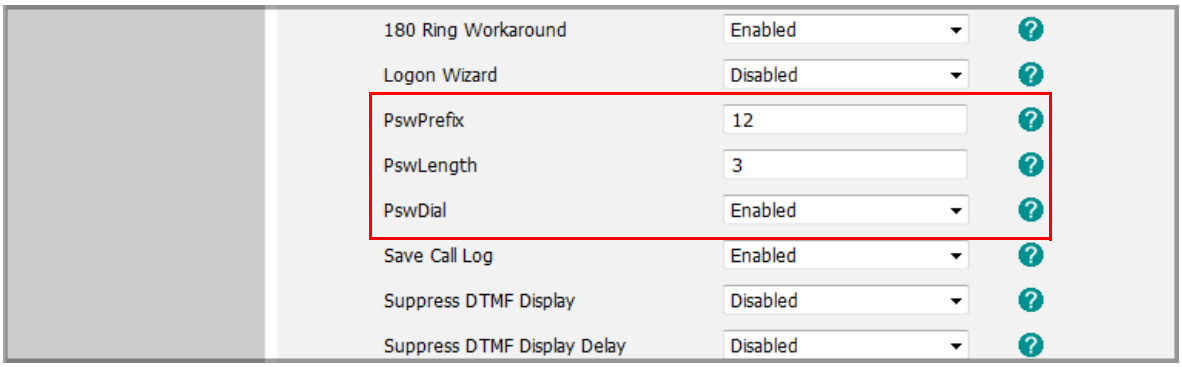
To configure password dial via web user interface:

- Click on **Features->General Information.**



- Enter the prefix of password dial in the **PswPrefix** field.
- Enter the desired number of hidden digits in the **PswLength** field.

- Select the desired value from the **PswDial** list



- Click **Confirm** to save the change.

ReCall

ReCall, also known as last call return, allows you to place a call to the last caller. Recall is implemented on the IP phone using a recall key. When you tap the recall key, the call will be placed to the phone number that last called you.

Procedure

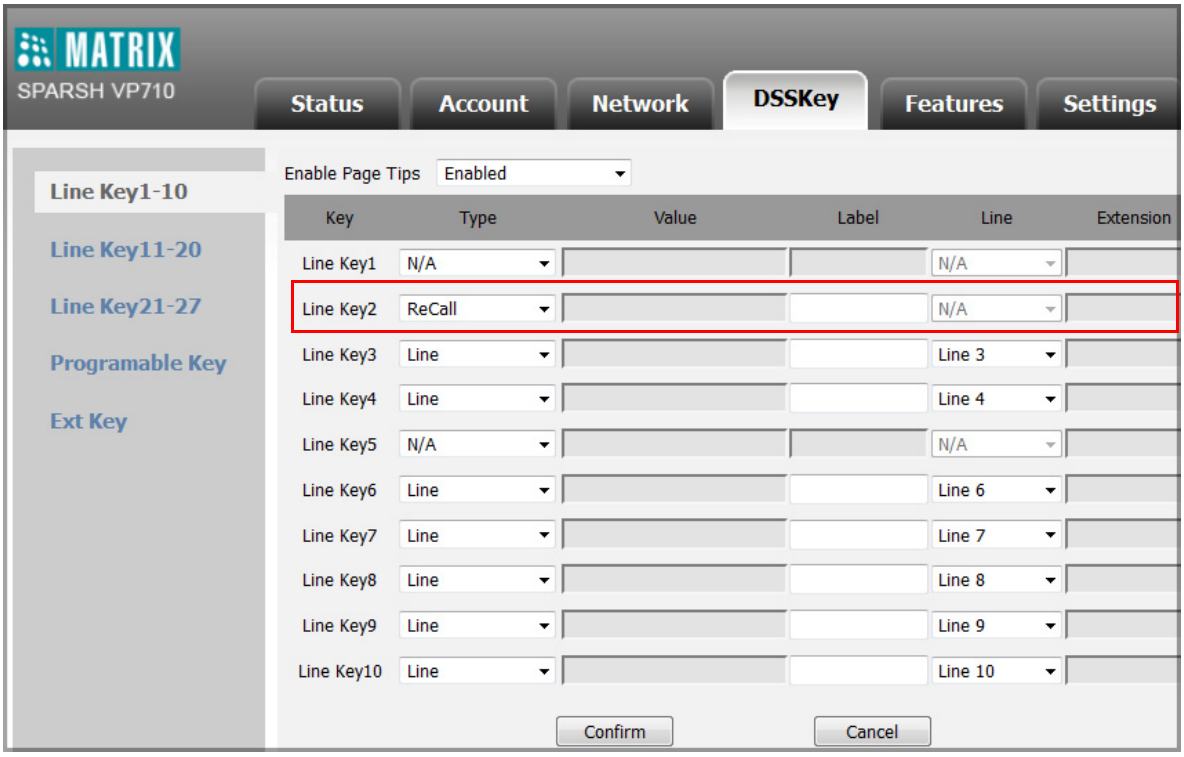
ReCall key can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Assign a recall key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Assign a recall key. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a recall key.

To configure a recall key via web user interface:

- Click on **DSSKey->Line Key (**or **Programmable Key).**

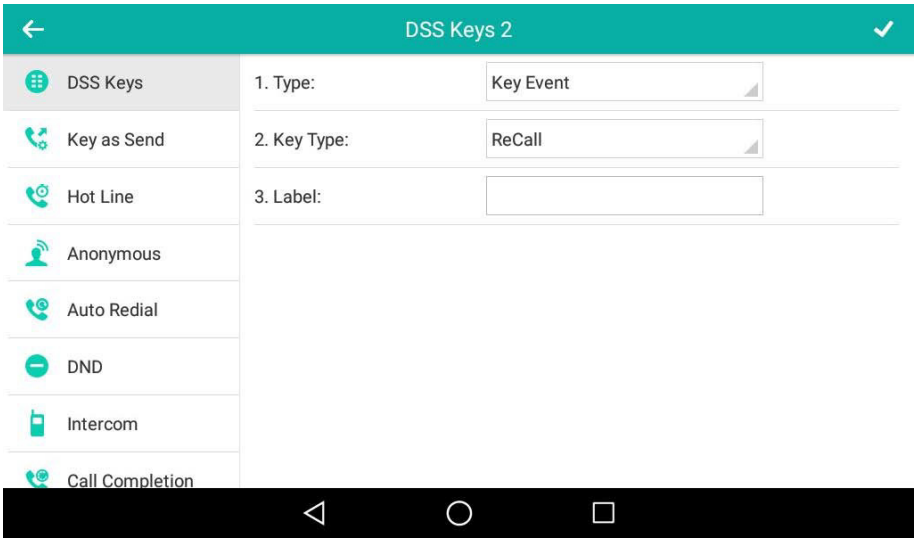
- In the desired DSS key field, select **ReCall** from **Type** list.



- In the **Label** field (optional), enter the string that will appear on the touch screen.
- Click **Confirm** to save the change.

To configure recall key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys.**
- Tap the desired line key.



- Tap the **Type** field. and tap to select **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field and tap to select **ReCall** in the pop-up dialog box.

- In the **Label** field (optional), enter the string that will appear on the touch screen.
- Tap  to accept the change or  to cancel

You can also tap  on the idle screen to configure a ReCall key.For further details refer "DSS Keys".

Recent Call In Dialing

Recent call in dialing feature allows you to view the placed calls list when the phone is on the pre-dialing screen. You can select an entry from the placed calls list to place a call.

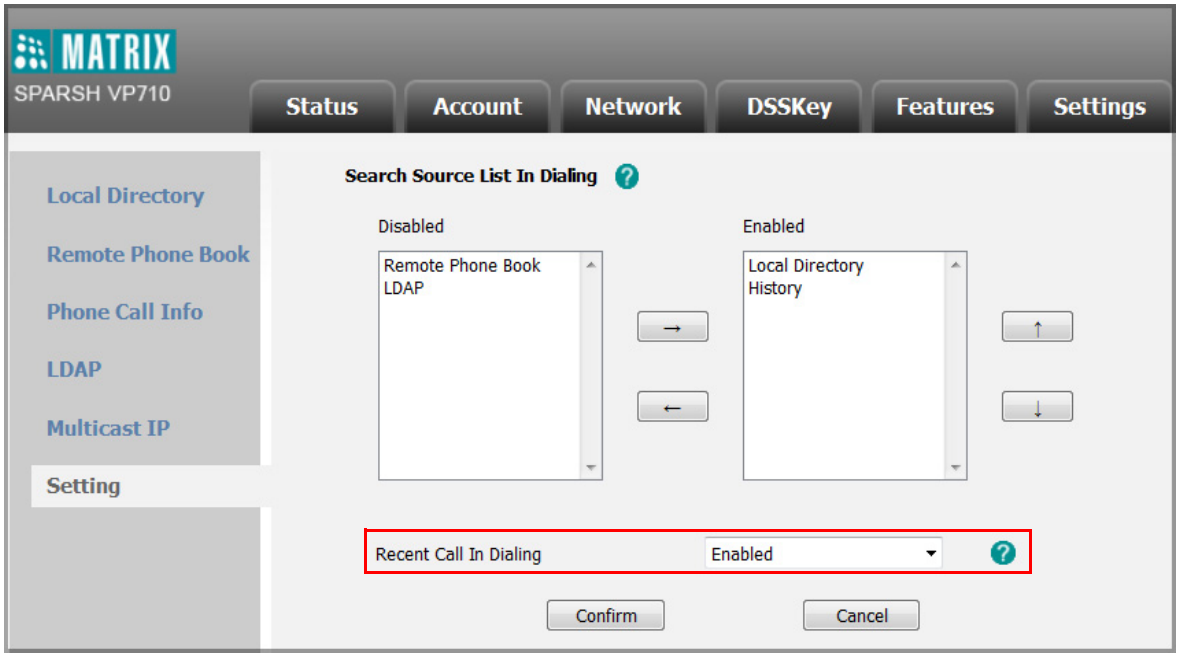
Procedure

Recent call in dialing can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure recent call in dialing feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure recent call in dialing feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=contacts-favorite&q=load

To configure recent call in dialing via web user interface:

- Click on **Directory->Setting**.
- Select **Enabled** from the **Recent Call In Dialing** list.



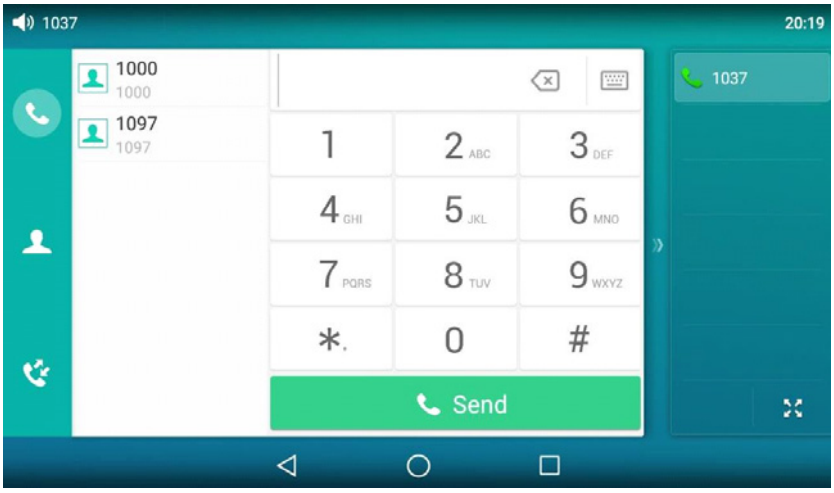
- Click **Confirm** to save the change.

Recent call in dialing is configurable via web user interface only.

To view placed calls list when the phone is on the dialing interface:

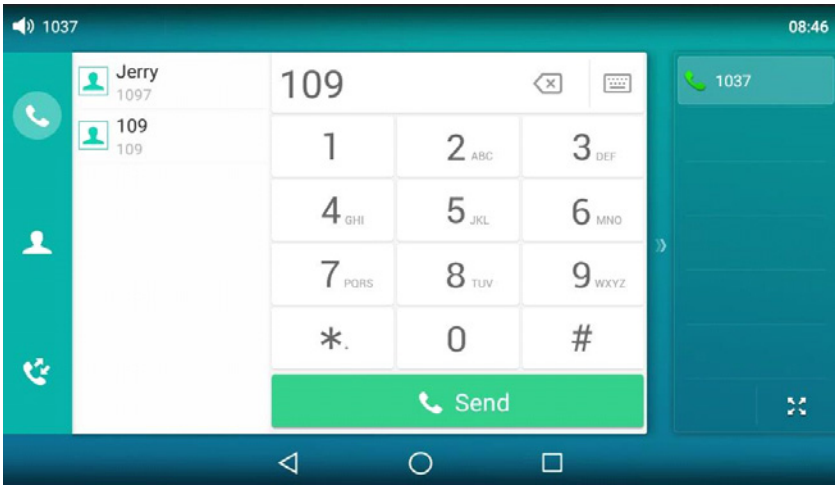
- Pick up the handset, press the Speakerphone key or tap the line key.

  The touch screen displays the placed calls list.



- Drag up and down to scroll through placed calls

  You can also enter a few continuous characters of the contact name or continuous numbers of the contact number (office, mobile or other number) to search from placed calls list.



Transfer Mode via Dsskey

Transfer mode via dsskey enables the IP phone to handle the current call differently via the DSS key.The IP phone supports three transfer modes: New Call, Blind Transfer and Attended Transfer.For more information on Blind Transfer and Attended Transfer, refer to "Call Transfer".The transfer mode via dsskey feature is available when the DSS key is assigned to the following features:

- Speed dial
- Transfer
- BLF/BLF List

For more information on how to configure the DSS Key, refer to "Configuring DSS Key".

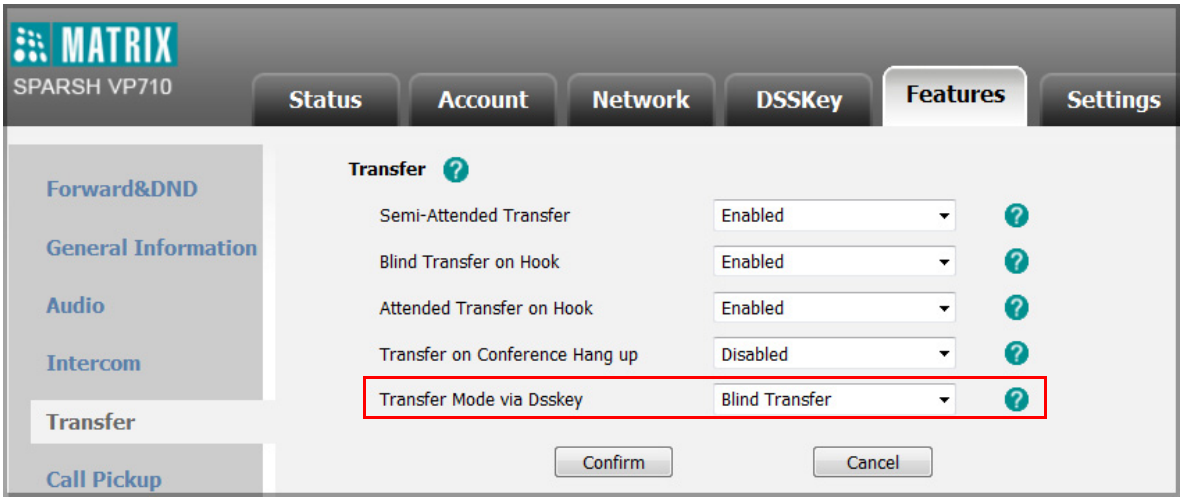
Procedure

Transfer mode via dsskey can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transfer mode via dsskey. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the transfer mode via dsskey. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-transfer&q=load

To configure transfer mode via dsskey via web user interface:

- Click on **Features->Transfer**.
- Select the desired value from the **Transfer Mode via Dsskey** list.



- Click **Confirm** to save the change.

CHAPTER 10

Accessing Advanced Features

This chapter provides operating instructions and information for making configuration changes applicable to the advanced features of the IP phone. Topics include:

- Action URI
- Action URL
- Answer By Hand
- Busy Lamp Field (BLF)
- Busy Tone Delay
- CDP
- Configuring the IP Phone for Use with a Firewall or NAT
- Distinctive Ring Tones
- DTMF
- Early Media
- Hot Desking
- Lightweight Directory Access Protocol (LDAP)
- Linker Layer Discovery Protocol (LLDP)
- Multicast Paging
- Quality of Service (QoS)
- Reboot in Talking
- Redial Tone
- Reserve # in User Name
- Return Code When Refuse
- Ringing Timeout
- Send User=phone
- Server Redundancy
- Session Timer
- Shared Call Appearance
- Send SIP Line
- Sip Send MAC
- SIP Session Timer
- Softkey Layout

- Static DNS Cache
- Tones
- Unregister When Reboot
- Use Outbound Proxy in Dialing
- Virtual Private Network (VPN)
- VLAN
- Web Server Type
- 180 Ring Workaround
- 802.1X Authentication
- 100 Reliable Retransmission

Action URI

HTTP/HTTPS GET Request

Action URI allows the IP phone to interact with web server application by receiving and handling an HTTP or HTTPS GET request. On receiving a GET request, the IP phone will perform the specified action and respond with a 200 OK message.

A GET request may contain variable named as "key" and variable value, which are separated by "="
The valid URI format is: *http(s)://<phoneIPAddress>/servlet?key=variable value*.

For example: http://10.10.20.10/servlet?key=SPEAKER.

MATRIX IP phones are compatible with other two old valid URI formats:

- *http(s)://<phoneIPAddress>/cgi-bin/ConfigManApp.com?key=variable value* and
- *http(s)://<phoneIPAddress>/cgi-bin/cgiServer.exx?key=variable value.*

SIP Notify Message

In addition, MATRIX IP phones support performing the specified action immediately by accepting a SIP NOTIFY message with the "Event: ACTION-URI" header from a SIP proxy server. The message body of the SIP NOTIFY message may contain variable named as "key" and variable value, which are separated by "=".

This method is useful for the users who are always working in the small/home office where a secure firewall may prevent the HTTP or HTTPS GET request from the external network.

If you want to only accept the SIP NOTIFY message from your SIP server and outbound proxy server, you have to enable the Accept SIP Trust Server Only feature. For more information, refer to "Accept SIP Trust Server Only".

Example of a SIP Notify with the variable value (SPEAKER):

Message Header

NOTIFY sip:3583@10.2.40.10:5062 SIP/2.0

Via: SIP/2.0/UDP 10.2.40.27:5063;branch=z9hG4bK4163876675

From: <sip:3586@10.2.1.48>;tag=2900480538

To: "3583" <sip:3583@10.2.1.48>;tag=490600926

Call-ID: 2923387519@10.2.40.10

CSeq: 4 NOTIFY

Contact: <sip:3586@10.2.40.27:5063>

Max-Forwards: 70

User-Agent: MATRIX SPARSH VP710 58.80.196.2 00:1b:09:04:8e:3c

Event: ACTION-URI

Content-Type: message/sipfrag

Content-Length: 6

Message Body

key=SPEAKER

The following table lists pre-defined variable values:

Variable Value	Phone Action
OK	Tap **Settings->Status**.
SPEAKER	Press the Speakerphone key.
F_TRANSFER	Transfer the call to another party.
VOLUME_UP	Increase the volume.
VOLUME_DOWN	Decrease the volume.
MUTE	Mute the call.
F_HOLD/HOLD	Place an active call on hold.
F_CONFERENCE	Tap the **conference** soft key.
Cancel/CANCEL/X	Cancel actions or reject incoming calls or end the call.
0-9/*/POUND	Press the keypad (0-9, * or #).
L1-LX	Tap the line keys (X=27).
MSG	Press the MESSAGE key.
HEADSET	Press the HEADSET key.
RD	Press the REDIAL key.

Reboot	Reboot the phone.
AutoP	Perform auto provisioning.
DNDOn	Activate the DND feature.
DNDOff	Deactivate the DND feature.
number=xxx&outgoing_uri=y	Place a call to xxx from SIP URI y. **Example:** http://10.3.20.10/servlet?key=number=1234&o utgoing_uri=1006@10.2.1.48 (1234 means the number you dial out; 1006@10.2.1.48 means the SIP URL you dial from)
OFFHOOK	Pick up the handset.
ONHOOK	Hang up the handset.
ANSWER/ASW/Asw	Answer the call.
Reset	Reset the phone.
ATrans=xxx	Perform a semi-attended/attended transfer to xxx.
BTrans=xxx	Perform a blind transfer to xxx.
CALLEND/ CallEnd	End the call.
CallWaitingOn	Activate the call waiting feature
CallWaitingOff	Deactivate the call waiting feature.
AlwaysFwdOn/BusyFwdOn/NoAnswFwdOn=xxx=n	Activate an always/busy/no answer forward feature to xxx for the IP phone ("xxx" means the destination number). The valid value of "n" means the duration time (seconds) before forwarding incoming calls (n is the times of 6, e.g., 24). It is only applicable to no answer forward feature. **Note:** It works only if the call forward mode is Phone, the always/busy/no answer forward feature will apply to all the accounts on the phone. **Example:** http://10.10.20.10/servlet?key=NoAnswFwdOn=1001=24
AlwaysFwdOff/BusyFwdOff/NoAnswFwdOff	Deactivate the always/busy/no answer forward feature for the IP phone. **Note:** It works only if the call forward mode is Phone, the always/busy/no answer forward feature will apply to all the accounts on the phone. **Example:** http://10.10.20.10/servlet?key=NoAnswFwdOff
ASW/CANCEL/HOLD/UNHOLD:xxx	Answer/end/hold/unhold a call (xxx refers to the call-id of the active call). **Example:** http://10.10.20.10/servlet?key=ASW:33093 **Note:** To get the call-id of the active call, configure the action URL: http(s)://<serverIPAddress>/help.xml?CallId=$call_id. For more information, refer to "Action URL".

phonecfg=get[&accounts=x][&dnd=x][&fw=x	Get firmware version, registration, DND or forward configuration information. The valid value of "x" is 0 or 1, 0 means you do not need to get configuration information. 1 means you want to get configuration information. **Note:** The valid URI is: http(s)://<phoneIPAddress>/servlet?phonecfg=get[&accounts=x][&dnd=x][&fw=x] **Example:** http://10.10.20.10/servlet?phonecfg=get[&accounts=1][&dnd=0][&fw=1]

- *The variable value is not applicable to all the events. For example, for an IP phone the variable value "MUTE" is only applicable during a call.*
- *When authentication is required, you must enter "p=login&q=login&username=xxx&pwd=yyy&jumpto=URI&" before the variable "key". xxx refers to the login user name and yyy refers to the login password.*

Matrix IP phones also support a combination of the variable values in the URI, but the order of the variable value is determined by the operation of the phone.
The valid URI format is: *http(s)://<phoneIPAddress>/servlet?key=variable value[;variable value]*. Variable values are separated by a semicolon from each other.

The following shows an example for answering an incoming call then mute the call immediately: http://10.3.20.10/servlet?key=ASW;MUTE.

Configuring Trusted IP Address for Action URI

By default, the IP phone do not handle HTTP/HTTPS GET requests for security reasons. For action URI you need to specify the trusted IP address. When the IP phone receives a GET request from the trusted IP address for the first time, the touch screen prompts the message "Allow Remote Control?". You can specify one or more trusted IP addresses on the IP phone, or configure the IP phone to receive and handle the URI from any IP address.

If you are using SIP NOTIFY message method, you do not need to specify the trusted IP address for action URI. But you should enable the IP phone to receive the action URI requests. When the IP phone receives a SIP NOTIFY message with the "Event: ACTION-URI" header from a SIP proxy server for the first time, the touch screen also prompts the message "Allow remote control?".

You can use action URI feature to capture the phone's current screen. For more information, refer to "Scenario A - Capturing the Current Screen of the Phone".

Procedure

Specify the trusted IP address for action URI using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the IP phone to receive the action URI requests. Configure the IP phone to pop up the Allow Remote Control prompt. Specify the trusted IP address(es) for sending the action URI to the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify the trusted IP address(es) for sending the action URI to the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-remotecontrl&q=load

To configure the trusted IP address(es) for action URI via web user interface:

- Click on **Features->Remote Control**.
- Enter the IP address or any in the **Action URI allow IP List** field.

  Multiple IP addresses are separated by commas. If you enter "any" in this field, the IP phone can receive

and handle GET requests from any IP address. If you leave the field blank, the IP phone cannot receive or handle any HTTP GET request.



- Click **Confirm** to save the change.

Scenario A - Capturing the Current Screen of the Phone

You can capture the screen display of the IP phone using the action URI.The IP phone supports handling an HTTP or HTTPS GET request. The URI format is: http(s)://<phoneIPAddress>/screencapture. The captured picture can be saved as a BMP or JPEG file.

You can also use the URI "http(s)://<phoneIPAddress>/screencapture/download" to capture the screen display first, and then download the image (which is saved as a JPG file and named with the phone model and the capture time) to the local system. Before capturing the phone's current screen, ensure that the IP address of the computer is included in the trusted IP address for Action URI on the phone. For more information on the trusted IP address, refer to "Configuring Trusted IP Address for Action URI".

When you capture the screen display, the IP phone may prompt you to enter the user name and password of the administrator if web browser does not remember the user name and password for web user interface login.

IP phone also support capturing the screen display using the old URI "http://<phoneIPAddress>/servlet?command=screenshot".

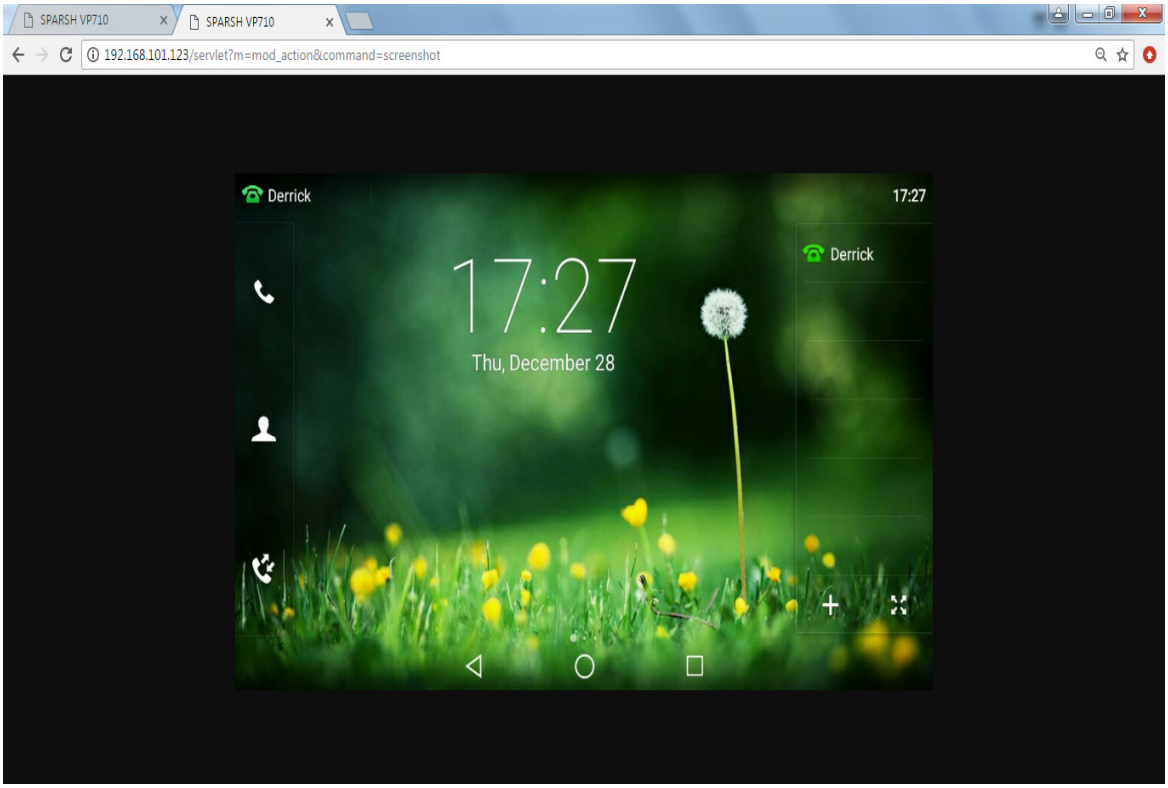
To capture the current screen of the phone:

- Enter request URI (e.g., http://10.10.20.233/servlet?m=mod_action&command=screenshot) in the browser's address bar and press the Enter key on the keyboard.
- Do one of the following:
  - If it is the first time you are capturing the current screen of the phone using the computer, the browser will display "**Remote control forbidden**", and the touch screen will prompt the message "A**llow remote control?**".
  - Tap **OK** on the phone to allow remote control.

- Refresh the web page.
  The browser will display an image showing the current screen of the phone. You can save the image to your local system.

OR

- Else, the browser will directly display an image showing the current screen of the phone.You can save the image to your local system.



Frequent capture may affect the phone performance. MATRIX recommend you to capture the phone screen display within a minimum interval of 4 seconds.

Scenario B - Placing a Call via Web User Interface

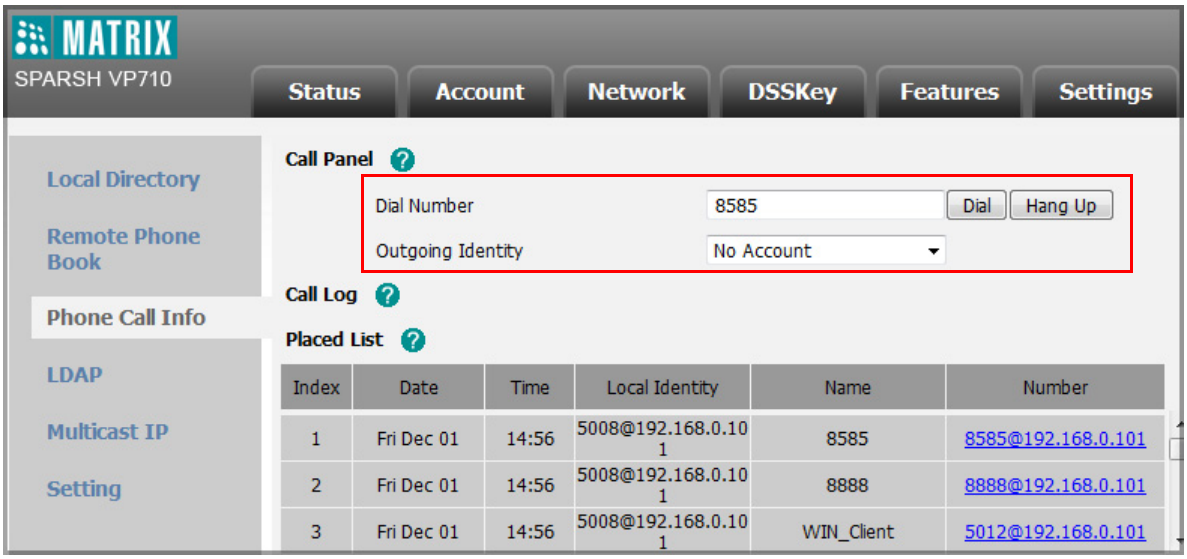
You can place a call via web user interface. Before doing it, ensure that the IP address of your computer is included in the trusted IP address for Action URI on the phone. For more information on the trusted IP address, refer to “Configuring Trusted IP Address for Action URI”.

If you place a call via web user interface but the trusted IP address has not been configured, the web user interface prompts “Call fail”.

To place a call via web user interface:

- Click on **Directory->Phone Call Info**.
- Select the desired account from the **Outgoing Identity** list.

- Enter the number you want to call in the **Dial Number** field.



- Click **Dial** to dial out the number.

  The web user interface prompts “Call Success” and the phone will automatically dial out the number. You can click **Cancel** to end the call.

  If it is the first time you place a call via web user interface, the touch screen will prompt the message “Allow remote control?”. Tap **OK** on the phone to allow remote control and then the phone will automatically dial out the number.

You can also place an IP direct call via web user interface. The IP phone supports either IPv4 or IPv6 address.

Action URL

Action URL allows the IP phone to interact with web server applications by sending an HTTP or HTTPS GET request. You can specify a URL that triggers a GET request when a specified event occurs. Action URL can only be triggered by the pre-defined events (e.g., Open DND). The valid URL format is: *http(s)://<serverIPAddress>/help.xml?*.

The following table lists the pre-defined events for action URL.

Event	Description
Setup Completed	When the IP phone completes startup.
Registered	When the IP phone successfully registers an account.
Unregistered	When the IP phone logs off the registered account.
Register Failed	When the IP phone fails to register an account.
Off Hook	When the IP phone is off hook.
On Hook	When the IP phone is on hook.
Incoming Call	When the IP phone receives an incoming call.
Outgoing Call	When the IP phone places a call.
Established	When the IP phone establishes a call.
Terminated	When the IP phone terminates a call.
Open DND	When the IP phone enables the DND mode.
Close DND	When the IP phone disables the DND mode.
Open Always Forward	When the IP phone enables the always forward.
Close Always Forward	When the IP phone disables the always forward.
Open Busy Forward	When the IP phone enables the busy forward.
Close Busy Forward	When the IP phone disables the busy forward.
Open No Answer Forward	When the IP phone enables the no answer forward.
Close No Answer Forward	When the IP phone disables the no answer forward
Transfer Call	When the IP phone transfers a call.
Blind Transfer	When the IP phone blind transfers a call.
Attended Transfer	When the IP phone performs the semi-attended/attended transfer.
Hold	When the IP phone places a call on hold.
UnHold	When the IP phone retrieves a hold call.
Held	When a call of the IP phone is held.

UnHeld	When a held call is resumed.
Mute	When the IP phone mutes a call.
UnMute	When the IP phone un-mutes a call.
Missed Call	When the IP phone misses a call.
IP Changed	When the IP address of the IP phone changes.
Forward Incoming Call	When the IP phone forwards an incoming call.
Reject Incoming Call	When the IP phone rejects an incoming call.
Answer New-In Call	When the IP phone answers a new call.
Transfer Finished	When the IP phone completes to transfer a call.
Transfer Failed	When the IP phone fails to transfer a call.
Idle To Busy	When the state of the IP phone changes from idle to busy
Busy To Idle	When the state of phone changes from busy to idle
Auto Finish	When the IP phone completes auto provisioning via power on.
Open Call Waiting	When the IP phone enables the call waiting.
Close Call Waiting	When the IP phone disables the call waiting.
Headset	When the IP phone presses the HEADSET key.
Handfree	When the IP phone presses the Speakerphone key
Cancel Call Out	When the IP phone cancels an outgoing call in the ring-back state.
Remote Busy	When an outgoing call is rejected.
Call Remote Canceled	When the remote party cancels the outgoing call in the ringing state.

An HTTP or HTTPS GET request may contain variable name and variable value, separated by "=". Each variable value starts with $ in the query part of the URL.

The valid URL format is: http(s)://<serverIPAddress>/help.xml?variable name=$variable value. Variable name can be customized by users, while the variable value is pre-defined.

For example -

A URL "*http://192.168.1.10/help.xml?mac=$mac*" is specified for the event Mute, $mac will be dynamically replaced with the MAC address of the IP phone when the IP phone mutes a call.

The following table lists pre-defined variable values.

Variable Value	Description
$mac	The MAC address of the IP phone
$ip	The IP address of the IP phone
$model	The IP phone model
$firmware	The firmware version of the IP phone
$active_url	The SIP URI of the current account when the IP phone places a call, receives an incoming call or establishes a call.
$active_user	The user part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
$active_host	The host part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
$local	The SIP URI of the caller when the IP phone places a call. The SIP URI of the callee when the IP phone receives an incoming call.
$remote	The SIP URI of the callee when the IP phone places a call. The SIP URI of the caller when the IP phone receives an incoming call.
$display_local	The display name of the caller when the IP phone places a call. The display name of the callee when the IP phone receives an incoming call.
$display_remote	The display name of the callee when the IP phone places a call. The display name of the caller when the IP phone receives an incoming call.
$call_id	The call-id of the active call.
$callerID	The display name of the caller when the IP phone receives an incoming call.
$calledNumber	The phone number of the callee when the IP phone places a call.

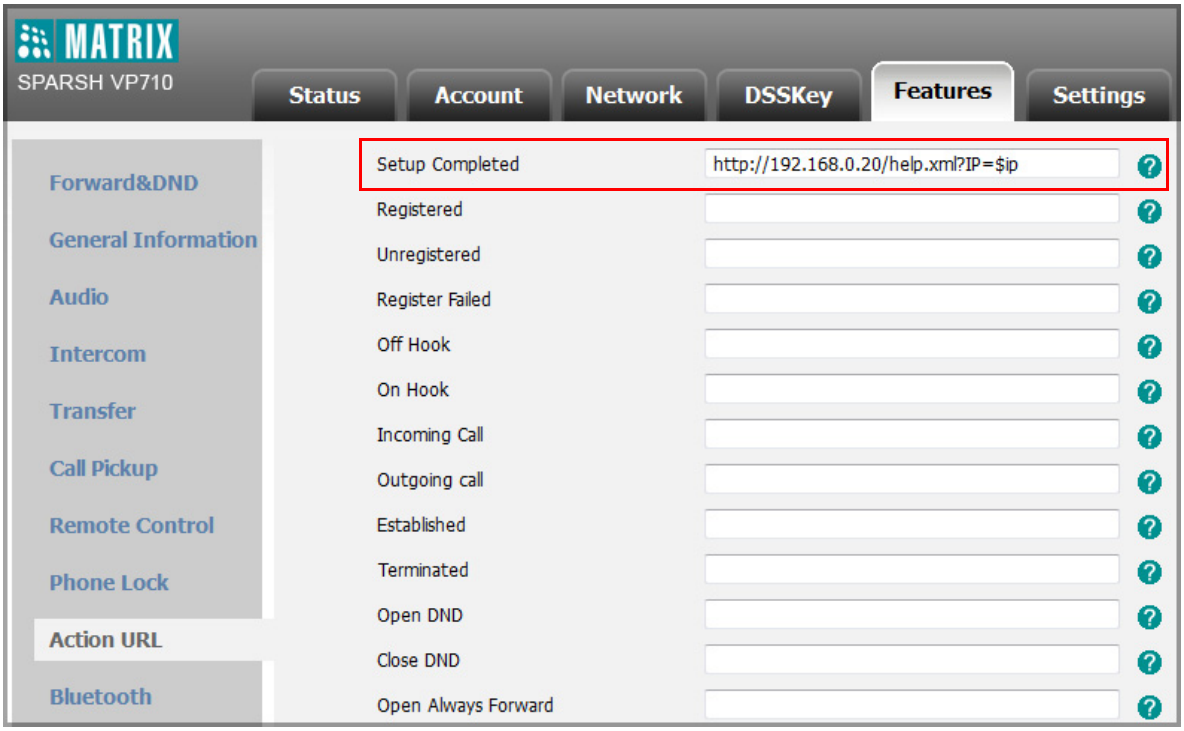
Procedure

Action URL can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure action URL. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure action URL. **Navigate to**: ttp://<phoneIPAddress>/servlet?m=mod_data&p=features-actionurl&q=load

To configure action URL via web user interface:

- Click on **Features->Action URL**.



- Enter the action URLs in the corresponding fields.
- Click **Confirm** to save the change.

Answer By Hand

Answer by hand allows you to answer an incoming call by picking up the handset, pressing the Speakerphone key or pressing the HEADSET key directly.If you disable answer by hand feature, you need to tap the corresponding line key or the **Answer** soft key to answer an incoming call after picking up the handset, pressing the Speakerphone key or pressing the HEADSET key.

Procedure

Answer by hand can be configured using the configuration files.

Configuration File	<MAC>.cfg	Configure answer by hand. For more information, refer "Appendix D - Configuration Parameters".

Busy Lamp Field (BLF)

You can use BLF to monitor a specific line for status changes on the phone. For example, you can configure a BLF key on the phone to monitor the status of a friend's line (busy or idle). The BLF key icon illuminates solid red when the friend's line is in use. For more information, contact your system administrator.

You can use BLF to monitor a specific line for status changes on the phone. BLF List allows a list of specific extensions to be monitored for status changes. It enables the monitoring phone to subscribe to a list of users, and receive notifications of the status of monitored users. Different indicators on the monitoring phone show the status of monitored users. The monitoring user can also be notified about calls being parked/no longer parked against any monitored user. For example, you can configure a BLF key on the phone to monitor the status of a friend's line (busy or idle)

The IP phone supports BLF list using a SUBSCRIBE/NOTIFY mechanism as specified in RFC 3265. This feature depends on support from a SIP server.

Visual Alert and Audio Alert for BLF Pickup

Visual and audio alert for BLF pickup allow the supervisor's phone to play an alert tone and display a visual prompt (e.g., "6001<-6002", 6001 is the monitored extension which receives an incoming call from 6002) when the monitored user receives an incoming call. In addition to the BLF key, visual alert for BLF pickup feature enables the supervisor to pick up the monitored user's incoming call by tapping the Dpickup key. The directed call pickup code must be configured in advance. For more information on how to configure the directed call pickup code for the Dpickup key, refer to "Directed Call Pickup".

Procedure

CLIP can be configured using the configuration files or locally.

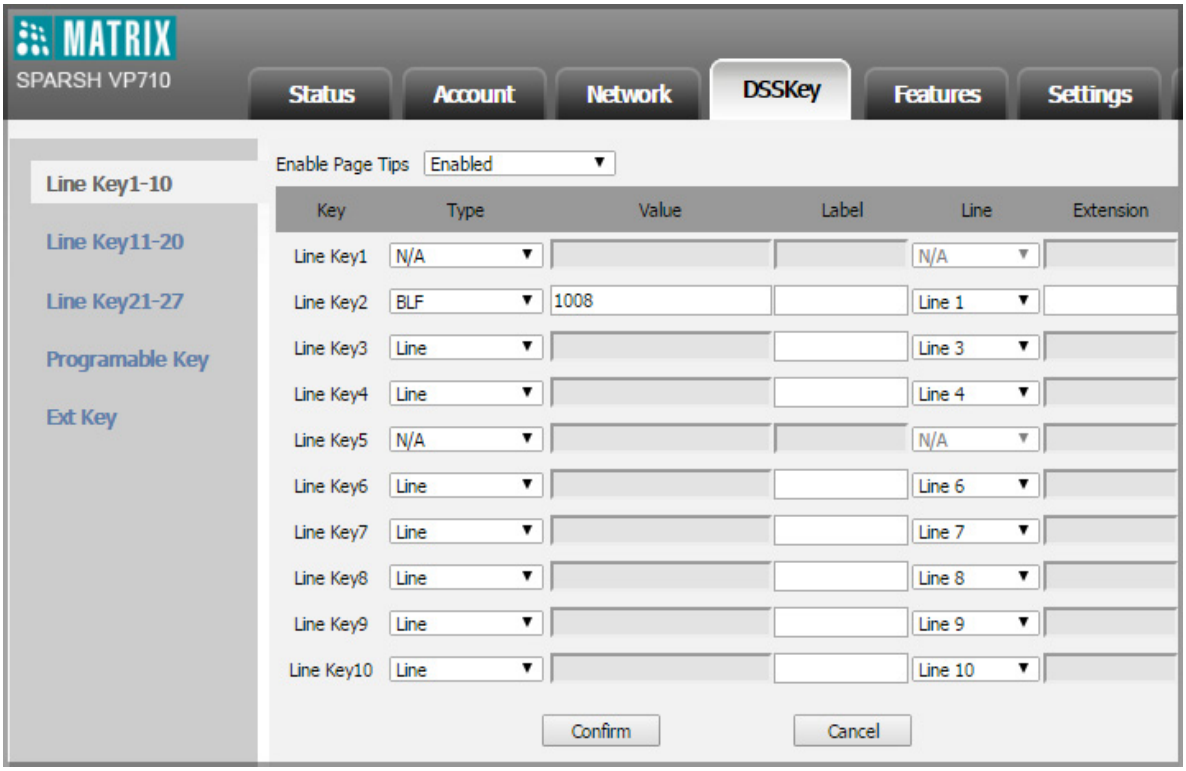
Configuration File	<MAC>.cfg	Specify whether to use visual alert and audio alert for BLF pickup. Assign a BLF key. Configure the period of the BLF subscription. Configure the event of the BLF subscription. Configure whether to handle NOTIFY messages out of the BLF dialog. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Assign a BLF key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load Specify whether to use visual alert and audio alert for BLF pickup **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-callpickup&q=load Configure the period of the BLF subscription. Configure whether to handle NOTIFY messages out of the BLF dialog. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0 Assign a BLF List key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a BLF key.

To configure a BLF key via web user interface:

- Click on **DSSKey->Line Key**.
- In the desired DSS key field, select **BLF** from the Type list.
- Enter the phone number or extension you want to monitor in the **Value** field.
- Select the desired line from the **Line** list.
- Enter the string that will appear on the touch screen in the **Label** field.

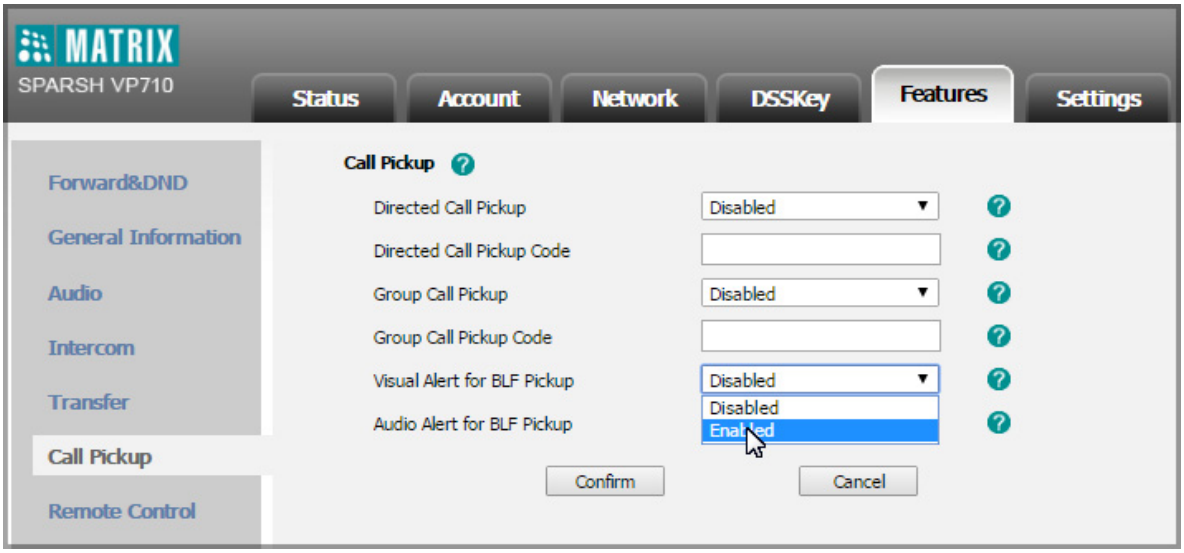
- Enter the directed call pickup code in the **Extension** field.



- Click **Confirm** to save the change.

To configure visual alert and audio alert for BLF pickup via web user interface:

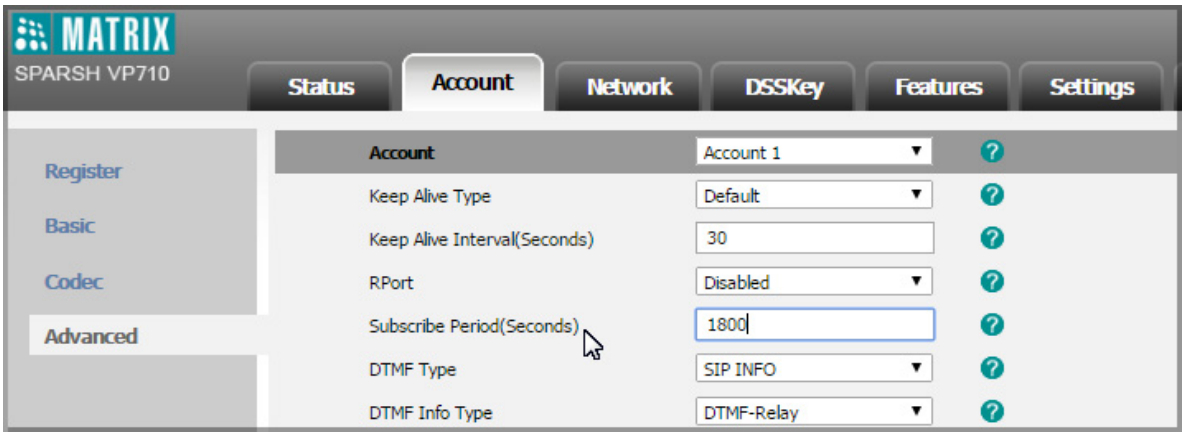
- Click on **Features->Call Pickup**.
- Select the desired value from the **Visual Alert for BLF Pickup** list.
- Select the desired value from the **Audio Alert for BLF Pickup** list.call pickup code in the **Extension** field.



- Click **Confirm** to save the change.

To configure BLF subscription via web user interface:

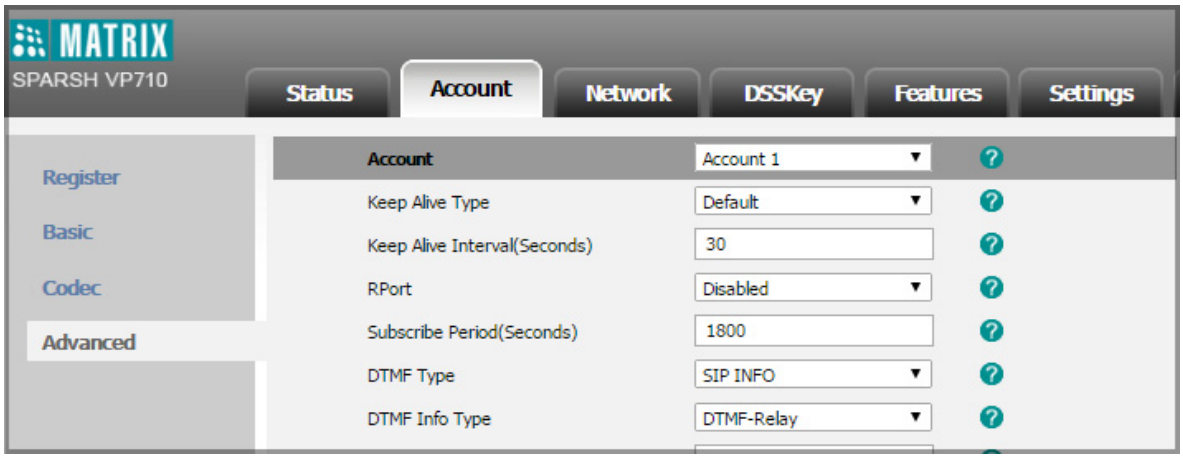
- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.
- Enter the desired period of BLF subscription in the **Subscribe Period(Seconds)** field.

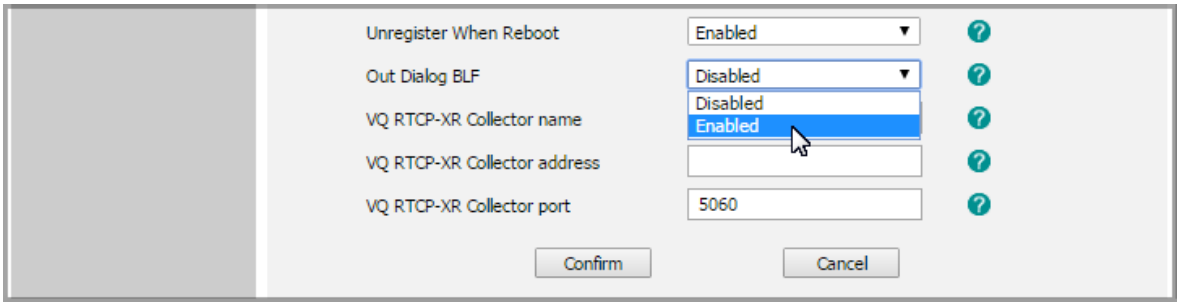


- Click **Confirm** to save the change.

To configure out dialog BLF via web user interface:

- Click on **Account->Advanced.**
- Select the desired account from the **Account** list.
- Select the desired value from the **Out Dialog BLF** list.

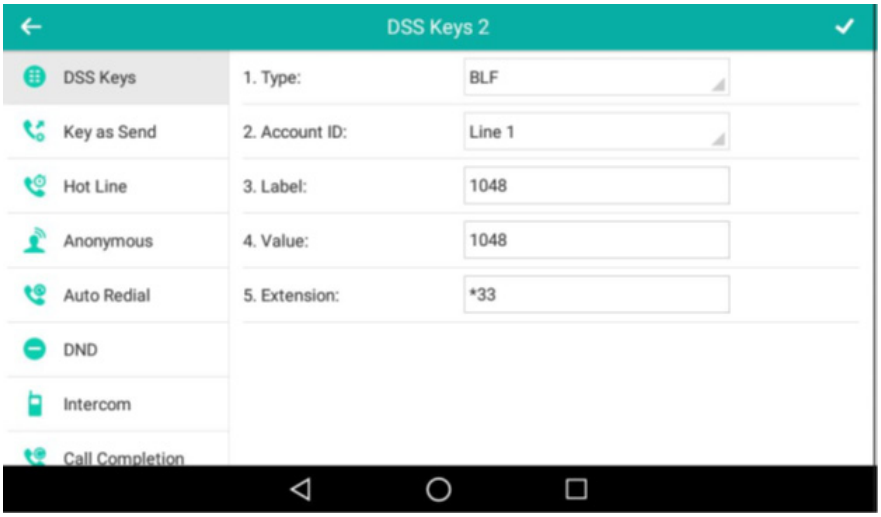




- Click **Confirm** to save the change.

To configure a BLF key via phone user interface:

- Tap **Settings->Features->DSS Keys.**
- Tap the desired DSS key.
- Tap the **Type** field.
- Tap **BLF** in the pop-up dialog box.
- Tap the **Account ID** field.
- Tap the desired line in the pop-up dialog box.
- Enter the string that will appear on the touch screen in the **Label** field.
- Enter the phone number or extension you want to monitor in the **Value** field.
- Enter the directed call pickup code in the **Extension** field.

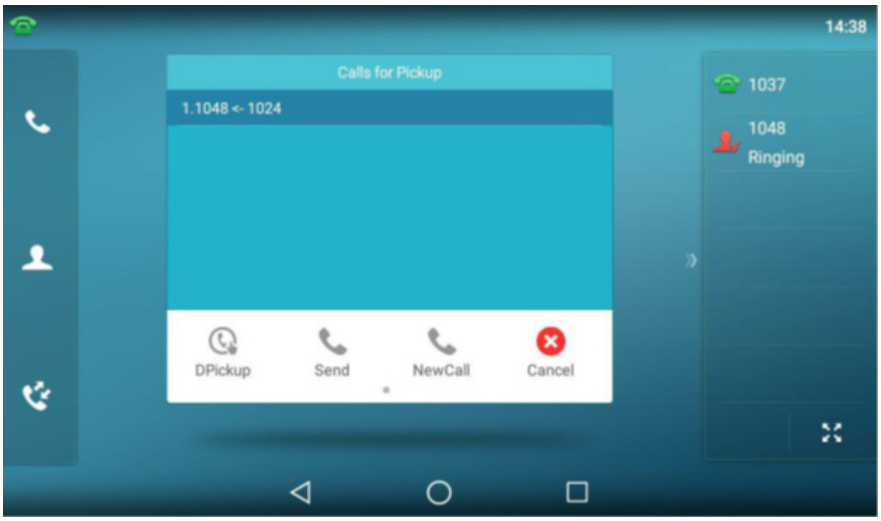


- Tap ✓ to accept the change or ◁ to cancel.

When the monitored line receives an incoming call, the following occurs on your phone:

- The phone plays a warning tone (if enabled).
- The BLF key icon flashes red.
- The caller ID appears on the touch screen (if enabled).

In the following figure, the touch screen shows an incoming call from 1024 on the monitored line.



You can do the following actions:

- Tap the **DPickup** soft key to pick up the call.
- Tap the **Send** soft key to dial out the monitored phone number.
- Tap the **NewCall** soft key to place a new call.
- Tap the **Cancel** soft key to exit the screen.

Busy Tone Delay

Busy tone is audible to the other party, indicating that the call connection has snapped when one party releases the call. Busy tone delay is defined as a period of time during which the busy tone is audible.

Procedure

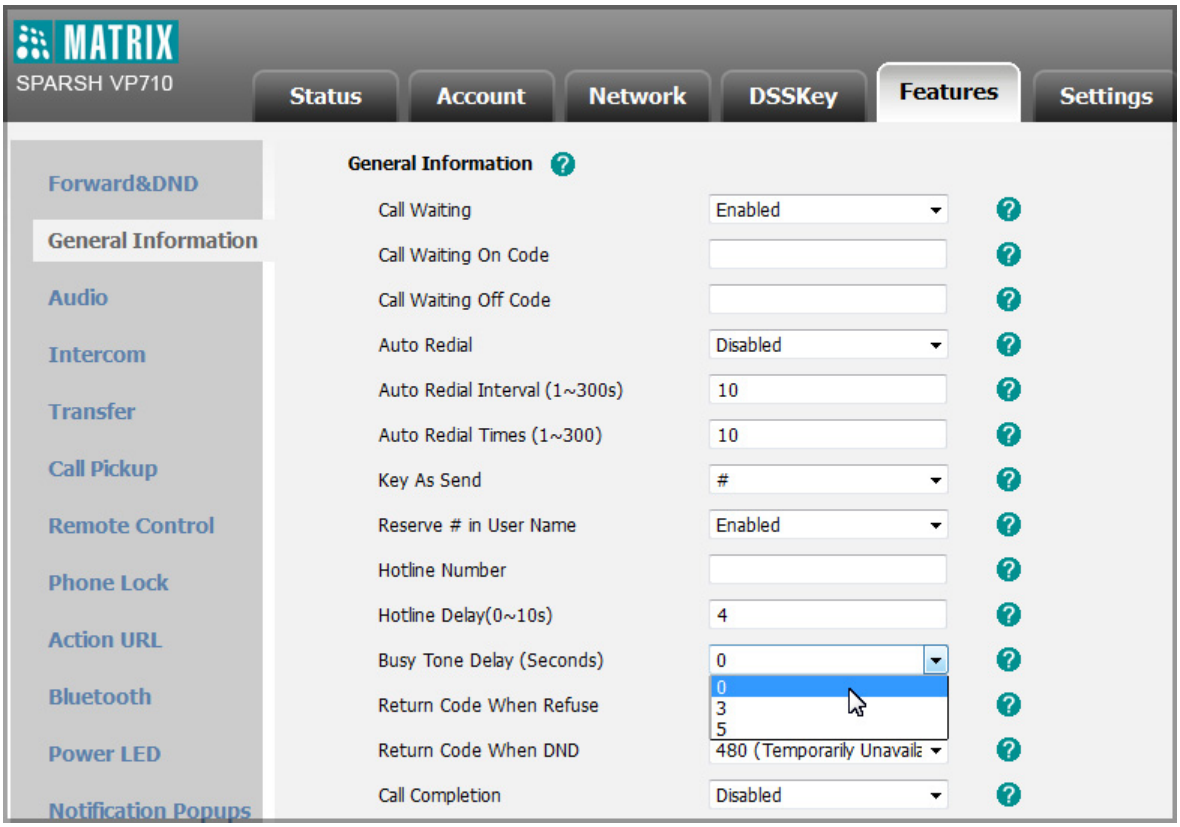
Busy tone delay can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure busy tone delay. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure busy tone delay. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure busy tone delay via web user interface:

- Click on **Features->General Information**.

- Select the desired value from the **Busy Tone Delay (Seconds)** list.



- Click **Confirm** to save the change.

CDP

CDP (Cisco Discovery Protocol) allows the IP phone to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol, and store the information about other devices.
When CDP feature is enabled on the IP phone, the IP phone periodically advertise its own information to the directly connected CDP-enabled switch. The IP phone can also receive CDP packets from the connected switch. When the VLAN configurations on the IP phone are different from the ones sent by the switch, the IP phone perform an update and reboot. This allows the IP phone to be plugged into any switch, obtain their VLAN IDs, and then start communications with the call control.

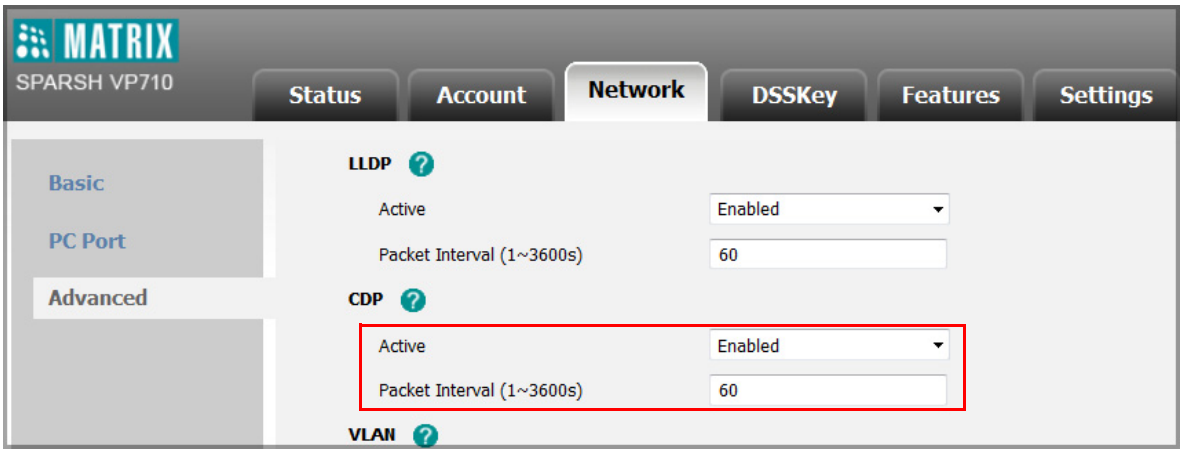
Procedure

CDP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure CDP feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure CDP feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure CDP feature.

To configure CDP via web user interface:

- Click on **Network->Advanced.**

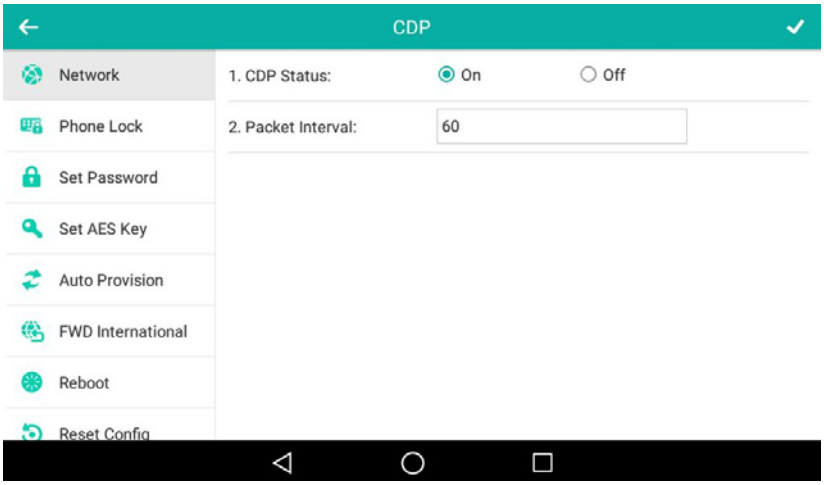


- In the **CDP** block, select the desired value from the **Active** list.
- Enter the desired time interval in the **Packet Interval (1~3600s)** field.
- Click **Confirm** to save the change.

- A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the phone.

To configure CDP feature via phone user interface:

- Tap **Settings->Advanced** (default password: admin) **->Network->CDP->CDP Status.**
- Tap the **On** radio box in the **CDP Status** field.
- Enter the priority value (1-3600s) in the **Packet Interval** field.



- Tap ✓ to save the change or ◁ to cancel.

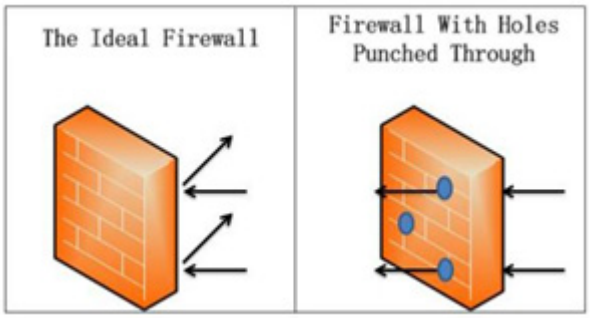
  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot

Configuring the IP Phone for Use with a Firewall or NAT

A firewall protects an organization's IP network by controlling data traffic from outside the network. If your IP phone communicates with other devices through a firewall, you must configure your firewall to allow incoming and outgoing traffic to the IP phone through the reserved ports and the required ports.



You must configure your firewall to allow incoming and outgoing traffic through the following ports:

Port	Port Type	Description
5060	UDP	SIP (default transport protocol)
5060	TCP	SIP (when selecting the TCP transport protocol)
5061	TCP	SIP (when selecting the TLS transport protocol)
50000-50249 (default range)	TCP/UDP	Reserved ports on the IP phone. For more information, refer to "Reserved Ports"

Reserved Ports

By default, the IP phone communicates through UDP ports in the 50000 - 50249 range for video and voice control. The phone uses only a small number of these ports during a call. The exact number depends on the number of participants in the call, the protocol used, and the number of ports required for the type of call: video or voice.

The following tables identify the number of ports required per connection by protocol and the type of call.

Required ports for a SIP two-way call:

Call Type	Number of Required Ports
Video	4 UDP Ports
Voice	2 UDP Ports
Each additional video participant requires 4 UDP ports.	
Each additional audio participant requires 2 UDP ports.	

The following table lists the number of UDP ports needed for the IP phone. This information can help you to determine the range of port number to be entered in the **Reserved Port** field.

Phone	Maximum Connections	Required Ports for a SIP Call	
Matrix SPARSH VP710	Three-way video call and two audio-only calls	16 UDP	50000-50015

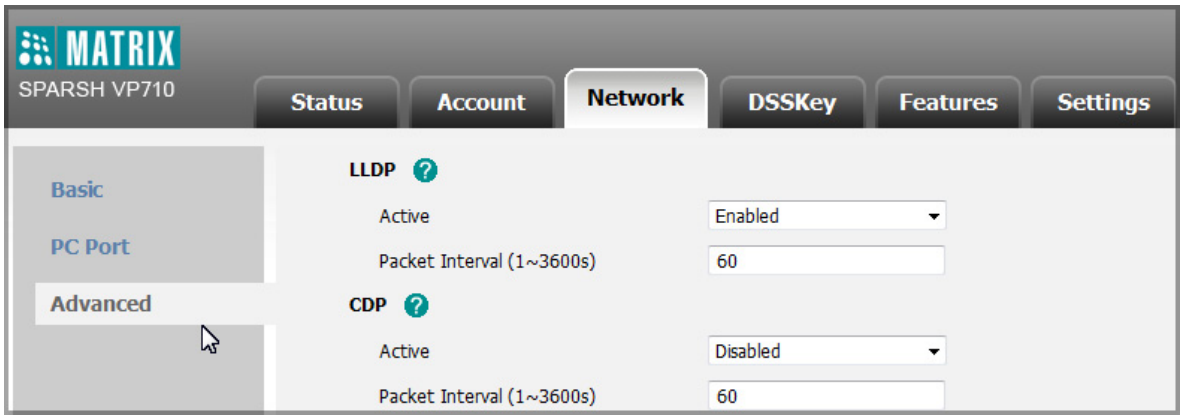
Procedure

Reserved ports can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the range of the UDP ports. Configure the range of the TCP ports. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the range of the UDP ports. Configure the range of the TCP ports. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load

To configure reserved ports via web user interface:

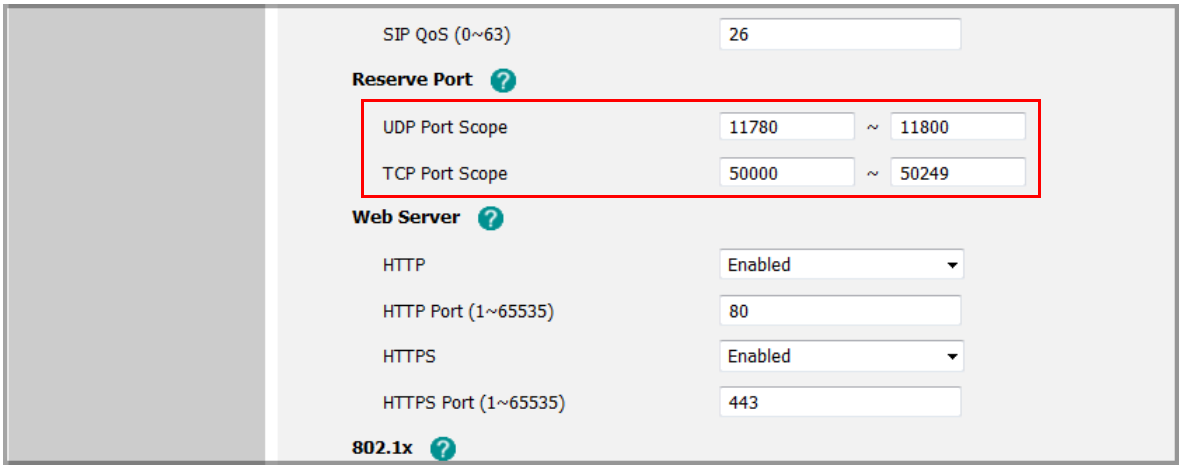
- Click on **Network->Advanced.**



- Enter the desired UDP port scope in the **UDP Port Scope** field.

  Make sure the port range consists of 20 ports for proper functioning of the IP phone.

- Enter the desired TCP port scope in the **TCP Port Scope** field.



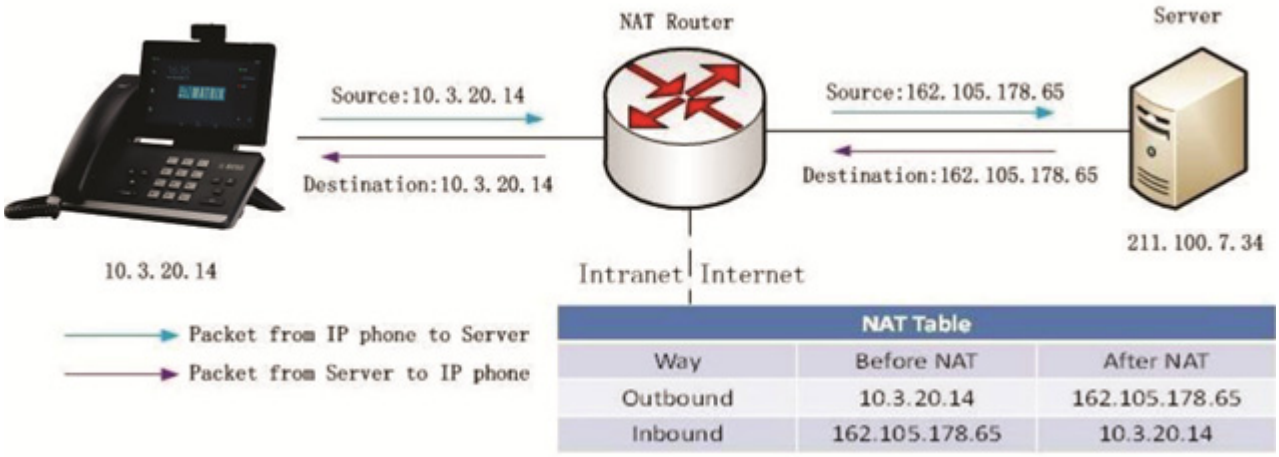
- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the phone.

Network Address Translation (NAT)

Network Address Translation (NAT) is essentially a translation table that maps public IP address and port combinations to private ones. This reduces the need for a large number of public IP addresses. NAT ensures security as each outgoing or incoming request must first go through a translation process. In the VoIP environment, NAT breaks end-to-end connectivity. **Image**



NAT Types

In **Symmetrical NAT**, the NAT router stores the address and port to which it sent the packet. Only packets coming from this address and port are forwarded back to the private address.

In **Full Cone NAT**, all the packets from a private address (e.g., iAddr: port1) to public network is send through a public address (e.g., eAddr: port2). Packets coming from the address of any server to eAddr: port2 will be forwarded back to the private address (e.g., iAddr: port1).

Address Restricted Cone NAT works similar like full cone NAT. A public host (hAddr: any) sends packets to iAddr: port1 through eAddr: port2 only if iAddr: port1 has previously sent a packet to hAddr: any. "Any" means the port number doesn't matter.

Port Restricted Cone NAT works similar like full cone NAT. A public host (hAddr: hPort) sends packets to iAddr: port1 through eAddr: port2 only if iAddr: port1 has previously sent a packet to hAddr: hPort**.**

NAT Traversal

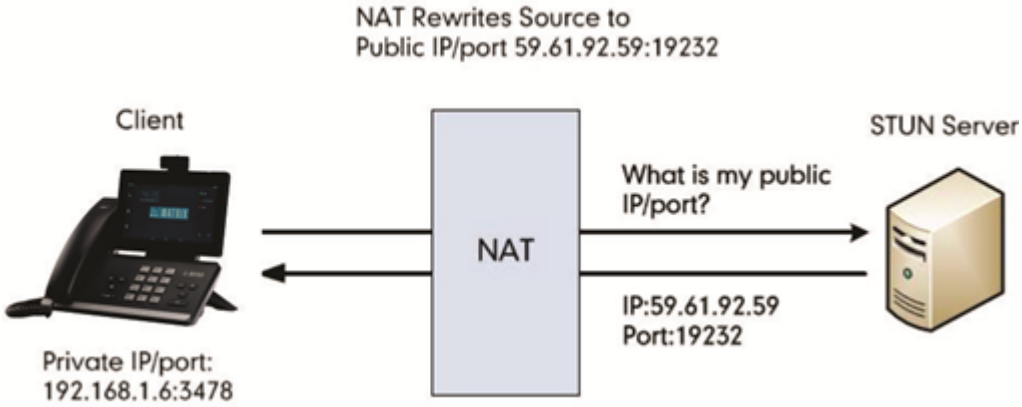
NAT traversal is a general term for techniques that establish and maintain IP connections traversing NAT gateways, typically required for client-to-client networking applications, especially for VoIP deployments.

STUN is one of the NAT traversal techniques supported by the IP phone.

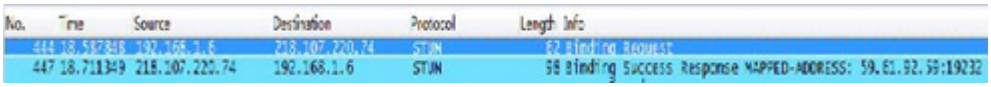
STUN (Simple Traversal of UDP over NATs)

STUN is a network protocol, used in NAT traversal for applications of real-time voice, video, messaging, and other interactive IP communications. The STUN protocol allows entities behind a NAT to first discover the presence and

the type of NAT. For more information on the NAT types, refer to “NAT Types” and to obtain the mapped (public) IP address and port number that the NAT has allocated for the UDP connections to remote parties. The protocol requires assistance from a third-party network server (STUN server) usually located on public Internet. The IP phone can be configured to act as a STUN client, to send exploratory STUN messages to the STUN server. The STUN server uses those messages to determine the public IP address and port used, and then informs the client.



Capturing packets after you enable the STUN feature, you can find that the IP phone sends Binding Request to the STUN server, and then mapped IP address and port is placed in the Binding Response: Binding Success Response MAPPED-ADDRESS: 59.61.92.59:19232.



SIP and TLS Source Ports for NAT Traversal

You can configure the SIP and TLS source ports on the IP Phone. Previously, the IP phone used default values (5060 for UDP/TCP and 5061 for TLS). In the configuration files, you can use the following parameters to configure the SIP and TLS source ports:

- Local SIP Port
- TLS SIP Port

If NAT is disabled, the port number is shown in the Via and Contact SIP headers of SIP messages. If NAT is enabled, the phone uses the NAT port number (and NAT IP address) in the Via and Contact SIP headers of SIP messages, but still use the configured source port.

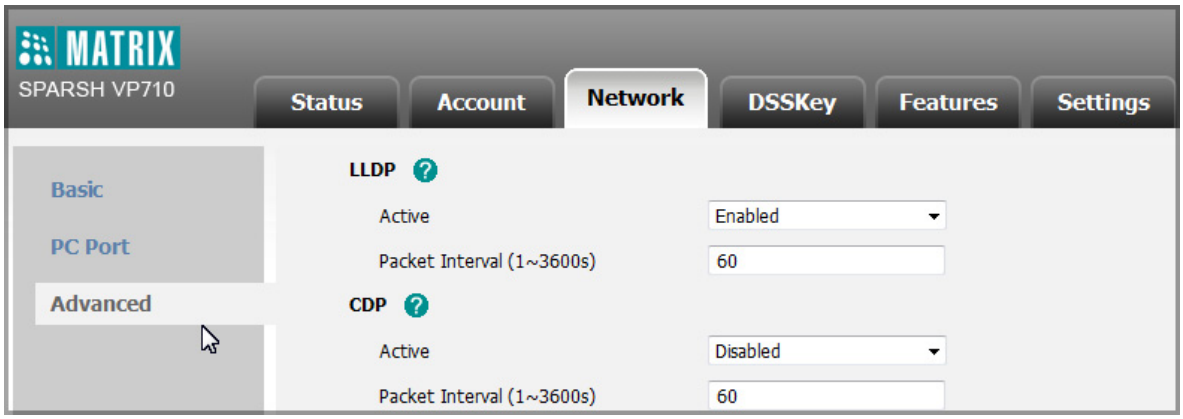
Procedure

NAT traversal and STUN server can be configured using the configuration files or locally.

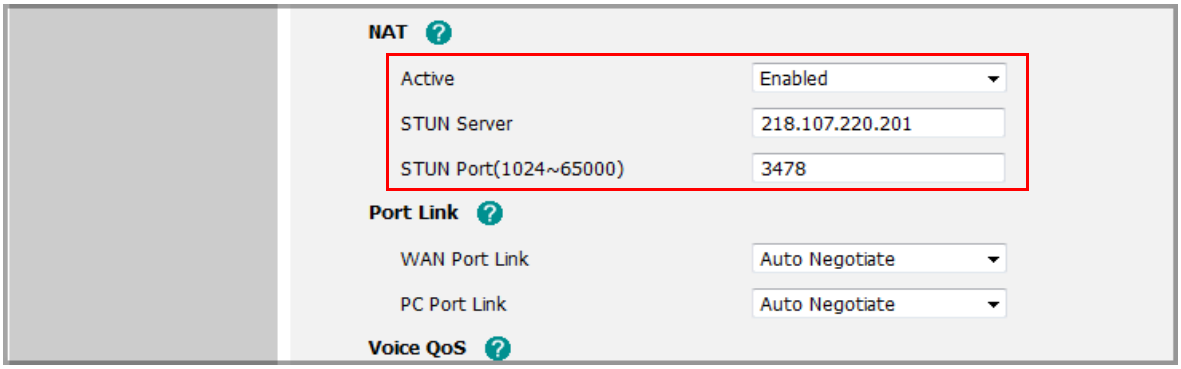
Configuration File	<MAC>.cfg	Configure NAT traversal and STUN server on the phone basis. Configure local SIP port and TLS SIP port. Configure NAT traversal on a per-line basis For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure NAT traversal and STUN server. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load Configure local SIP port and TLS SIP port. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-sip&q=load Configure NAT traversal on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-register&q=load&acc=0
	Phone User Interface	Configure NAT traversal and STUN server on a phone basis. Configure NAT traversal on a per-line basis.

To configure NAT traversal and STUN server via web user interface:

- Click on **Network->Advanced**.



- In the **NAT** block, select the desired value from the **Active** list.
- Enter the IP address or the domain name of the STUN server in the **STUN Server** field.
- Enter the port of the STUN server in the **STUN Port(1024~65000)** field.



- Click **Confirm** to save the change.

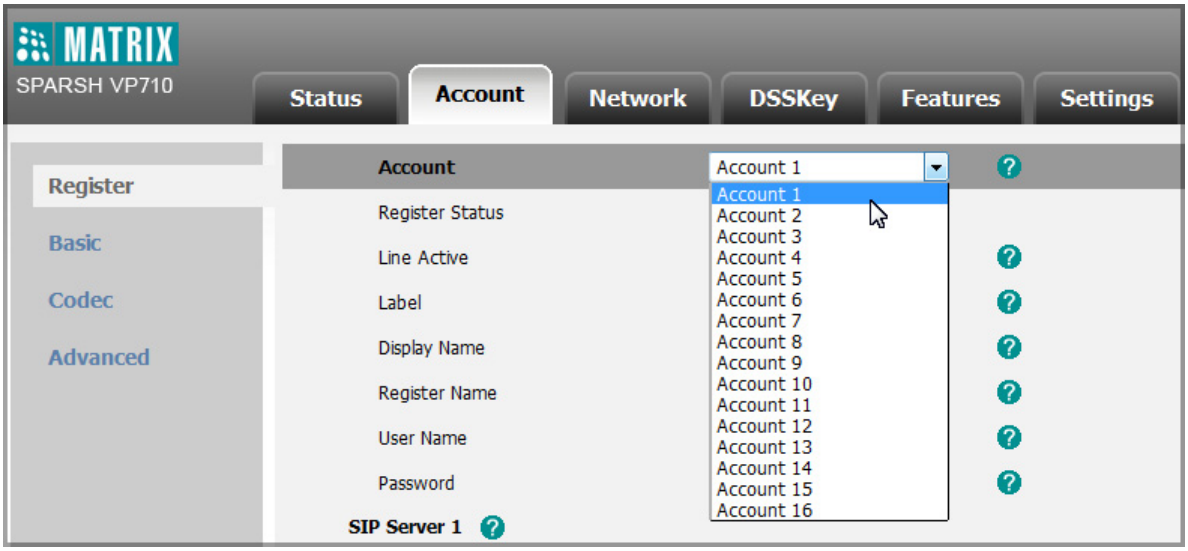
  A dialog box pops up to prompt that settings will take effect after a reboot

- Click OK to reboot the phone

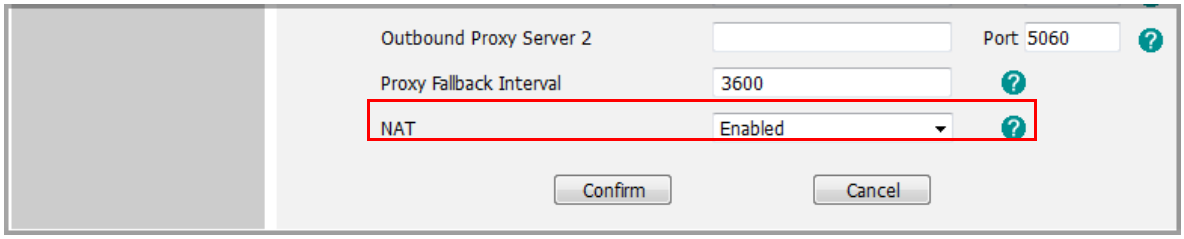
To configure NAT traversal for account via web user interface:

- Click on **Account->Register**.

- Select the desired account from the **Account** list.



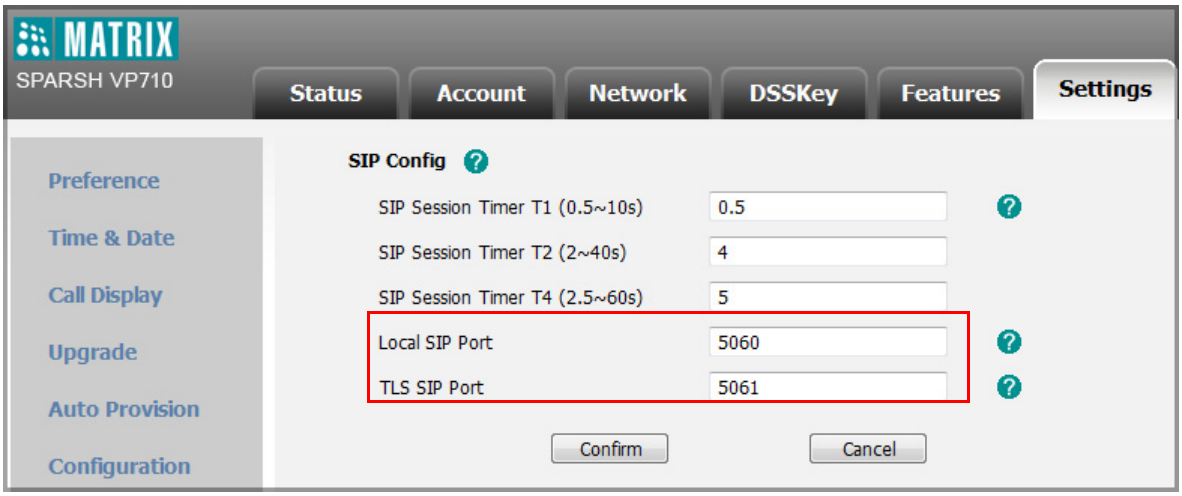
- Select **Enabled** from the **NAT** list.



- Click **Confirm** to save the change.

To configure local SIP port and TLS SIP port via web user interface:

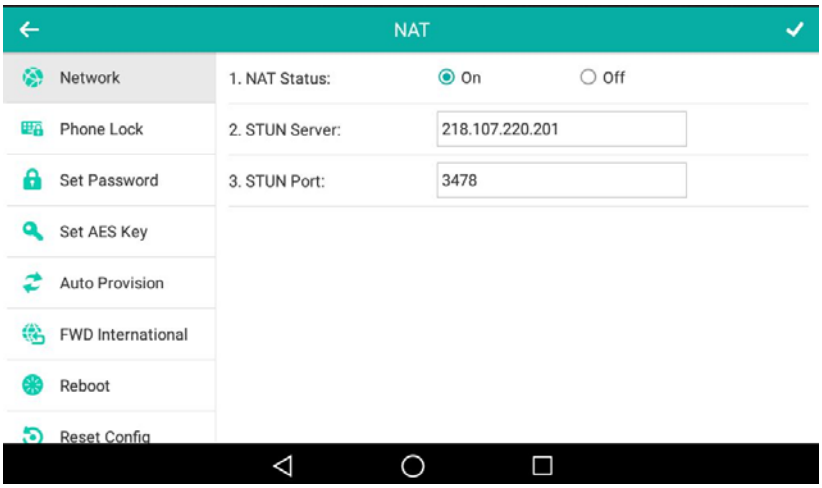
- Click on **Settings->SIP.**
- Enter the desired local SIP port in the **Local SIP Port** field.
- Enter the desired TLS SIP port in the **TLS SIP Port** field.



- Click **Confirm** to save the change.

To configure NAT traversal and STUN server via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->NAT->NAT Status**.
- Tap the **On** radio box in the **NAT Status** field.
- Enter the IP address or the domain name of the STUN server in the **STUN Server** field.
- Enter the port of the STUN server in the **STUN Port** field.



- Tap to save the change or to cancel.

  The phone prompts you to reboot the phone.

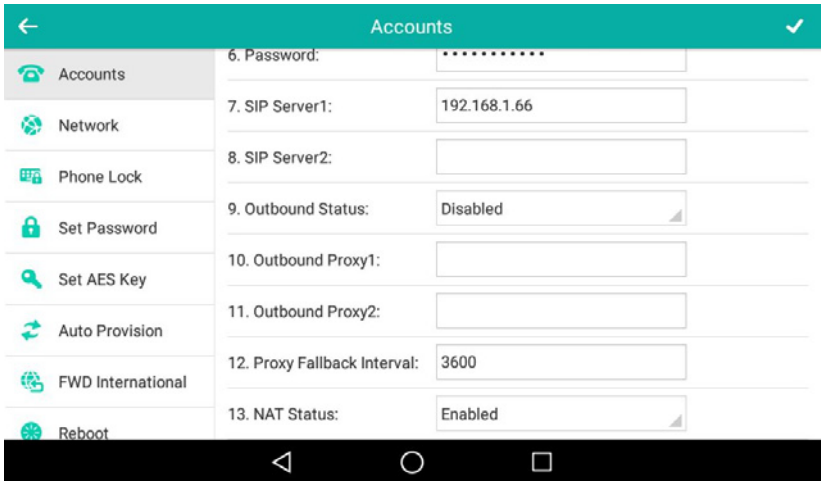
- Tap OK to reboot the phone.

  The settings will take effect after a reboot

To configure NAT traversal for a specific account via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Accounts.**
- Tap the desired account.
- Tap the **NAT Status** field.

- Tap **Enabled** in the pop-up dialog box.



- Tap ✓ to save the change or ◁ to cancel.

Keep Alive

IP phones can send keep-alive packets to the NAT device for keeping the communication port open.

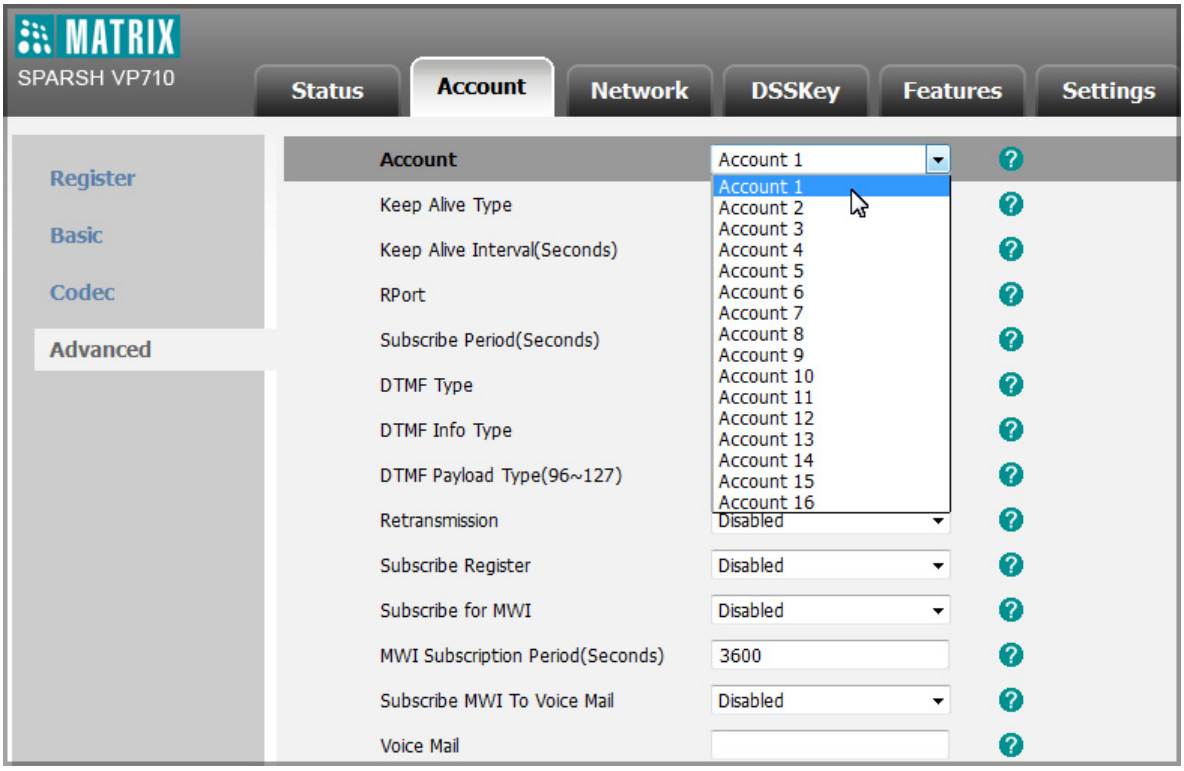
Procedure

Keep alive feature can be configured using the configuration files or locally.

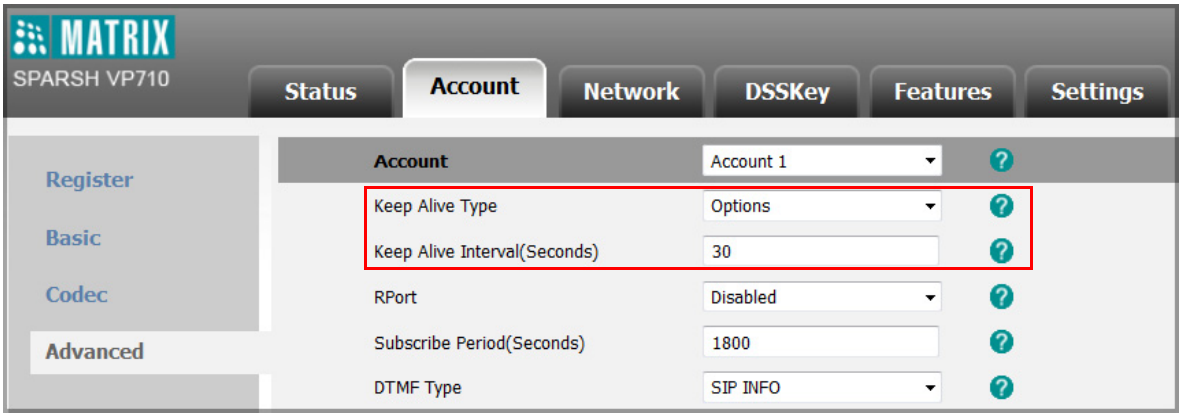
Configuration File	<MAC>.cfg	Configure the type of keep-alive packets on a per-line basis. Configure the keep-alive interval on a per-line basis. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the type of keep-alive packets on a per-line basis. Configure the keep-alive interval on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure the type of keep-alive packets and keep-alive interval via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Keep Alive Type** list.
- Enter the keep-alive interval in the **Keep Alive Interval(Seconds)** field.



- Click **Confirm** to save the change.

Rport

The Session Initiation Protocol (SIP) operates over UDP and TCP. When used with UDP, responses to requests are returned to the source address the request came from, and returned to the port written into the topmost “Via” header of the request message. However, this behavior is not desirable when the client is behind a Network Address Translation (NAT) or firewall. So a new parameter “rport” for the “Via” header field is required.

Rport described in RFC 3581, allows a client to request that the server sends the response back to the source port from which the request came. Rport feature depends on support from a SIP server.

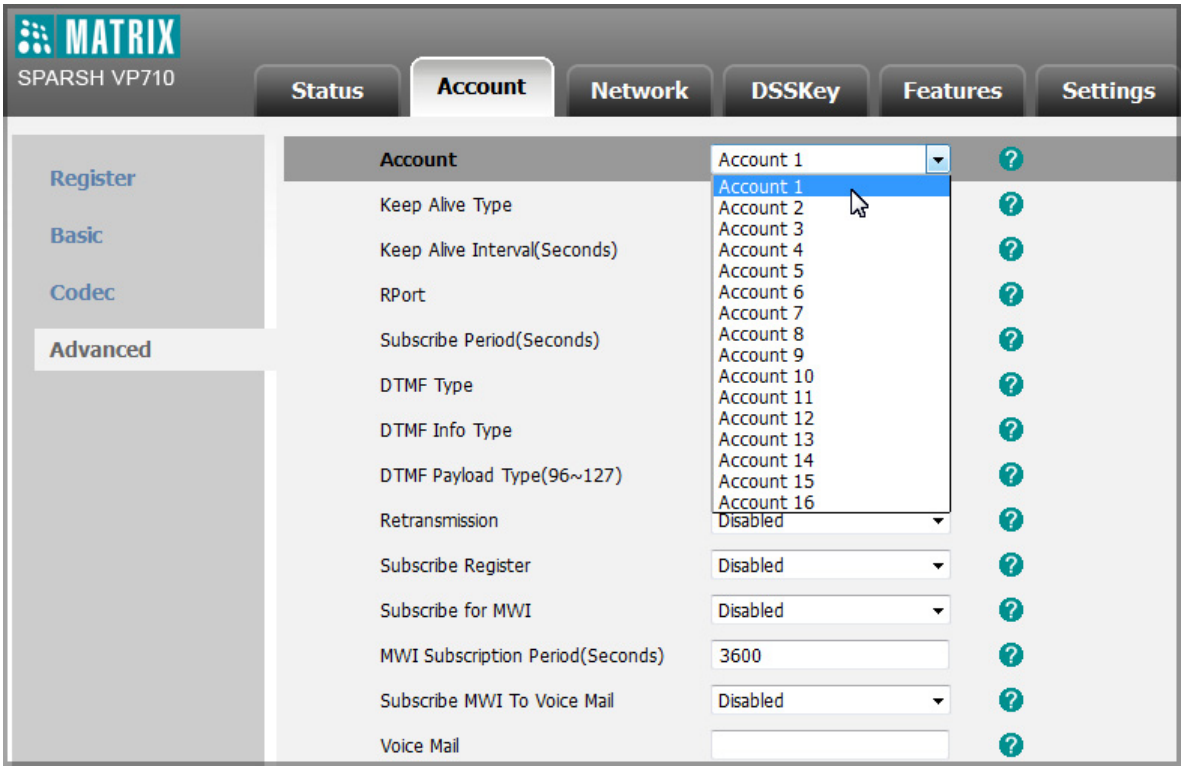
Procedure

Rport feature can be configured using the configuration files or locally.

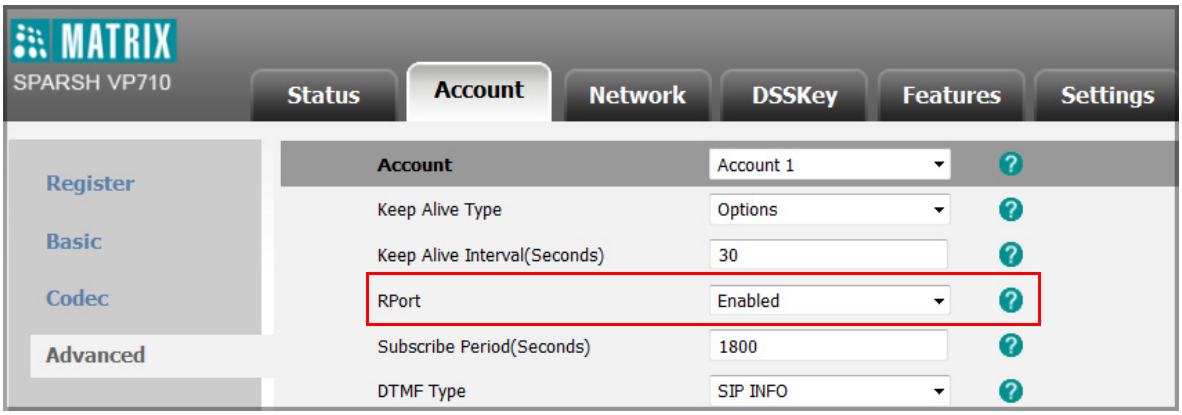
Configuration File	<MAC>.cfg	Configure NAT Rport feature on a per-line basis. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure NAT Rport feature on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure Rport feature via web user interface:

- Click on **Account->Advanced.**
- Select the desired account from the **Account** list.



- Select the desired value from the **RPort** list.



- Click **Confirm** to save the change.

Distinctive Ring Tones

This feature allows certain incoming calls to trigger the IP phone to play Distinctive ring tones.

The IP phone inspects the INVITE request for an "Alert-Info" header when receiving an incoming call. If the INVITE request contains an "Alert-Info" header, the IP phone strips out the URL or keyword parameter and maps it to the appropriate ring tone.



If the caller is present in the local directory, the ring tone assigned to the caller should be preferentially played.

Following is the format of Alert-Info headers:

- Alert-Info: Bellcore-drN
- Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)
- Alert-Info: <URL>
- Alert-Info: info=info text;x-line-id=0

Alert-Info: Bellcore-drN

If the Alter-Info header contains the keyword “Bellcore-drN”, the IP phone will play the desired ring tone.
The following table identifies the corresponding ring tone:

Value of N	Ring Tone (features.alert_info_tone = 1)	Ring Tone (features.alert_info_tone = 0)
1	Bellcore-dr1	Ring1.wav
2	Bellcore-dr2	Ring2.wav
3	Bellcore-dr3	Ring3.wav
4	Bellcore-dr4	Ring4.wav
5	Bellcore-dr5	Ring5.wav
6	Ring6.wav	
7	Ring7.wav	
8	Ring8.wav	
9	Silent.wav	
10	Splash.wav	
N<1 or N>10	Ring1.wav	

Examples:

- Alert-Info: http://127.0.0.1/Bellcore-dr1

- Alert-Info: test/Bellcore-dr1
- Alert-Info: Bellcore-dr1
- Alert-Info: Bellcore-dr1;x-line-id=1
- Alert-Info: <http://10.1.0.31>;info=Bellcore-dr1

The following table identifies the different Bellcore ring tone patterns and cadences.
(These ring tones are designed for the BroadWorks server).

Bellcore Tone	Pattern ID	Pattern	Cadence	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
Bellcore-dr1 (standard)	1	Ringing		1800	2000	2200
		Silent		3600	4000	4400
Bellcore-dr2	2	Ringing	Long	630	800	1025
		Silent		315	400	525
		Ringing	Long	630	800	1025
		Silent		3475	4000	4400
Bellcore-dr3	3	Ringing	Short	315	400	525
		Silent		145	200	525
		Ringing	Short	315	400	525
		Silent		145	200	525
		Ringing	Long	630	800	1025
		Silent		2975	4000	4400
Bellcore-dr4	4	Ringing	Short	200	300	525
		Silent		145	200	525
		Ringing	Long	800	1000	1100
		Silent		145	200	525
		Ringing	Short	200	300	525
		Silent		2975	4000	4400
Bellcore-dr5	5	Ringing		450	500	550



If you are waiting for a call, "Bellcore-dr5" is a ring splash tone that reminds you that the DND or Always Call Forward feature is enabled on the server side.

Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)

If the Alter-Info header contains the keyword “ringtone-N” or “MyMolodyN”, the IP phone will play the corresponding local ring tone (RingN.wav), or play the first local ring tone (Ring1.wav) in about 10 seconds if “N” is greater than 10 or less than 1.

Examples:

- Alert-Info: ringtone-2
- Alert-Info: ringtone-2;x-line-id=1
- Alert-Info: <http://10.1.0.31>;info=ringtone-2
- Alert-Info: <http://127.0.0.1/ringtone-2>
- Alert-Info: MyMelody2
- Alert-Info: MyMelody2;x-line-id=1
- Alert-Info: <http://10.1.0.31>;x-line-id=0;info=MyMelody2

The following table identifies the corresponding local ring tone:

Value of N	Ring Tone
1	Ring1.wav
2	Ring2.wav
3	Ring3.wav
4	Ring4.wav
5	Ring5.wav
6	Ring6.wav
7	Ring7.wav
8	Ring8.wav
9	Silent.wav
10	Splash.wav
N<1 or N>10	Ring1.wav

Alert-Info: <URL>

If the Alert-Info header contains a remote URL, the IP phone will try to download the WAV ring tone file from the URL and then play the remote ring tone
if the value of the parameter “account.X.alert_info_url_enable” is set to 1 (or “Distinctive Ring Tones” is Enabled on the web user interface), or play the preconfigured local ring tone in about 10 seconds if the value of the parameter “account.X.alert_info_url_enable” is set to 0 or if the IP phone fails to download the remote ring tone.

Example:

- Alert-Info: http://192.168.0.12:8080/Custom.wav

Alert-Info: info=info text;x-line-id=0

If the Alert-Info header contains an info text, the IP phone will map the text with the Internal Ringer Text preconfigured (or the value of the parameter “distinctive_ring_tones.alert_info.X.text” is configured) on the IP phone, and then play the ring tone associated with the Internal Ringer Text (the ring tone can be configured by the parameter “distinctive_ring_tones.alert_info.X.ringer”). If no internal ringer text maps, the IP phone will play the preconfigured local ring tone in about 10 seconds.

Example:

- Alert-Info: info=family;x-line-id=0
- Alert-Info: <http://10.1.0.31>;info=family
- Alert-Info: <http://10.1.0.31>;info=family;x-line-id=0

The IP phone will answer the incoming calls automatically without playing the ring tone, if the INVITE request contains the following type of strings:

- Alert-Info: Auto Answer
- Alert-Info: info = alert-autoanswer
- Alert-Info: answer-after = 0 (or Alert-Info: Answer-After = 0)

If enable auto answer tone feature is enabled, the phone plays a warning tone to alert you before answering the incoming call. For more information on Enable auto answer tone, refer to “Auto Answer”.

If the Alert-Info header contains multiple types of keywords, the IP phone will process the keywords in the following order: AutoAnswer>URL>info

Procedure

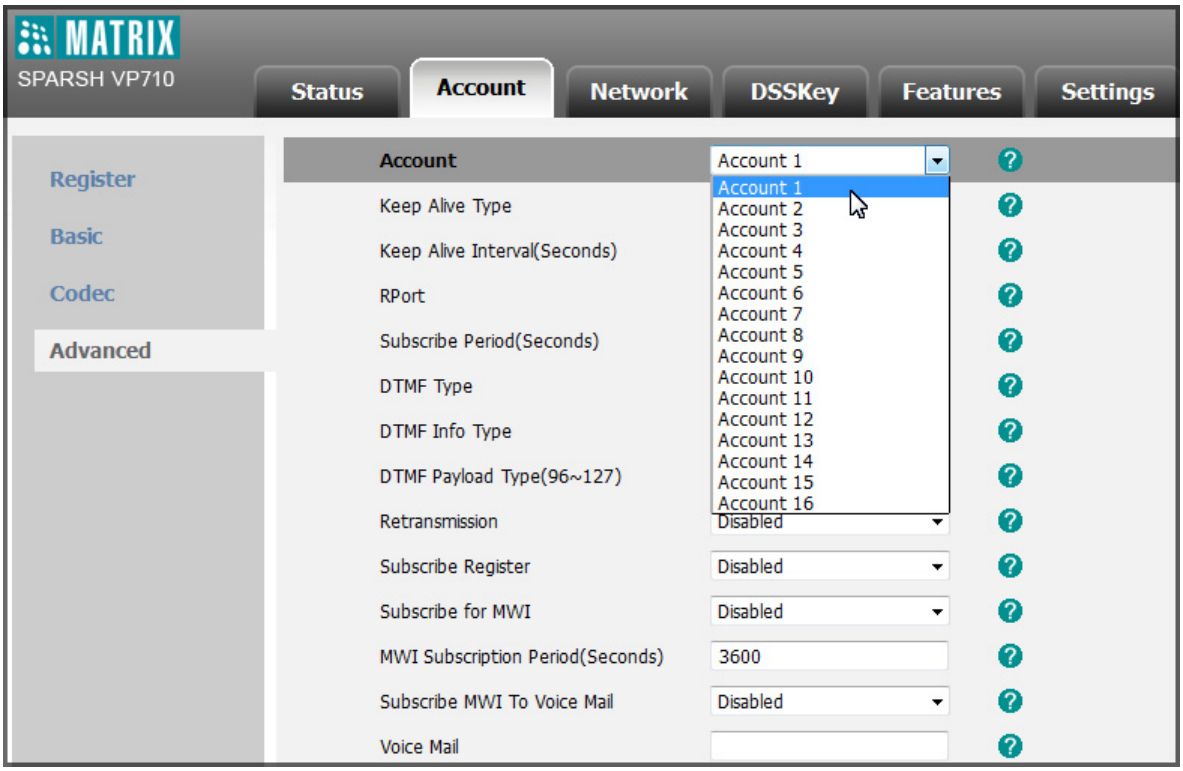
Distinctive ring tones can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure distinctive ring tones. Configure the internal ringer text and internal ringer file. For more information, refer “Appendix D - Configuration Parameters”.

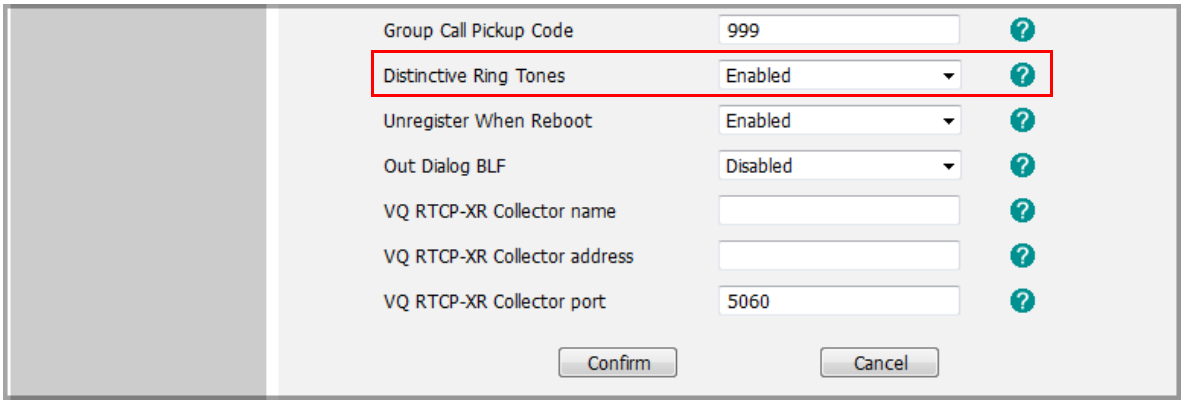
Local	Web User Interface	Configure distinctive ring tones. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_data&p=account-adv&q=load&acc=0 Configure the internal ringer text and internal ringer file. **Navigate to:** http://<phoneIPAddress>/ servlet?m=mod_data&p=settings-ring&q=load

To configure distinctive ring tones via web user interface:

- Click on **Account-> Advanced**.
- Select the desired account from the **Account** list.



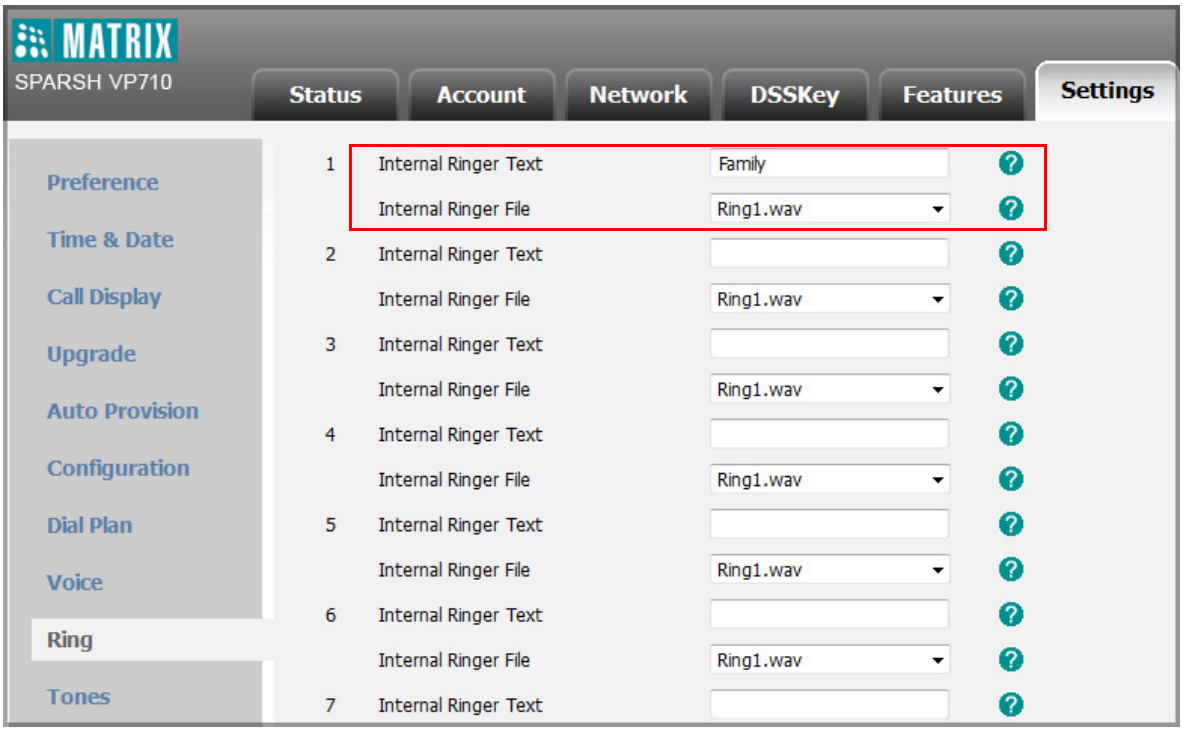
- Select the desired value from the **Distinctive Ring Tones** list.



- Click **Confirm** to save the change.

To configure the internal ringer text and internal ringer file via web user interface:

- Click on **Settings->Ring**
- Enter the keywords in the **Internal Ringer Text** fields.
- Select the desired ring tones for each text from the **Internal Ringer File** list.



- Click **Confirm** to save the change.

DTMF

DTMF (Dual Tone Multi-frequency) also known as touch-tone is used for telecommunication signaling over analog telephone lines in the voice-frequency band. DTMF is the signal sent from the IP phone to the network, which is generated when the keypad of the IP phone is pressed during a call.
Each key pressed on the IP phone generates one sinusoidal tone of two frequencies. One is generated from a high frequency group and the other from a low frequency group.

The DTMF keypad is laid out in a 4 × 4 matrix, with each row representing a low frequency, and each column representing a high frequency. Pressing a digit key (such as '1') will generate a sinusoidal tone for each of the two frequencies (697 and 1209 hertz (Hz)).

DTMF Keypad Frequencies:

	1209 Hz	1336 Hz	1447 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Methods of Transmitting DTMF Digits on SIP Calls:

The method of transmitting DTMF digits is configurable on a per-line basis.

- **RFC 2833** -- DTMF digits are transmitted by RTP Events compliant to RFC 2833.
- **INBAND** -- DTMF digits are transmitted in the voice band.
- **SIP INFO** -- DTMF digits are transmitted by SIP INFO messages.

RFC 2833

DTMF digits are transmitted using the RTP Event packets that are sent along with the voice path. These packets use RFC 2833 format and must have a payload type that matches what the other end is listening for. The payload type for RTP Event packets is configurable. IP phones default to 101 for the payload type, which use the definition to negotiate with the other end during call establishment.

The RTP Event packet contains 4 bytes. The 4 bytes are distributed over several fields denoted as Event, End bit, R-bit, Volume and Duration. If the End bit is set to 1, the packet contains the end of the DTMF event. You can configure the sending times of the end RTP Event packet.

INBAND

DTMF digits are transmitted within the audio of the IP phone conversation. It uses the same codec as your voice and is audible to conversation partners.

SIP INFO

DTMF digits are transmitted by the SIP INFO messages when the voice stream is established after a successful SIP 200 OK-ACK message sequence. The SIP INFO message is sent along the signaling path of the call. The SIP INFO message can transmit DTMF digits in three ways: DTMF, DTMF-Relay and Telephone-Event.

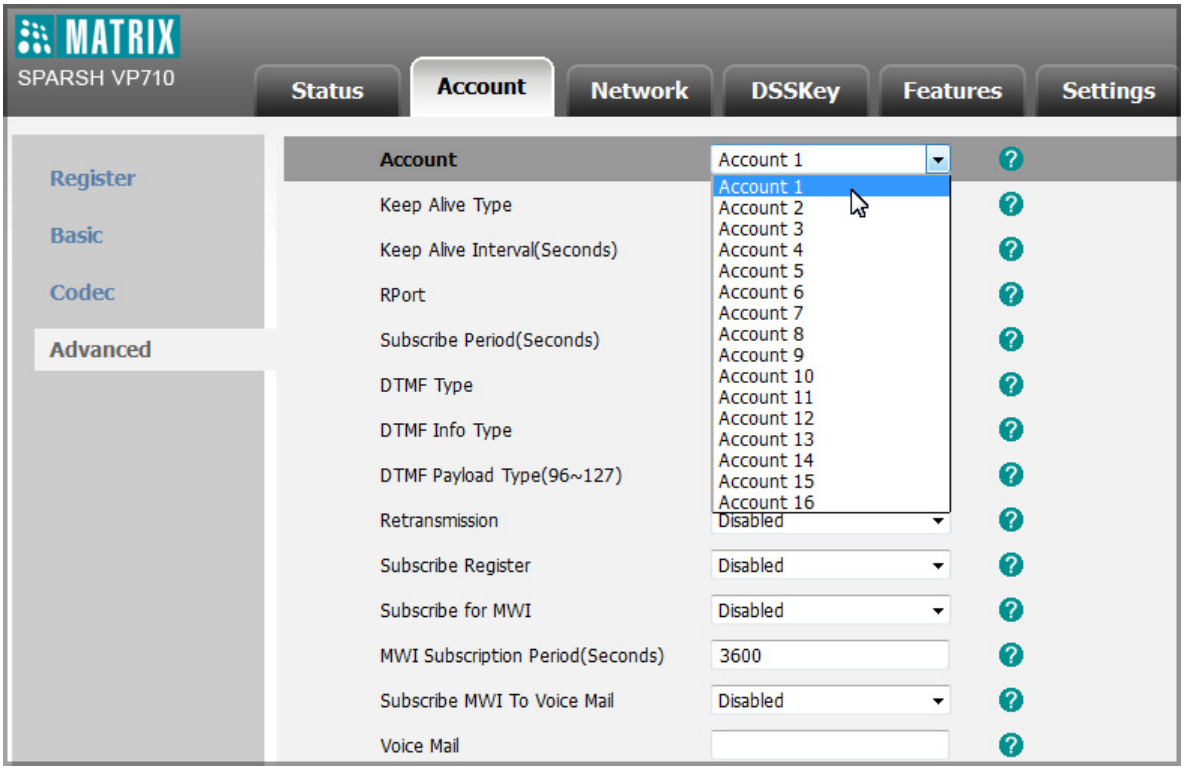
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the method of transmitting DTMF digit and the payload type. Configure the number of times for the IP phone to send the end RTP Event packet. Configure the duration time for DTMF. Configure the frequency level of DTMF digits. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the method of transmitting DTMF digits and the payload type. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0 Configure the number of times for the IP phone to send the end RTP Event packet. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure the method of transmitting DTMF digits via web user interface:

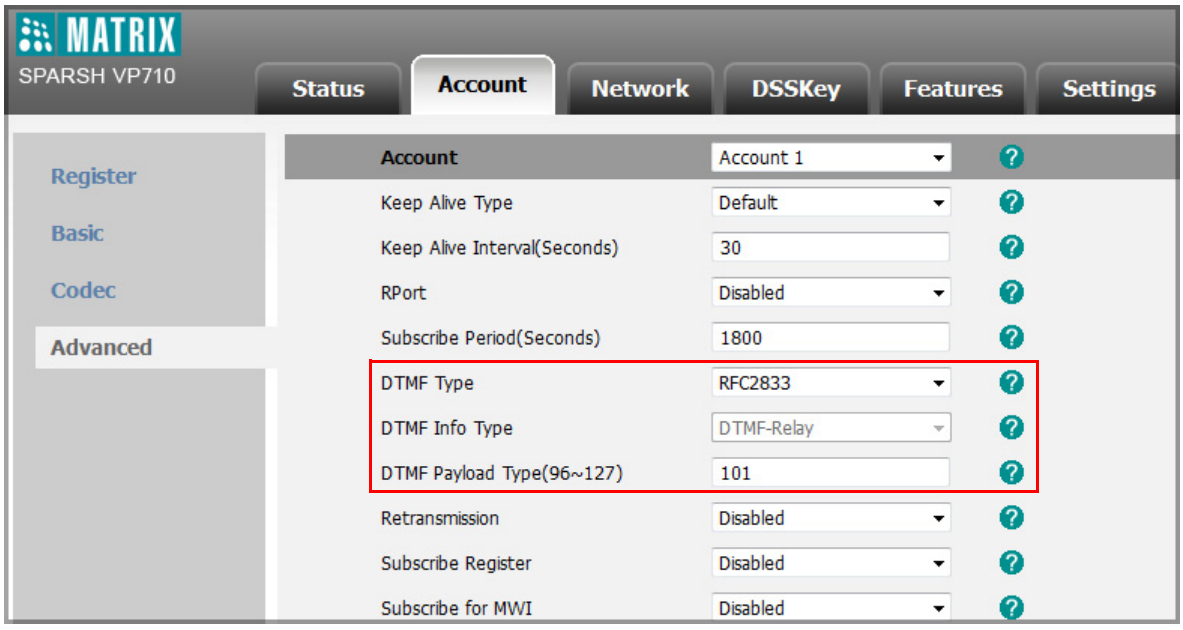
- Click on **Account-> Advanced**.
- Select the desired account from the of **Account** list.



- Select the desired value from the **DTMF Type** list.

  If **SIP INFO** or **RFC2833 + SIP INFO** is selected, select the desired value from the **DTMF Info Type** list.

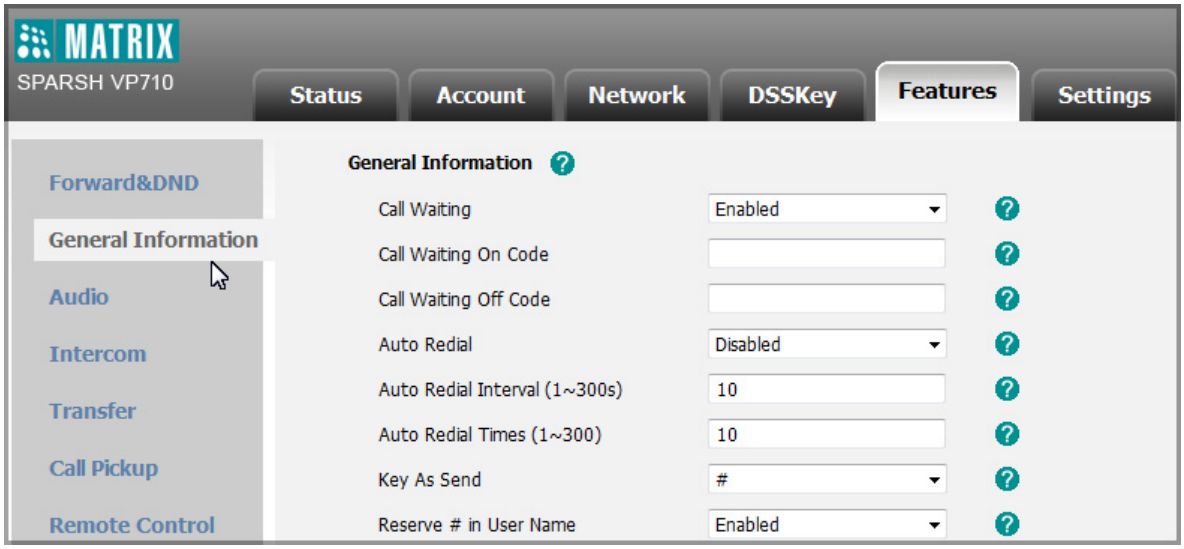
- Enter the desired value in the **DTMF Payload Type (96~127)** field.



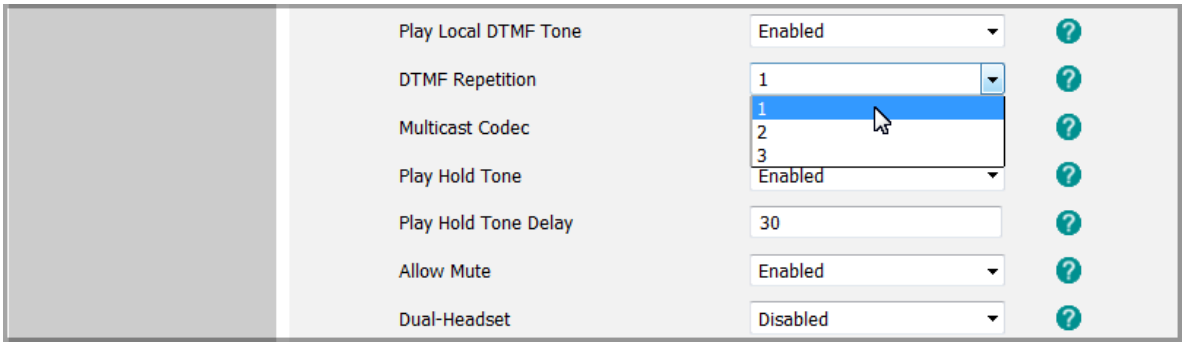
- Click **Confirm** to save the change.

To configure the number of times to send the end RTP Event packet via web user interface:

- Click on **Features->General Information**.



- Select the desired value (1-3) from the **DTMF Repetition** list.



- Click **Confirm** to save the change.

Suppress DTMF Display

Suppress DTMF display allows the IP phone to suppress the display of DTMF digits during an active call. DTMF digits are displayed as “*” on the touch screen.
Suppress DTMF display delay defines whether to display the DTMF digits for a short period of time before displaying as “*”.

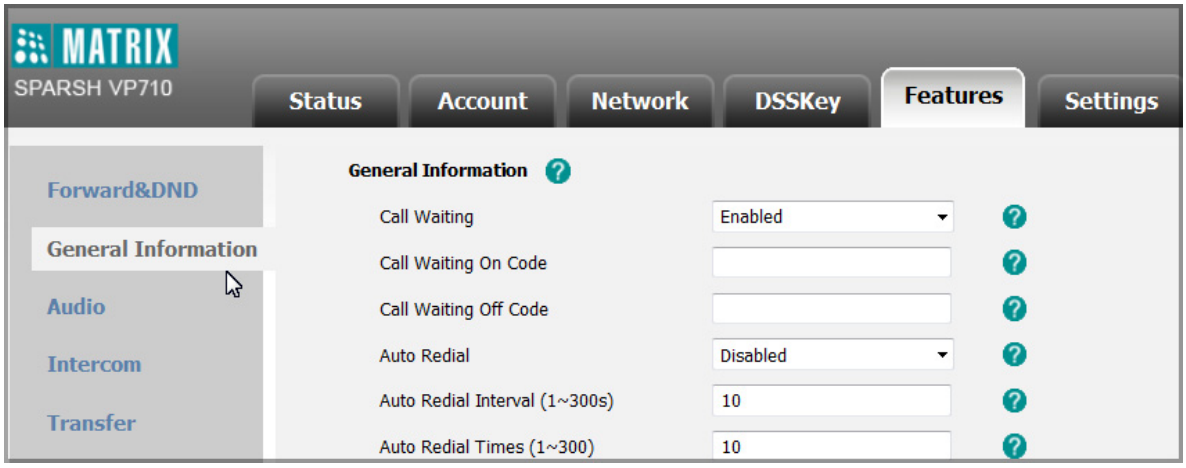
Procedure

Configuration changes can be performed using the configuration files or locally.

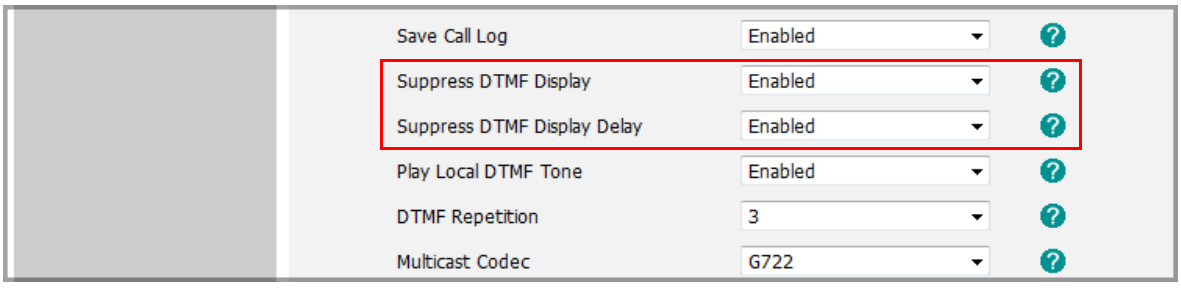
Configuration File	<MAC>.cfg	Configure suppress DTMF display and suppress DTMF display delay. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure suppress DTMF display and suppress DTMF display delay. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure suppress DTMF display and suppress DTMF display delay via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Suppress DTMF Display** list.
- Select the desired value from the **Suppress DTMF Display Delay** list.



- Click **Confirm** to save the change.

Transfer via DTMF

Call transfer is implemented via DTMF on some traditional servers. The IP phone sends specified DTMF digits to the server for transferring calls to third parties.

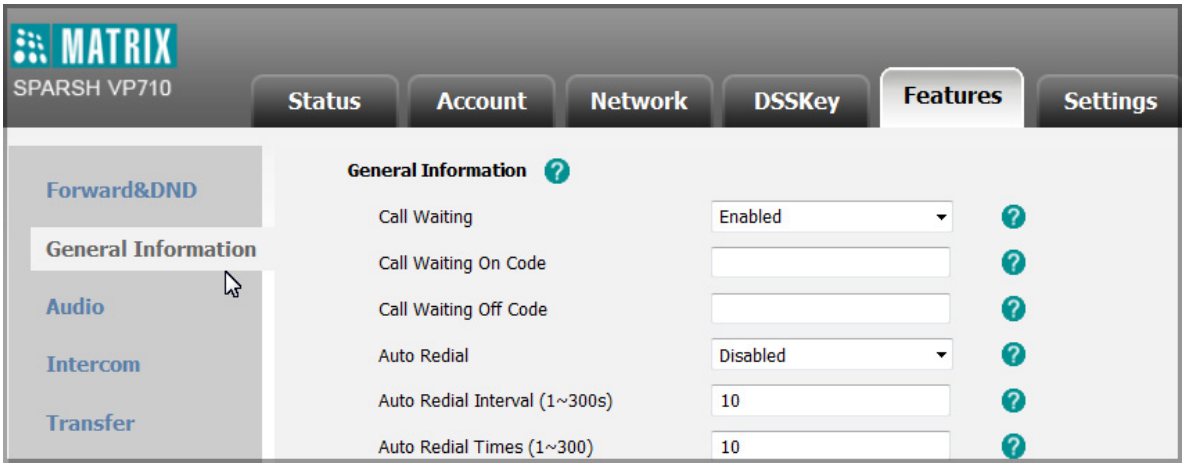
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure transfer via DTMF. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure transfer via DTMF. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

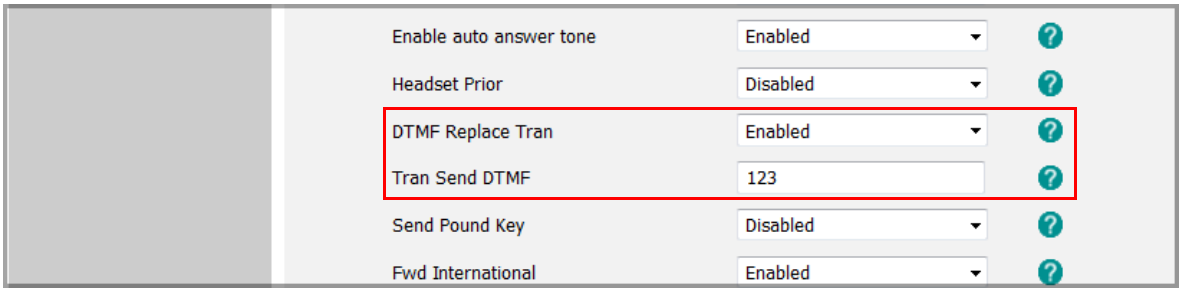
To configure transfer via DTMF via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **DTMF Replace Tran** list.

- Enter the specified DTMF digits in the **Tran Send DTMF** field.



- Click **Confirm** to save the change.

Play Local DTMF Tone

Play local DTMF tone allows the IP phone to play a local DTMF tone during an active call. If this feature is enabled, you can hear the DTMF tone when the keypad of the IP phone is pressed during a call.

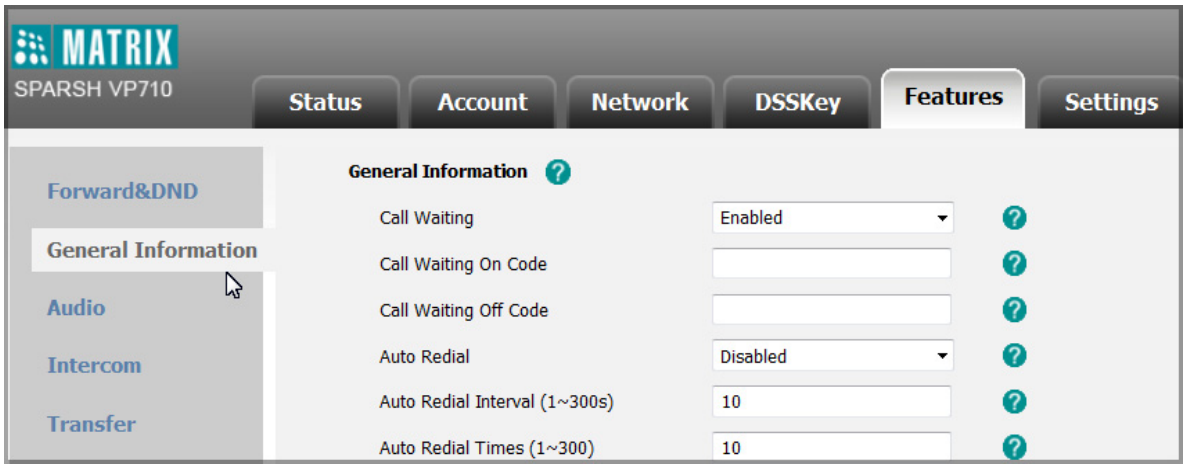
Procedure

Configuration changes can be performed using the configuration files or locally.

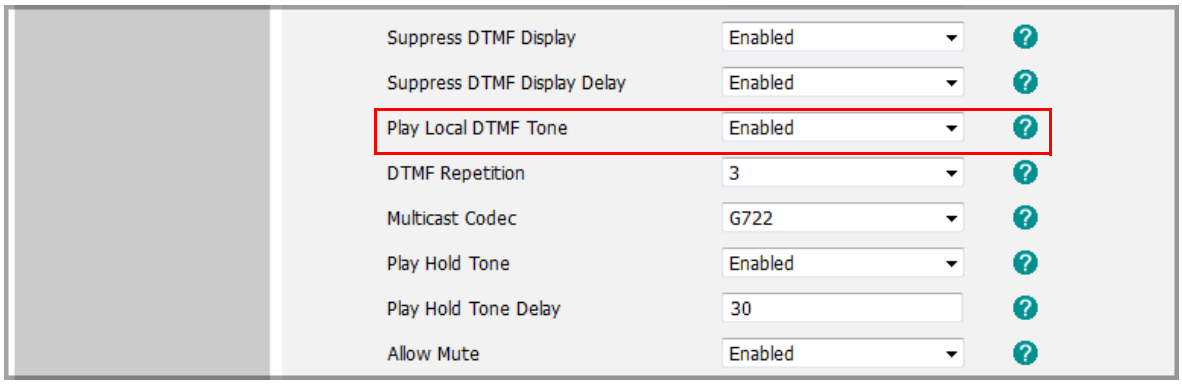
Configuration File	<MAC>.cfg	Configure play local DTMF tone. For more information, refer "Appendix D - Configuration Parameters".
Local	**Web User Interface**	Configure play local DTMF tone. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure play local DTMF tone via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Play Local DTMF Tone** list.



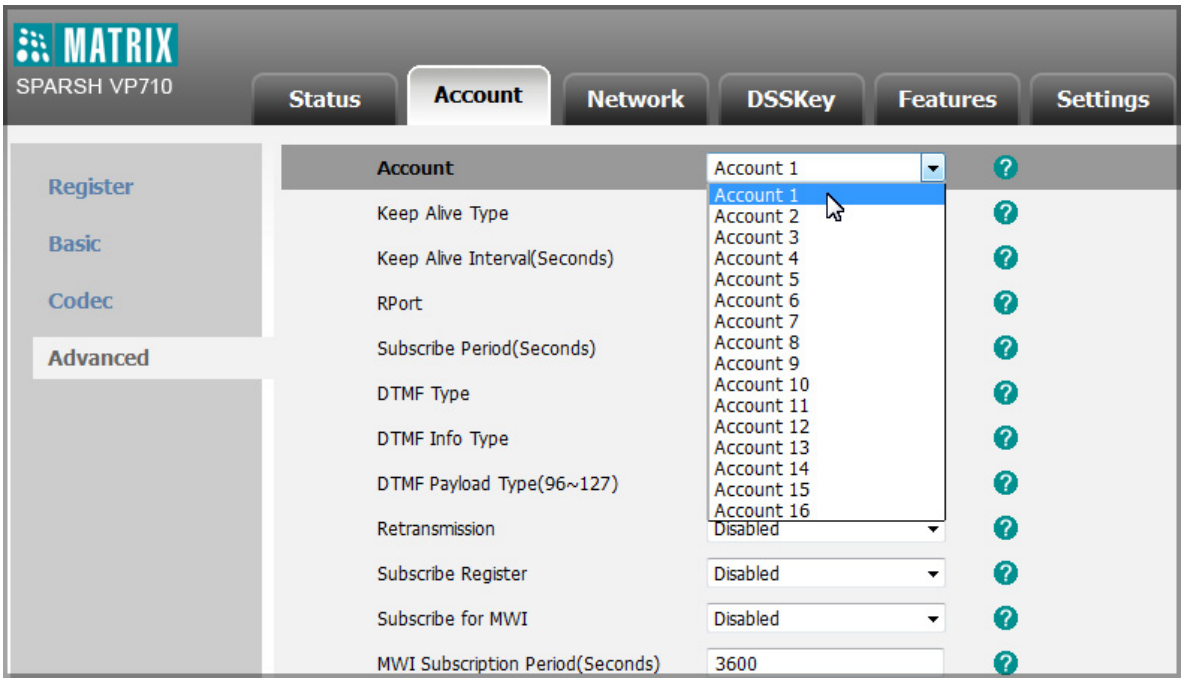
- Click **Confirm** to save the change.

Early Media

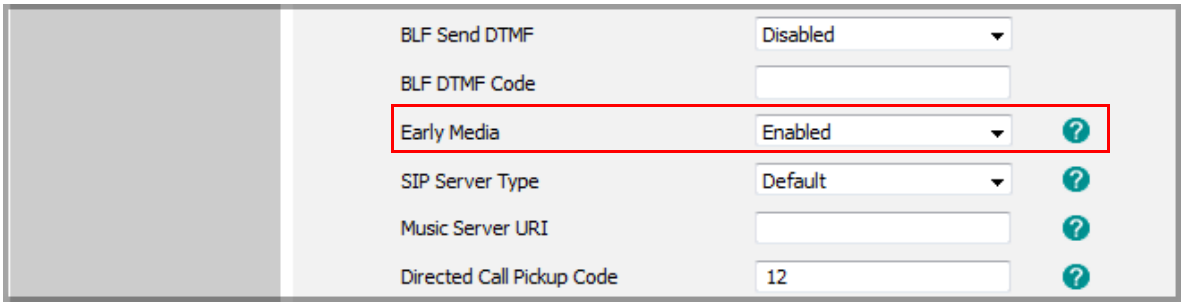
Early media refers to media (e.g., audio and video) played to the caller before a SIP call is actually established. Current implementation supports early media through the 183 message. When the caller receives a 183 message with SDP before the call is established, a media channel is established. This channel is used to provide the early media stream for the caller.

To configure Early media via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Early Media** list.



- Click **Confirm** to save the change.

Hot Desking

Hot desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee.

A primary motivation for hot desking is cost reduction. Hot desking is regularly used in places where not all employees are in the office at the same time, or not in the office for a long time, which means actual personal offices would often be vacant, consuming valuable space and resources.

You can use hot desking on the IP phone to log out of the existing account and then log into a new account, that is, many users can share the phone resource in different time. To use this feature, you need to configure a hot desking key in advance.

Hot Desking is useful for people who are often away from their own desks and must work from another. Hot Desking allows them to use all the features and facilities of their own extension from another.



Hot desking is not available on all servers. For more information, contact your system administrator.

Procedure

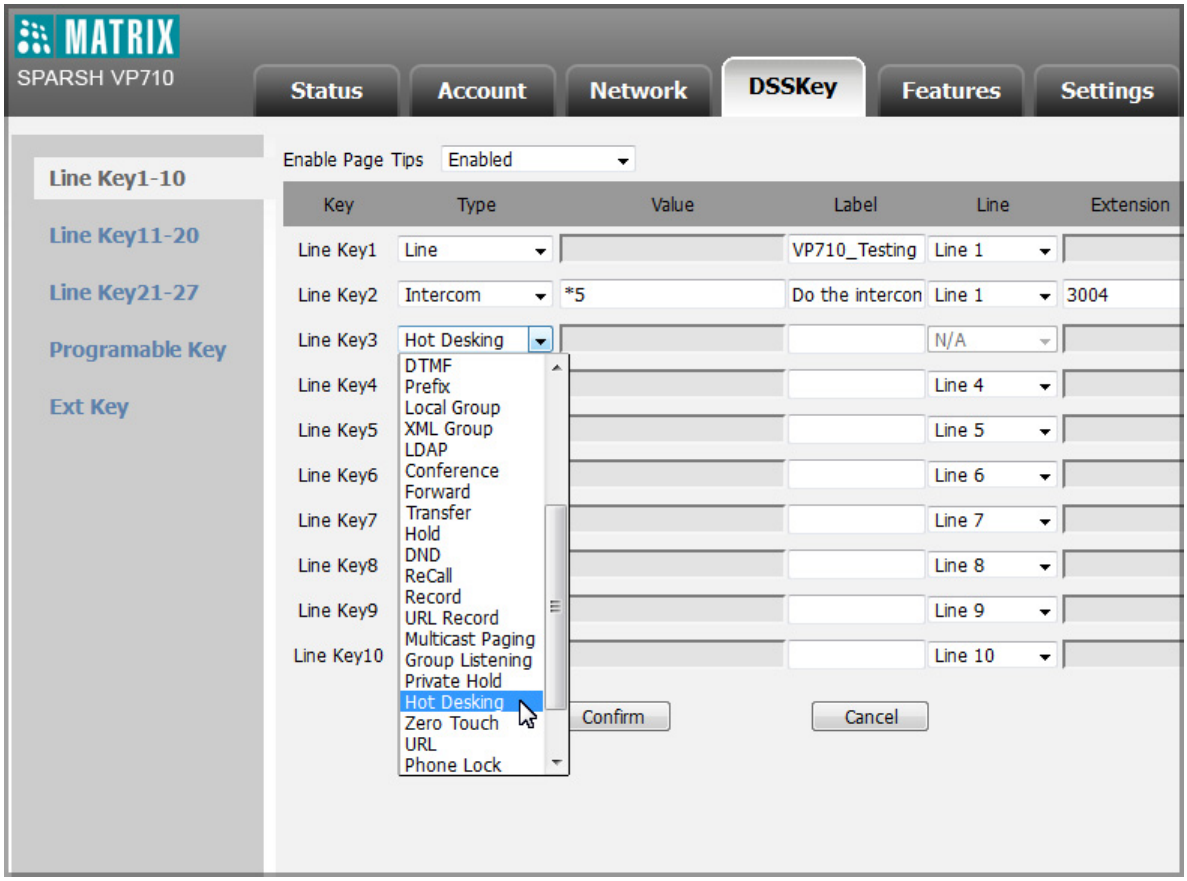
Hot desking key can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the hot desking login wizard. Assign a hot desking key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Assign a hot desking key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a hot desking key.

To configure a hot desking key via web user interface:

- Click on **DSSKey->Line Key (**or **Programable Key/Ext Key)**.
- In the desired DSS key field, select **Hot Desking** from the **Type** list.

- In the **Label** field (optional), enter the string that will appear on the touch screen.

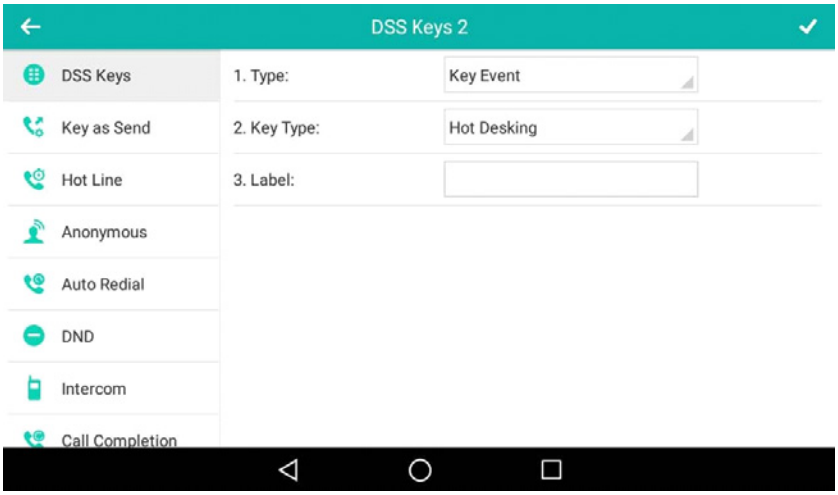


- Click **Confirm** to save the change.

To configure a hot desking key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired line key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **Hot Desking** in the pop-up dialog box.

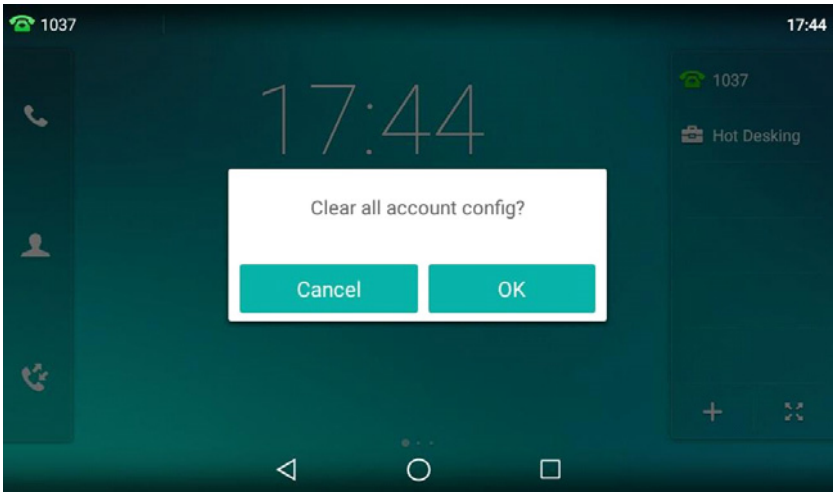
- In the **Label** field (optional), enter the string that will appear on the touch screen



- Tap ✓ to save the change or ◁ to cancel.

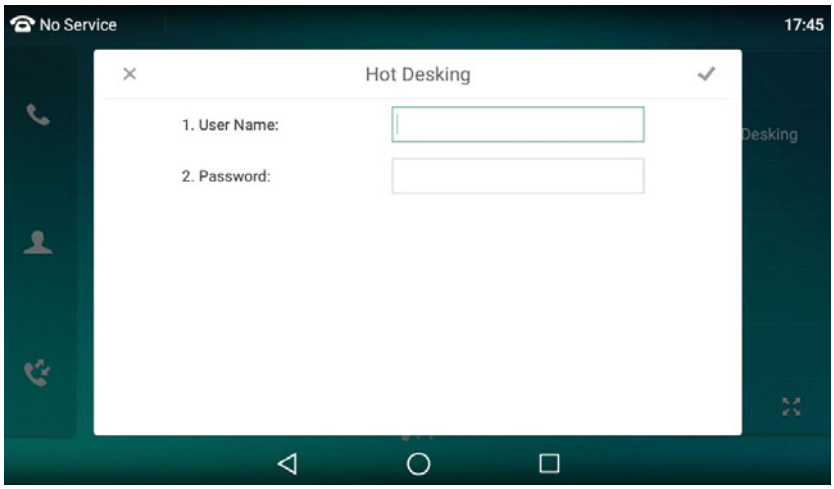
To use hot desking:

- Tap the hot desking key when the phone is idle.
- The touch screen prompts the following warning:.



- Tap **OK**.
- Registration configurations of all accounts on the phone will be cleared immediately.

The login wizard will be displayed as below:



- Enter the login information in each field.
- Tap ✓ to save the change or ◁ to cancel.

For details, refer “Logon Wizard”.

Lightweight Directory Access Protocol (LDAP)

LDAP (Lightweight Directory Access Protocol) is an application protocol for accessing and maintaining information services for the distributed directory over an IP network. The IP phone can be configured to interface with a corporate directory server that supports LDAP version 2 or 3.

The following LDAP servers are supported:

- Microsoft Active Directory
- Sun ONE Directory Server
- Open LDAP Directory Server
- Microsoft Active Directory Application Mode (ADAM)

The biggest advantage for LDAP is that you can access the central LDAP directory of the corporation using the IP phone. Therefore, you do not have to maintain the local directory. You can search and dial out from the LDAP directory, and save LDAP entries to the local directory.You can only read the LDAP entries displayed on the IP phone. You cannot add, edit or delete these entries.
When an LDAP server is properly configured, the IP phone can look up entries from the LDAP server in a wide variety of ways. The LDAP server indexes all the data in its entries, and “filters” can be used to select the desired entry or group, and return the desired information.Configurations on the IP phone limit the amount of the displayed entries when querying from the LDAP server, and decide how attributes are displayed and sorted.

You can configure DSS key for a LDAP key, and then tap the LDAP key to enter the LDAP search screen when the IP phone is idle.

LDAP Attributes

The following table lists the most common attributes used to configure the LDAP lookup on the IP phone.

Abbreviation	Name	Description
gn	givenName	First name
cn	commonName	LDAP attribute is made up from given name joined to surname.
sn	surname	Last name or family name
dn	distinguishedName	Unique identifier for each entry
dc	dc	Domain component
-	company	Company or organization name
-	telephoneNumber	Office phone number.
mobile	mobilephoneNumber	Mobile or cellular phone number
ipPhone	IPphoneNumber	Home phone number

Procedure

LDAP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure LDAP Assign an LDAP key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure LDAP. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=contacts-LDAP&q=load Assign an LDAP key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign an LDAP key.

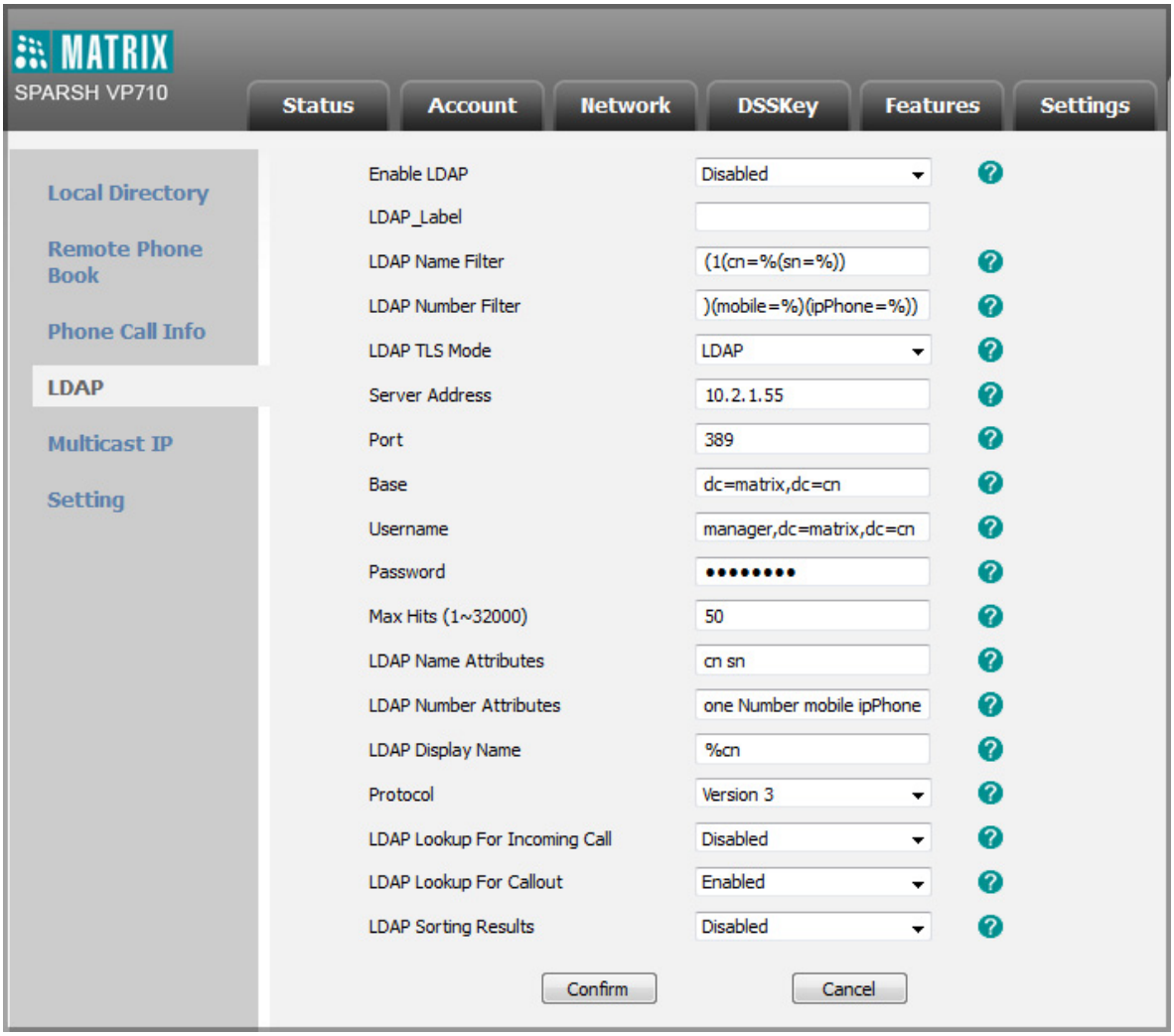
LDAP Key

For more information on how to configure DSS Key for LDAP key, refer "DSS Keys".

To configure LDAP feature via web user interface:

- Click on **Directory->LDAP**.
- Enter the values in the corresponding fields.

- Select the desired values from the corresponding lists.

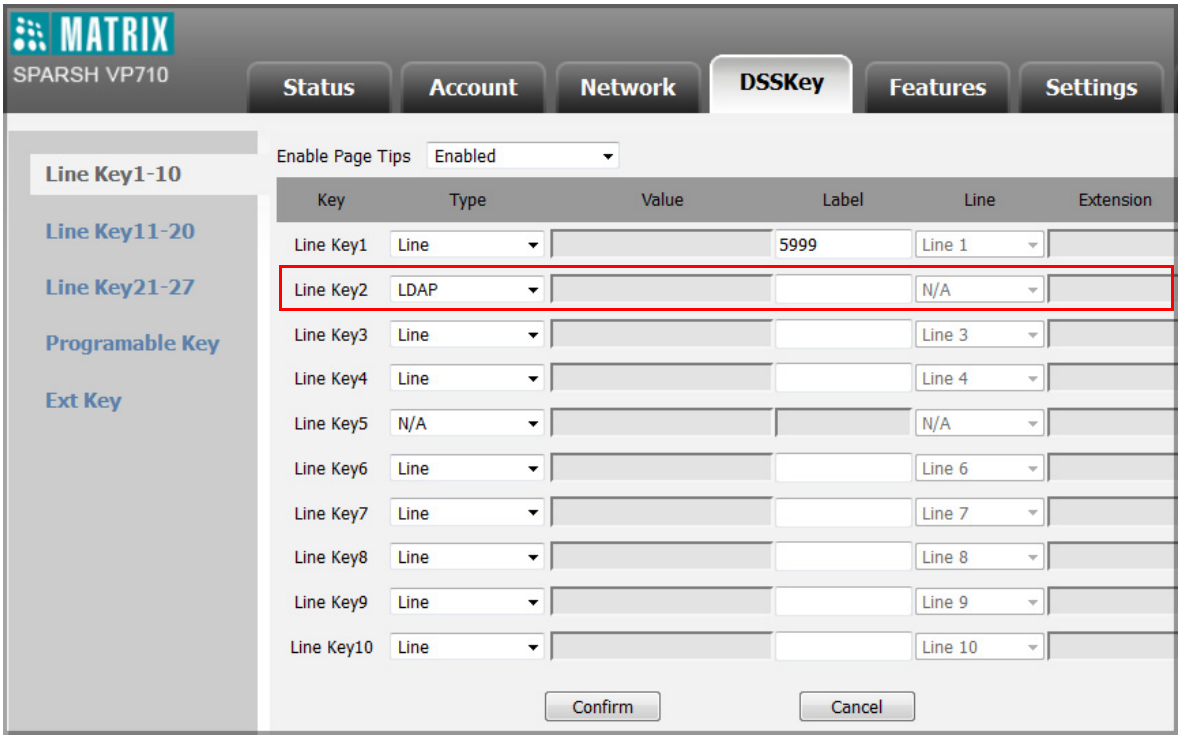


- Click **Confirm** to save the change.

To configure an LDAP key via web user interface:

- Click on **DSSKey->Line Key (or Programable Key).**
- In the desired DSS key field, select **LDAP** from the **Type** list.

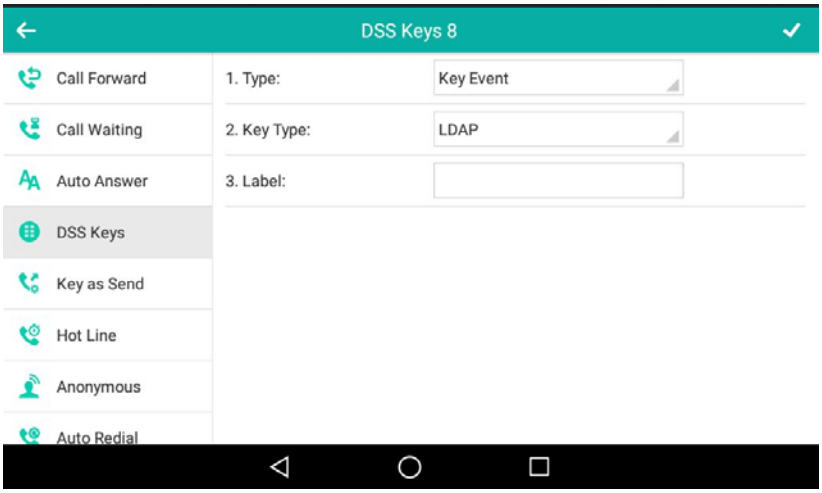
- In the **Label** field (Optional), enter the string that will appear on the touch screen.



- Click **Confirm** to save the change.

To configure LLDP feature via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired DSS key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **LDAP** in the pop-up dialog box.
- In the **Label** field (Optional), enter the string that will appear on the touch screen.



- Tap ✓ to save the change or ◁ to cancel.

Linker Layer Discovery Protocol (LLDP)

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol, which allows IP phone to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol and store the information about other devices.

When LLDP feature is enabled, the IP phone advertise its own information periodically to the directly connected LLDP-enabled switch.The IP phone may also receive LLDP packets from the connected switch. When the application type is “voice”, the IP phone decides whether to update the VLAN configurations obtained from the LLDP packets or not. If the VLAN configurations on the IP phone is different from the ones sent by the switch, the IP phone performs an update and reboot. This allows the IP phone to plug into any switch, obtain their VLAN IDs, and then start communication with the call control.

LLDP-MED (Media Endpoint Discovery)

LLDP-MED is published by the Telecommunications Industry Association (TIA). It is an extension to LLDP that operates between endpoint devices and network connectivity devices. LLDP-MED specifically provides support for voice over IP (VoIP) applications and provides the following capabilities:

- Capabilities Discovery -- allows the IP phone to determine the capabilities that the connected switch supports and has enabled.
- Network Policy -- provides voice VLAN configuration to notify the IP phone which VLAN to use and QoS-related configuration for voice data. It provides a “plug and play” network environment.
- Power Management -- provides information related to how the IP phone are powered, power priority, and how much power IP phone need.
- Inventory Management -- provides a means to effectively manage the IP phone and their attributes such as model number, serial number and software revision.

TLVs supported by IP phones are summarized in the following table:

TLV Type	TLV Name	Description
Mandatory TLVs	Chassis ID	The network address of the IP phone.
	Port ID	The MAC address of the IP phone.
	Time To Live	Seconds until data unit expires.
	End of LLDPDU	Marks end of LLDPDU.

Optional TLVs	System Name	Name assigned to the IP phone.
	System Description	Description of the IP phone.
	System Capabilities	The supported and enabled capabilities of the IP phone. The supported capabilities are Bridge and Telephone. The enabled capabilities are Bridge and Telephone by default.
	Port Description	Description of port that sends data unit. The default value is “WAN PORT”.
IEEE Std 802.3 Organizationally Specific TLV	MAC/PHY Configuration/ Status	Duplex and bit rate settings of the IP phone. The Auto Negotiation is supported and enabled by default. The advertised capabilities of PMD. Auto-Negotiation is: 100BASE-TX (full duplex mode), 100BASE-TX (half duplex mode), 10BASE-T (full duplex mode), or 10BASE-T (half duplex mode).

TIA Organizationally Specific TLVs	Media Capabilities	The MED device type of the IP phone and the supported LLDP-MED TLV type can be encapsulated in LLDPDU. The supported LLDP-MED TLV types are: LLDP-MED Capabilities, Network Policy, Extended Power via MDI-PD and Inventory.
	Network Policy	Port VLAN ID, application type, L2 priority and DSCP value.
	Extended Power-via-MDI	Power type, source, priority and value.
	Inventory – Hardware Revision	Hardware revision of the IP phone.
	Inventory – Firmware Revision	Firmware revision of the IP phone.
	Inventory – Software Revision	Software revision of the IP phone.
	Inventory – Serial Number	Serial number of the IP phone.
	Inventory – Manufacturer Name	Manufacturer name of the IP phone.
	Inventory – Model Name	Model name of the IP phone.
	Asset ID	Assertion identifier of the IP phone. The default value is “asset”.

Procedure

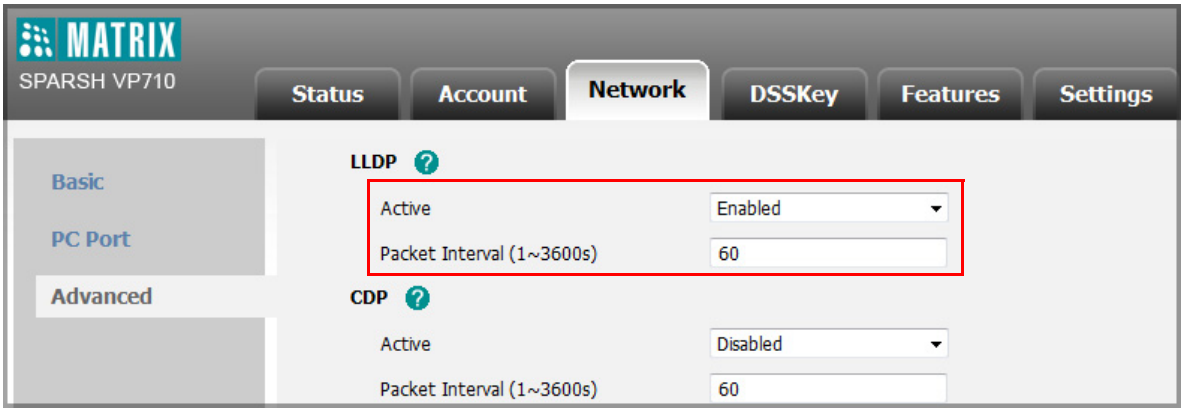
LLDP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure LLDP feature. For more information, refer “Appendix D - Configuration Parameters”.

Local	Web User Interface	Configure LLDP feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure LLDP feature.

To configure LLDP feature via web user interface:

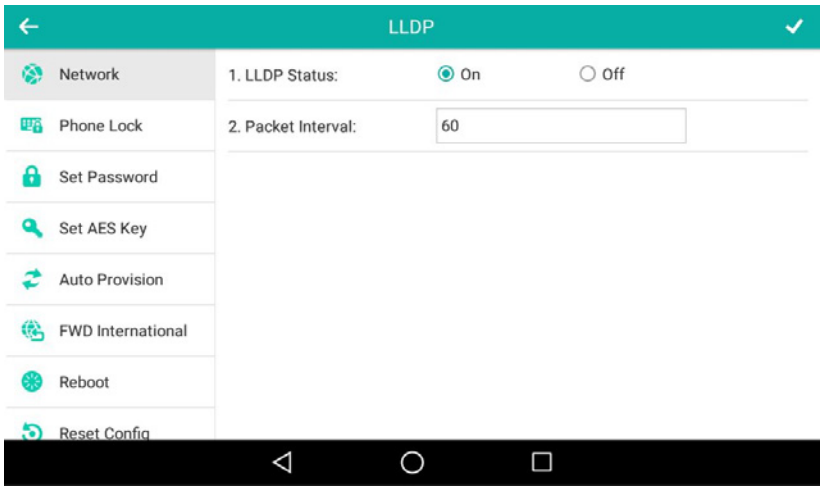
- Click on **Network->Advanced**.
- In the **LLDP** block, select the desired value from the **Active** list.
- Enter the desired time interval in the **Packet Interval (1~3600s)** field.



- Click **Confirm** to save the change.
- A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the IP phone.

To configure LLDP feature via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->LLDP**.
- Tap the **On** radio box in the **LLDP Status** field.
- Enter the priority value (1-3600s) in the **Packet Interval** field.



- Tap ✓ to save the change or ◁ to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone

  The settings will take effect after a reboot

Multicast Paging

Multicast paging allows you to broadcast time sensitive announcements quickly and efficiently to the people who are listening to a specific multicast group or a specific channel.

You can configure a multicast paging key or a paging list key on the IP phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address(es) and channel(s) without involving SIP signaling.

You can configure the phone to receive an Real-time Transport Protocol (RTP) streams to/ from pre-configured multicast listening address(es) and channel(s) without involving SIP signaling.
You can specify up to 31 multicast listening addresses and channel(s) on the IP phone.

The following describes 31 paging channels:

- 0: Matrix IP phones running old firmware version (old paging mechanism) and third-party devices (e.g., Cisco IP phones) supporting paging with Matrix IP phones running old firmware version are all grouped into the channel 0. This is for compatibility with the old Matrix IP phones and third-party devices.

- 1 to 25 - Each Channel corresponds to the Polycom¡¦s channel 1 to 25 respectively. This is for compatibility with the Polycom IP phones.

- 26 to 30 - These channel are for separate communication among the Matrix IP phones running new firmware version (new paging mechanism).

The IP phone will automatically ignore all incoming multicast paging calls on the different channel.

Sending RTP Stream

You can send an RTP stream without involving SIP signaling by tapping a configured multicast paging key or a paging list key. A multicast address (IP: Port) and a channel (0 to 30) should be assigned to the multicast paging key, which is defined to transmit RTP stream to a group of designated IP phones on the desired channel.

When the IP phone sends the RTP stream to a pre-configured multicast address belongs to a desired channel, each IP phone preconfigured to listen to the multicast address on the same channel can receive the RTP stream. When the originator stops sending the RTP stream, the subscribers stop receiving it.

Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify a multicast codec for the IP phone to send the RTP stream. Configure the multicast IP address and port number for a paging list key. Configure the channel of the multicast paging group for a paging list key. Configure the multicast paging group name for a paging list key. Assign a multicast paging key. Assign a paging list key. Specify a multicast codec for the IP phone to send the RTP stream. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Specify a multicast codec for the IP phone to send the RTP stream. **Navigate to:** http://<phoneIPAddress>/ servlet?m=mod_data&p=features-general&q=load Configure the multicast IP address and port number for a paging list key. Configure the multicast paging group name for a paging list key. Configure the channel of the multicast paging group for a paging list key. **Navigate to:** http://<phoneIPAddress>/ servlet?m=mod_ data&p=contacts-multicastIP&q=load Assign a multicast paging key or a paging list key. **Navigate to:** http://<phoneIPAddress>/ servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Configure the multicast IP address and port number for a paging list key. Configure the channel of the multicast paging group for a paging list key. Configure the multicast paging group name for a paging list key. Assign a multicast paging key or a paging list key.

You can specify a multicast codec for the IP phone to use for multicast RTP using the Configuration File.

Multicast Paging Key

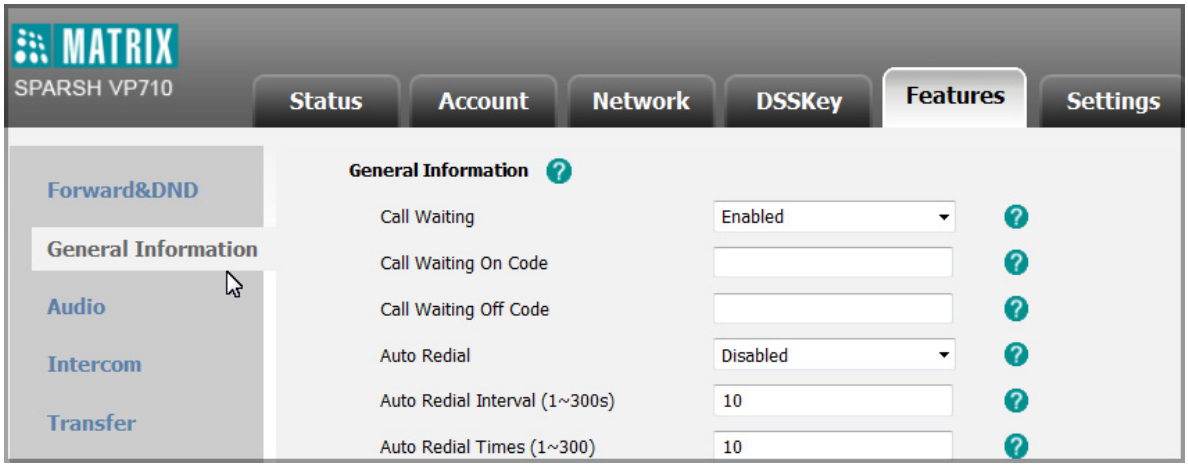
For more information on how to configure the DSS Key, refer to Appendix D: Configuring DSS Key.

Paging List key

For more information on how to configure the DSS Key, refer to Appendix D: Configuring DSS Key.

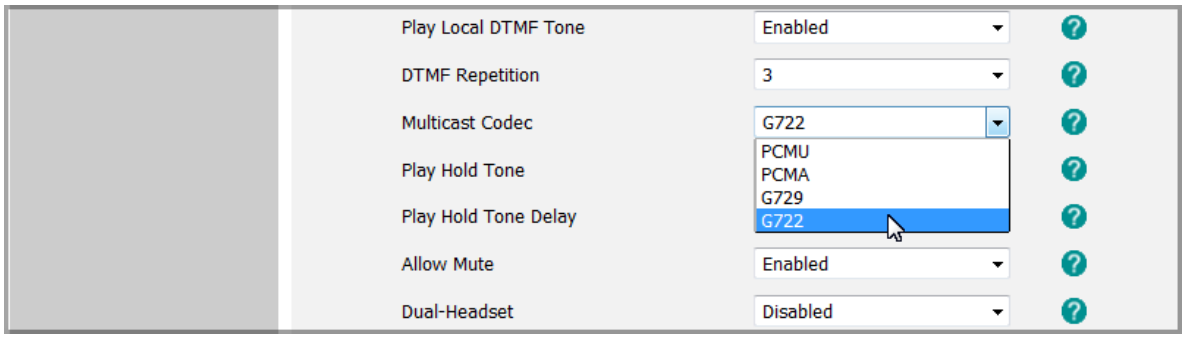
To configure a codec for multicast paging via web user interface:

- Click on **Features->General Information**.



- Select the desired codec from the. **Multicast Codec** list.

  The default codec is G722

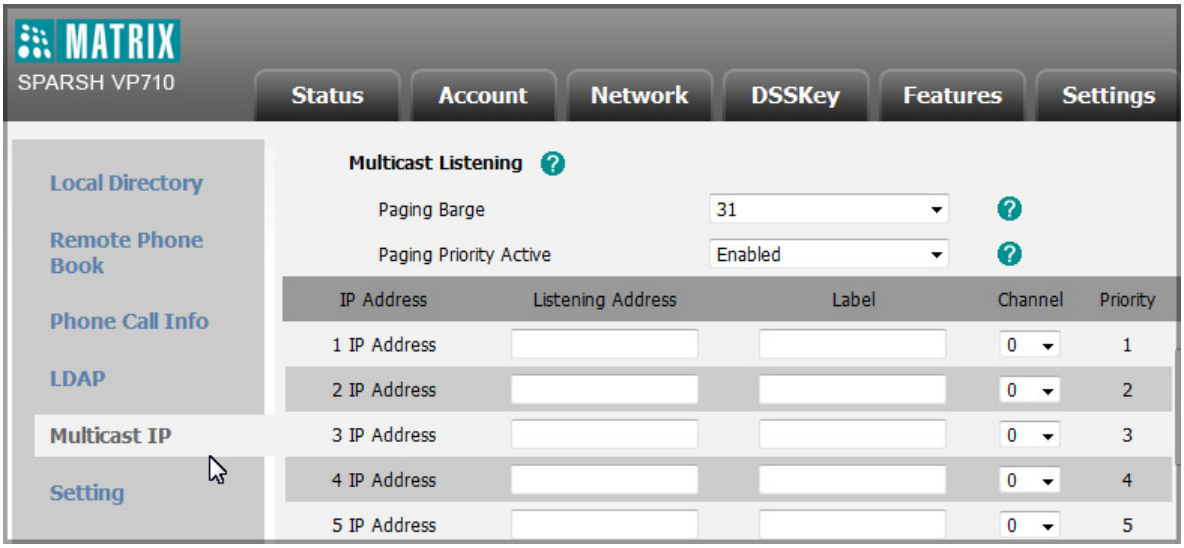


- Click **Confirm** to save the change.

- *If G722 codec is used for multicast paging, the touch screen will display the icon* HD *to indicate that it is providing high definition voice.*
- *Default codec for multicast paging is configurable via web user interface only.*

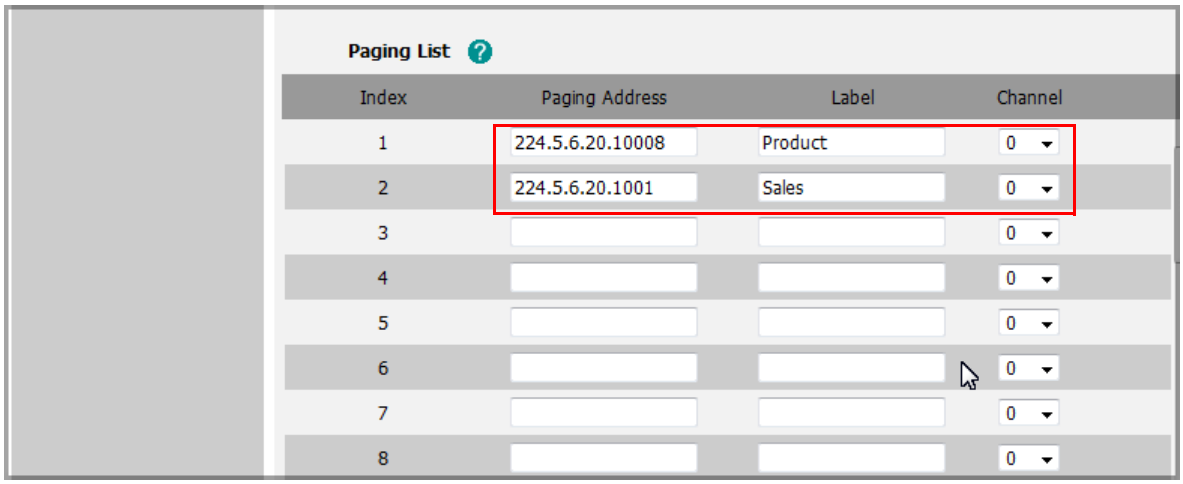
To configure two sending multicast addresses via web user interface:

- Click on **Directory->Multicast IP**.



- Enter the sending multicast address and port number in the **Paging Address** field.
- Enter the label in the **Label** field.

  The label will appear on the touch screen when sending the RTP multicast.
- Select the desired channel from the **Channel** list.



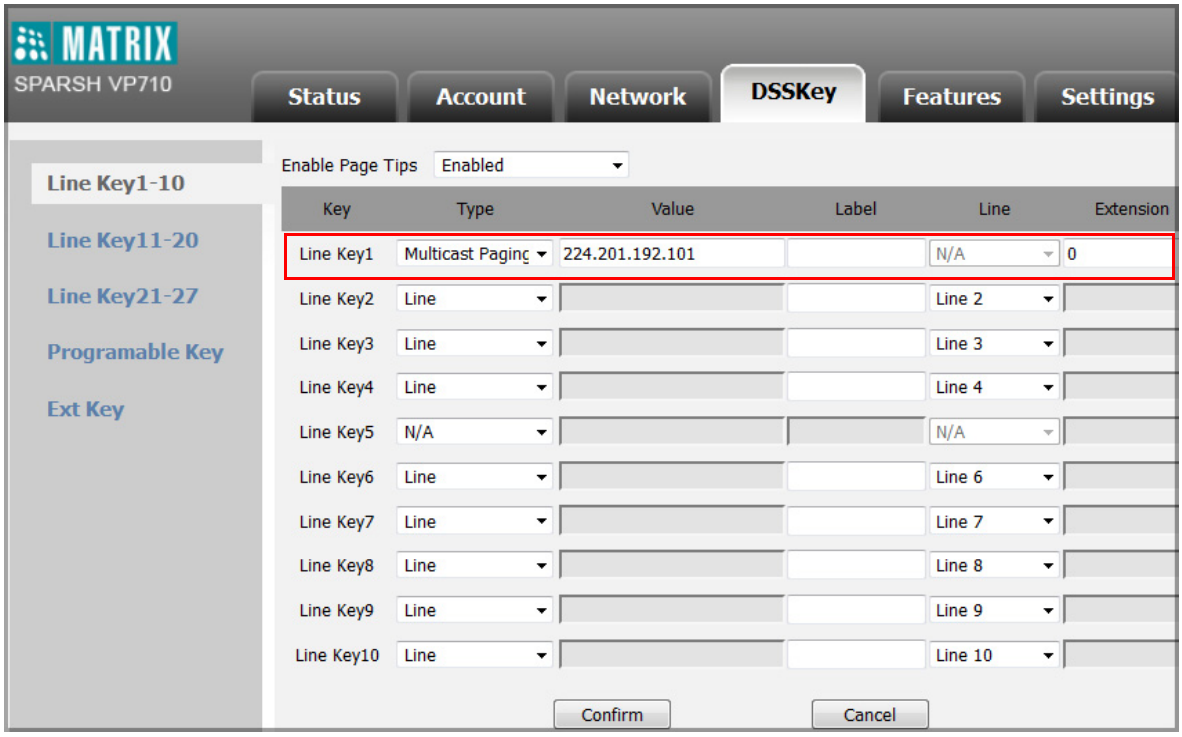
- Click **Confirm** to save the change.

To configure a multicast paging key via web user interface:

- Click on **DSSKey->Line Key**.
- In the desired DSS key field, select **Multicast Paging** from the **Type** list.
- Enter the multicast IP address and port number in the **Value** field.

  The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
- In the **Label** field (Optional), enter the string that will appear on the touch screen.
- Enter the desired channel in the Extension field.
- The valid channel ranges from 0 to 30.

- Select the desired codec from the **Multicast Codec** list.

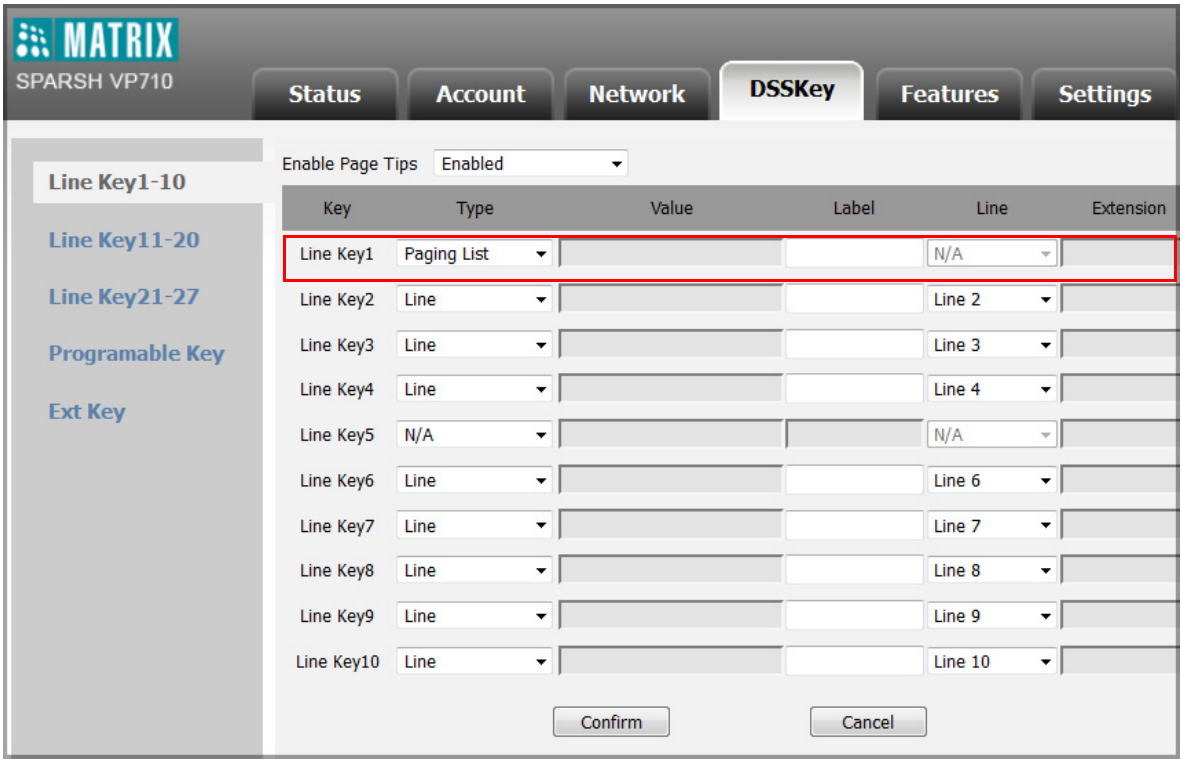


- Click **Confirm** to save the change.

To configure a paging list key via web user interface:

- Click on **DSSKey->Line Key (or Programable Key)**.
- In the desired DSS key field, select **Paging List** from the **Type** list.

- In the **Label** field (optional), enter the string that will appear on the touch screen.



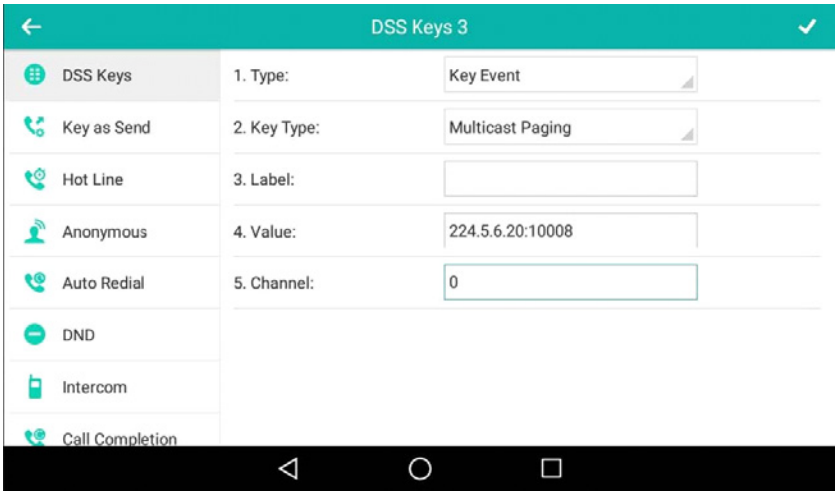
- Click **Confirm** to save the change.

To configure a multicast paging key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired DSS key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **Multicast Paging** in the pop-up dialog box.
- In the **Label** field (optional), enter the string that will appear on the touch screen.
- Enter the multicast IP address and port number in the **Value** field.T

  The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

- Enter the desired channel in the **Channel** field.

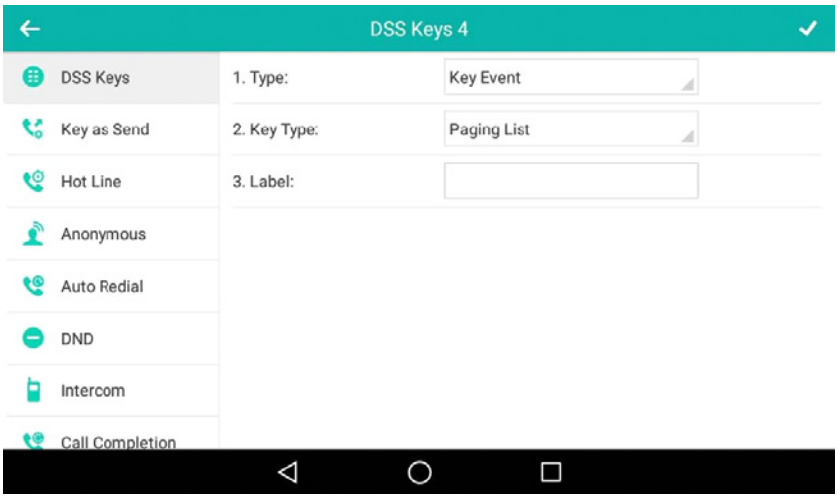


  The valid channel ranges from 0 to 30.

- Tap ✓ to save the change or ◁ to cancel.

To configure a paging list key via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Features->DSS Keys**.
- Tap the desired line key.
- Tap the **Type** field.
- Tap **Key Event** in the pop-up dialog box.
- Tap the **Key Type** field.
- Tap **Paging List** in the pop-up dialog box.
- In the **Label** field (optional), enter the string that will appear on the touch screen.

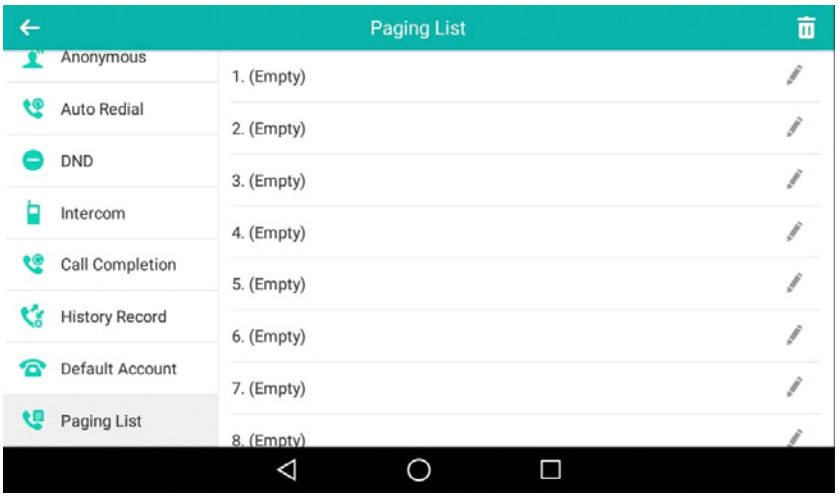


- Tap ✓ to save the change or ◁ to cancel.

To configure paging list via phone user interface:

- Tap the paging list key when the phone is idle.

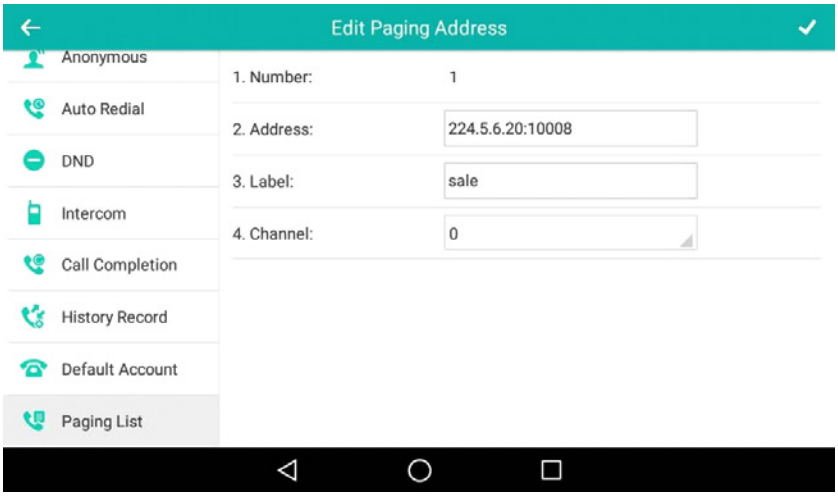
  If the paging list key is not configured, you can also swipe down from the top of the screen or swipe left/ right to go to the second idle screen.
  Then tap **Settings->Features->Paging List** to configure the paging list.



- Tap  after the desired paging group.
- Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Address** field.

  The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
- Enter the paging group name in the **Label** field.
- Enter the desired channel in the **Channel** field

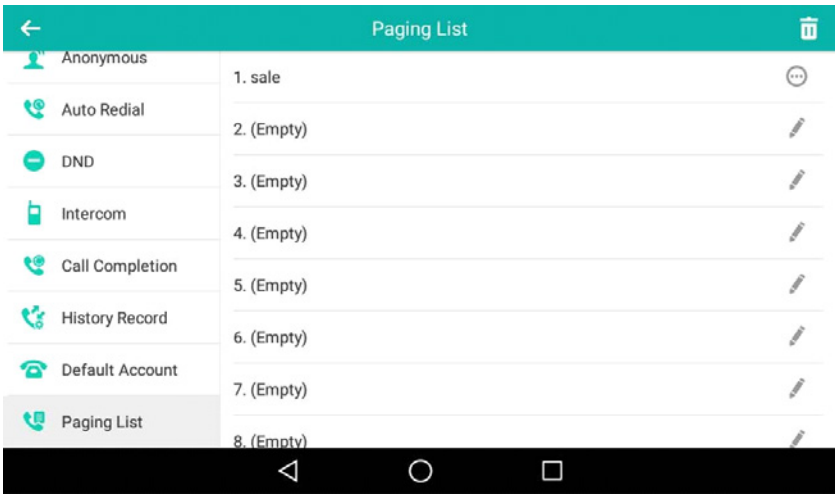
  The valid channel ranges from 0 to 30.



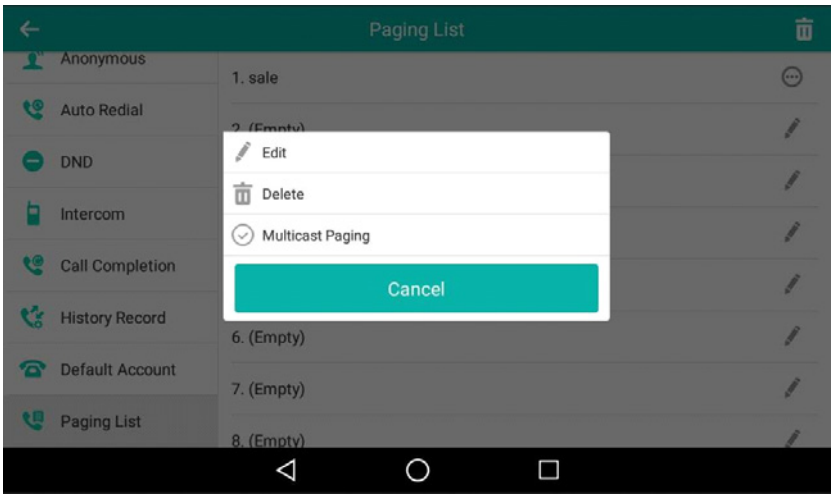
- Tap  to save the change or  to cancel.
- Repeat the above5 steps to add more paging groups.

To delete paging group via phone user interface:

- Tap the paging list key when the phone is idle.

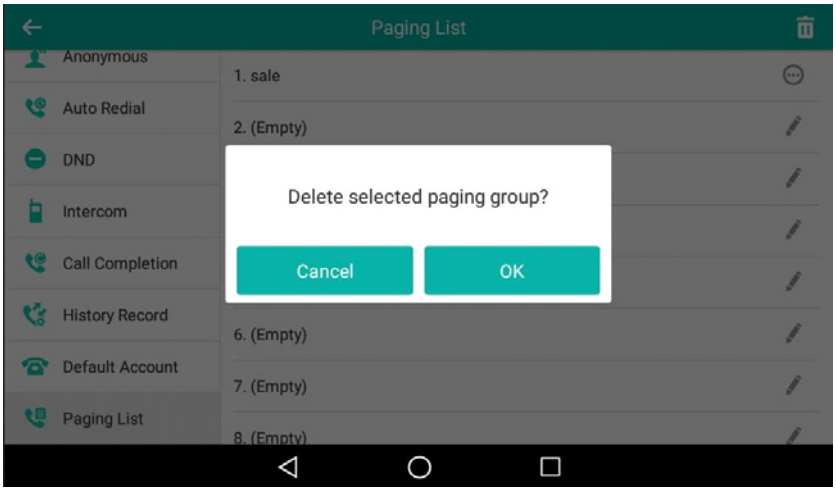


- Tap ⋯ after the desired paging group.



The icon ⋯ only appears after the configured paging group.

- Tap **Delete**.



- Tap **OK** to save the change or **Cancel** to cancel.
- If you want to delete all paging groups, you can tap on the top-right of the touch screen.
- Click **Confirm** to save the change.

Receiving RTP Stream

IP phone can receive an RTP stream from the pre-configured multicast address(es) without involving SIP signaling, and can handle the incoming multicast paging calls differently depending on the configurations of Paging Barge and Paging Priority Active.
You can specify up to 31 multicast addresses and channel(s) that the phone listens to on the network.

- *RTP stream is listened in the hands-free (speakerphone) mode by default. If you want to listen the RTP stream using the engaged audio device (speakerphone, handset or headset), contact your system administrator for more information.*
- *Fixed volume to play RTP stream for specified paging group is configurable by your system administrator.*

Paging Barge

Paging barge parameter defines the priority of the voice call in progress. If the priority of an incoming multicast paging call is lower than that of the active call, it will be ignored automatically. If this parameter is disabled in the Paging Barge list, the voice call in progress takes precedence over all the incoming multicast paging calls.
Valid values in the Paging Barge list:

- **1 to 31:** Defines the priority of an active call,1 with the highest priority and 31 with the lowest.
- **Disabled:** The voice call in progress will take precedence over all the incoming paging calls.

.
Paging Priority Active

Paging priority active parameter decides how the IP phone handles the incoming multicast paging calls when there is an ongoing multicast paging call. If this parameter is enabled, the IP phone automatically answers the incoming call with higher or equal priority and the previous multicast paging call is put on hold. The incoming multicast paging calls with lower priorities will be ignored. If the parameter is disabled, the IP phone will automatically ignore all

incoming multicast paging calls

Procedure

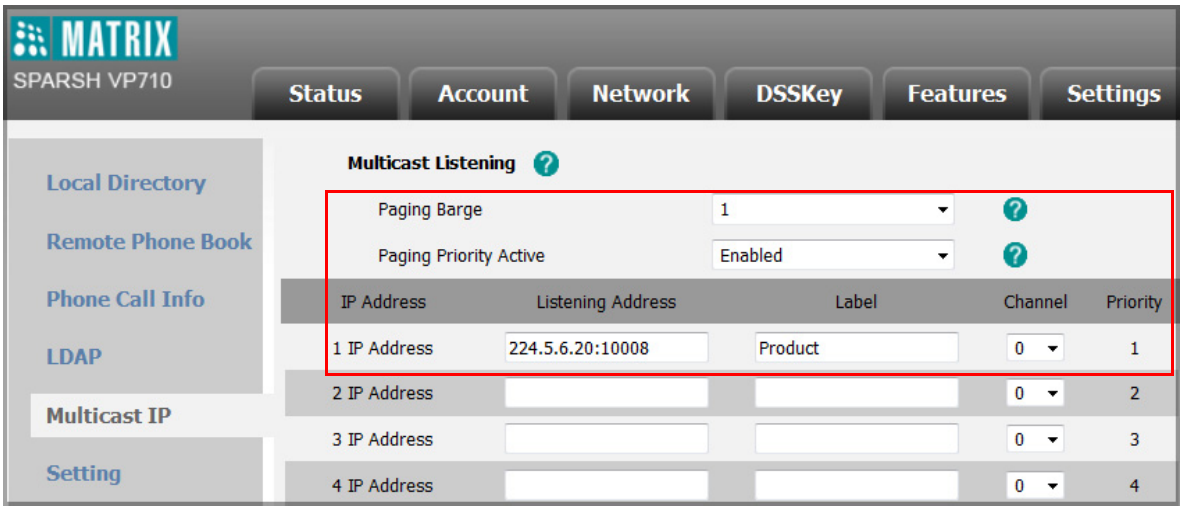
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the listening multicast address. Configure Paging Barge and Paging Priority Active features. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the listening multicast address. Configure Paging Barge and Paging Priority Active features. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_data&p=contacts-multicastIP&q=load

To configure multicast listening addresses via web user interface:

- Click on **Directory->MulticastIP**.
- Select the desired value from the **Paging Barge** list.
- Select the desired value from the **Paging Priority Active** list.
- Enter the multicast IP address(es) and port number (for example, 224.5.6.20:10008) which the phone listens for incoming RTP multicast in the **Listening Address** field.
  1 is the highest priority and 31 is the lowest priority.
- Enter the label in the **Label** field.
  Label will appear on the touch screen when receiving the RTP multicast.

- Select the desired channel from the **Channel** list.



Click **Confirm** to save the change.

- *The priorities of listening addresses are predefined: 1 with the highest priority, 31 with the lowest.*
- *The phone plays a warning tone when receiving incoming multicast paging calls.*
- *The phone will not receive a new incoming multicast paging call when there are two calls on the phone.*
- *Multicast listening addresses is configurable via web user interface only.*

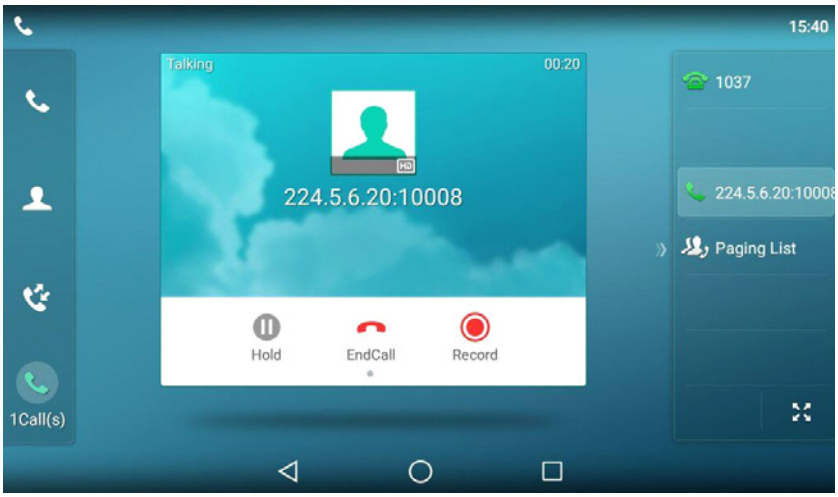
Using Multicast Paging

To send RTP stream via a multicast paging key when the receiver's phone is idle:

- Tap the multicast paging key when the phone is idle.

  The phone sends RTP to a preconfigured multicast address (IP: Port).Both the sender's and receiver's phones play a warning tone and the receiver automatically answers the multicast RTP session in the hands-free (speakerphone) mode.The multicast paging key icon will become and solid.

  The following figure shows a multicast RTP session on the IP phone:



- Your can do the following:
  - Tap the **Hold** soft key to place the current multicast RTP session on hold.
    The sender¡¦s phone places the multicast RTP session on hold and receiver¡¦s phone releases the session.
    To resume the held multicast RTP session, tap the **Resume** soft key.
    The multicast RTP session is established again.
  - Tap the **EndCall** soft key to cancel the multicast RTP session.
  - Tap the **Record** soft key to record a call.
    To stop recording, tap the **Stop** soft key



- *Multicast RTP is only one way from the sender to the multicast address(es) (receiver).*
- *For outgoing RTP multicasts, all other existing calls on the phone will be placed on hold.*

To send RTP stream via a paging key list when the receiver's phone is idle:

- Tap paging list key when the phone is idle.
- Do one of the following to send RTP:
  - Tap the desired paging group.
  - Tap  after the desired paging group, and then tap **Multicast Paging**.

- Your can do the following:

  - Tap the **Hold** soft key to place the current multicast RTP session on hold.
    The sender¡¦s phone places the multicast RTP session on hold and receiver¡¦s phone releases the session.
    To resume the held multicast RTP session, tap the Resume soft key.
    The multicast RTP session is established again.
  - Tap the **EndCall** soft key to cancel the multicast RTP session.
  - Tap the **Record** soft key to record a call.
    To stop recording, tap the **Stop** soft key

Quality of Service (QoS)

Quality of Service (QoS) is the ability to provide different priorities to different packets in the network, allowing the transport of traffic with special requirements. When the network capacity is insufficient, QoS guarantees are important for applications that require fixed bit rate and are delay sensitive. Four major QoS factors to be considered when configuring a modern QoS implementation are bandwidth, delay, jitter and loss.

QoS provides better network service through the following features:

- Supporting dedicated bandwidth
- Improving loss characteristics
- Avoiding and managing network congestion
- Shaping network traffic
- Setting traffic priorities across the network

The Best-Effort service is the default QoS model in IP networks. It provides no guarantee for data delivering, which means delay, jitter, packet loss and bandwidth allocation are unpredictable.

Differentiated Services (DiffServ or DS) is the most widely used QoS model. It provides a simple and scalable mechanism for classifying and managing network traffic and providing QoS on modern IP networks. Each router on the network can provide QoS simply based on the DiffServ class.

Differentiated Services Code Point (DSCP) is used to define DiffServ classes and stored in the first six bits of the ToS (Type of Service) field. The DSCP value ranges from 0 to 63 with each DSCP specifying a particular per-hop behavior (PHB) applicable to a packet.

A PHB refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet.

Four standard PHBs are available to construct a DiffServ-enabled network and achieve QoS:

- **ClassSelector PHB** -- backwards compatible with IP precedence. Class Selector code points are of the form “xxx000”. The first three bits are the IP precedence bits. These class selector PHBs retain almost the same forwarding behavior as the nodes that implement IP precedence-based classification and forwarding.
- **Expedited Forwarding PHB** -- It is the key ingredient in DiffServ model for providing a low-loss, low-latency, low-jitter and assured bandwidth service.
- **Assured Forwarding PHB** -- It defines a method by which BAs (Bandwidth Allocations) can be given different forwarding assurances.
- **Default PHB** -- It specifies that a packet marked with a DSCP value of “000000” gets the traditional best effort service from a DS-compliant node.

VoIP is extremely bandwidth- and delay-sensitive. QoS is a major issue in VoIP implementations, regarding how to guarantee that packet traffic will not be delayed or dropped due to interference of other lower priority traffic. VoIP can guarantee high-quality QoS only if the voice and the SIP packets are given priority over other kinds of network traffic. The IP phone supports the DiffServ model of QoS.

Voice QoS

In order to make VoIP transmissions intelligible to receivers, voice packets should not be dropped, excessively delayed, or made to suffer varying delay. DiffServ model can guarantee high-quality voice transmission when the voice packets are configured to a higher DSCP value.

Video QoS

To ensure acceptable visual quality for video, video packets emanated from the IP phone should be configured with a high transmission priority.

SIP QoS

SIP protocol is used for creating, modifying and terminating two-party or multi-party sessions. To ensure good voice quality, SIP packets emanated from IP phone should be configured with a high transmission priority.
DSCPs for voice and SIP packets can be specified respectively.

Wi-Fi QoS

Wi-Fi Multimedia (WMM) is based on the IEEE 802.11e standard and provides basic Quality of service (QoS) features to wireless networks. QoS enables Wi-Fi access points to prioritize the traffic and optimizes the way shared network resources are allocated among different applications.



For voice and SIP packets, the IP phone obtains DSCP info from the network policy if LLDP feature is enabled, which takes precedence over the manual settings. For more information on LLDP, refer to "Linker Layer Discovery Protocol (LLDP)"

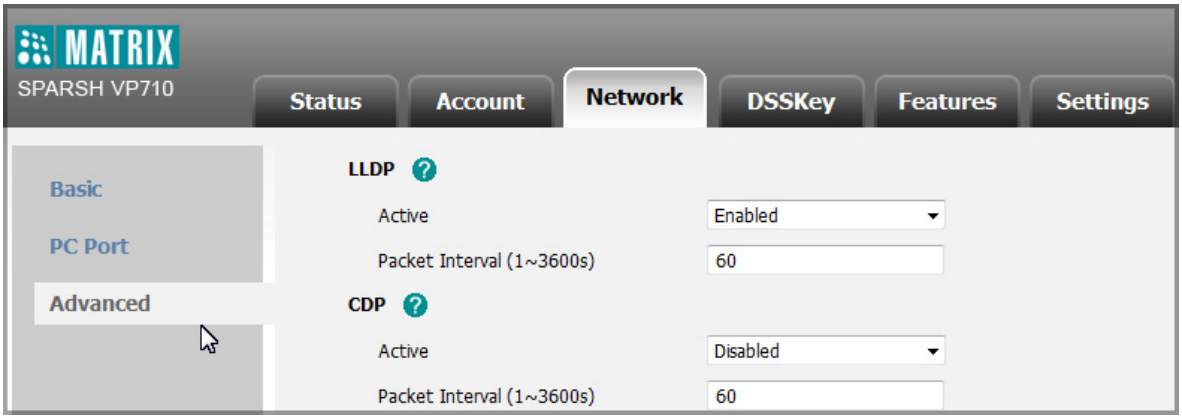
Procedure

QoS can be configured using the configuration files or locally.

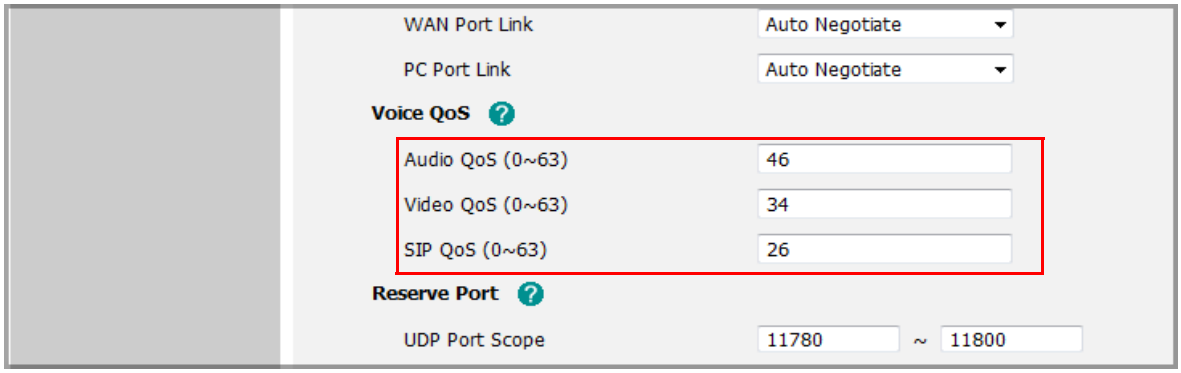
Configuration File	<MAC>.cfg	Configure the DSCPs for voice packets, SIP packets and video packet. Configure the WMM feature in the wireless network. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the DSCPs for voice packets, SIP packets and video packets. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load

To configure DSCPs for voice packets and SIP packets via web user interface:

- Click on **Network->Advanced**.



- Enter the desired value in the **Audio QoS(0~63)** field.
- Enter the desired value in the **Voice QoS(0~63)** field.
- Enter the desired value in the **SIP QoS(0~63)** field.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the phone.

Reboot in Talking

Reboot in talking allows the IP phone to reboot during an active call when it receives a reboot request by action URI. For more information on action URI, refer to "Action URI".

By default, IP phone do not receive and handle HTTP/HTTPS GET requests. To use this feature, you need to specify the trusted IP address(es) for action URI in advance. For more information, refer to "Configuring Trusted IP Address for Action URI".

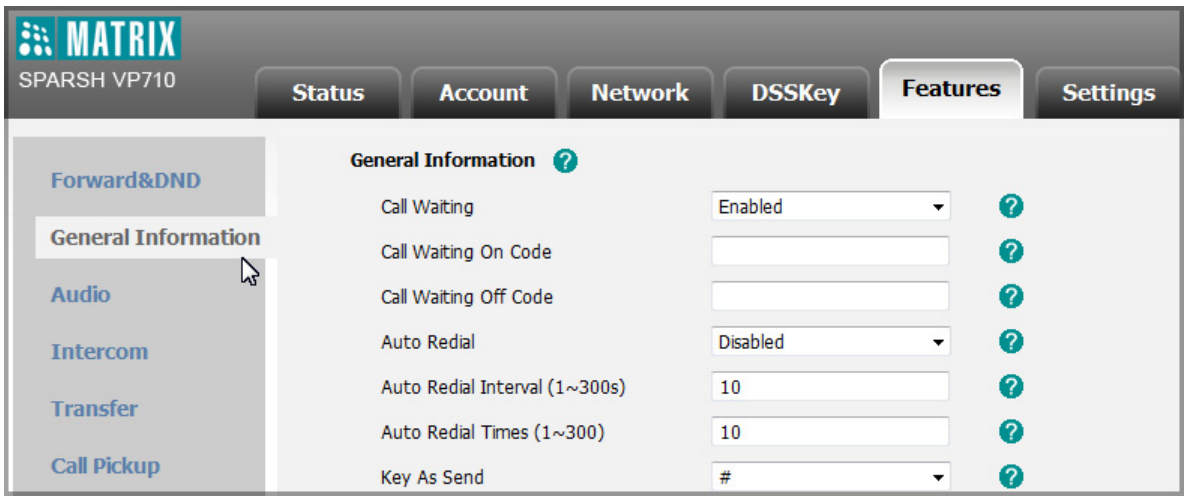
Procedure

Reboot in talking can be configured using the configuration files or locally.

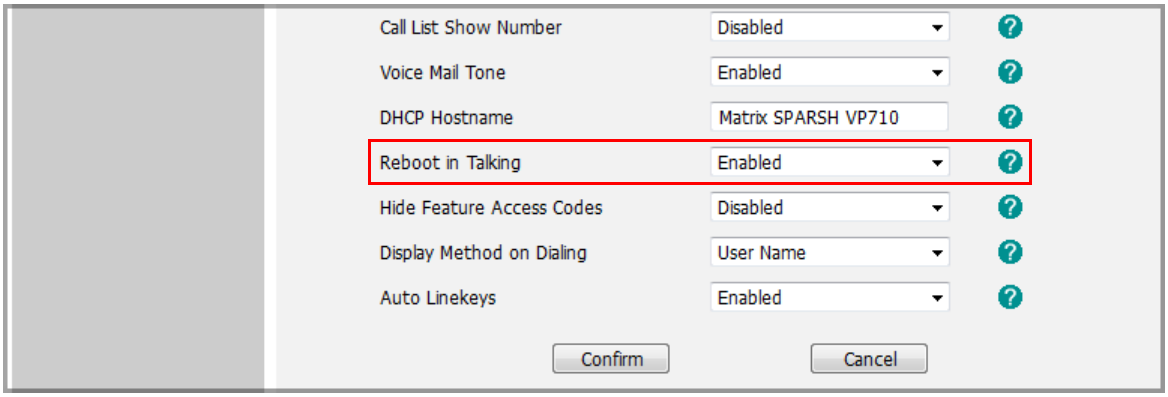
Configuration File	<MAC>.cfg	Configure reboot in talking. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure reboot in talking. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure reboot in talking via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Reboot in Talking** list.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the phone.

Redial Tone

Redial tone allows the IP phone to continue playing the dial tone after inputting the preset numbers on the pre-dialing screen.

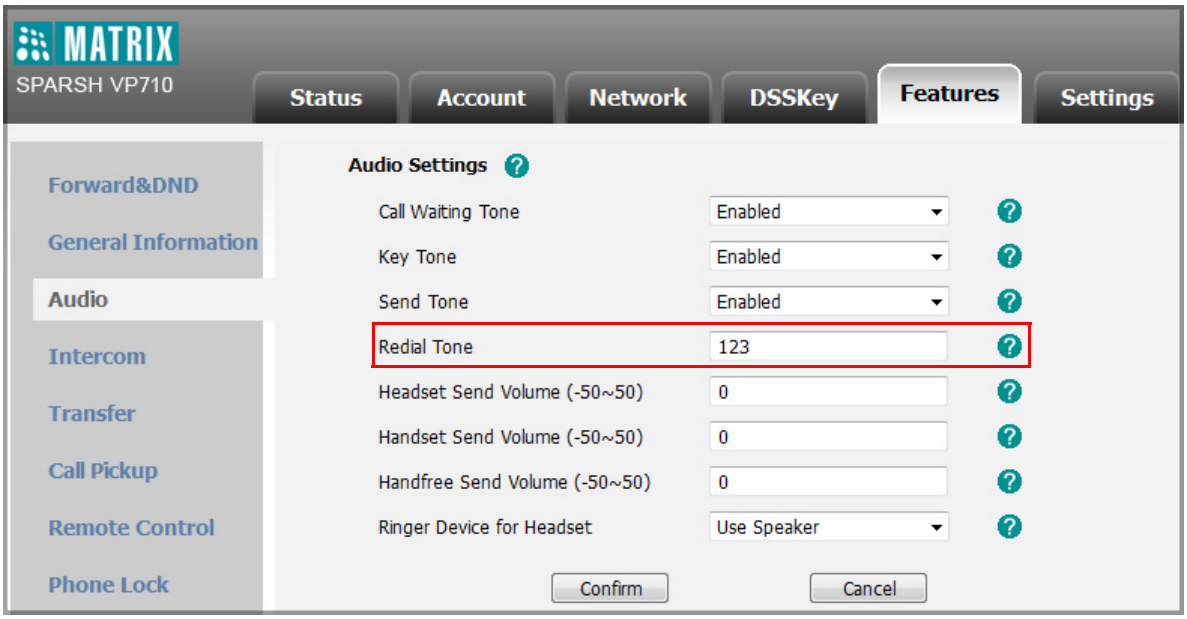
Procedure

Redial tone can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure redial tone feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure redial tone feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-audio&q=load

To configure redial tone via web user interface:

- Click on **Features->Audio.**
- Enter the desired value in the **Redial Tone** field.



- Click **Confirm** to save the change.

Reserve # in User Name

Reserve # in User Name allows the IP phone to reserve “#” in user name. When Reserve # in User Name feature is disabled, “#” will be converted into “%23”.

For example: the user registers an account (user name: 1010#) on the phone, the phone will send 1010%23 instead of 1010# in the REGISTER message or INVITE message to SIP server.

Example of a SIP REGISTER message:

```
INVITE sip:2@10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.6:5060;branch=z9hG4bK1867789050
From: "1010" <sip:1010%23@10.2.1.48:5060>;tag=1945988802
To: <sip:2@10.2.1.48:5060>
Call-ID: 0_2336101648@10.3.20.6
CSeq: 1 INVITE
Contact: <sip:1010%23@10.3.20.6:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Allow-Events: talk,hold,conference,refer,check-sync
Content-Length: 300
```

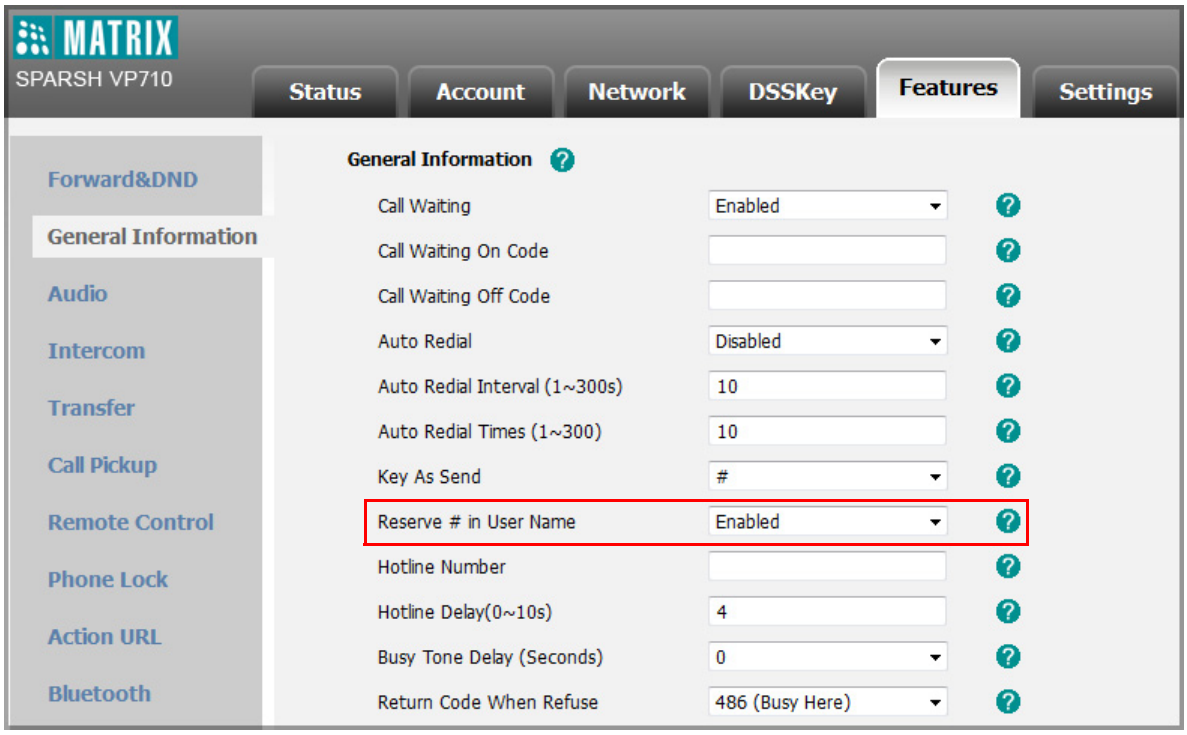
Procedure

Reserve # in User Name can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure reserve # in user name. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Configure reserve # in user name. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure reserve # in user name via web user interface:

- Click on **Features->General Information.**



- Select the desired value from the **Reserve # in User Name** list.
- Click **Confirm** to save the change.

Return Code When Refuse

Return code when refuse defines the return code and reason of the SIP response message for the refused call. The caller's touch screen displays the reason according to the received return code. Available return codes and reasons are:

- 404 (Not found)
- 480 (Temporarily not available)
- 486 (Busy here)
- 603 (Decline)

Procedure

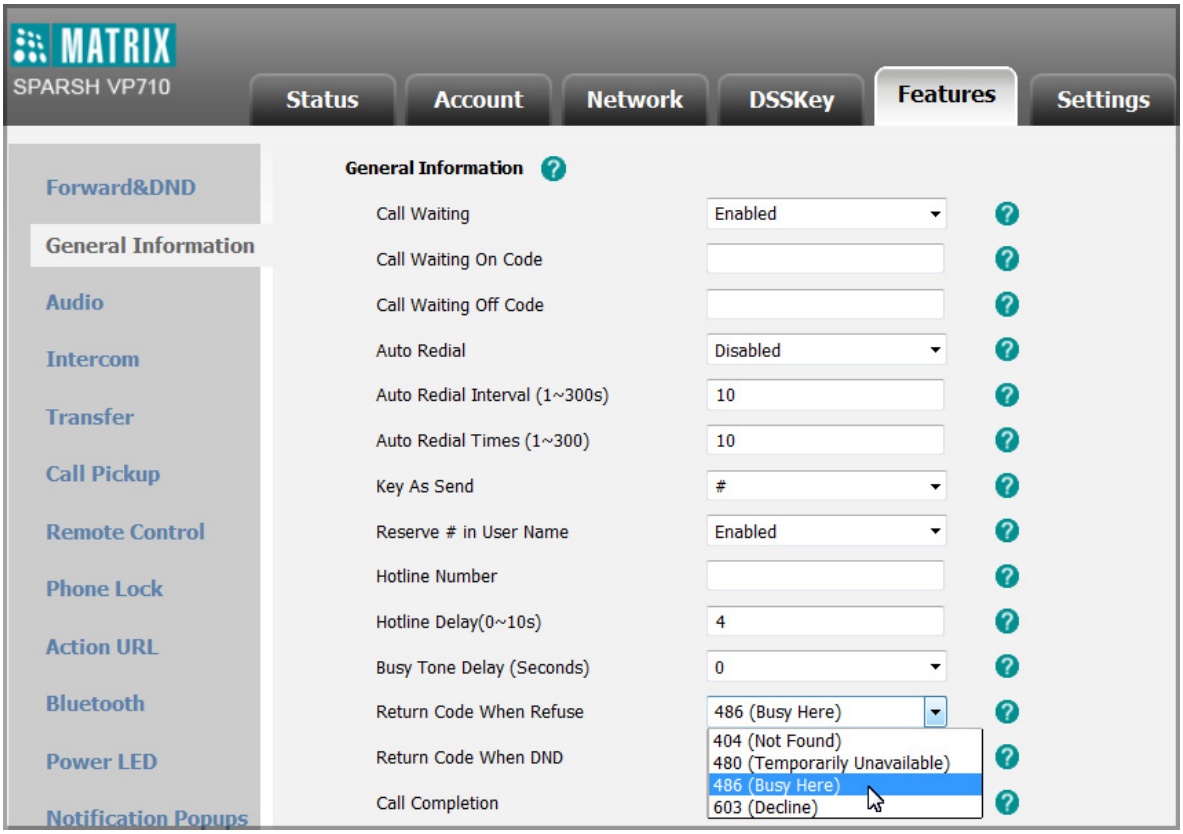
Return code for refused call can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify the return code and the reason of the SIP response message when refusing a call. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify the return code and the reason of the SIP response message when refusing a call. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To specify the return code and the reason when refusing a call via web user interface:

- Click on **Features->General Information**.

- Select the desired value from the **Return Code When Refuse** list.



- Click **Confirm** to save the change.

Ringing Timeout

Ringing timeout defines a specific period of time within which the IP phone will stop ringing if the call is not answered.

Procedure

Ringing timeout can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the duration time in the ringing state. For more information, refer "Appendix D - Configuration Parameters".

Send User=phone

When placing a call, the IP phone will send an INVITE request to the proxy server. Send user=phone feature allows adding user=phone to the SIP header of the INVITE message.

Example of a SIP INVITE message:

```
INVITE sip:101@10.2.1.48:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.3.20.6:5060;branch=z9hG4bK2475812834
From: "1010" <sip:1010@10.2.1.48:5060>;tag=3747068208
To: <sip:101@10.2.1.48:5060;user=phone>
Call-ID: 0_4008470062@10.3.20.6
CSeq: 1 INVITE
Contact: <sip:1010@10.3.20.6:5060>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Allow-Events: talk,hold,conference,refer,check-sync
Content-Length: 300
```

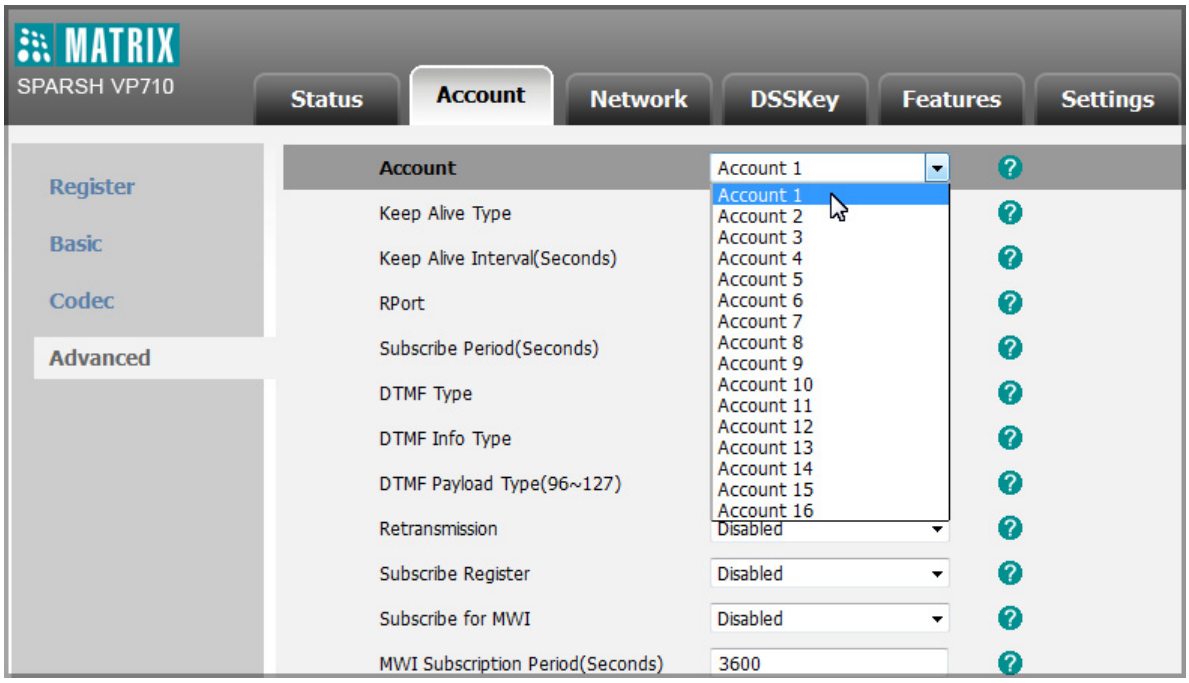
Procedure

Send User= phone can be configured using the configuration files or locally.

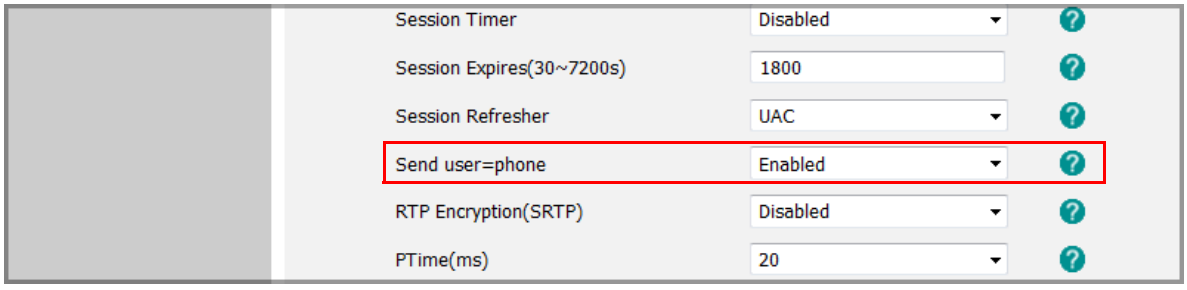
Configuration File	<MAC>.cfg	Configure send user=phone feature on a per-line basis. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure send user=phone feature on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure send user=phone feature via web user interface:

- Click on **Account->Advanced.**
- Select the desired account from the **Account** list.



- Select the desired value from the **Send user=phone** list.



- Click **Confirm** to save the change.

Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of the phone service when the connection between the IP phone and the server fails or the server fails and therefore the server needs to be taken offline for maintenance.

Two types of redundancy are possible and in some cases, a combination of the two may be deployed:

- **Failover:** In this mode, the full phone functionality is preserved by having a secondary call server with equivalent capability which takes over the server that has gone down/off-line.This mode of operation should be done using the DNS mechanism from the primary to the secondary server. Therefore, if you want to use this mode, the server must be configured with a domain name.

- **Fallback:** In this mode, a secondary less featured call server with SIP capability takes over the call control to provide basic calling capability, but without some advanced features (for example, shared line and MWI) offered by the working server. IP phone supports configuration of two servers per SIP registration for fallback purpose.

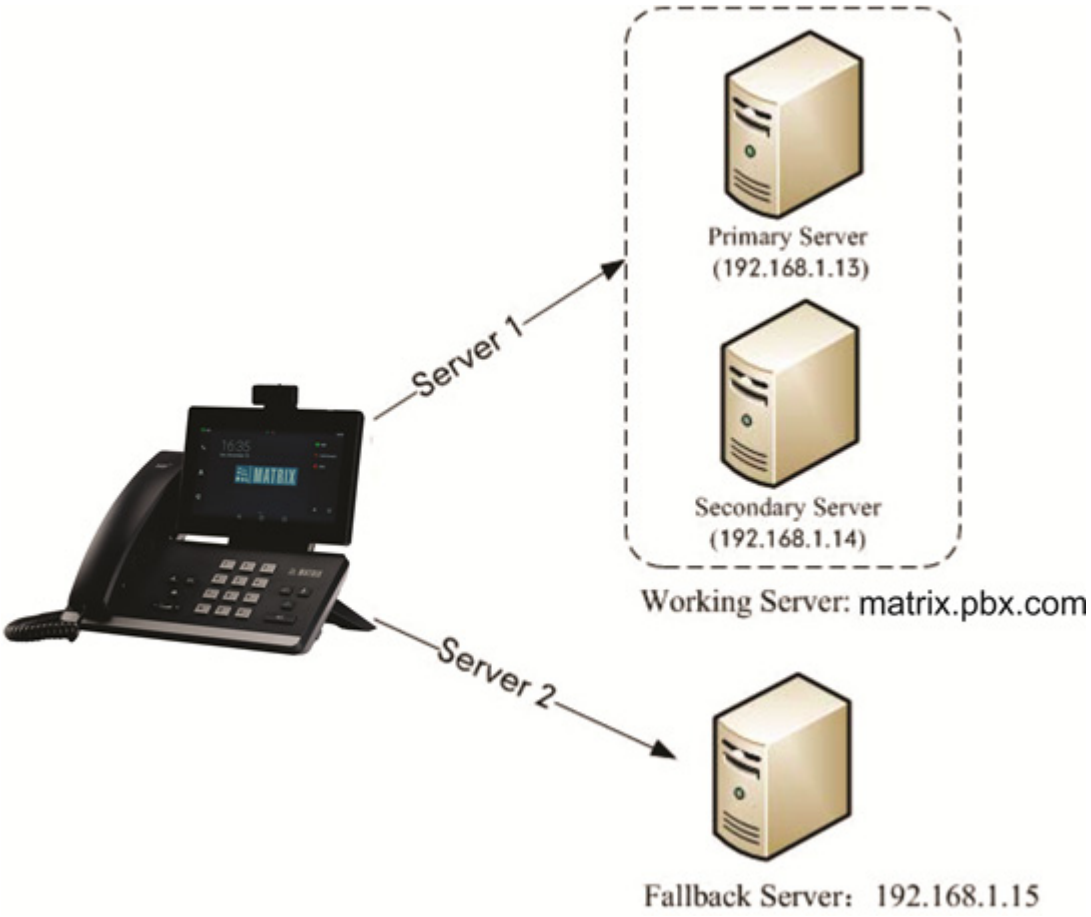


- *When using some advanced features in concurrent registration mode, the IP phone has certain limitations and for successive registration mode, the phone service may have a brief interrupt when the server fails. Therefore it is recommended to use failover mode for server redundancy as it ensure the continuity of the phone service and allow accessing of all the call features when the server fails*.

Phone Configuration for Redundancy Implementation

To assist in explaining the redundancy behavior, an illustrative example of how an IP phone may be configured is shown below.

In the example, server redundancy for fallback and failover purposes is deployed. Two separate servers (a working server and a fallback server) are configured for per account (line) registration.



- **Working Server:** Server 1 is configured with the domain name of the working server. For example: matrix.pbx.com. DNS mechanism is used such that the working server is resolved to multiple servers with different IP addresses for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server (e.g., 192.168.1.13) has the highest priority in the cluster of servers resolved by the DNS server. The secondary server (e.g., 192.168.1.14) backs up the primary server when the primary server fails and offers the same functionality as the primary server.

- **Fallback Server:** A fallback server offers less functionality than the working server.Server 2 is configured with the IP address of the fallback server.For example, 192.168.1.15.

Outgoing Call When the Working Server Connection Fails

When you initiates a call, the IP phone will go through the following steps to connect the call:

- The IP phone sends the INVITE request to the primary server.
- If the primary server does not respond correctly to the INVITE (that is, the primary server responds to the INVITE with 503 message or the request for responding with 100 Trying message times out (64*T1 seconds, defined in RFC 3261)), then the IP phone tries to make the call using the secondary server.
- If the secondary server is also unavailable, the IP phone will try to make the call through fallback server until it either succeeds in making a call or exhausts all servers at which point the call will fail.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below:

- If TCP is used, then the signaling fails if the connection or the send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list (this list contains all the server addresses resolved by the DNS server) and this is the last server, then the signaling fails after the complete UDP timeout defined in RFC 3261. If it is not the last server in the list, the maximum number of retries depends on the configured retry counts (configured by the parameter “account.X.sip_server.Y.retry_counts”).

Phone Registration

Registration method of the failover mode:

The IP phone must always register to the primary server first except in failover conditions. If this is unsuccessful, the phone will re-register as many times as configured until the registration is successful. When the primary server registration is unavailable, the secondary server will serve as the working server. As soon as the primary server registration succeeds, it returns to be the working server.

Registration methods of the fallback mode include (not applicable to outbound proxy servers):

- **Concurrent registration:** The IP phone registers to SIP server 1 and SIP server 2 (working server and fallback server) at the same time. Although the IP phone registers to two SIP servers, only one server works at the same time. In the failure situation, a fallback server can take over the basic calling capability, but without some advanced features (for example, shared lines and MWI) offered by the working server.By default,

- **Successive registration:** The IP phone only registers to one server at a time. The IP phone first registers to the working server. In the failure situation, the IP phone registers to the fallback server which can take over all calling capabilities.

When registering to the working server, the IP phone must always register to the primary server first except in failover conditions. When the primary server registration is unavailable, the secondary server will serve as the working server.

Fallback Scenario

The following introduces a REGISTER fallback scenario. The SIP server 1 (working server) and SIP server 2 (fallback server) are configured with the IP address respectively for account 1. The parameter “account.1.fallback.redundancy_type” is configured as 1 (Successive Registration).

REGISTER Fallback

The phone has ability to fail over to a fallback server when the working server has no response to a REGISTER request.

- The phone sends a REGISTER request to the working server.
- The phone retries REGISTER requests to the working server (three times by default).

- After no response from the working server, the phone sends a REGISTER request to the fallback server.
- The fallback server responds with 200 OK to the REGISTER request.

The phone sends REGISTER requests to the working server to detect whether the server is available at intervals defined by the "account.1.fallback.timeout" parameter after failing over to the fallback server. When the working server recovers, the phone has ability to fail back next REGISTER request to the working server.

The following introduces an INVITE fallback scenario. The SIP server 1 (working server) and SIP server 2 (fallback server) are configured with the IP address respectively for account 1. The parameter "account.1.fallback.redundancy_type" is configured as 0 (Concurrent Registration).

INVITE Fallback

The phone has ability to fail over to a fallback server when the working server has no response to an INVITE request.

- Phone A places a call to Phone B.
- Phone B answers the call.

  The following SIP messages appear:

  - Phone A sends an INVITE request to the working server.
  - Phone A retries INVITE requests to the working server (three times by default).
  - After no response from the working server, the phone sends an INVITE request to the fallback server.
  - The fallback server responds with 200 OK to the INVITE request.

Phone A sends REGISTER requests to the working server to detect whether the server is available. When the working server recovers, the phone has ability to fail back the INVITE request to the working server.

Failover Scenario

The following introduces a REGISTER failover scenario. The SIP server 1 is configured with the domain name of the working server for account 1. The working server is resolved to two SIP servers (primary server and secondary server) using the DNS mechanism. The parameter "account.1.sip_server.1.failback_mode" is configured as 0 (newRequests) and "account.1.sip_server.1.register_on_enable" is configured as 0 (Disabled).

REGISTER Failover

The phone has ability to fail over to a secondary server when the primary server has no response to a REGISTER request.

- The phone sends REGISTER request to the primary server.
- The phone retries REGISTER requests to the primary server (three times by default).
- After no response from the primary server, the phone sends a REGISTER request to the secondary server.
- The secondary server responds with 200 OK to the REGISTER request.

The phone waits until next REGISTER attempt and then sends next REGISTER request to the primary server. When the primary server recovers, the phone has ability to fail back next REGISTER request to the primary server.

INVITE Failover

The phone has ability to fail over to a secondary server when there is no response to an INVITE request.

- Phone A places a call to Phone B.
- Phone B answers the call.

  The following SIP messages appear:

  - Phone A sends an INVITE request to the primary server.
  - Phone A retries INVITE requests to the primary server (three times by default).
  - After no response from the primary server, the phone sends an INVITE request to the secondary server.
  - The secondary server responds with 200 OK to the INVITE request.

When phone A places a call to Phone B again, the phone sends an INVITE request to the primary server first. When the primary server recovers, the phone has ability to immediately fail back INVITE request to the primary server after failing over to the secondary server.

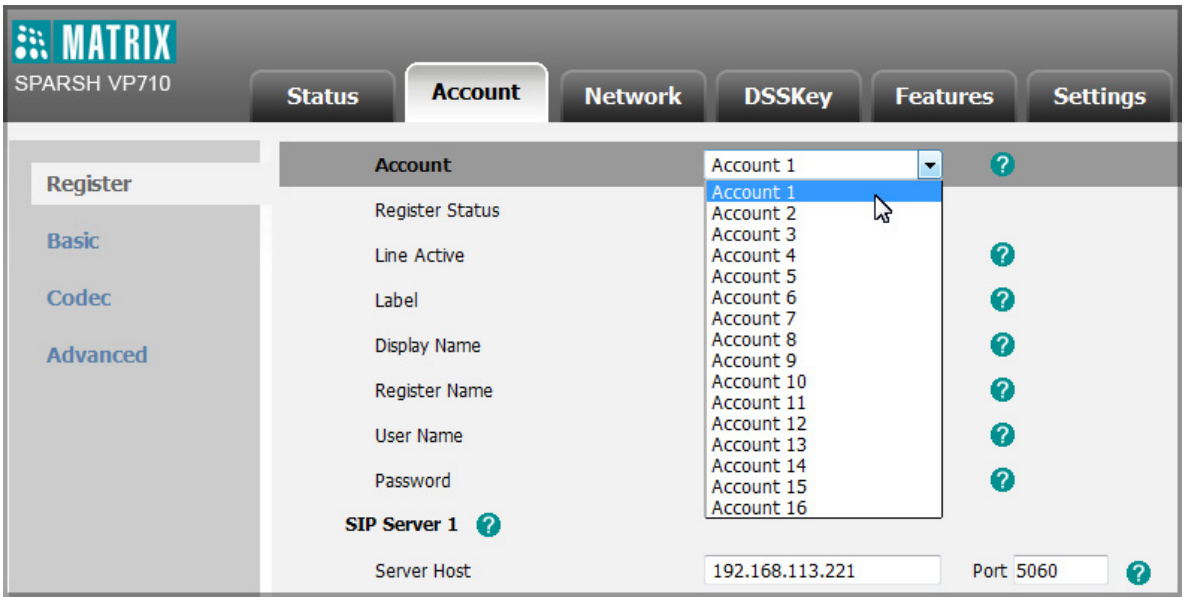
Procedure

Server redundancy can be configured using the configuration files or locally.

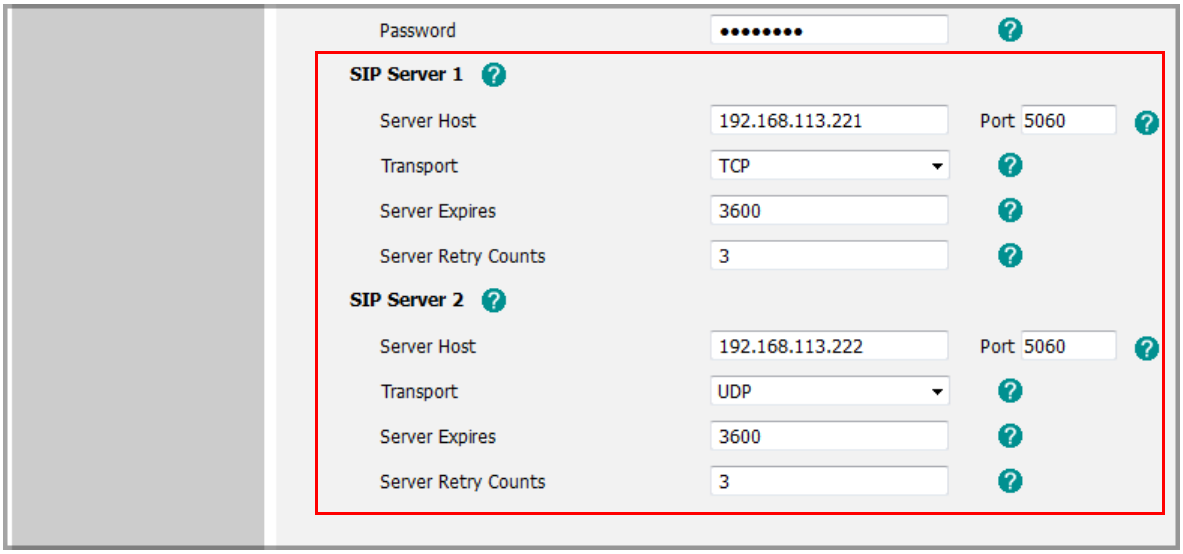
Configuration File	<MAC>.cfg	Configure the SIP server redundancy. Configure the outbound proxy server redundancy. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the server redundancy on the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-register&q=load&acc=0

To configure server redundancy for fallback purpose via web user interface:

- Click on **Account->Register**.
- Select the desired account from the **Account** list.

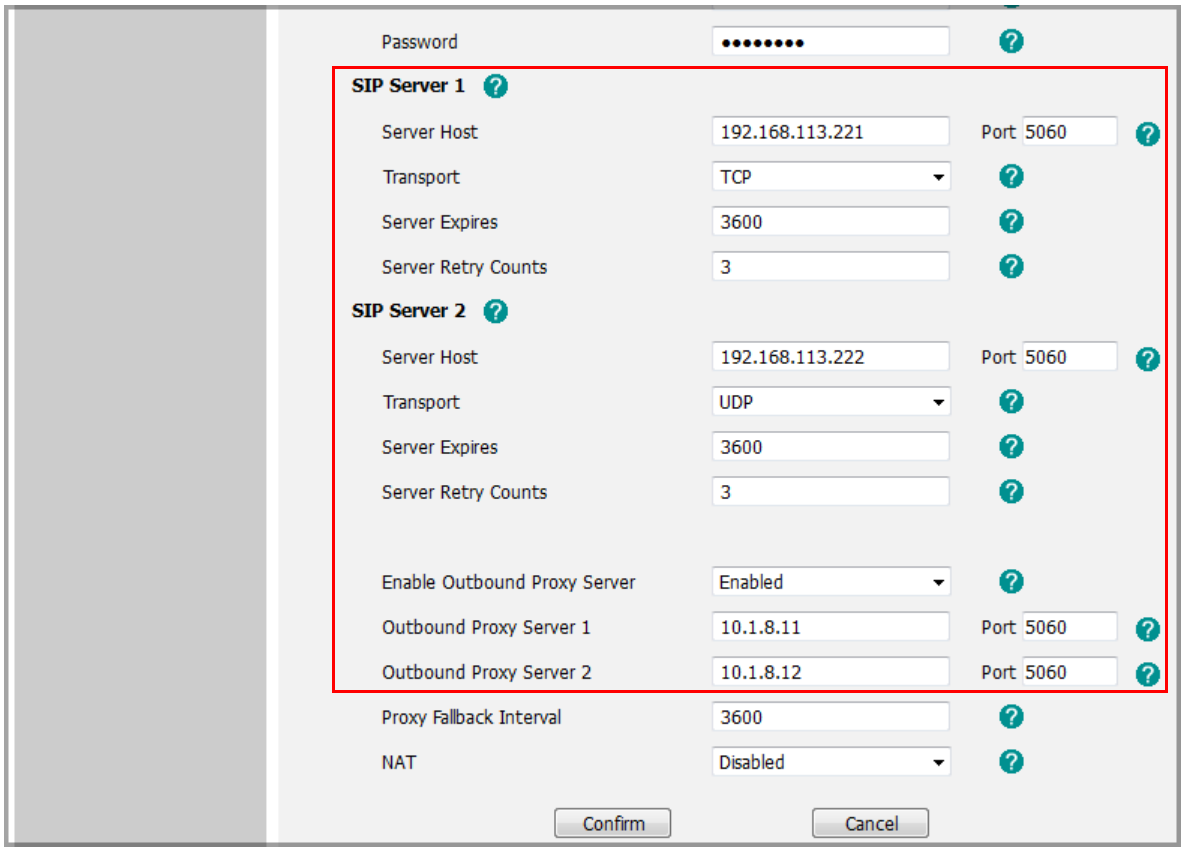


- Configure registration parameters of the selected account in the corresponding fields
- Configure parameters of **SIP Server 1** and **SIP Server 2** in the corresponding fields.



- If you use outbound proxy servers, do the following:
  - Select **Enabled** from the **Enable Outbound Proxy Server** list.

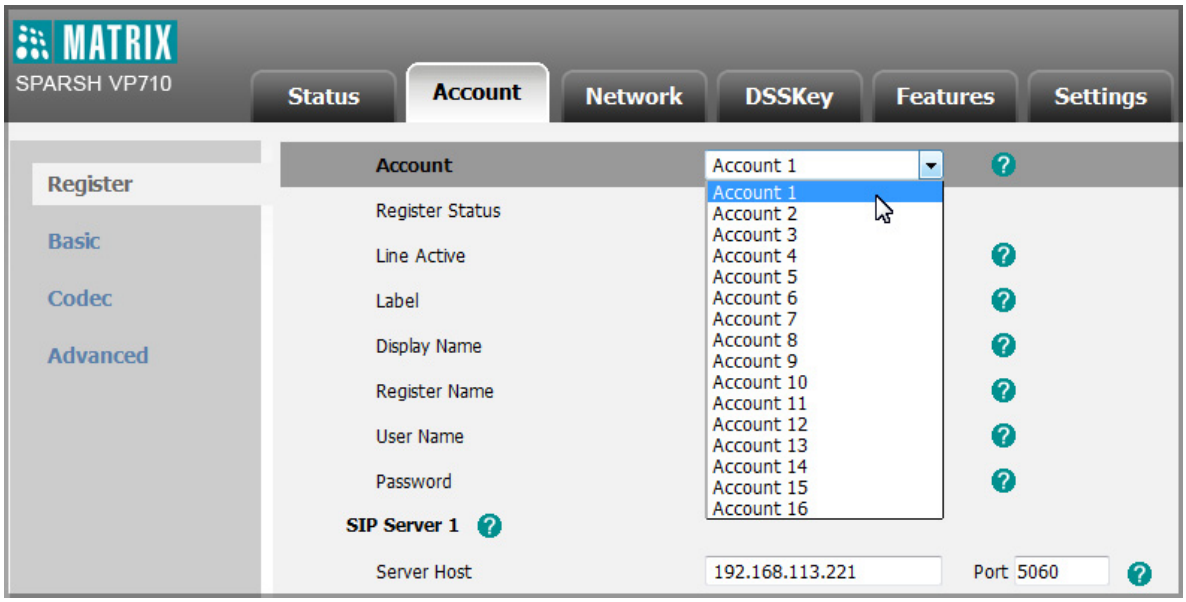
- Configure parameters of **Outbound Proxy Server 1** and **Outbound Proxy Server 2** in the corresponding fields.



- Click **Confirm** to save the change.

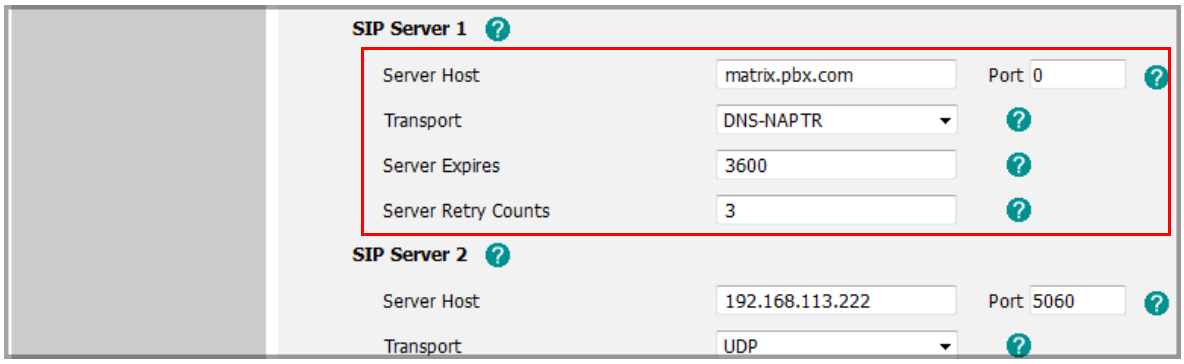
To configure server redundancy for failover purpose via web user interface:

- Click on **Account->Register**.
- Select the desired account from the **Account** list.

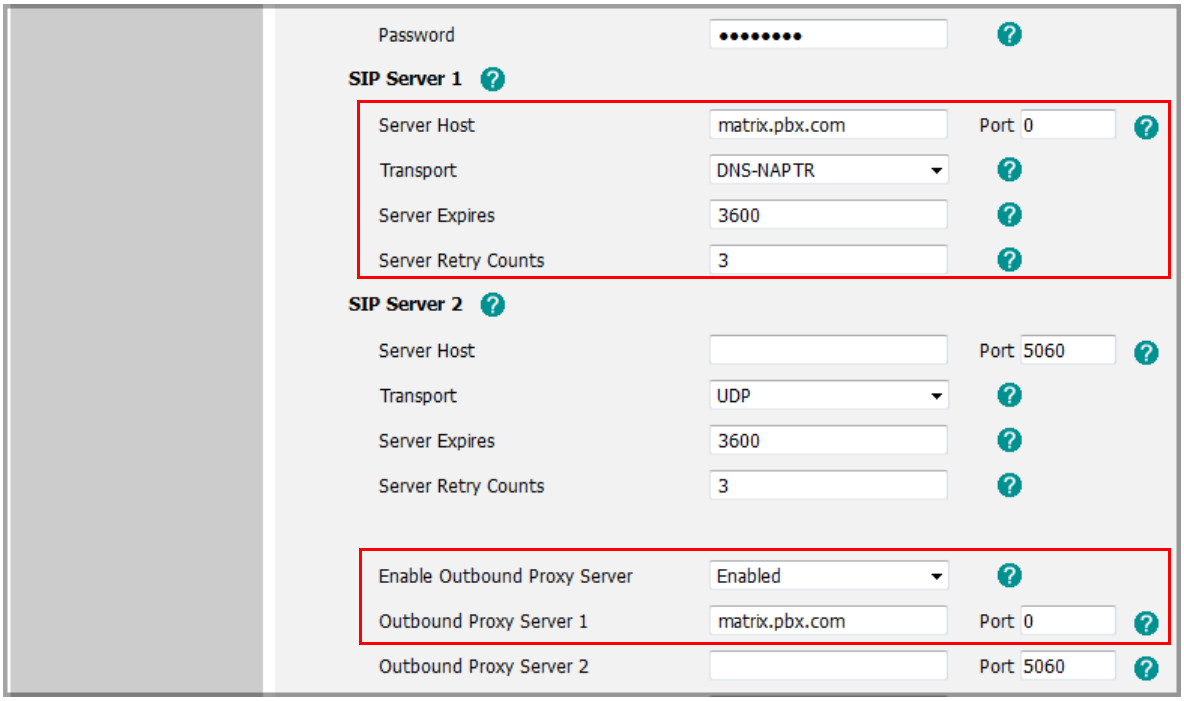


- Configure registration parameters of the selected account in the corresponding fields.
- Configure parameters of the **SIP Server 1** or **SIP Server 2** in the corresponding fields.

  You must set the **Port** of SIP server to 0 for NAPTR, SRV and A queries
- Select **DNS-NAPTR** from the **Transport** list.



- If you use outbound proxy servers, do the following
  - Select Enabled from the **Enable Outbound Proxy Server** list.
  - Configure parameters of outbound proxy server 1/2 in the corresponding fields.
    You must set the **Port** of outbound proxy server to 0 for NAPTR, SRV and A queries.



- Click **Confirm** to save the change.

Server Domain Name Resolution

If a domain name is configured for a SIP server, the IP address(es) associated with that domain name will be resolved through DNS as specified by RFC 3263. The DNS query involves NAPTR, SRV and A queries, which allows the IP phone to adapt to various deployment environments.

- NAPTR query: IP phone sends the NAPTR query to get the NAPTR pointer and transport protocol (UDP, TCP and TLS).
- SRV query: IP phone performs SRV query on the record returned from the NAPTR for the target domain name and the port number.
- A query: IP phone performs A query for the IP addresses.If an explicit port (except 0) is specified and the transport type is set to DNS-NAPTR, A query will be performed only.

If a SIP server port is set to 0 and the transport type is set to DNS-NAPTR, NAPTR and SRV queries will be tried before falling to A query. If no port is found through the DNS query, 5060 will be used.

The following details the procedures of DNS query for the IP phone to resolve the domain name (for example, matrix.pbx.com) of working server into the IP address, port and transport protocol.

NAPTR (Naming Authority Pointer)

First, the IP phone sends NAPTR query to get the NAPTR pointer and transport protocol.

Example of NAPTR records:

```
              order  pref  flags  service      regexp        replacement

IN NAPTR      90     50    "s"    "SIP+D2T"    ""      _sip._tcp.matrix.pbx.com
IN NAPTR      100    50    "s"    "SIP+D2U"    ""      _sip._udp.matrix.pbx.com
```

Parameters are explained in the following table:

Parameter	Description
order	Specify preferential treatment for the specific record. The order is from lowest to highest, lower order is more preferred.
pref	Specify the preference for processing multiple NAPTR records with the same order value. Lower value is more preferred.
flags	The flag “s” means to perform an SRV lookup.

service	Specify the transport protocols: SIP+D2U: SIP over UDP SIP+D2T: SIP over TCP SIP+D2S: SIP over SCTP SIPS+D2T: SIPS over TCP
regexp	Always empty for SIP services.
replacement	Specify a domain name for the next query.

The IP phone picks the first record, because its order of 90 is lower than 100.

The pref parameter is unimportant as there is no other record with order 90.
The flag "s" indicates performing the SRV query next.
TCP will be used, targeted to a host determined by an SRV query of "_sip._tcp.matrix.pbx.com". If the flag of the NAPTR record returned is empty, the IP phone will perform NAPTR query again according to the previous NAPTR query result.

SRV (Service Location Record)

The IP phone performs an SRV query on the record returned from the NAPTR for the host name and the port number. Example of SRV records:

	Priority	Weight	Port	Target
IN SRV	0	1	5060	server1.matrix.pbx.com
IN SRV	0	2	5060	server2.matrix.pbx.com

Parameters are explained in the following table:

Parameter	Description
Priority	Specify preferential treatment for the specific host entry. Lower priority is more preferred.
Weight	When priorities are equal, weight is used to differentiate the preference. The preference is from highest to lowest. Keep the same to load balance.
Port	Identify the port number to be used.
Target	Identify the actual host for an A query.

SRV query returns two records.

The two SRV records point to different hosts and have the same priority 0. The weight of the second record is higher than the first one, so the second record is picked first. The two records also contain a port “5060”, the IP phone uses this port. If the Target is not a numeric IP address, the IP phone performs A query. So in this case, the IP phone uses “server1.matrix.pbx.com" and “server2.matrix.pbx.com" for the A query.

A (Host IP Address)

The IP phone performs an A query for the IP address of each target host name.
Example of A records:

Server1.matrix.pbx.com IN A 192.168.1.13
Server2.matrix.pbx.com IN A 192.168.1.14

The IP phone picks the IP address "192.168.1.14" first.

Procedure

SIP Server Domain Name Resolution can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transport method on the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the transport type on the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-register&q=load&acc=0

Session Timer

Session timer allows SIP sessions to refresh periodically through an UPDATE request, to determine whether a SIP session is still active or not. Session timer is specified in RFC 4028.

IP phone supports two refresher modes: UAC and UAS

Whether the endpoint functions as a UAC or a UAS depends on the UA that initiates the SIP request. If the initiator is configured as UAC, the other client or the SIP server will function as a UAS. If the initiator is configured as UAS, the other client or the SIP server will function as a UAC.

The session expiration is negotiated via the Session-Expires header in the INVITE message. The negotiated refresher is always the UAC and it will send an UPDATE request at the negotiated session expiration. The value “refresher=uac” included in the UPDATE message means that the UAC performs the refresh.

Example of UPDATE message (UAC mode):

```
UPDATE sip:1058@10.10.20.34:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.20.32:5060;branch=z9hG4bK2104991394
From: "10111" <sip:10111@10.2.1.48:5060>;tag=2170397024
To: <sip:1058@10.2.1.48:5060>;tag=200382096
Call-ID: 4_1556494084@10.10.20.32
CSeq: 2 UPDATE
Contact: <sip:10111@10.10.20.32:5060>
Max-Forwards: 70
User-Agent: User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Session-Expires: 90;refresher=uac
Supported: timer
Content-Length: 0
```

Procedure

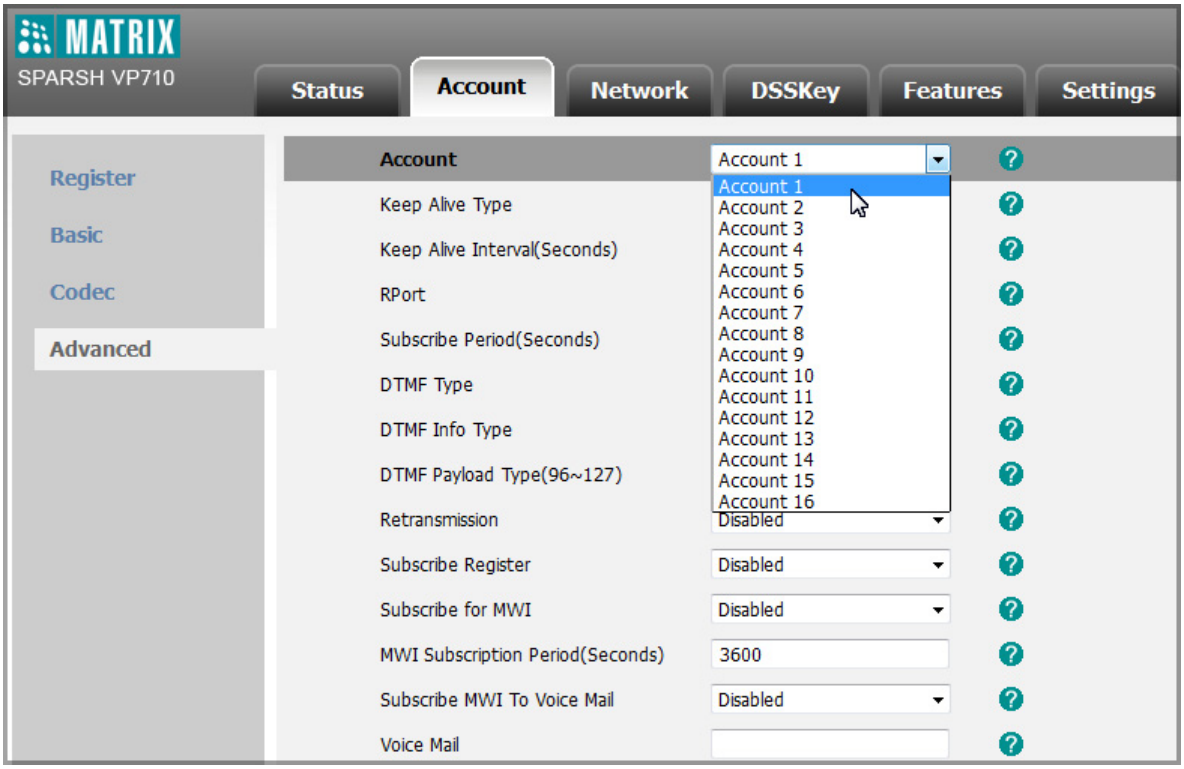
Session timer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure session timer. For more information, refer “Appendix D - Configuration Parameters”.

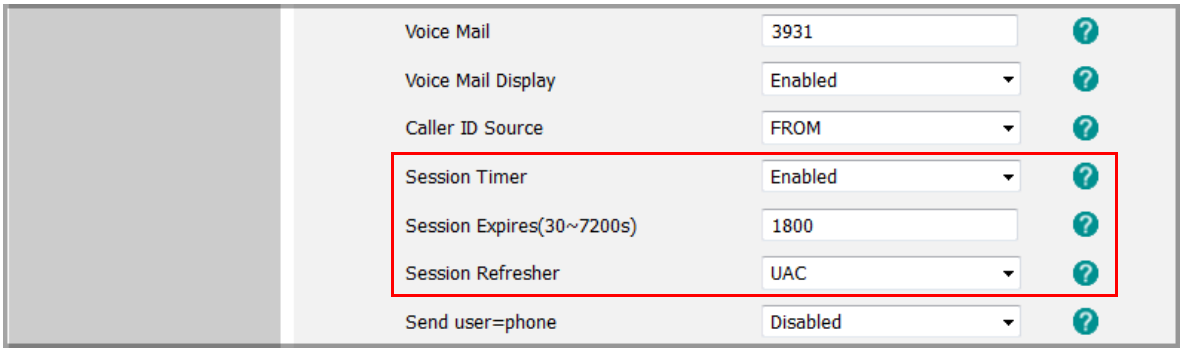
Local	Web User Interface	Configure session timer. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure session timer via web user interface:

- Click on **Account-> Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Session Timer** list.
- Enter the desired time interval in the **Session Expires (30~7200s)** field.
- Select the desired refresher from the **Session Refresher** list.



- Click **Confirm** to save the change.

Shared Call Appearance

SCA allows users to share an extension which can be registered on two or more IP phones at the same time. For more information on how to register accounts, refer to “Account Registration”. If you want to customize multiple DSS keys to associate with an account, refer to “Multiple Line Keys per Account”.

Any IP phone can be used to originate or receive calls on the shared line. An incoming call can be presented to multiple phones simultaneously. The incoming call can be answered on any IP phone but not all. A call that is active on one IP phone will be presented visually to other IP phones that share the call appearance. IP phones support SCA using a SUBSCRIBE/NOTIFY mechanism as specified in RFC 3265. The events used are:

- “call-info” for call appearance state notification
- “line-seize” for the IP phone to ask to seize the line

SCA supports the IP phones barging in an active call. In addition, SCA has the call pull capability. Call pull feature allows users to retrieve an existing call from another shared phone that is in active or public hold status.

If the call is placed on public hold, the held call is available for any shared party to retrieve. If the call is placed on private hold, the held call is only available for the hold party to retrieve. You need to configure either the private hold soft key or a private hold key before you place the call on private hold.

Using SCA Feature on the IP Phone

This section provides you with detailed information on using the phone in a SCA Hybrid Key System scenario. In a SCA Hybrid Key System scenario, the status of the line key icon which associates with a shared line will change. For more information on line key icon, refer “Icon Instructions”.

You can do the following using the IP phone in a SCA Hybrid Key System scenario:

- Place calls
- Answer calls
- Place a call on hold
- Retrieve a held call

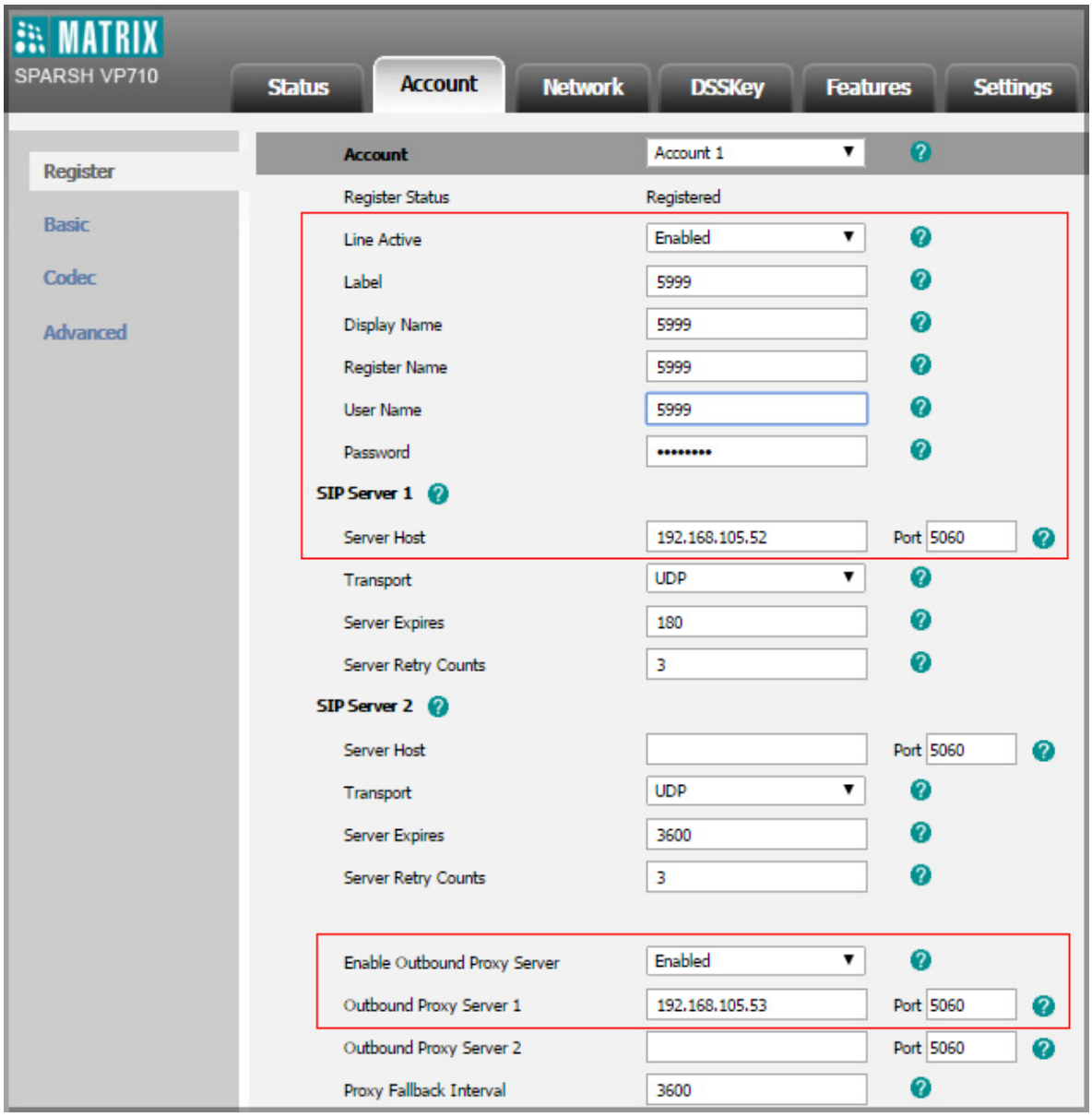
Procedure

CLIP can be configured using the configuration files or locally.

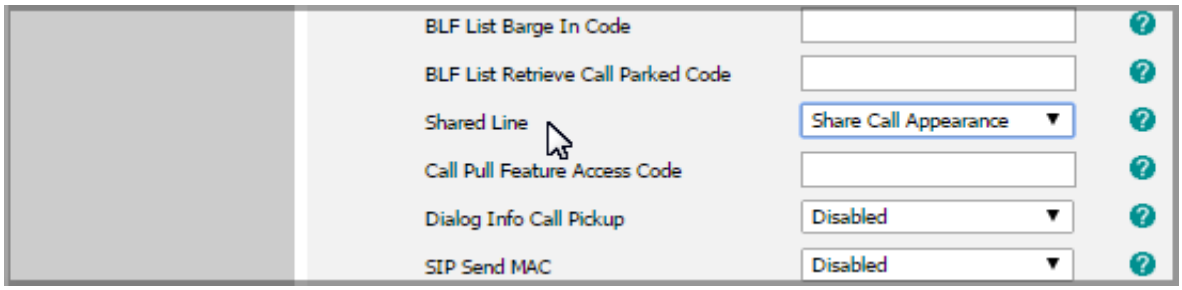
Configuration File	<MAC>.cfg	Configure the registration line type. Configure the call pull feature access code. Configure the private hold soft key. Assign a private hold key. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the registration line type. Configure the call pull feature access code. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0 Configure the private hold soft key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-softkey&q=load Assign a private hold key. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=dsskey&q=load
	Phone User Interface	Assign a private hold key.

To configure the shared line settings on the primary phone via web user interface:

- Register the primary account (e.g., 5999).



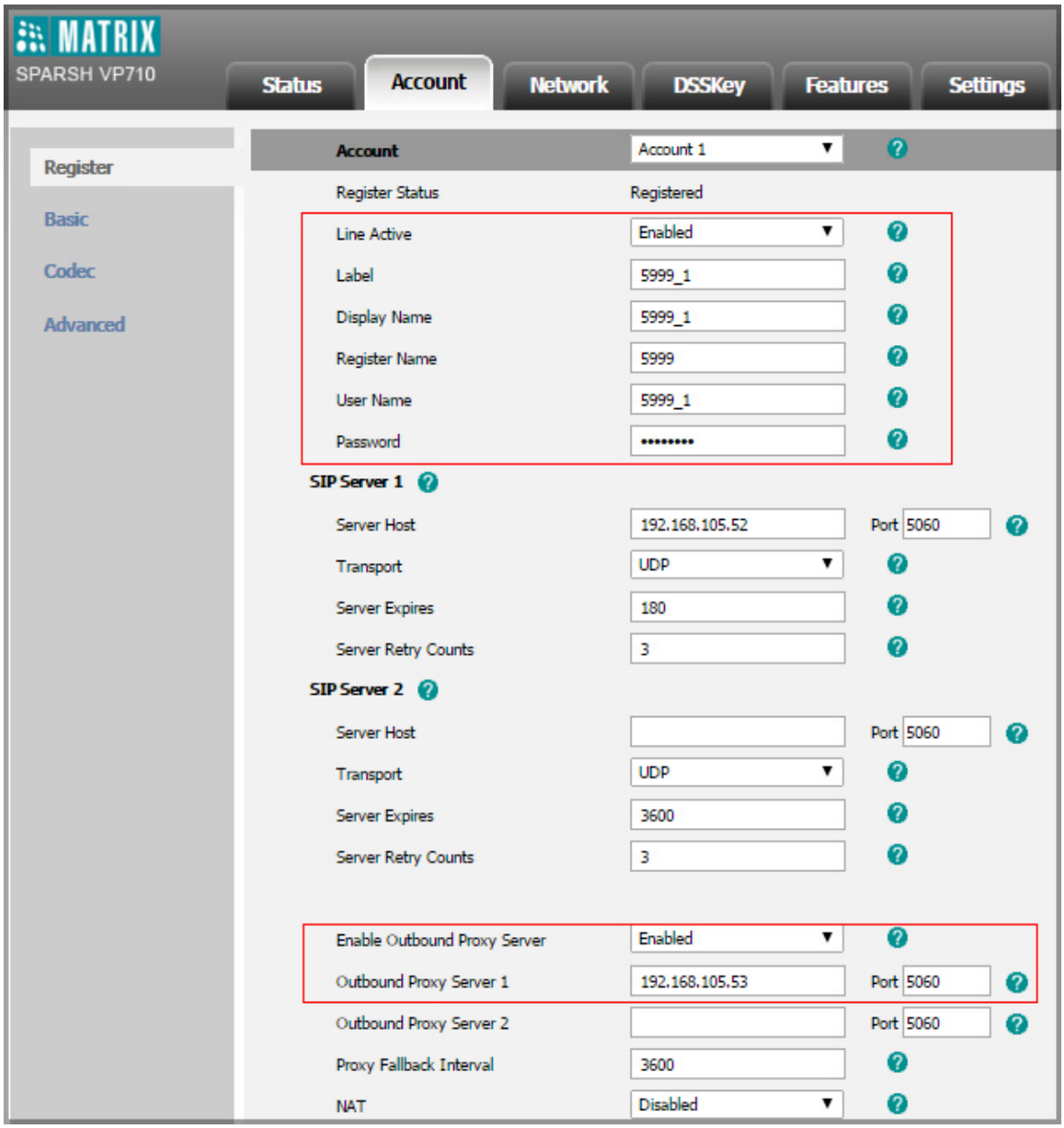
- Click on **Advanced**, select **Shared Call Appearance** from the **Shared Line** list.



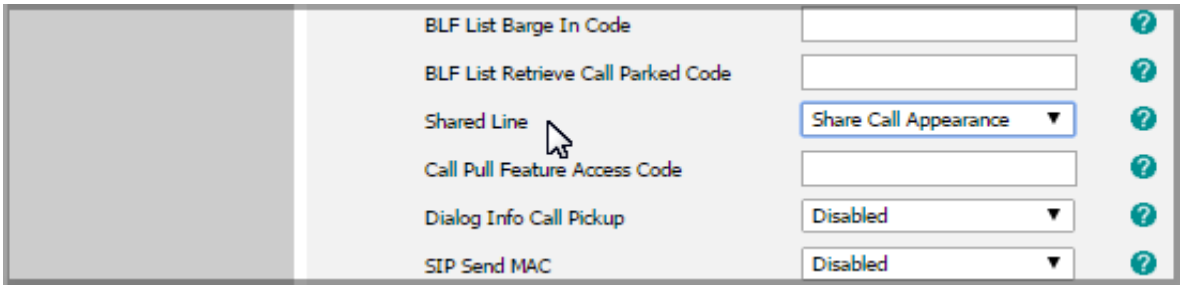
- Click **Confirm** to save the change.

To configure the shared line settings on alternate phone via web user interface:

- Register the alternate account (e.g., 4609_1).
  (Enter the primary account 4609 in the Register Name field.)



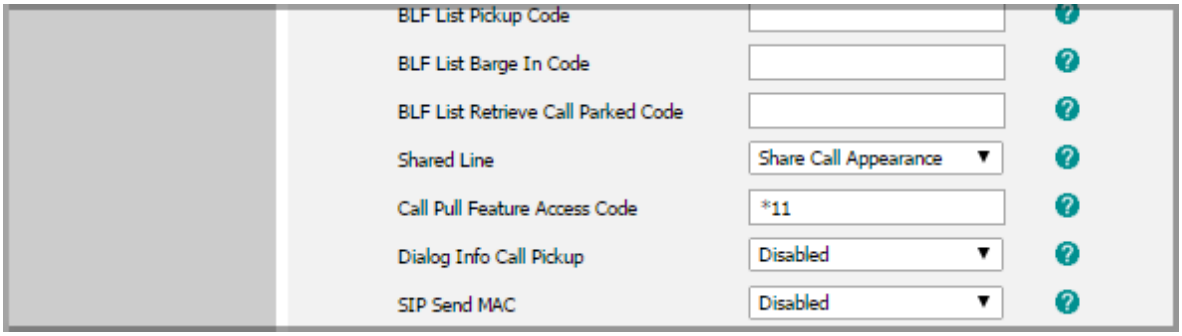
- Click on **Advanced**, select **Shared Call Appearance** from the **Shared Line** list.



- Click **Confirm** to save the change.

To configure the call pull feature access code via web user interface:

- Click on **Account->Advanced.**
- Select the desired account from the **Account** list.
- Enter the call pull feature access code (e.g., *11) in the **Call Pull Feature Access Code** field.



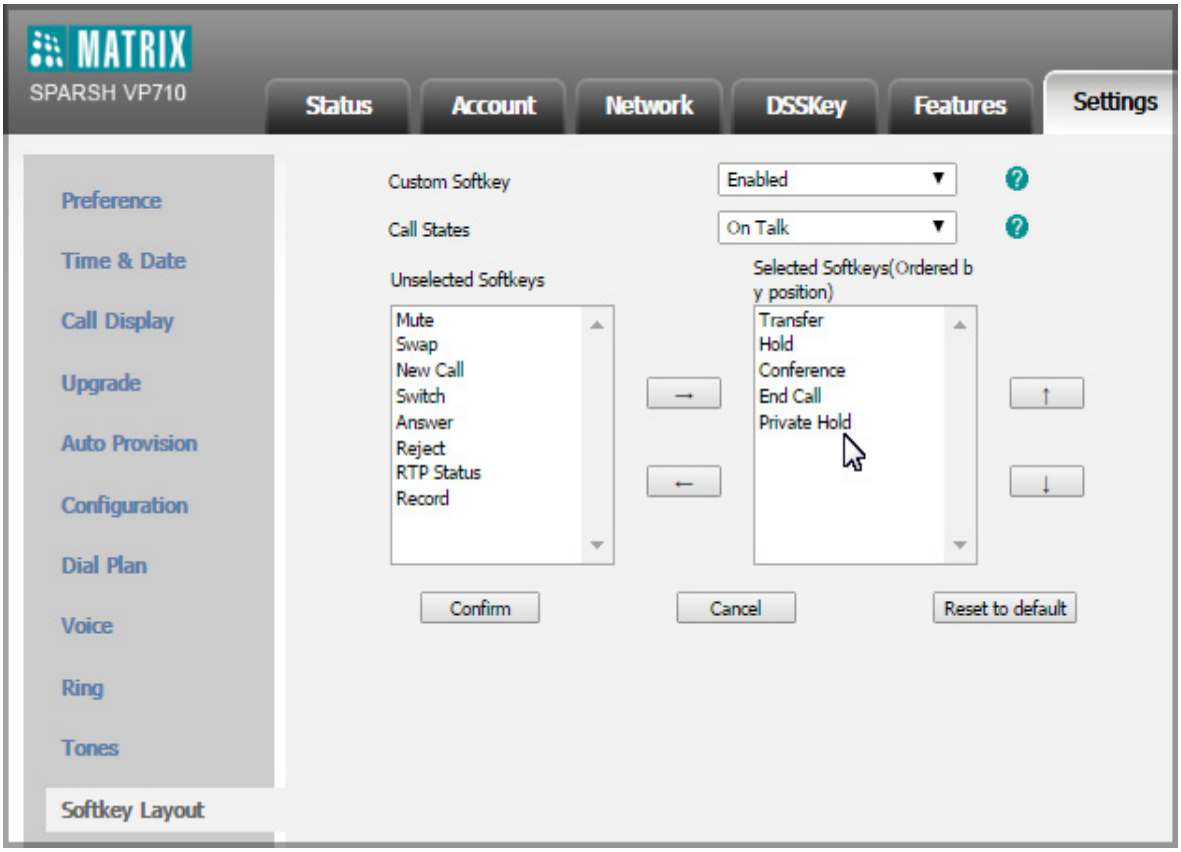
- Click **Confirm** to save the change.

.

To configure the private hold soft key via web user interface:

- Click on **Settings->Softkey Layout.**
- Select **Enabled** from the **Custom Softkey** list.
- Select **On Talk** from the **Call States** list.
- Select **Private Hold** from the Unselected Softkeys column and then click → .

The Private Hold appears in the **Selected Softkeys** column.

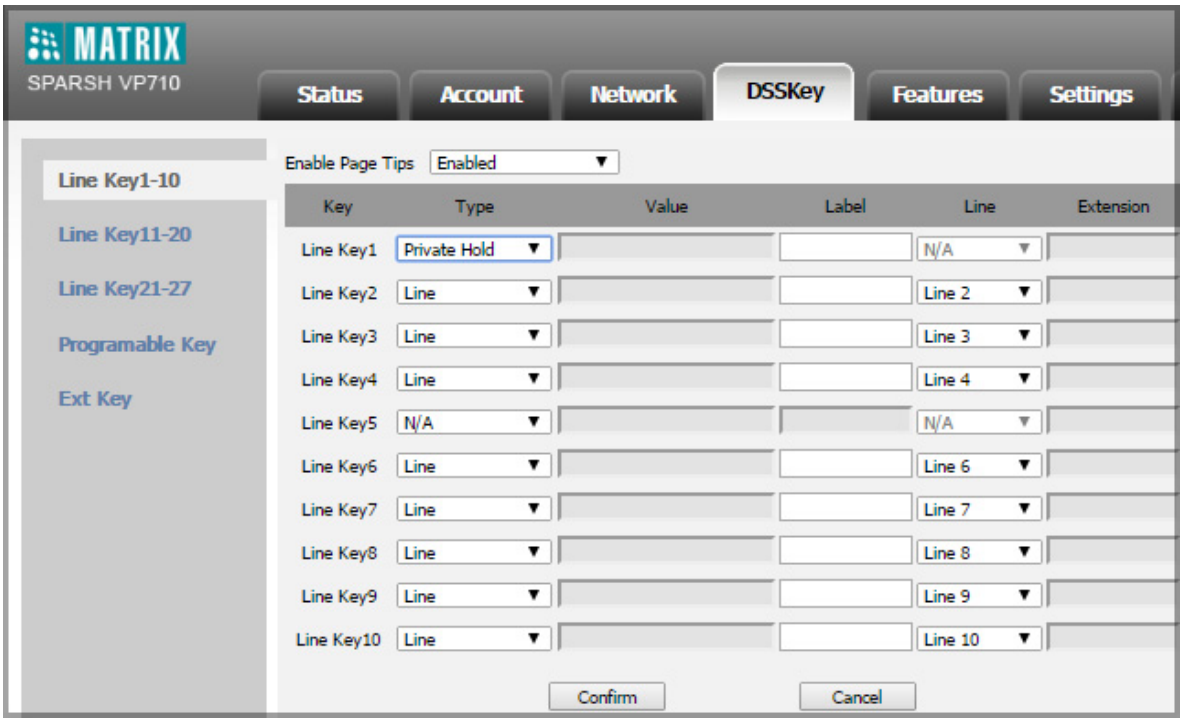


- Click **Confirm** to save the change.

To configure a private hold key via web user interface:

- Click on **DSSKey->Line Key (or Programable Key)**.
- In the desired **DSS key** field, select **Private Hold** from the **Type** list.

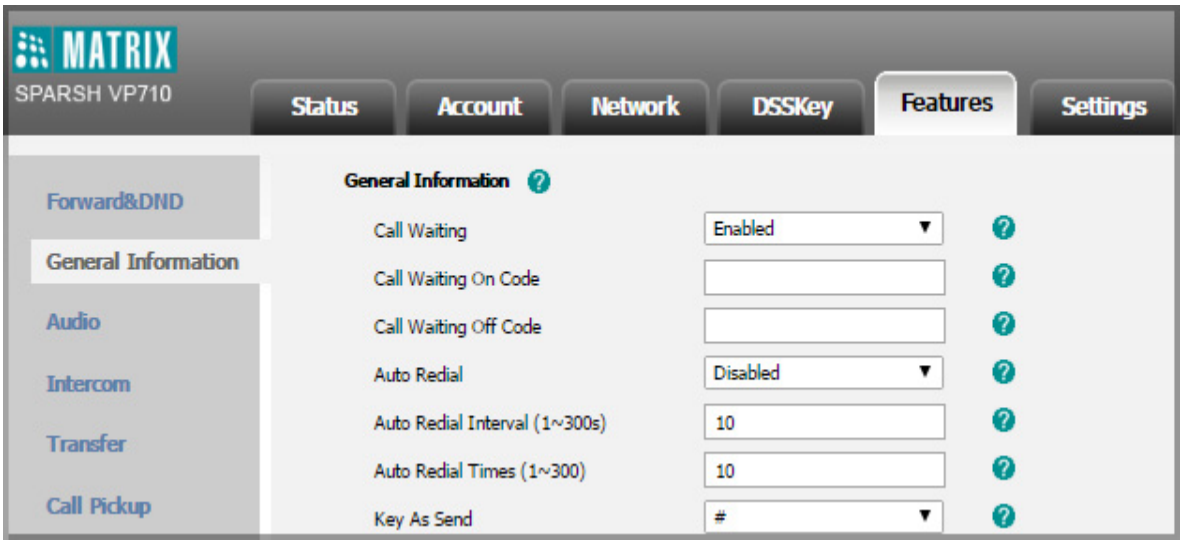
- Enter the string that will appear on the touch screen in the **Label** field.



- Click **Confirm** to save the change.

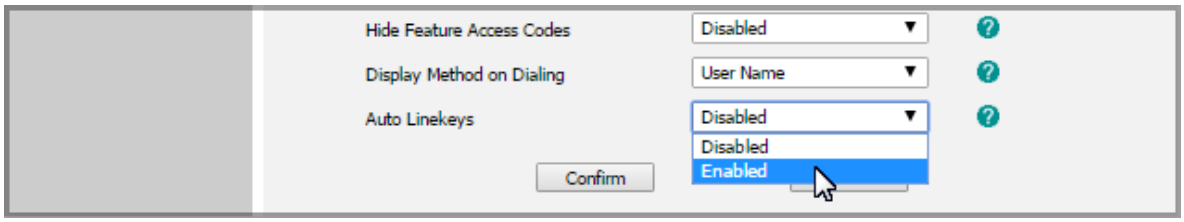
To configure the auto linekeys feature via web user interface:

- Click on **Features-> General Information.**



- Select **Enabled** from the**Auto LineKeys** list**.**

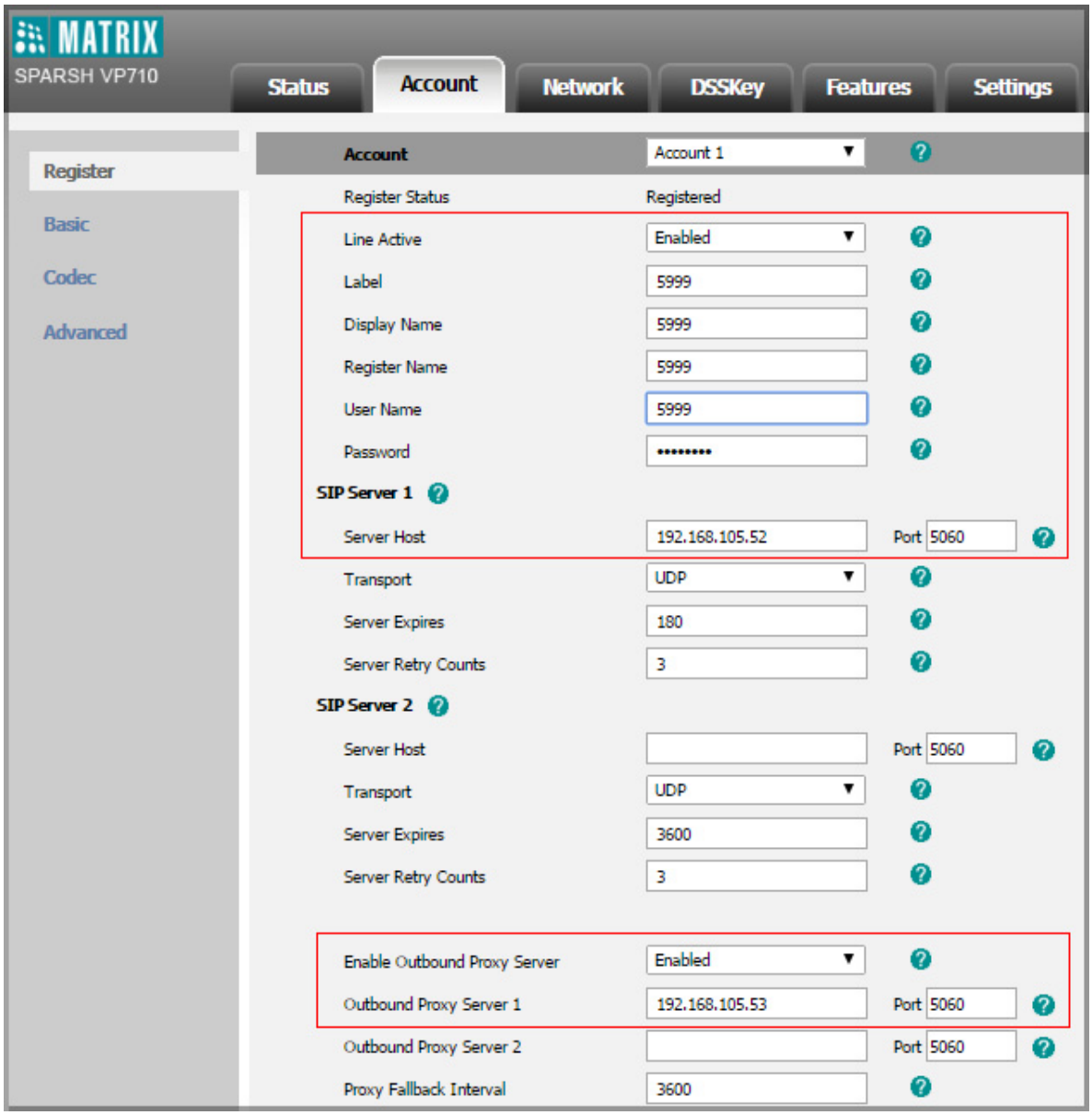
If Auto LineKeys is enabled, you can automatically assign multiple DSS keys with Line type for a registered shared line on the phone.



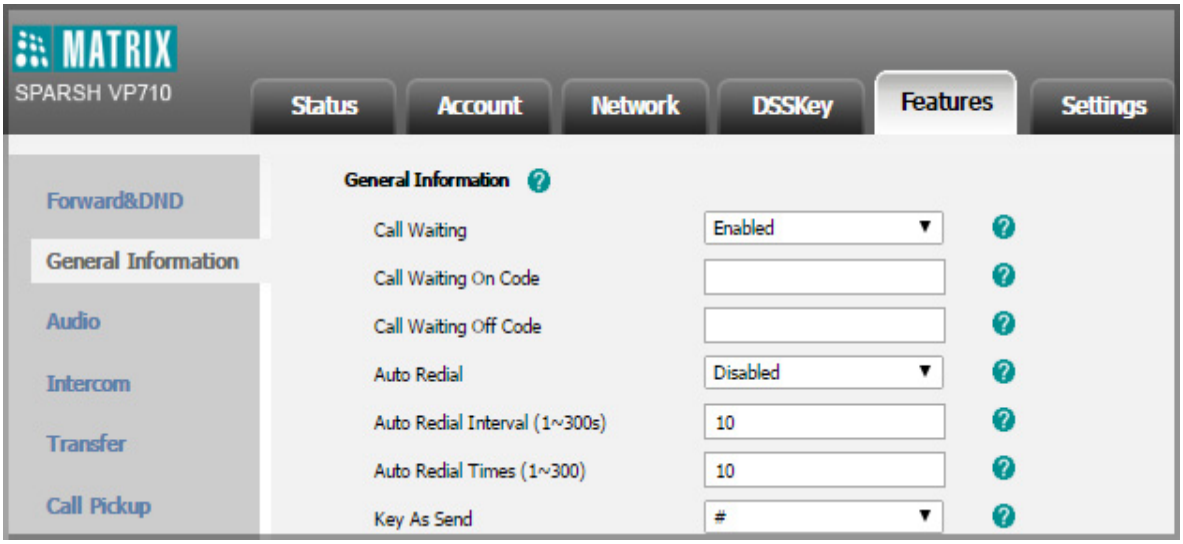
- Click **Confirm** to save the change.

To configure the shared line settings and the number of line key on phone A via web user interface:

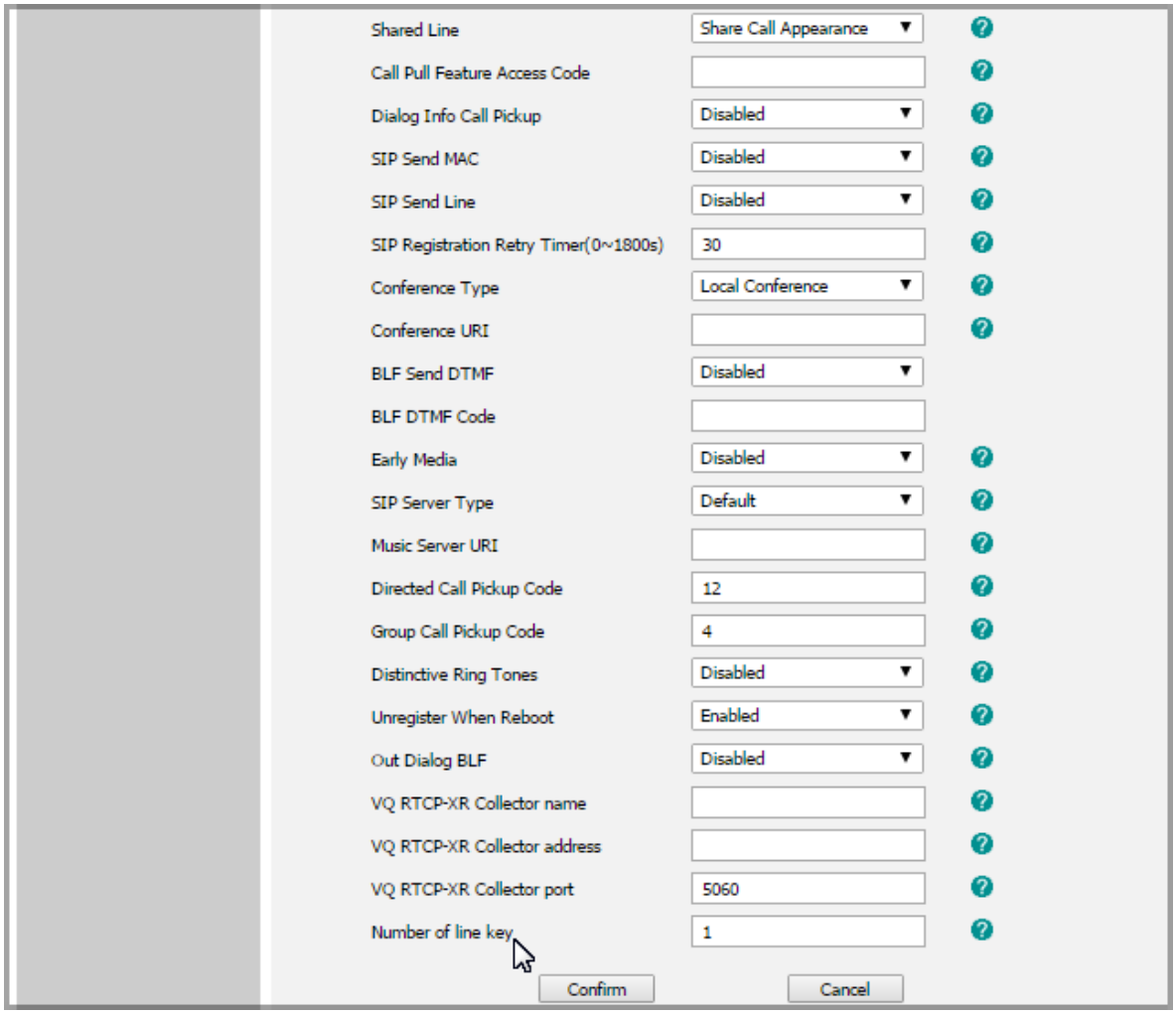
- Register the primary account 4603.



- Click on **Advanced**, and then select **Share Call Appearance** from the **Shared Line** list.



- Enter the desired number in the **Number of line key** field.



This field appears only if Auto Linekeys feature is enabled.

Placing Calls

You can have one call or multiple calls on the shared line.

To place a call on the shared line:

Do one of following:

- Enter the desired number using the keypad or the dial pad on the touch screen when the phone is idle.
- Press #SEND, or tap **Send**.

  The phone will dial the entered number using the first line key.

OR

- Tap the line key when the phone is idle to enter the dialing screen.
- Enter the desired number using the keypad or the dial pad on the touch screen and then press #SEND, or tap **Send**.

  The phone will dial the entered number using the selected line key.

To place multiple calls on the shared line:

You can have more than one call on the shared line. To place a new call when there is an active call on the line key 1 of phone A, do one of the following on phone A:

- Press  or tap the Hold soft key. The original call is placed on hold.
- Tap the **NewCall** soft key to enter the dialing screen.
- Enter the desired number using the keypad or the dial pad on the touch screen.
- Press #SEND, or tap **Send**.
- Phone A will dial the entered number using the line key 2 automatically.

OR

- Tap the line key. The original call is placed on hold.
- Enter the desired number using the keypad or the dial pad on the touch screen.
- Press #SEND, or tap **Send**.
- Phone A will dial the entered number using the selected line key.

Answering Calls

You can have one call or multiple calls on the shared line. Incoming calls will be distributed evenly among the available line keys.

To answer a call on the shared line:

When an incoming call arrives on the shared line, the phone A and phone B will ring simultaneously, and the icon indicators of the line key 1 on both phone A and phone B will become  and flashing. You can answer the incoming call on either phone A or phone B but not both.

Do one of the following on phone A or phone B:

- Tap the line key 1, or tap the Answer soft key on phone A.

  The icon indicators of the line key 1 on phone A and phone B will become and solid ..

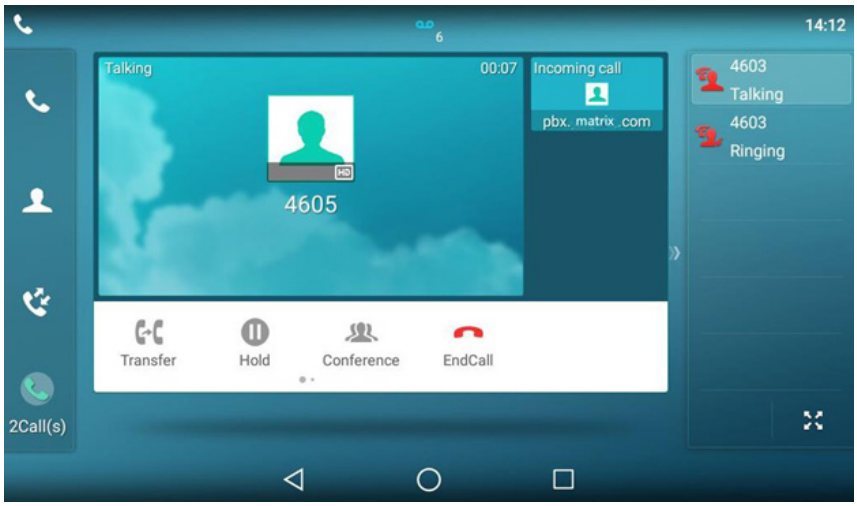
OR

- Tap the line key 1, or tap the Answer soft key on phone B.

  The icon indicators of the line key 1 on phone A and phone B will become and solid .

To answer multiple calls on the shared line:

An incoming call arrives on the shared line when there is an active call on phone A's line key 1. The icon indicators of the line key 2 on both phone A and phone B will become and flashing. You can answer the incoming call on either phone A or phone B. The touch screen of phone A displays the information of the incoming call.



Make sure call waiting feature is enabled on phone A.

Do one of the following on phone A:

- Tap the line key 2. Phone B stops ringing.

OR

- Tap the **Answer** soft key. Phone B stops ringing.

OR

- Tap the incoming call prompt box.

  Tap the **Answer** soft key. Phone B stops ringing.

The incoming call is answered and the original call is placed on hold. The icon indicators of the line key 1 on phone A and phone B will become and solid indicating that there is the held call on the line key 1. The icon indicators of the line key 2 on phone A and phone B will become and solid indicating that there is an active call on the line key 2.

You can also answer the call on phone B. Do one of the following on phone B:

- Tap the line key 2. Phone A stops ringing.

OR

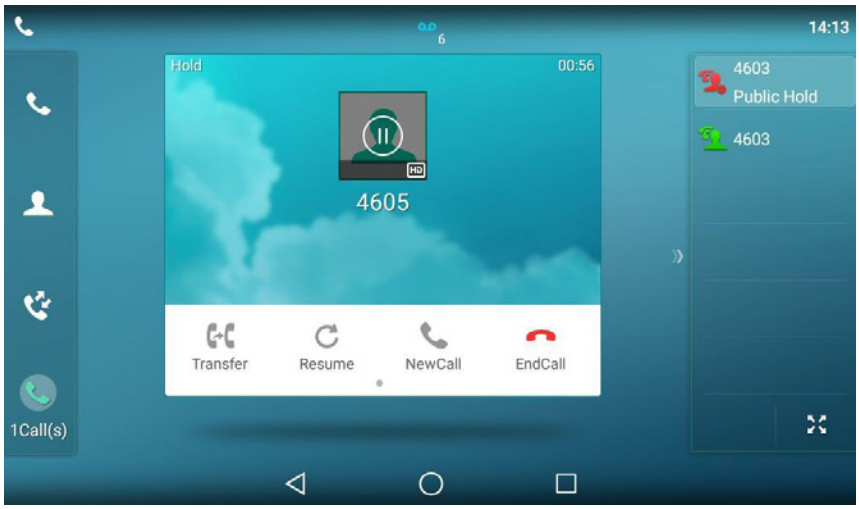
- Tap the **Answer** soft key. Phone A stops ringing.

  The icon indicators of the line key 2 on phone A and phone B will become and solid indicating that there is an active call on the line key 2. Meanwhile, the icon indicators of the line key 1 phone A and phone B will become and solid indicating that there is the held call on the line key 1.

Placing a Call on Hold

To place a call on public hold:

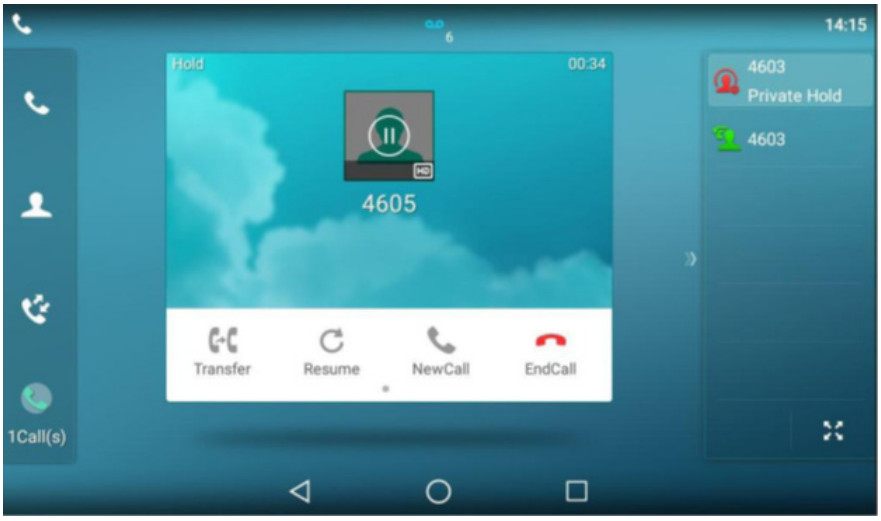
- Press or tap the **Hold** soft key on phone A when party A and party C are talking.



The icon indicators of line key 1 on phone A and phone B will become .

To place a call on private hold:

- Press the **PriHold** soft key or private hold key on phone A when there is an active call on the shared line (You may need to swipe the soft key area left to see the **PriHold** soft key).



- The icon indicators of line key 1 on phone A will become when the shared line call is placed on private hold.
- The icon indicators of line key 1 on phone B will become when the shared line call is placed on private hold.

Retrieving a Held Call

To retrieve a call placed on public hold:

You can retrieve the public held call on either phone A or phone B.

To retrieve the call on phone A:

- Tap , press or tap the **Resume** soft key.

  The conversation between phone A and phone C is retrieved.

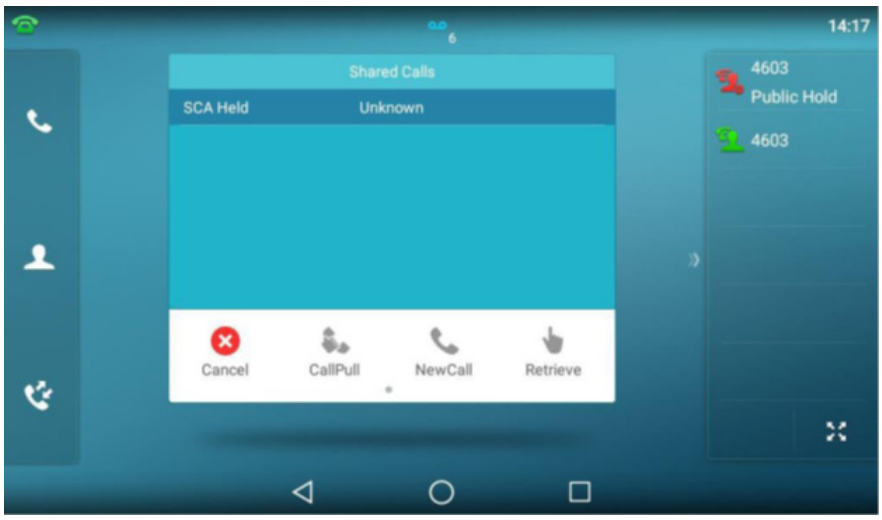
To retrieve the call on phone B:

Do one of the following:

- Tap .

OR

- Long tap the desired line key. The **Cancel**, **CallPull**, **NewCall** and **Retrieve** soft keys appear on the touch screen.



- Tap the **Retrieve** soft key to retrieve the call.

  The conversation is established between phone B and phone C, phone A disconnects the call. And the line key icon on phone A and phone B will become .

To retrieve a call placed on private hold:

The private held call can be only retrieved by the hold party (party A).

Do the following:

- Tap , press or tap the **Resume** soft key on phone A.

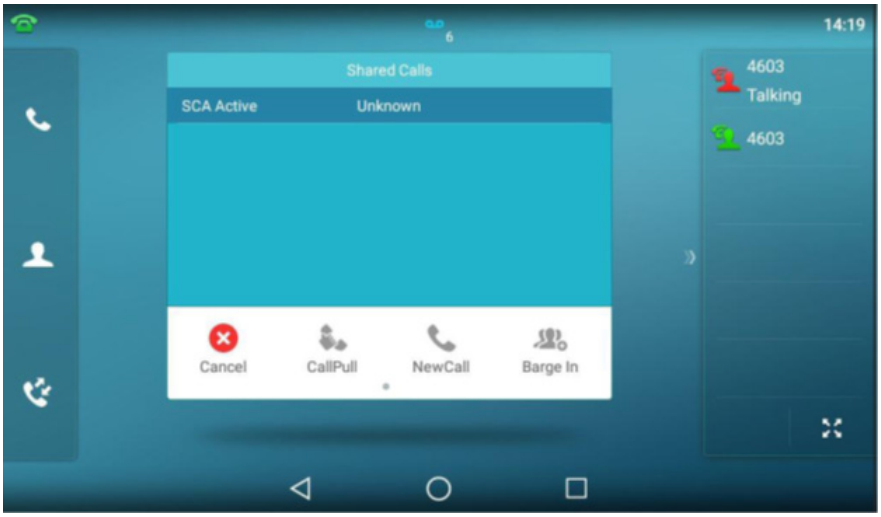
  The conversation between phone A and phone C is retrieved.

Barging In an Active Call

To interrupt the active call on the shared line:

If phone A has only one active call, do the following:

- Long tap the desired line key on phone B.
  The **Cancel, CallPull, NewCall** and **Barge** In soft keys appear on the touch screen of phone B.



- Tap the **Barge In** soft key to interrupt the active call of phone A.
  Party B will set up a conference call with the other parties in the active call.

- Press , tap the **Hold** soft key or the **EndCall** soft key.

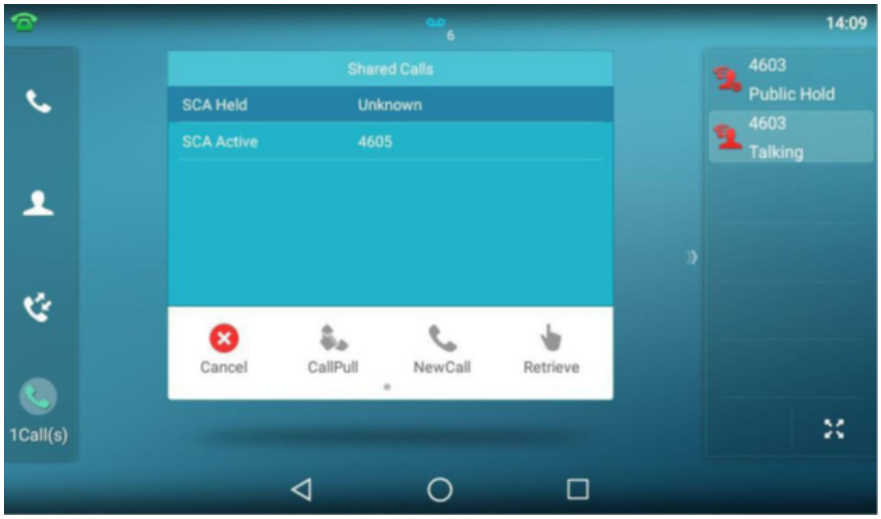
  - If any party in the conference call presses or taps the **Hold** soft key, two-way voice can be heard between the remaining parties.

  - If party A or party B taps the **EndCall** soft key, the remaining parties are still connected. If the other party (not the shared line party) taps the **EndCall** soft key, the conference call is ended.

- You can also tap the line key with the solid red icon indicator on phone B to set up a conference call with the other parties in the active call.

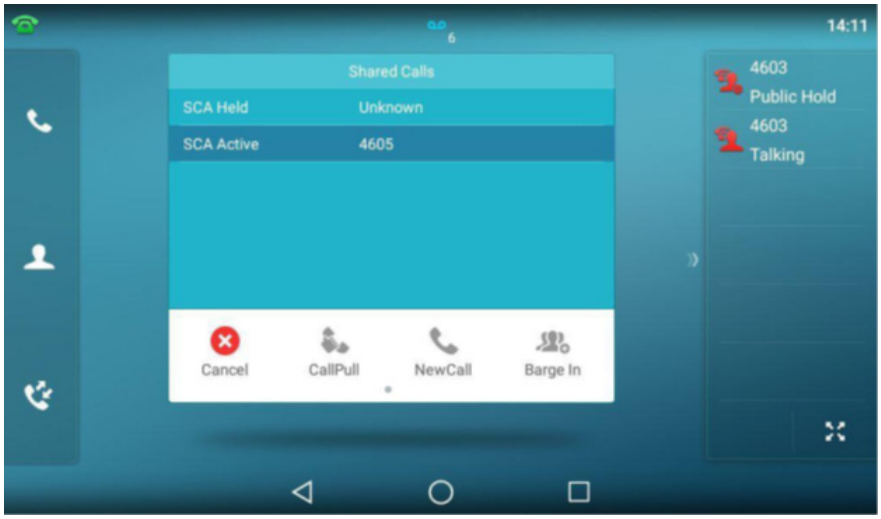
If phone A has more than one call, do the following:

- Long tap the desired line key on phone B.
  The list of calls appears on the touch screen of phone B.



- Tap the item to select the active call.

  The **Cancel**, **CallPull**, **NewCall** and **Barge In** soft keys appear when the active call is highlighted.



- Tap the **Barge In** soft key to interrupt the active call of phone A.

  Party B may hear a warning tone and then set up a conference call with the other parties of the active call.

- 4. Press , tap the **Hold** soft key or the **EndCall** soft key.

  - If any party in the conference call presses, taps the **Hold** soft key, two-way voice can be heard between the remaining parties.
  - If party A or party B taps the **EndCall** soft key, the remaining parties are still connected. If the other party (not the shared line party) taps the **EndCall** soft key, the conference call is ended.

Call Pull

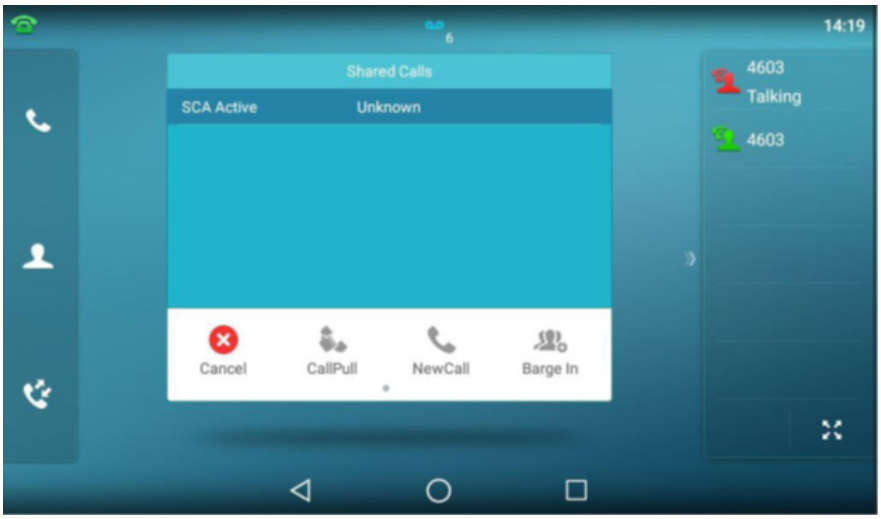
Call pull feature allows users to retrieve an existing call from another shared phone that is in active or hold status. For example, when there is a call between phone A and phone C, you can use call pull feature on phone B to retrieve this call from phone A. Then the call is established between phone B and phone C.

To retrieve a call from another shared phone:

If there is an active call between phone A and phone C, do the following:

- Long tap the desired line key on phone B.

  The **Cancel**, **CallPull**, **NewCall** and **Barge In** soft keys appear on the touch screen of phone B.



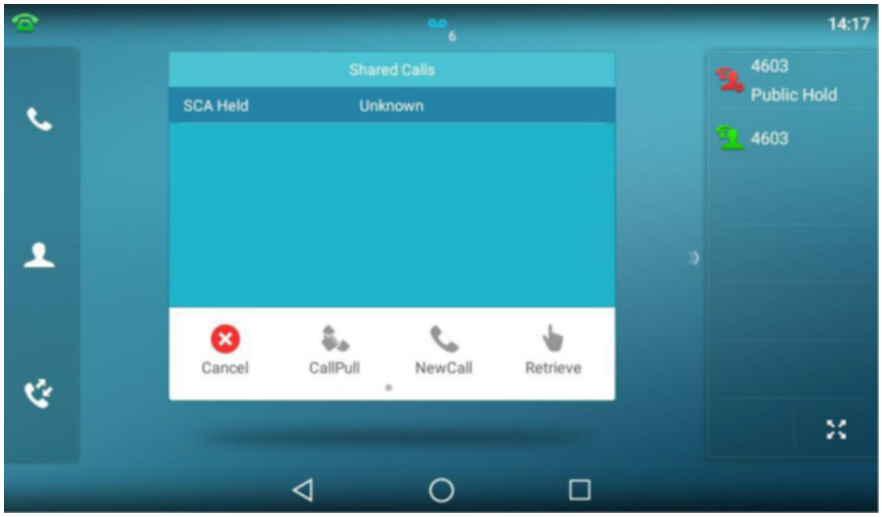
- Tap the **CallPull** soft key.

  The active call has been retrieved from the phone A successfully.

If there is a held call between phone A and phone C, do the following:

- Long tap the desired line key on phone B.

  The **Cancel**, **CallPull**, **NewCall** and **Retrieve** soft keys appear on the touch screen of phone B.



- Tap the **CallPull** soft key.

  The held call has been retrieved from the phone A successfully.

SIP Send Line

The IP phone can send the line number in the REGISTER message. SIP send line allow adding “Line:<linenumber>”(e.g., Line: 1) to the SIP header of the REGISTER message. The line number is a number between 0 and 15.

The following table lists line number values for the IP phone model.

Phone Model	Line Number	Description
SPARSH VP710	0~15	0~15 stand for line 1~line 16

Example of a SIP REGISTER message:

```
REGISTER sip:10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3990593443
From: "11" <sip:11@10.2.1.48:5060>;tag=255071842
To: "11" <sip:11@10.2.1.48:5060>
Call-ID: 1_2369214377@10.3.20.14
CSeq: 2 REGISTER
Contact: <sip:11@10.3.20.14:5060;line=1da6aa8d7254654>
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Expires: 0
Allow-Events: talk,hold,conference,refer,check-sync
Line: 1
Content-Length: 0
```

Procedure

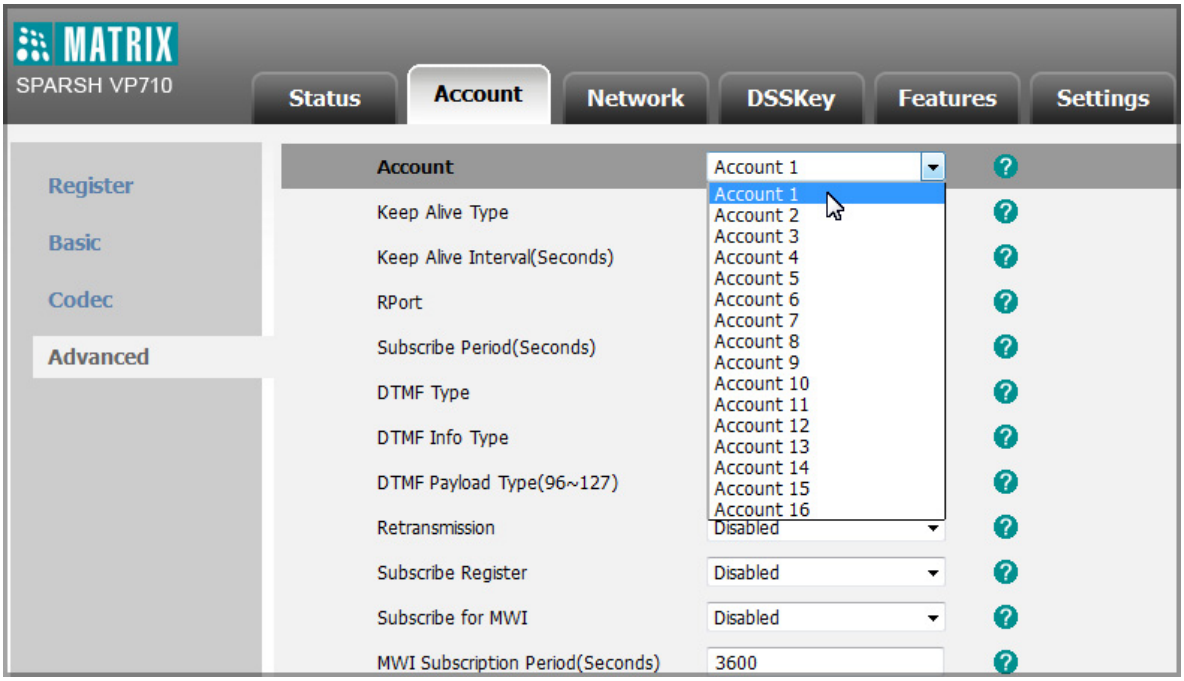
SIP Send Line can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure SIP send line on a per-line basis. For more information, refer “Appendix D - Configuration Parameters”.

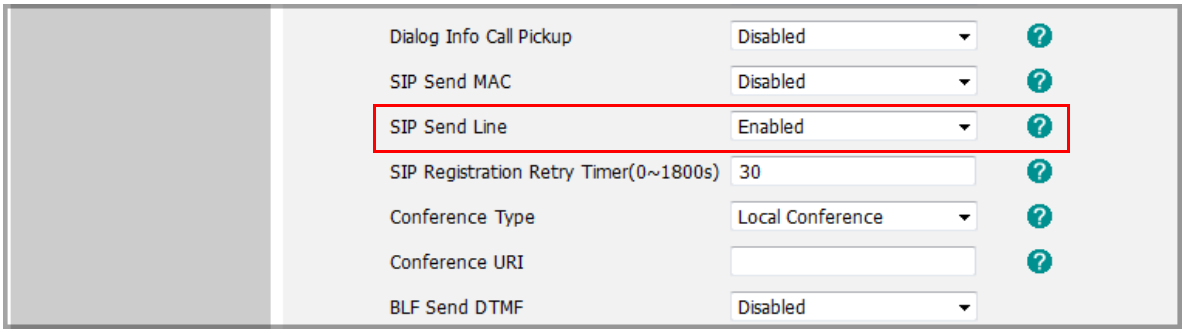
Local	Web User Interface	Configure SIP send line on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure SIP send line via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **SIP Send Line** list.



- Click **Confirm** to save the change.

SIP Send MAC

The IP phone can send the MAC address in the REGISTER message. SIP send MAC allow adding “Mac:<PhoneMACAddress>” (e.g., Mac: 00:15:65:74:b1:50) to the SIP header of the REGISTER message.

Example of a SIP REGISTER message:

```
REGISTER sip:10.2.1.48:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.20.14:5060;branch=z9hG4bK3593117201
From: "11" <sip:11@10.2.1.48:5060>;tag=2788360609
To: "11" <sip:11@10.2.1.48:5060>
Call-ID: 1_1863786852@10.3.20.14
CSeq: 2 REGISTER
Contact: <sip:11@10.3.20.14:5060;line=cc75882e976e208>
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Expires: 0
Allow-Events: talk,hold,conference,refer,check-sync
Mac: 00:1b:09:04:8e:3c
Content-Length: 0
```

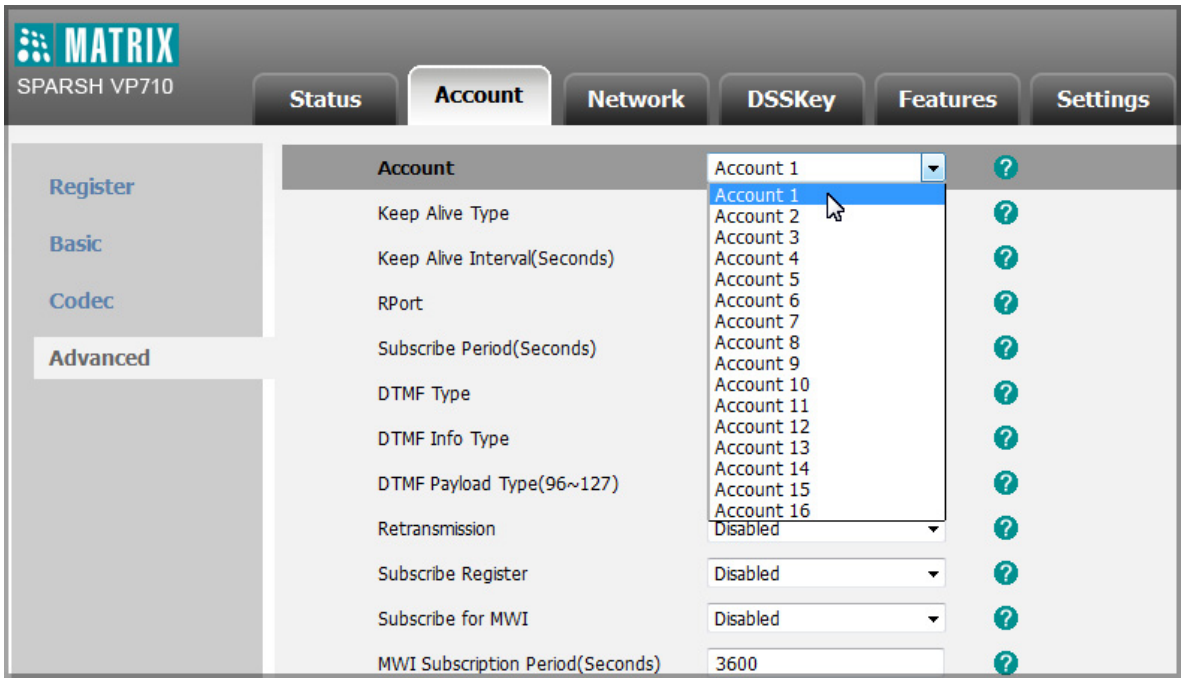
Procedure

SIP send MAC can be configured using the configuration files or locally.

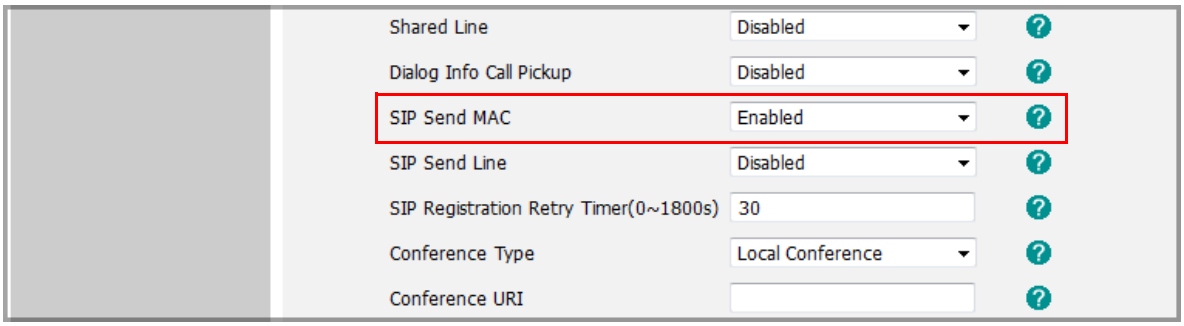
Configuration File	<MAC>.cfg	Configure SIP send MAC on a per-line basis. For more information, refer “Appendix D - Configuration Parameters”.
Local	Web User Interface	Configure SIP send MAC on a per-line basis. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure SIP Send MAC feature via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **SIP Send MAC** list.



- Click **Confirm** to save the change.

SIP Session Timer

SIP session timers T1, T2 and T4 are SIP transaction layer timers defined in RFC 3261. These session timers are configurable for the IP phone.

Timer T1 - It is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server.

Timer T2 - It represents the maximum retransmitting time of any SIP request message. The retransmitting and doubling of T1 will continue until the retransmitting time reaches the T2 value.

Timer T4 - It represents the time the network will take to clear messages between the SIP client and server.

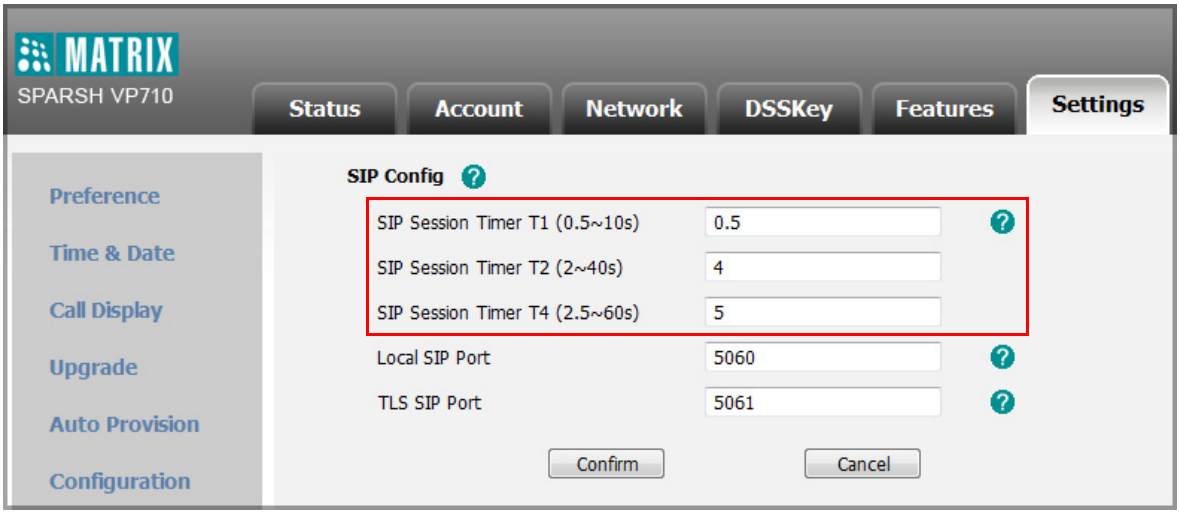
Procedure

SIP session timer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure SIP session timer. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure SIP session timer. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-sip&q=load

To configure SIP session timer via web user interface:

- Click on **Settings-> SIP**.
- Enter the desired value in the **SIP Session Timer T1 (0.5~10s)** field.
- Enter the desired value in the **SIP Session Timer T2 (2~40s)** field.
- Enter the desired value in the **SIP Session Timer T4 (2.5~60s)** field.



- Click **Confirm** to save the change.

Softkey Layout

Softkey layout is used to customize the soft keys at the bottom of the touch screen to meet the users' requirements. In addition to specify which soft keys to display, you can determine their display order. It can be configured based on call states.

The following table lists soft keys available for IP phones in different call states.

Call State		Default Soft Keys	Optional Soft Keys
CallFailed		NewCall	End Call
CallIn		Answer Forward Silence Reject	Switch
Connecting	Connecting	End Call	Switch
	SemiAttendTrans	Transfer End Call	Switch
RingBack	RingBack	End Call	Switch
	SemiAttendTransBack	Transfer End Call	Switch

Talking	Talk	Transfer Hold Conference End Call	Mute SWAP NewCall Switch Answer Reject PriHold Park GPark RTP Status Record
	Hold	Transfer Resume NewCall End Call	Switch Answer Reject Record
	Held	End Call	Switch Answer Reject NewCall Record
	Conferenced	Hold Split End Call	Switch Answer Reject Mute RTP Status Record

Procedure

Softkey layout can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the softkey layout. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure the softkey layout. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-softkey&q=load

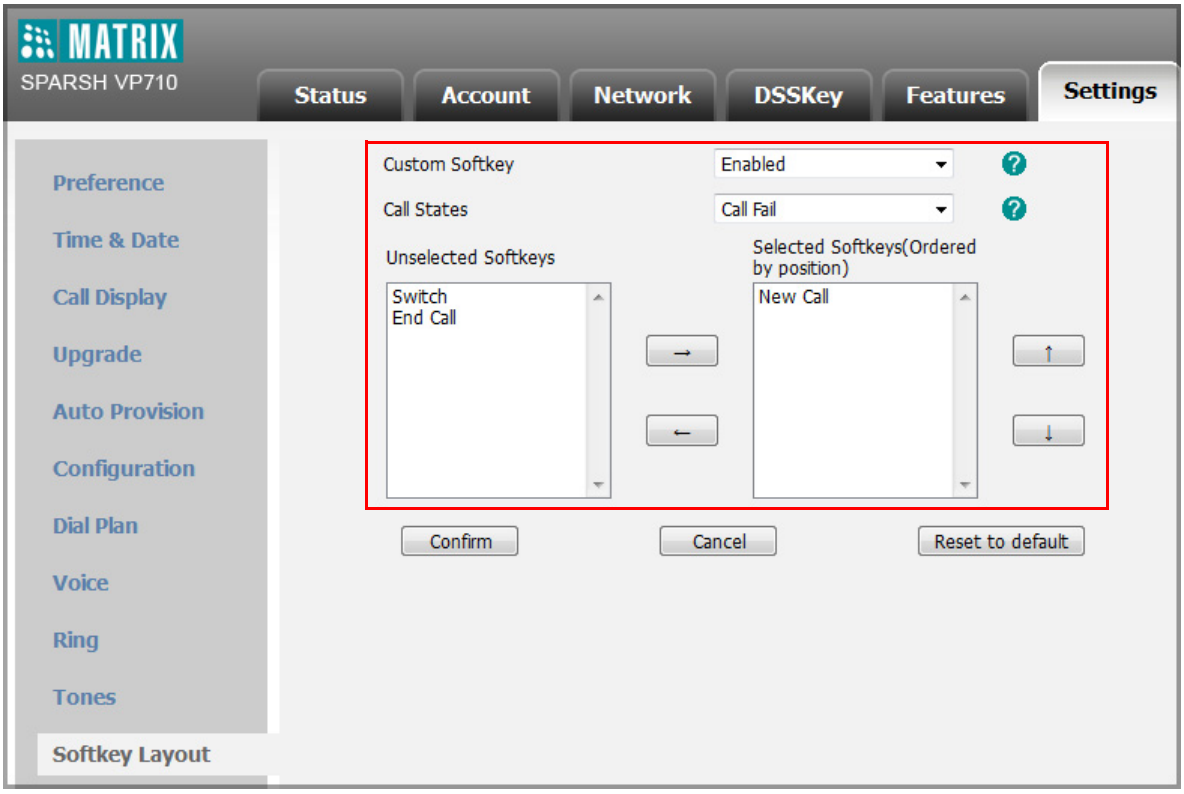
To configure softkey layout via web user interface:

- Click on **Settings->Softkey Layout**.
- Select the desired value from the **Custom Softkey** list.
- Select the desired state from the **Call States** list.
- Select the desired soft key from the **Unselected Softkeys** column and then click .

  The selected soft key appears in the **Selected Softkeys** column.

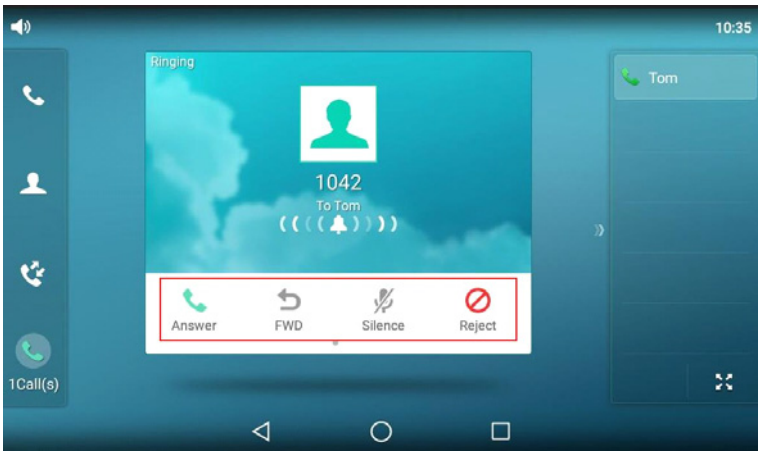
  If more than four soft keys are selected, the selected soft keys will be displayed in two pages. Swipe left or right to see more soft keys.
- Repeat the above step to add more soft keys to the **Selected Softkeys** column.
- To remove the soft key from the **Selected Softkeys** column, select the desired soft key and then click .
- To adjust the display order of soft keys, select the desired soft key and then click or .

  The touch screen displays the soft keys in the adjusted order.



- Click **Confirm** to save the change.

The following shows the softkeys displaying on the phone in the CallIn state:



Static DNS Cache

Failover redundancy can only be utilized when the configured domain name of the server is resolved to multiple IP addresses. If the IP phone is not configured with a DNS server, or the DNS query returns no result from a DNS server, you can statically configure a set of DNS NAPTR/SRV/A records into the IP phone. The IP phone will attempt to resolve the domain name of the SIP server with static DNS cache.

When the IP phone is configured with a DNS server, it will behave as follows to resolve domain name of the server:

- The IP phone performs a DNS query to resolve the domain name from the DNS server.
- If the DNS query returns no results for the domain name, or the returned record cannot be contacted, the values in the static DNS cache (if configured) are used when their configured time intervals are not elapsed.
- If the configured time interval is elapsed, the IP phone will attempt to perform a DNS query again.
- If the DNS query returns a result, the IP phone will use the returned record from the DNS server and ignore the statically configured cache values.

When the IP phone is not configured with a DNS server, it will behave as follows:

- The IP phone attempts to resolve the domain name within the static DNS cache.
- The IP phone will always use the results returned from the static DNS cache.

Support for negative caching of DNS queries as described in RFC 2308 is also provided to allow faster failover when prior DNS queries have returned no results from the DNS server. IP phones can be configured to use static DNS cache preferentially. Static DNS cache is configurable on a per-line basis.

Procedure

Static DNS cache can be configured only using the configuration files.

Configuration File	<MAC>.cfg	Configure NAPTR/SRV/A records. Configure the IP phone whether to cache the additional DNS records. Configure the IP phone whether to use static DNS cache preferentially. For more information, refer "Appendix D - Configuration Parameters".

Tones

When receiving a message, the IP phone will play a warning tone. You can customize tones or select specialized tone sets (vary from country to country) to indicate different conditions of the IP phone. Available tone sets for IP phones:

- Australia
- Austria
- Brazil
- Belgium
- China
- Czech
- Denmark
- Finland
- France
- Germany
- Great Britain
- Greece
- Hungary
- Lithuania
- India
- Italy
- Japan
- Mexico
- New Zealand
- Netherlands
- Norway
- Portugal
- Spain
- Switzerland
- Sweden
- Russia
- United States
- Chile
- Czech ETSI

Configured tones can be heard on the IP phone for the following conditions.

Condition	Description
Dial	When in the pre-dialing interface

Secondary Dial	When adding a comma "," to the digit map (For more information on digit map, refer to Dial Plan using "Dial Plan using Digit Map String Rules")
Ring Back	Ring-back tone
Busy	When the callee is busy
Congestion	When the network is congested
Call Waiting	Call waiting tone (For more information on call waiting, refer to "Call Waiting").
Dial Recall	When receiving a call back
Info	When receiving a special message
Stutter	When receiving a voice mail (For more information on voice mail tone, refer to "Voice Mail Tone")
Auto Answer	When automatically answering a call. (For more information on auto answer, refer to "Auto Answer").

Procedure

Tones can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the tones for the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the tones for the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-tones&q=load

To configure tones via web user interface:

- Click on **Settings->Tones**.
- Select the desired value from the **Select Country** list.

  If you select **Custom**, you can customize a tone for each condition of the IP phone.



- Click **Confirm** to save the change.

Unregister When Reboot

Unregister when reboot allows the IP phone to unregister first before re-registering the account when finishing a reboot.

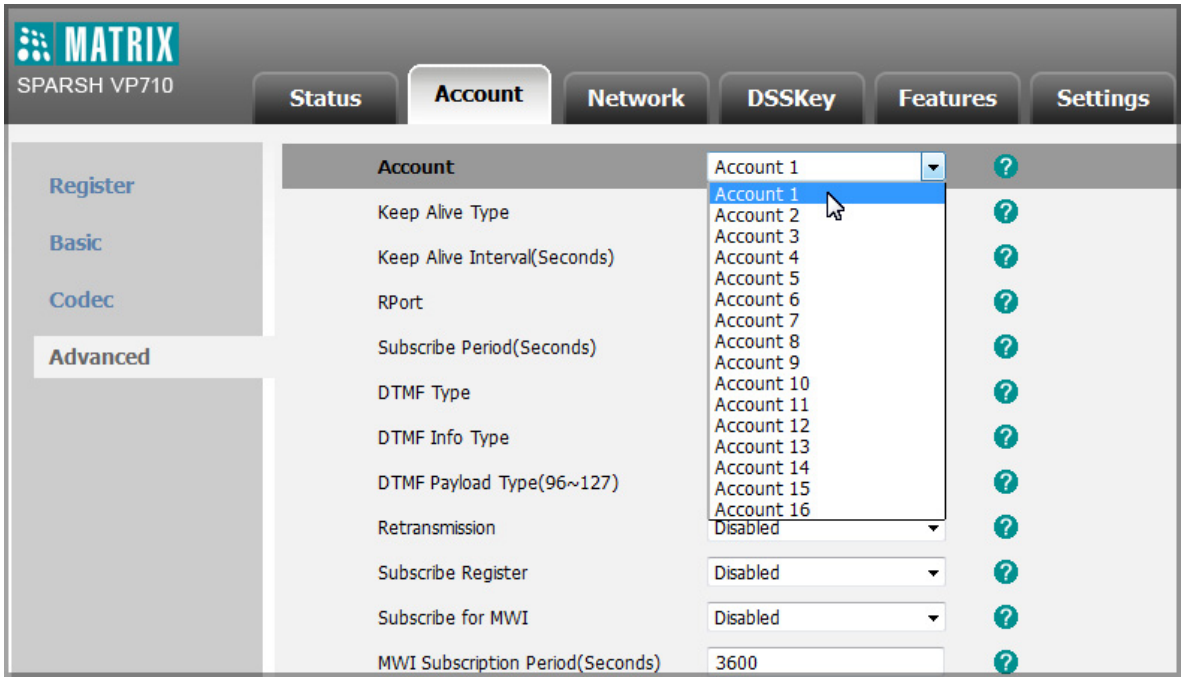
Procedure

Unregister when reboot can be configured using the configuration files or locally.

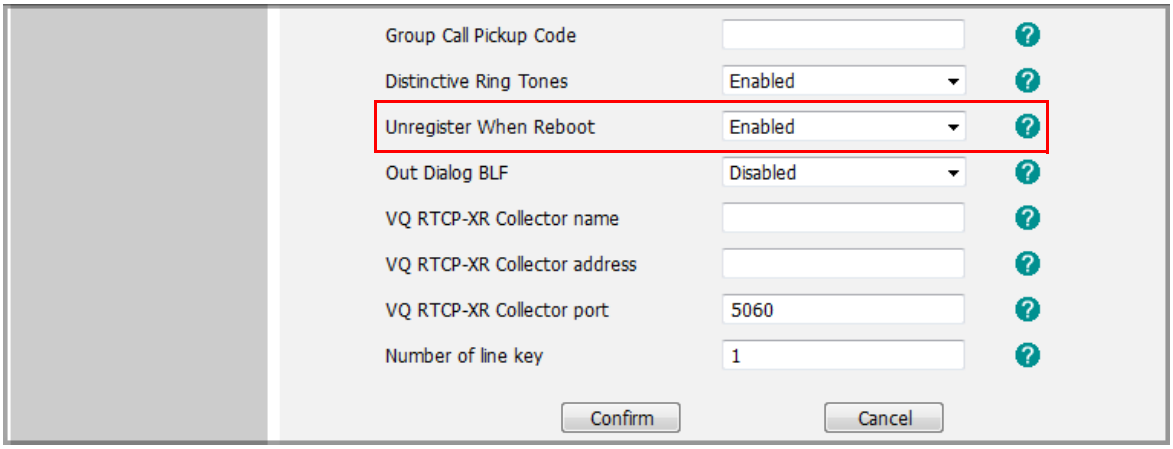
Configuration File	<MAC>.cfg	Configure unregister when reboot. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure unregister when reboot. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure unregister when reboot via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Unregister When Reboot** list.



- Click **Confirm** to save the change.

Use Outbound Proxy in Dialog

An outbound proxy server can receive all initiating request messages and route them to the designated destination. If the IP phone is configured to use an outbound proxy server within a dialog, all SIP request messages from the IP phone will be sent to the outbound proxy server forcibly.

To use this feature, make sure the outbound server has been correctly configured on the IP phone. For more information on how to configure outbound server, refer to "Account Registration".

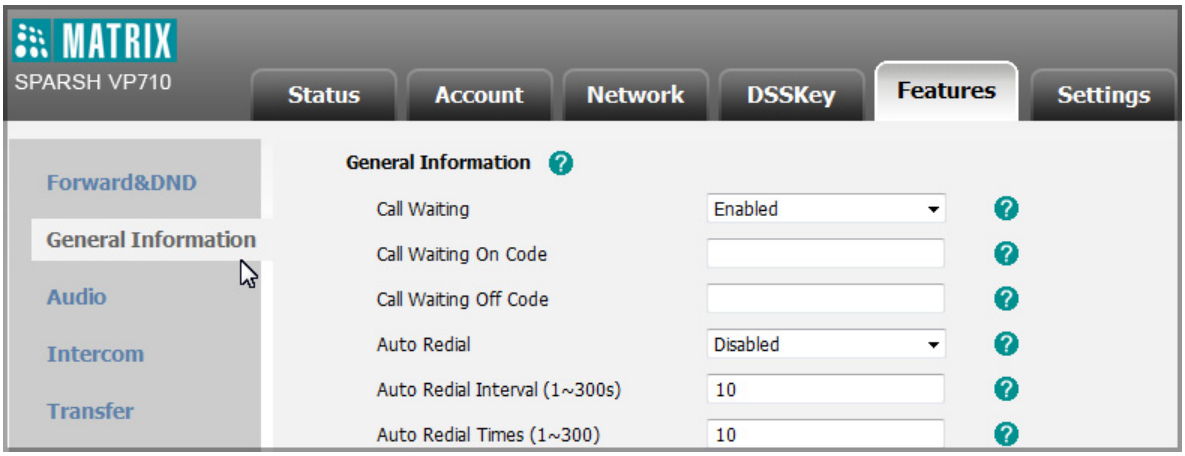
Procedure

Use outbound proxy in dialog can be configured using the configuration files or locally.

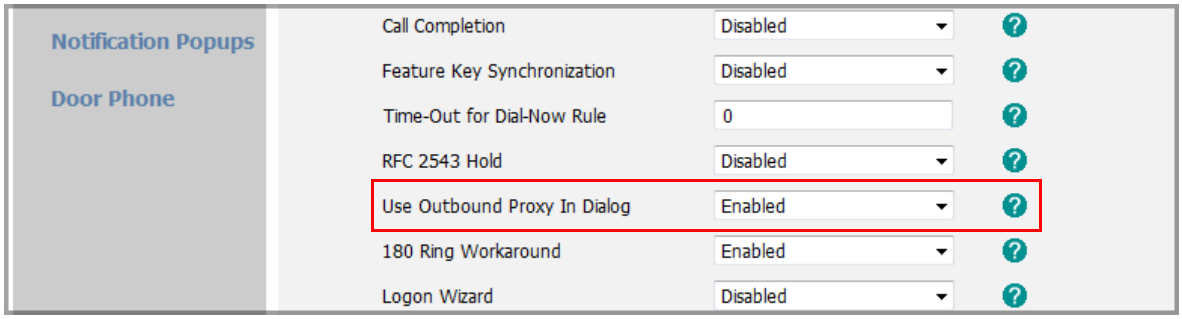
Configuration File	<MAC>.cfg	Specify whether to use outbound proxy in a dialog. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Specify whether to use outbound proxy in a dialog. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure use outbound proxy in dialog via web user interface:

- Click on **Features->General Information**.



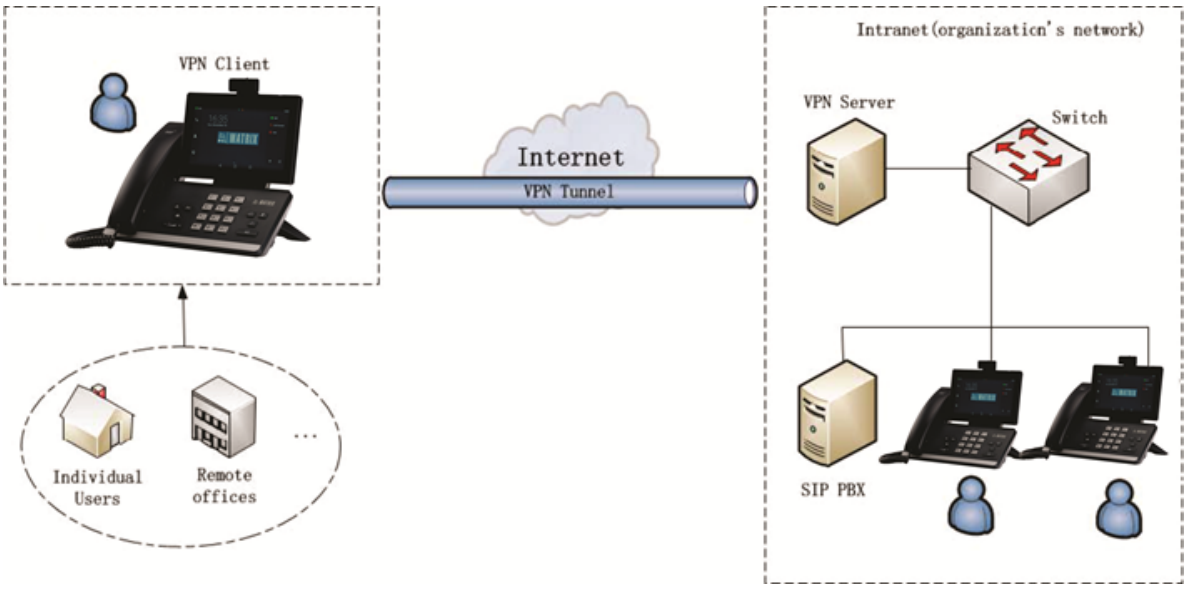
- Select the desired value from the **Use Outbound Proxy In Dialog** list.



- Click **Confirm** to save the change.

Virtual Private Network (VPN)

Virtual Private Network is a secured private network connection built on top of the public telecommunication infrastructure, such as the Internet. It has become more prevalent due to the benefits of scalability, reliability, convenience and security. VPN provides remote offices or individual users with secure access to their organization's network. **Image**



There are two types of VPN access:

- Remote-access VPN
- Site-to-Site VPN

Remote-access VPN: Remote-access VPN connects an individual device to a network. It allows employees to access their company's intranet from home or outside the office.

Site-to-Site VPN: Site-to-Site VPN connect two networks together.It allows employees in geographically separated offices to share one cohesive virtual network.

VPN can be also classified by the protocols used to tunnel the traffic. It provides security through tunneling protocols: IPSec, SSL, L2TP and PPTP.

VPN Technology

IP phone supports SSL VPN, which provides remote-access VPN capabilities through SSL.OpenVPN is a full featured SSL VPN software solution. It creates secure connections in remote access facilities, designed to work with the TUN/TAP virtual network interface.
TUN and TAP are virtual network kernel devices. TAP simulates a link layer device and provides a virtual point-to-point connection, while TUN simulates a network layer device and provides a virtual network segment.

IP phone use OpenVPN to achieve VPN feature. To prevent disclosure of private information, tunnel endpoints must authenticate each other before secure VPN tunnel is established.

After VPN feature is configured properly on the IP phone, the IP phone acts as a VPN client and uses the certificates to authenticate the VPN server.
To use VPN, the compressed package of VPN-related files should be uploaded to the IP phone in advance. The file format of the compressed package must be *.tar.

The related VPN files are: certificates (ca.crt and client.crt), key (client.key) and the configuration file (vpn.cnf) of the VPN client.

The following table lists the unified directories of the OpenVPN certificates and key in the configuration file (vpn.cnf) for the IP phone:

VPN files	Description	Unified Directories
ca.crt	CA certificate	/config/openvpn/keys/ca.crt
client.crt	Client certificate	/config/openvpn/keys/client.crt
client.key	Private key of the client	/config/openvpn/keys/client.key

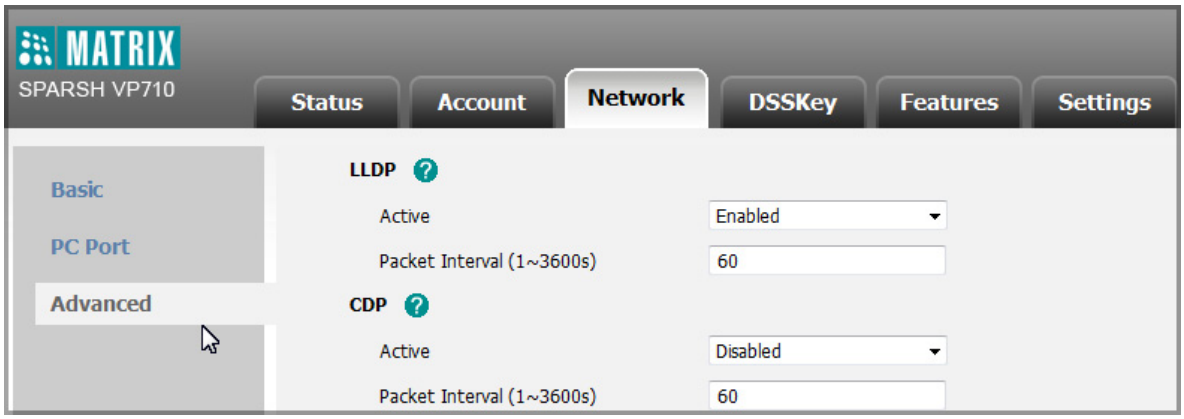
Procedure

VPN can be configured using the configuration files or locally.

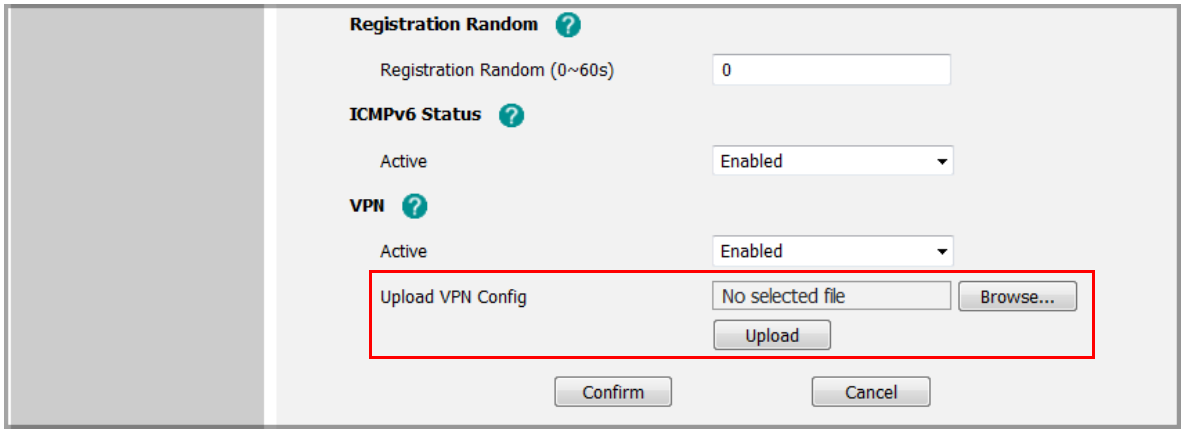
Configuration File	<MAC>.cfg	Configure VPN feature and upload a TAR file to the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure VPN feature and upload a TAR file to the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure VPN feature.

To upload a TAR file and configure VPN via web user interface:

- Click on **Network->Advanced**.



- In the **VPN** block, click **Browse** to select and **Upload** the desired the TAR file from the local system.
  The web user interface prompts the message “Operating, Please Wait…”.
- Select the desired value from the **Active** list in the **VPN** block.

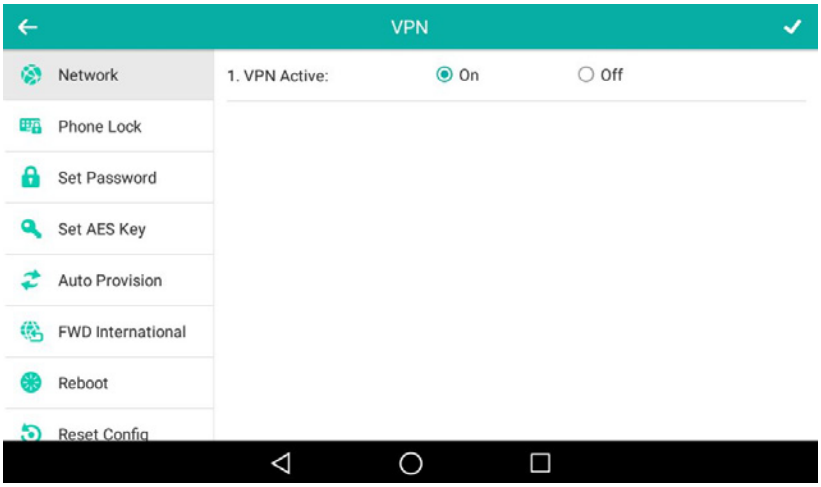


- Click **Confirm** to save the change.
  A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the phone.

To configure VPN via phone user interface after uploading a TAR file:

- Tap **Settings->Advanced** (default password: admin) **->Network->VPN**.
- Tap the **On** radio box in the **VPN Active** field.

You must upload the OpenVPN TAR file using configuration files or via web user interface in advance.



- Tap ✓ to save the change or ◁ to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

VLAN

VLAN (Virtual Local Area Network) is used to logically divide a physical network into several broadcast domains. VLAN membership can be configured through software instead of physically relocating devices or connections. Grouping devices with a common set of requirements regardless of their physical location can greatly simplify network design. VLANs can address issues such as scalability, security and network management.

The purpose of VLAN configurations on the IP phone is to insert a tag with VLAN information to the packets generated by the IP phone. When VLAN is properly configured for the ports (Internet port and PC port) on the IP phone, the IP phone will tag all packets from these ports with the VLAN ID. The switch receives and forwards the tagged packets to the corresponding VLAN according to the VLAN ID in the tag as described in IEEE Std 802.3.

VLAN on the IP phone allows simultaneous access for a regular PC. This feature allows a PC to be daisy chained to an IP phone and the connection for both PC and IP phone to be trunked through the same physical Ethernet cable.

In addition to manual configuration, the IP phone also supports automatic discovery of VLAN via LLDP, CDP or DHCP. The assignment takes effect in this order: assignment via LLDP/CDP, manual configuration, then assignment via DHCP.

Procedure

VLAN assignment method can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the VLAN assignment method. For more information, refer "Appendix D - Configuration Parameters".

Manual Configuration for VLAN in the Wired Network

By default, VLAN is disabled on the IP phone. You can configure VLAN for the Internet port and PC port manually. Before configuring VLAN on the IP phone, you need to obtain the VLAN ID from your network administrator.

Procedure

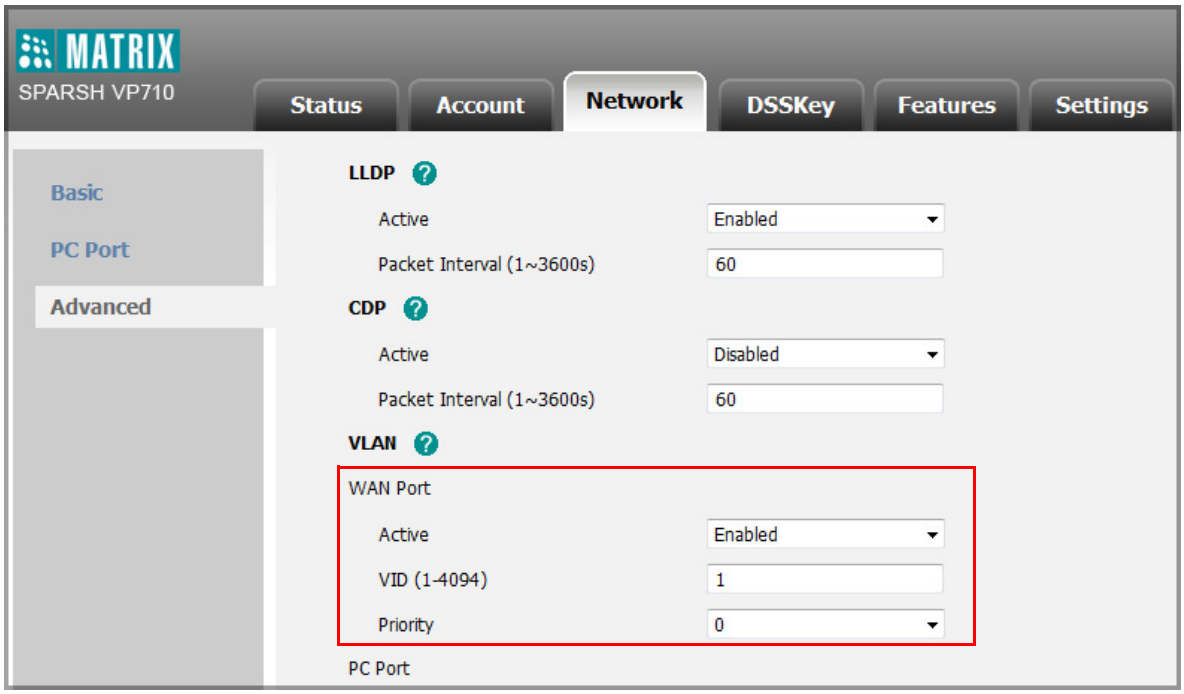
VLAN can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the VLAN assignment method. Configure VLAN for the Internet port and PC port manually. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure VLAN for the Internet port and PC port manually. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure VLAN for the Internet port and PC port manually.

To configure VLAN for Internet port via web user interface:

- Click on **Network->Advanced**.
- In the **WAN Port** block, select the desired value from the **Active** list.
- Enter the VLAN ID in the **VID (1-4094)** field.

- Select the desired value (0-7) from the **Priority** list.



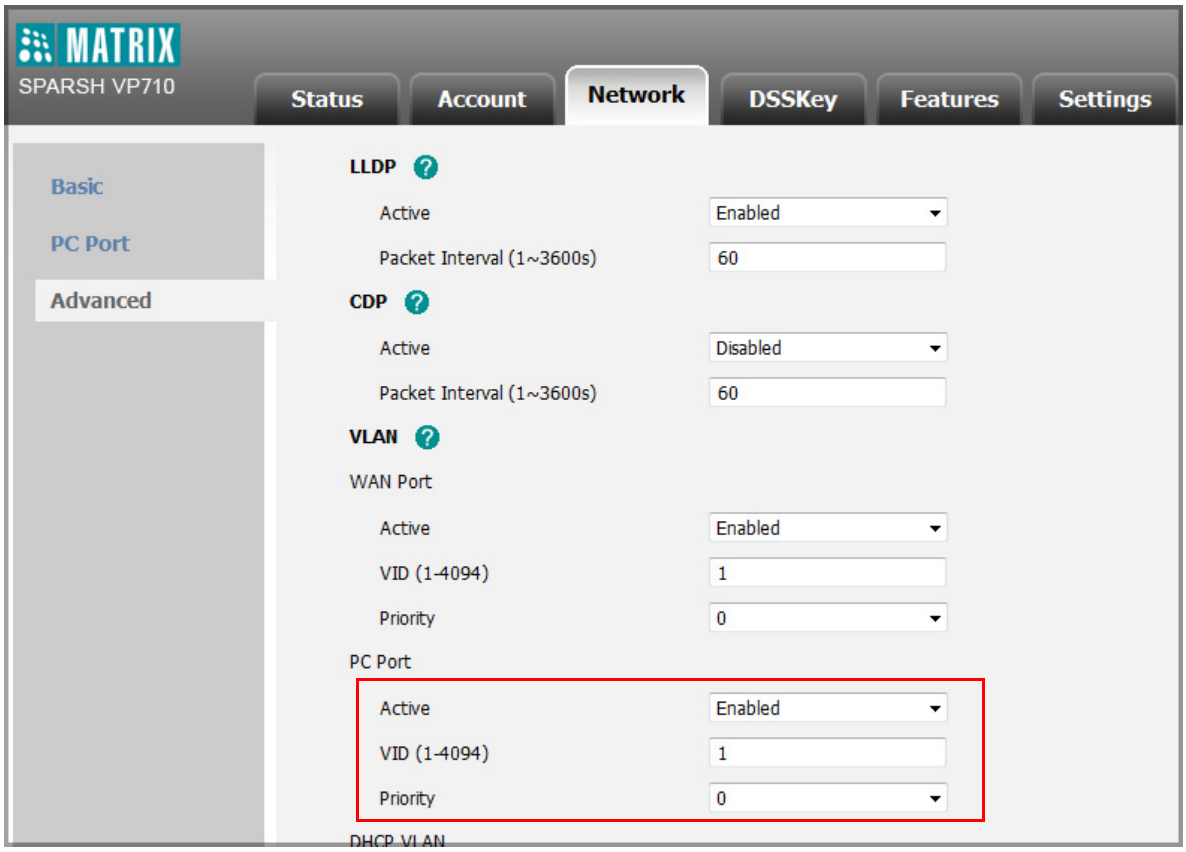
- Click **Confirm** to save the change.

  A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the IP phone.

To configure VLAN for PC port via web user interface:

- Click on **Network->Advanced**.
- In the **PC Port** block, select the desired value from the **Active** list.
- Enter the VLAN ID in the **VID (1-4094)** field.

- Select the desired value (0-7) from the **Priority** list.



- Click **Confirm** to save the change.

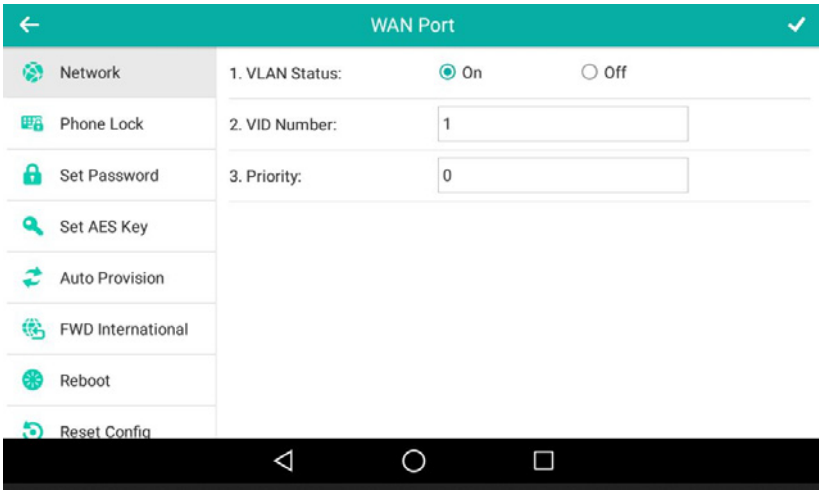
  A dialog box pops up to prompt that the settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To configure VLAN for Internet port (or PC port) via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) ->**Network->VLAN->WAN Port** (or **PC Port**).
- Tap the **On** radio box in the **VLAN Status** field.
- Enter the VLAN ID (1-4094) in the **VID Number** field.

- Enter the priority value (0-7) in the **Priority** field.



- Tap ✓ to save the change or ◁ to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

DHCP VLAN

IP phone supports VLAN discovery via DHCP. When the VLAN Discovery method is set to DHCP, the IP phone examines DHCP option for a valid VLAN ID. The predefined option 132 is used to supply the VLAN ID by default. You can customize the DHCP option used to request the VLAN ID.

Procedure

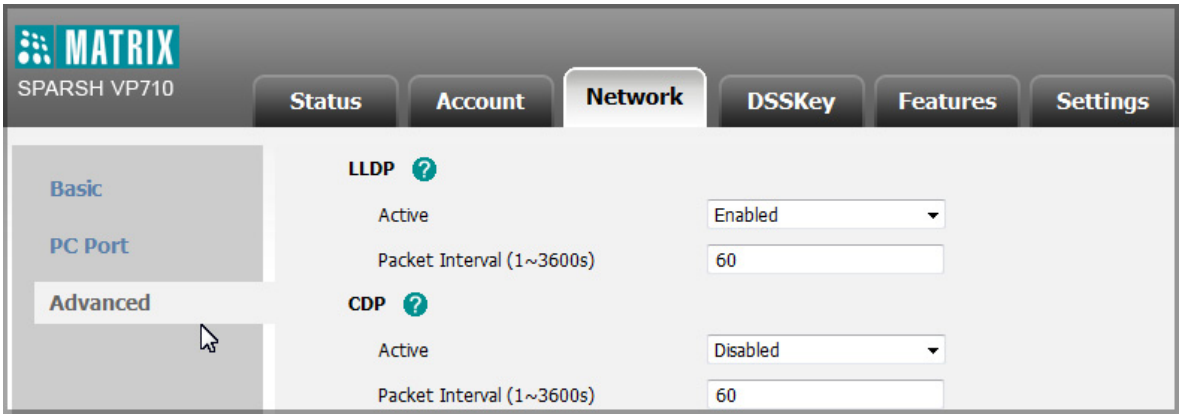
VLAN can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure DHCP VLAN discovery feature. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure DHCP VLAN discovery feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure DHCP VLAN discovery feature.

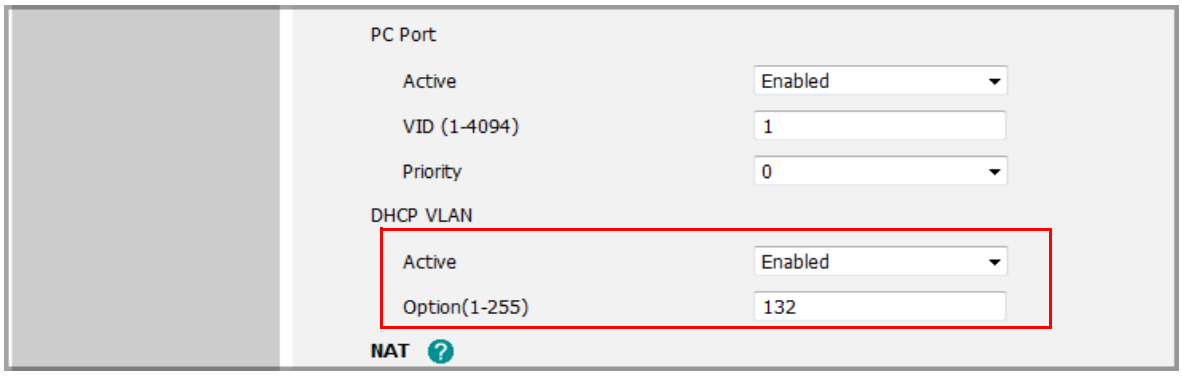
To configure DHCP VLAN discovery via web user interface:

- Click on **Network->Advanced**.



- In the **DHCP VLAN** block, select the desired value the **Active** list.
- Enter the desired option in the **Option(1-255)** field.

  The default option is 132.

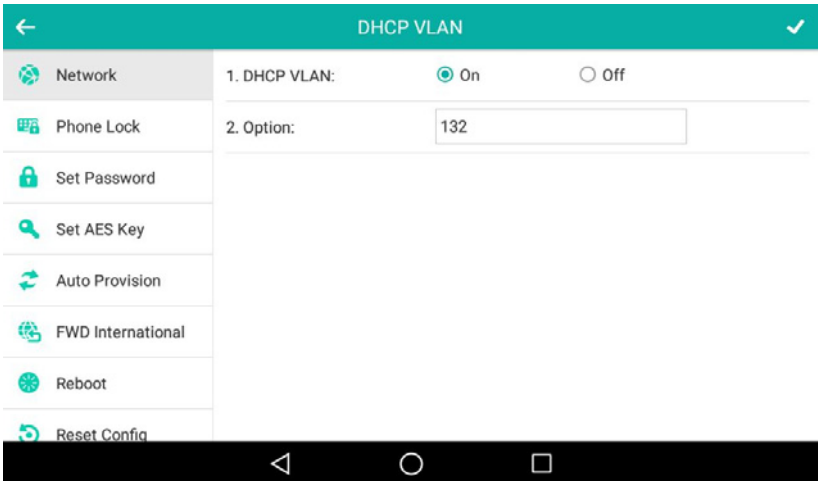


- Click **Confirm** to save the change.

  A dialog box pops up to prompt that the settings will take effect after a reboot.
- Click **OK** to reboot the phone.

To configure DHCP VLAN discovery via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) ->**Network->VLAN->DHCP VLAN**.
- Tap the **On** radio box in the **DHCP VLAN** field.
- Enter the desired option in the **Option** field



- Tap ✓ to save the change or ◁ to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

Web Server Type

You can configure the user or administrator features of the phone via web user interface. Web server type determines access protocol of the IP phone's web user interface. IP phone supports both HTTP and HTTPS protocols for accessing the web user interface through a web browser.This can be disabled when it is not needed or when it poses a security threat. For more information on accessing the web user interface, refer to"Web User Interface".

HTTP is an application protocol that runs on top of the TCP/IP suite of protocols.HTTPS is a web protocol that encrypts and decrypts user page requests as well as pages returned by the web server.Both HTTP and HTTPS port numbers are configurable.

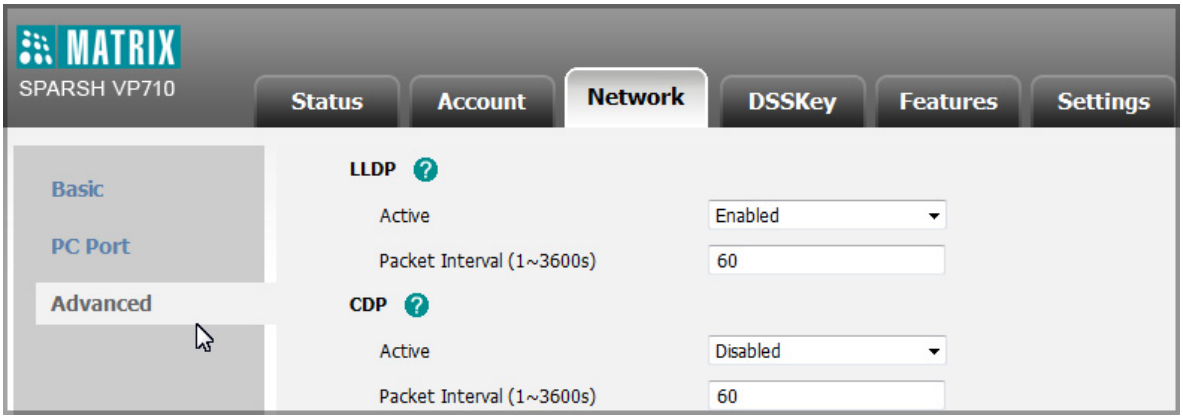
Procedure

Web server type can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the web access type, HTTP port and HTTPS port. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the web access type, HTTP port and HTTPS port. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure the web access type, HTTP port and HTTPS port.

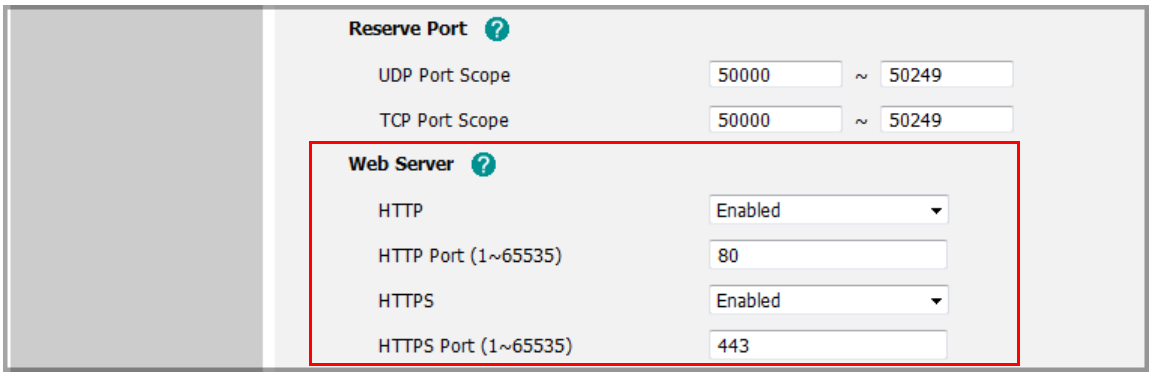
To configure web server type via web user interface:

- Click on **Network->Advanced**.



- Select the desired value from the **HTTP** list.
- Enter the HTTP port number in the **HTTP Port (1~65535)** field.

  The default HTTP port number is 80.
- Select the desired value from the **HTTPS** list.
- Enter the HTTPS port number in the **HTTPS Port (1~65535)** field.
- The default HTTPS port number is 443.

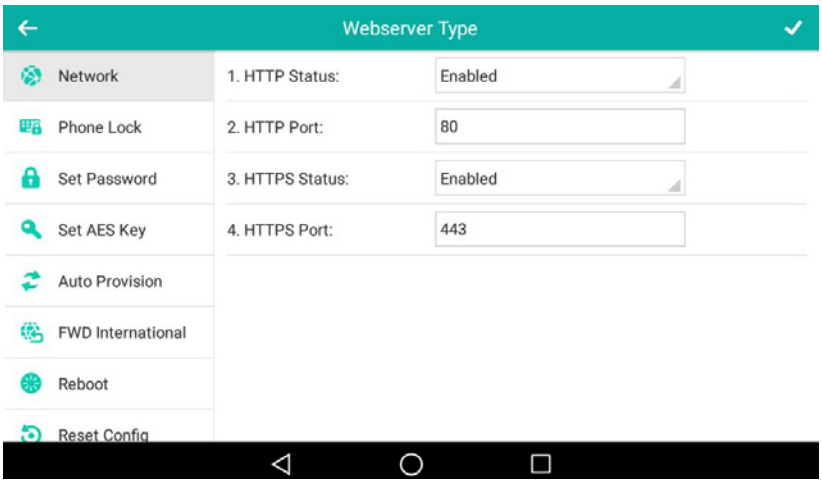


- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.
- Click **OK** to reboot the IP phone.

To configure web server type via phone user interface:

- Tap **Settings->Advanced** (default password: admin) **->Network->Webserver Type**.
- Tap the **HTTP Status** field.
- Tap **Enabled** in the pop-up dialog box.
- Enter the desired HTTP port number in the **HTTP Port** field.
- Tap the **HTTPS Status** field.
- Tap **Enabled** in the pop-up dialog box.
- Enter the desired HTTPS port number in the **HTTPS Port** field.



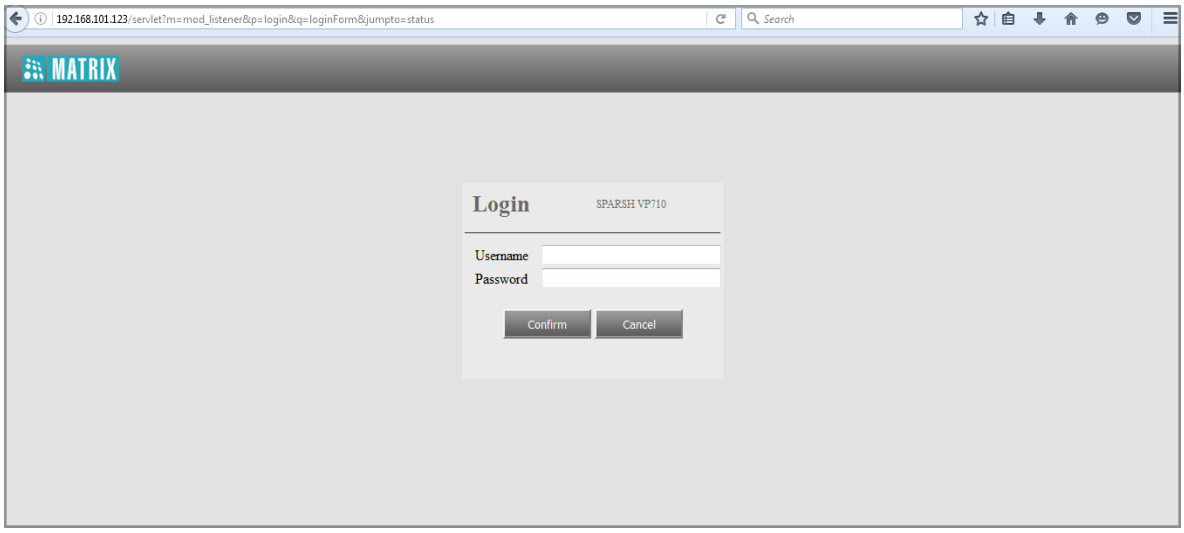
- Tap ✓ to save the change or ◁ to cancel.

The phone prompts you to reboot the phone

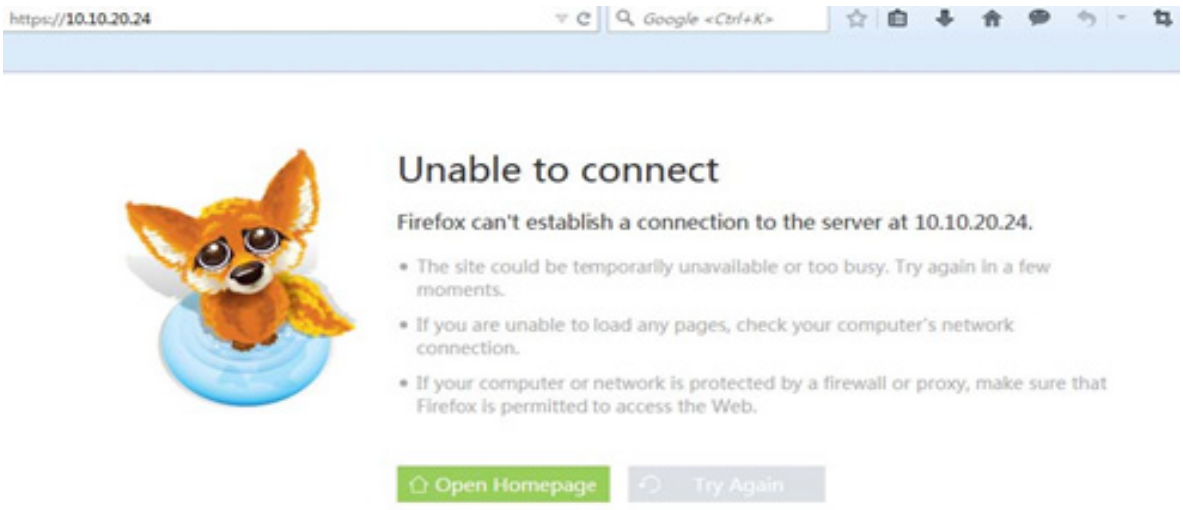
- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

When you enable user to access web user interface of the IP phone using the HTTP/HTTPS protocol (For example: HTTPS protocol):



When you disable user to access web user interface of the IP phone using the HTTP/HTTPS protocol (For example: HTTPS protocol):



180 Ring Workaround

180 ring workaround defines whether to deal with the 180 message received after the 183 message. When the caller receives a 183 message, it suppresses any local ring-back tone and begins to play the media received. 180 ring workaround allows the IP phone to resume and play the local ring-back tone upon a subsequent 180 message received.

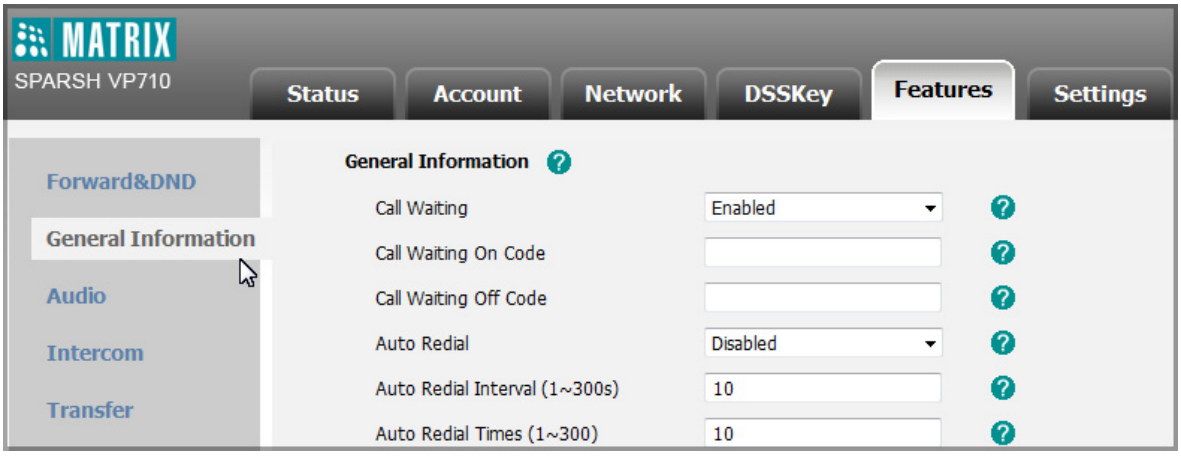
Procedure

180 ring workaround can be configured using the configuration files or locally.

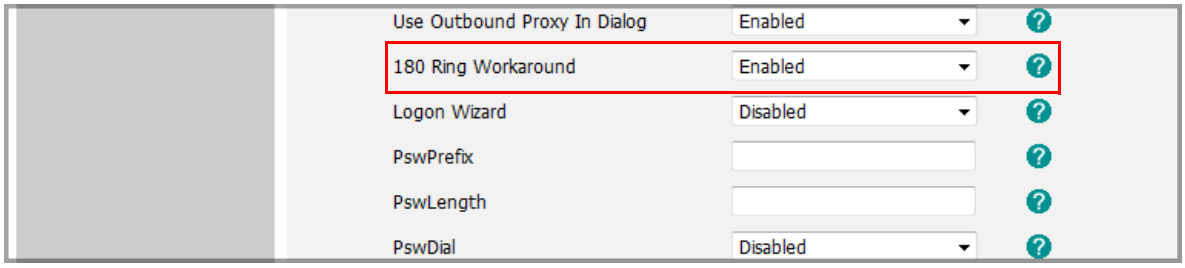
Configuration File	<MAC>.cfg	Configure 180 ring workaround. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure 180 ring workaround. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure 180 ring workaround via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **180 Ring Workaround** list.



- Click **Confirm** to save the change.

802.1X Authentication

IEEE802.1X authentication is an IEEE standard for the Port-based Network Access Control (PNAC), part of IEEE 802.1group of networking protocols.It offers an authentication mechanism for devices to connect/link to a LAN or WLAN.

The 802.1X authentication involves three parties:

- Supplicant
- Authenticator
- Authentication Server

The supplicant is the IP phone that wishes to attach to a LAN or WLAN.

With 802.1X port-based authentication, the IP phone provides credentials, such as username and password, for the authenticator, and then the authenticator forwards the credentials to the authentication server for verification.

If the authentication server determines the credentials to be valid, the IP phone is allowed to access the resources located on the protected side of the network.



The IP phones support the following protocols for 802.1X authentication:

- EAP-MD5
- EAP-TLS (requires Device and CA certificates, requires no password)
- EAP-PEAP/MSCHAPv2 (requires CA certificates)
- EAP-TTLS/EAP-MSCHAPv2 (requires CA certificates)
- EAP-PEAP/GTC (requires CA certificates)
- EAP-TTLS/EAP-GTC (requires CA certificates)

- EAP-FAST (requires CA certificates)

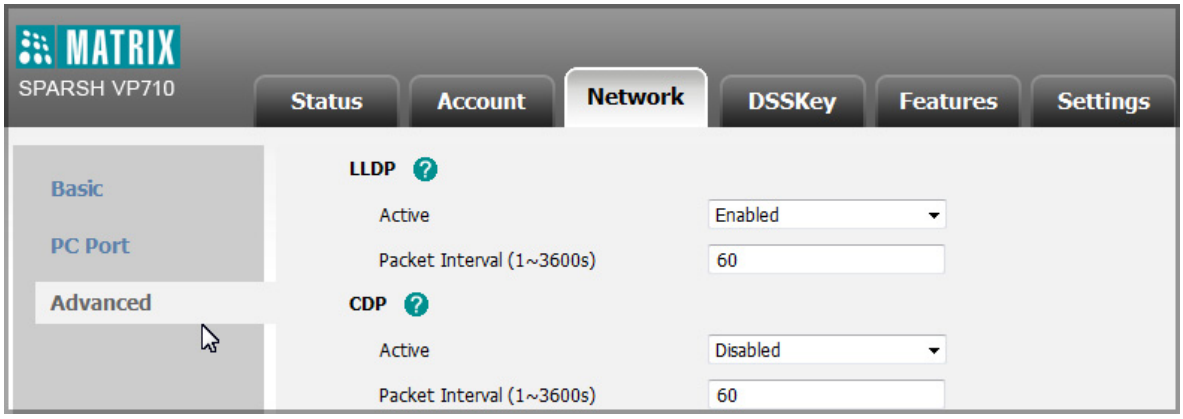
Procedure

802.1X authentication can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the 802.1X authentication. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the 802.1X authentication. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load
	Phone User Interface	Configure the 802.1X authentication

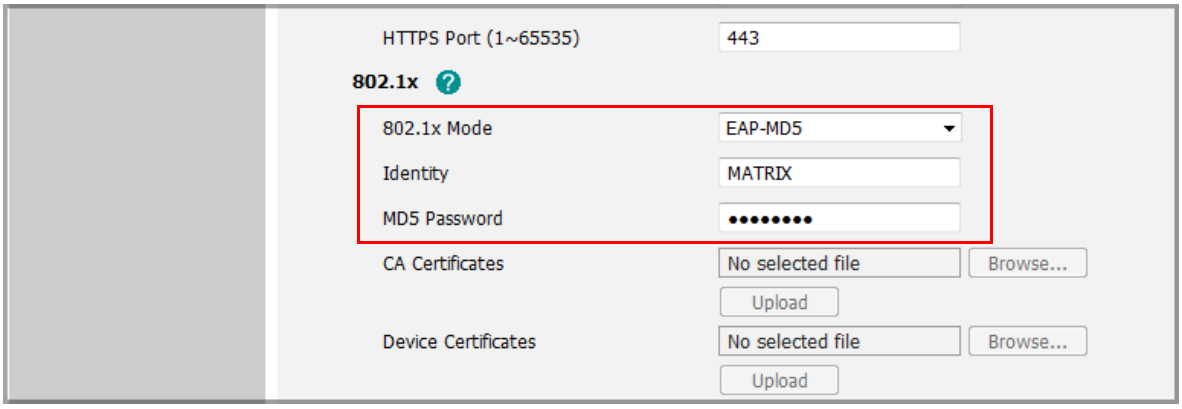
To configure the 802.1Xauthentication via web user interface:

- Click on **Network->Advanced**.

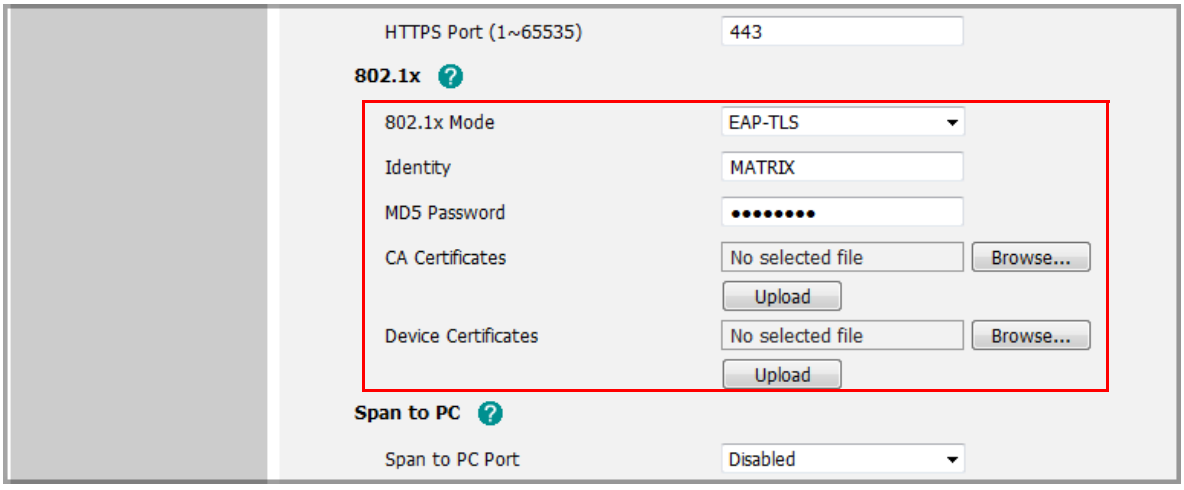


- In the **802.1x** block, select the desired protocol from the **802.1x Mode** list.
  - If you select **EAP-MD5**:
    - Enter the username for authentication in the **Identity** field.

- Enter the password for authentication in the **MD5 Password** field.

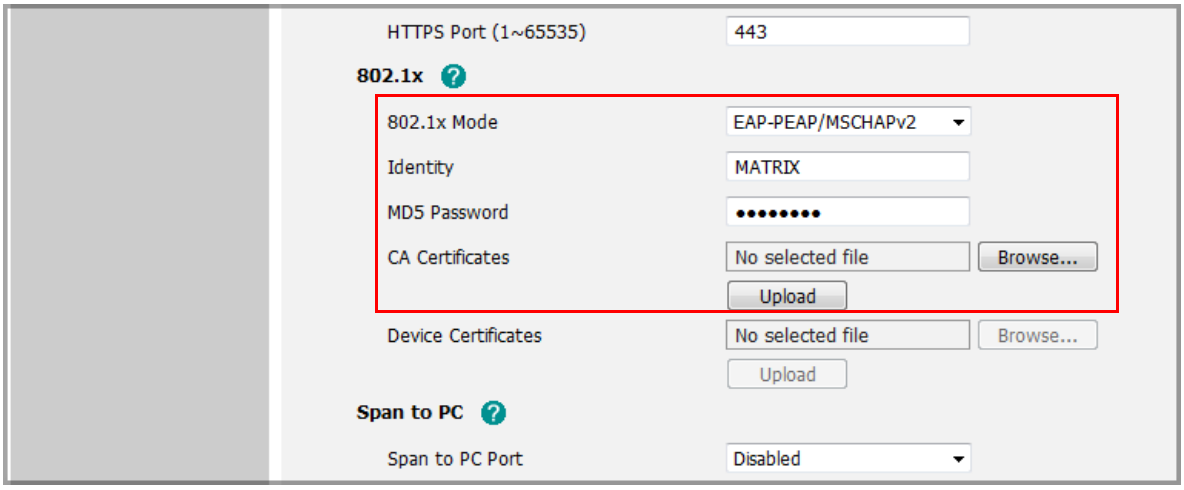


- If you select **EAP-TLS**:
  - Enter the username for authentication in the **Identity** field.
  - Leave the **MD5 Password** field blank.
  - In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
  - In the **Device Certificates** field, click **Browse** to select and **Upload** desired client (*.pem or *.cer) certificate from your local system.

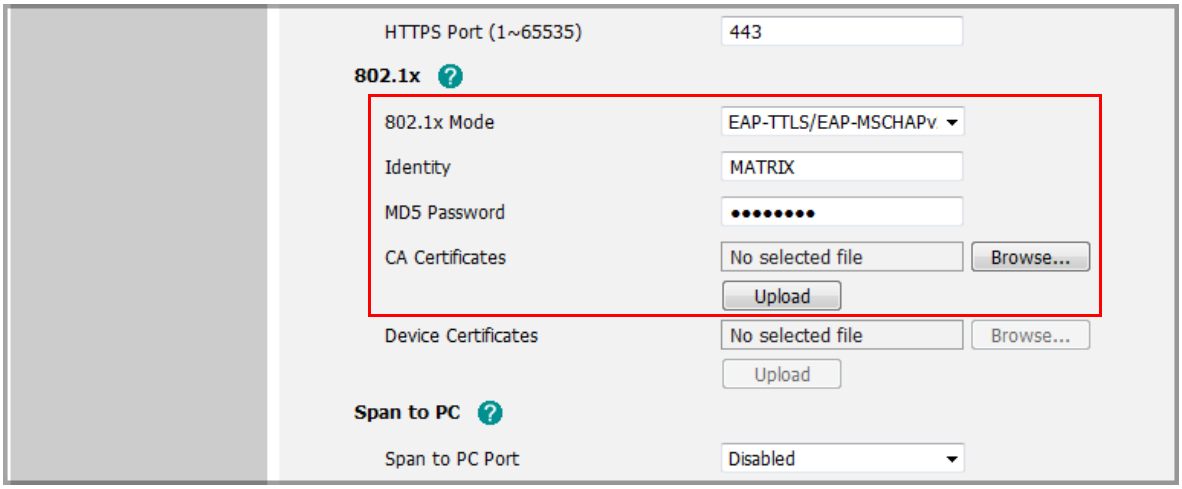


- If you select **PEAP-MSCHAPv2**:
  - Enter the username for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.

- In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.

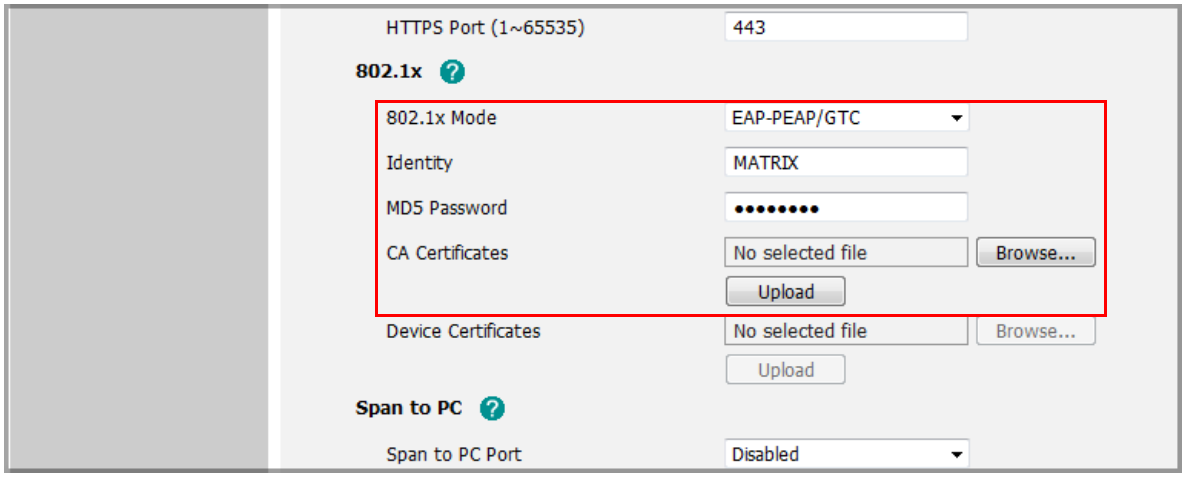


- If you select **EAP-TTLS/EAP-MSCHAPv2**:
  - Enter the username for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.
  - In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.

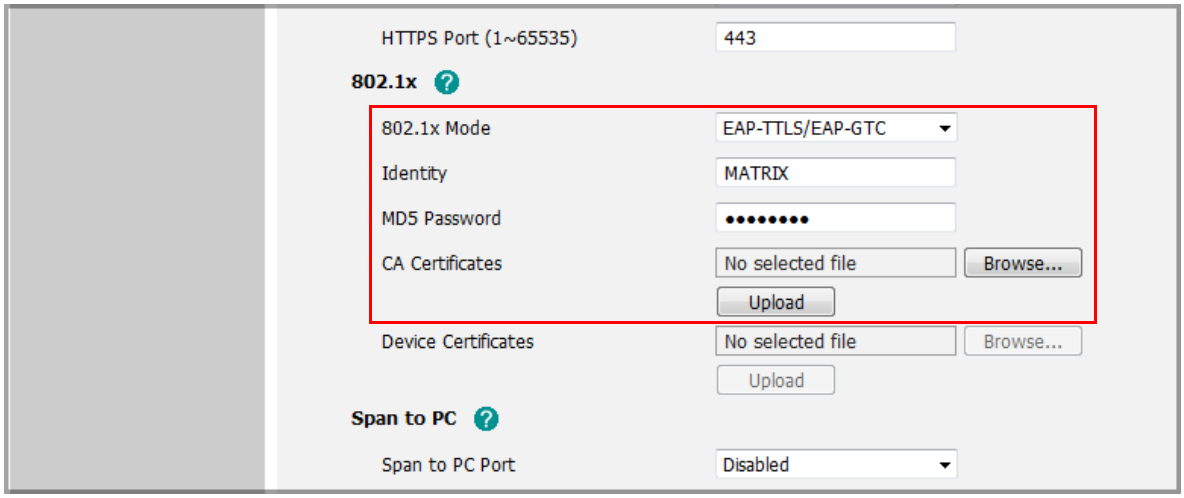


- If you select **EAP-PEAP/GTC:**
  - Enter the user name for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.

- In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CA certificate(*.pem, *.crt, *.cer or *.der) from your local system.

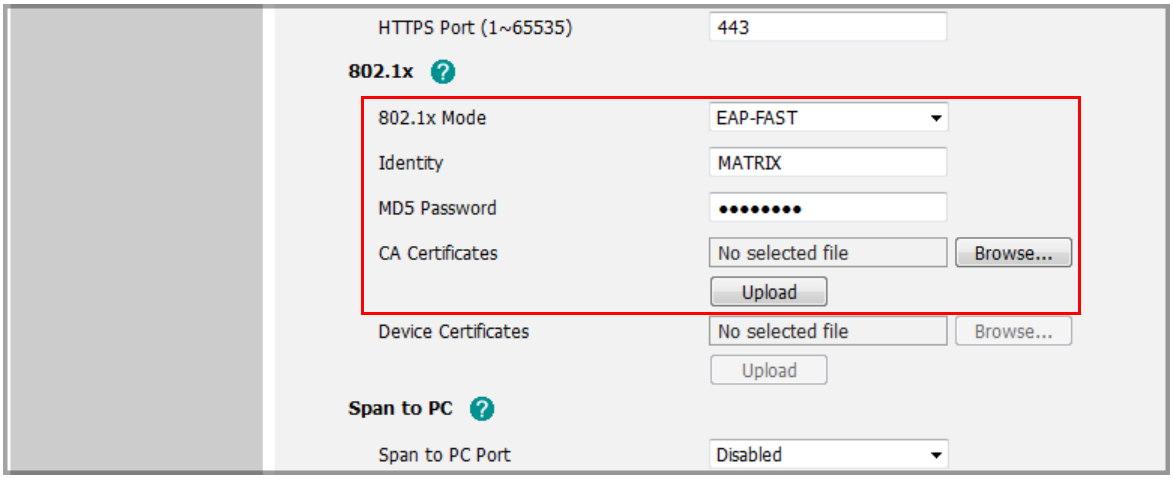


- If you select **EAP-TTLS/EAP-GTC**:
  - Enter the user name for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.
  - In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CAcertificate (*.pem, *.crt, *.cer or *.der) from your local system.



- If you select **EAP-FAST:**
  - Enter the user name for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.

- In the **CA Certificates** field, click **Browse** to select and **Upload** the desired CA certificate(*.pem, *.crt, *.cer or *.der) from your local system.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the IP phone.

To configure the 802.1X authentication via phone user interface:

- Swipe down from the top of the screen or swipe left/right to go to the second idle screen.
- Tap **Settings->Advanced** (default password: admin) **->Network->802.1**x.
- Tap the **802.1x Mode** field

  - If you select **EAP-MD5**:
    - Enter the username for authentication in the **Identity** field.
    - Enter the password for authentication in the **MD5 Password** field.

  - If you select **EAP-TLS**:
    - Enter the username for authentication in the **Identity** field.
    - Leave the **MD5 Password** field blank.

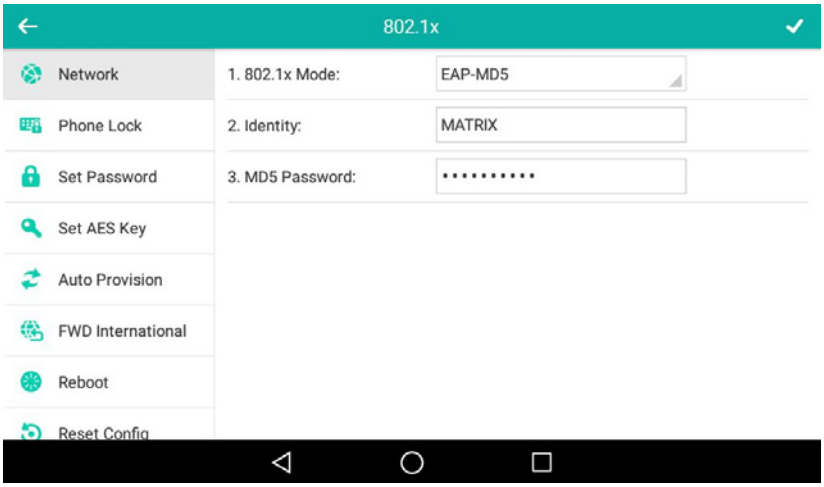
  - If you select **EAP-PEAP/MSCHAPv2:**
    - Enter the user name for authentication in the **Identity** field.
    - Enter the password for authentication in the **MD5 Password** field.

  - If you select **EAP-TTLS/EAP-MSCHAPv2:**
    - Enter the username for authentication in the **Identity** field.
    - Enter the password for authentication in the **MD5 Password** field.

  - If you select **EAP-PEAP/GTC:**
    - Enter the username for authentication in the **Identity** field.
    - Enter the password for authentication in the **MD5 Password** field.

- If you select **EAP-TTLS/EAP-GTC:**
  - Enter the username for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.

- If you select **EAP-FAST:**
  - Enter the username for authentication in the **Identity** field.
  - Enter the password for authentication in the **MD5 Password** field.



- Tap ✓ to save the change or ◁ to cancel.

  The phone prompts you to reboot the phone.

- Tap **OK** to reboot the phone.

  The settings will take effect after a reboot.

100 Reliable Retransmission

As described in RFC 3262, 100rel tag is for the reliability of provisional responses. When present in a Supported header, it indicates that the IP phone can send or receive reliable provisional responses. When present in a Require header in a reliable provisional response, it indicates that the response is to be sent reliably.

Example of a SIP INVITE message:

```
INVITE sip:1024@pbx.matrix.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.3.6.197:5060;branch=z9hG4bK1708689023
From: "1025" <sip:1025@pbx.matrix.com:5060>;tag=1622206783
To: <sip:1024@pbx.matrix.com:5060>
Call-ID: 0_537569052@10.3.6.197
CSeq: 2 INVITE
Contact: <sip:1025@10.3.6.197:5060>
Authorization: Digest username="1025", realm="pbx.matrix.com",
nonce="BroadWorksXi5stub71Ts2nb05BW",
uri="sip:1024@pbx.matrix.com:5060", response="f7e9d35c55af45b3f89beae95e913171",
algorithm=MD5,
cnonce="0a4f113b", qop=auth, nc=00000001
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH,UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: MATRIX SPARSH VP710 58.80.196.2  00:1b:09:04:8e:3c
Supported: 100rel
Allow-Events: talk,hold,conference,refer,check-sync
Content-Length: 302
```

Procedure

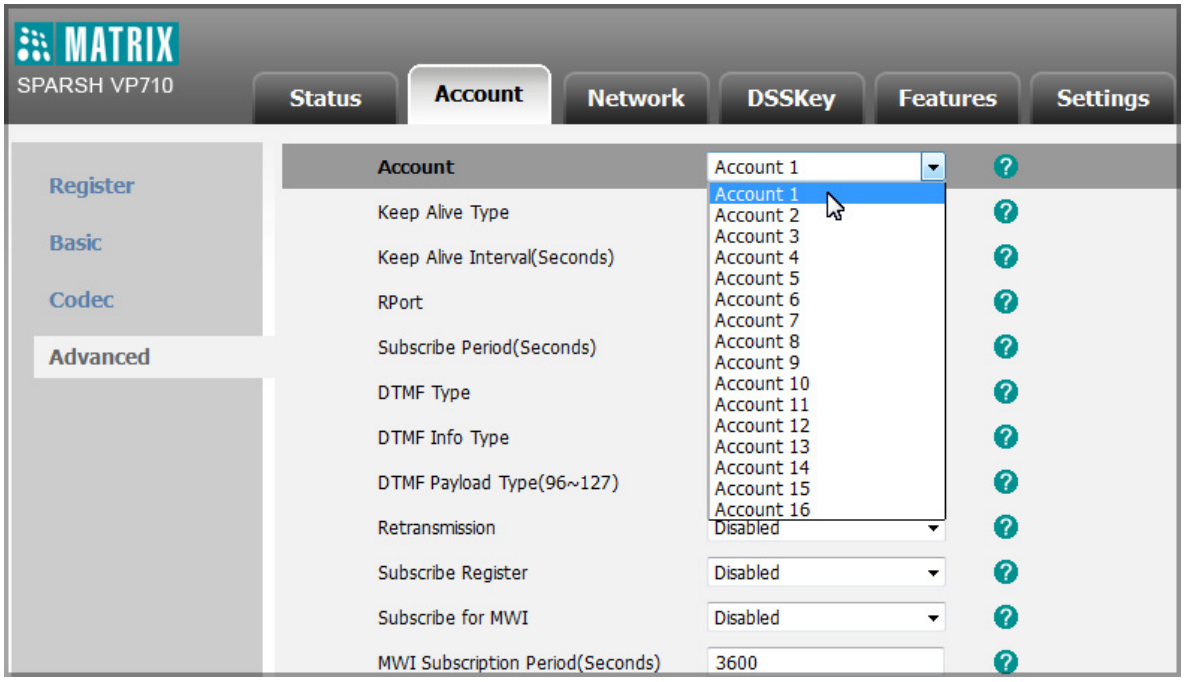
100 Reliable Retransmission can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the 100 reliable retransmission feature. For more information, refer "Appendix D - Configuration Parameters".

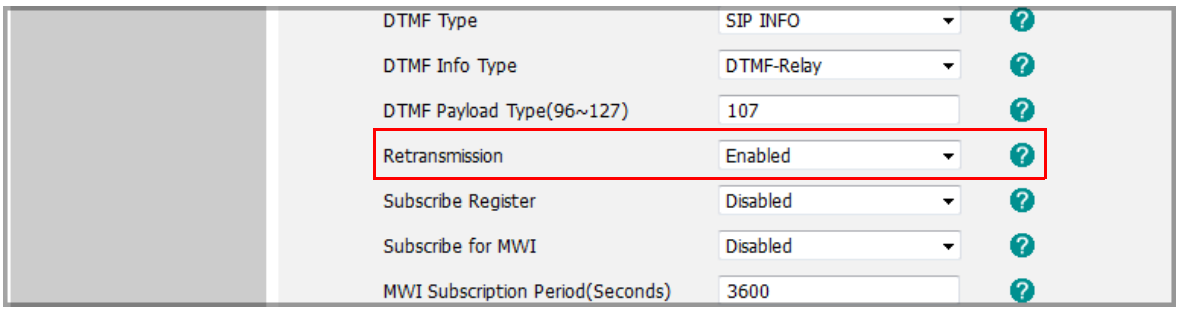
Local	Web User Interface	Configure the 100 reliable retransmission feature. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure 100 reliable retransmission via web user interface:

- Click on **Account->Advanced**.
- Select the desired account from the **Account** list.



- Select the desired value from the **Retransmission** list.



- Click **Confirm** to save the change.

CHAPTER 11

Configuring Audio Features

This chapter provides information for making configuration changes for the following audio features:

- Headset Use
  - Headset Mode Activation/De-activation
  - Headset Prior
- Audio Codecs
- Acoustic Clarity Technology

Headset Use

Physically connect your headset and activate the headset mode for use. For more information on physically connecting a headset, refer “Phone Installation”.

If you want to use a Bluetooth headset, activate the Bluetooth mode and then pair and connect the Bluetooth headset with your phone. For more information on using a Bluetooth headset, refer “Bluetooth”.

This section provides an introduction to wired headset use.

The IP phone supports two types of headsets: USB headset and Bluetooth headset. You can connect up to two types of headsets at once. The IP phone will use the headset by the priority. If both a Bluetooth headset and a USB headset are connected, only the last connected headset can be used. You can use USB headset to answer an incoming call directly without activating the headset mode.

Headset Mode Activation/De-activation

To activate the headset mode:

- Press  on the phone.

  The HEADSET key LED illuminates solid green, and the icon  appears on the status bar.

  When you tap the line key or the **Answer** soft key to answer an incoming call, the call is connected to your headset automatically. For more information on using the headset to answer a call, refer to “Answering Calls”.

With the handset on-hook, enter the desired number and tap **Send**, the phone will then place a call using the headset automatically. For more information on using the headset to place a call, refer to "Placing Calls".

To deactivate the headset mode:

- Press  again on the phone.

  The HEADSET key LED goes out, and the icon  disappears from the status bar.

Ringer Device for Headset

The IP phones support either or both speaker and headset ringer devices. Ringer Device for Headset feature allows users to configure which ringer device to be used when receiving an incoming call. For example, if the ringer device is set to Headset, ring tone will be played through your headset.

If the ringer device is set to Headset or Headset & Speaker, the headset (wired headset, Bluetooth headset or USB headset) should be connected to the IP phone and the headset mode also should be activated in advance. You can press the HEADSET key to activate the headset mode.

Procedure

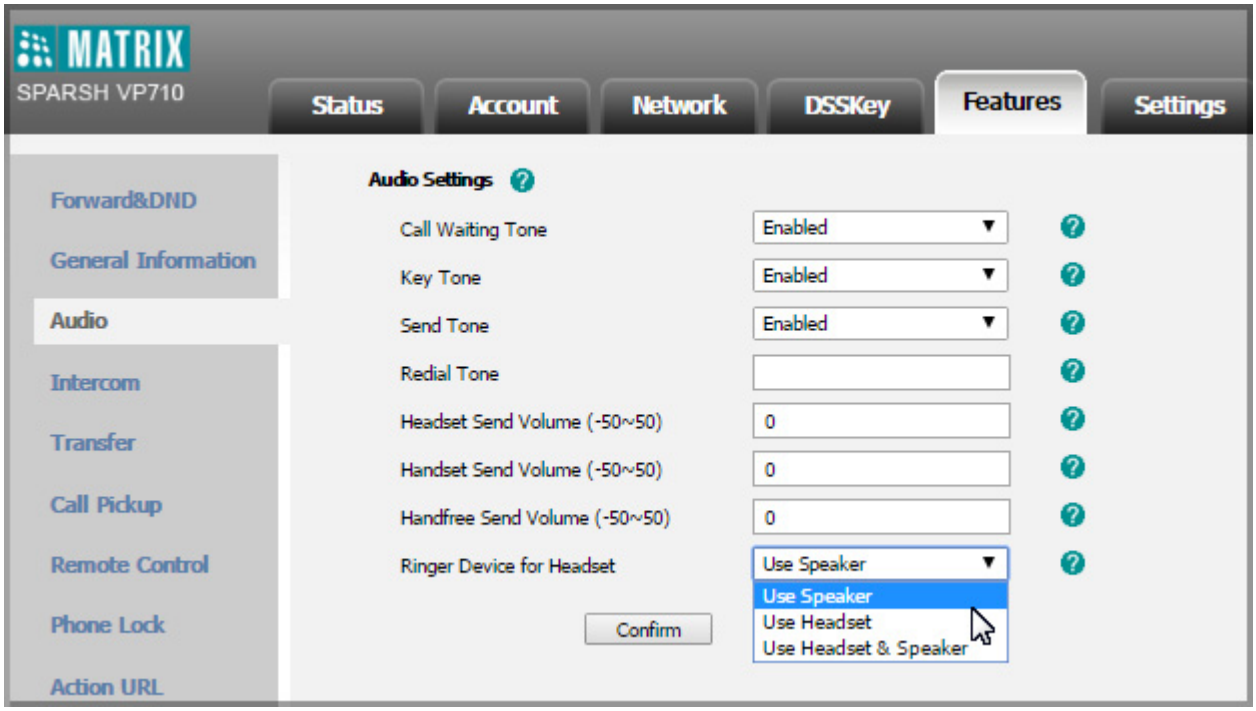
Ringer device for headset can be configured using the following methods

Configuration File	<MAC>.cfg	Configure the ringer device for the IP phone. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the ringer device for the IP phone. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure ringer device for headset via web user interface:

- Click on **Features->Audio.**

- Select the desired value from the **Ringer Device for Headset** list.



- Click **Confirm** to save the change.

Headset Prior

Headset prior allows you to use headset preferentially if a headset is physically connected to the IP phone. This feature is especially useful for permanent or full-time headset users.

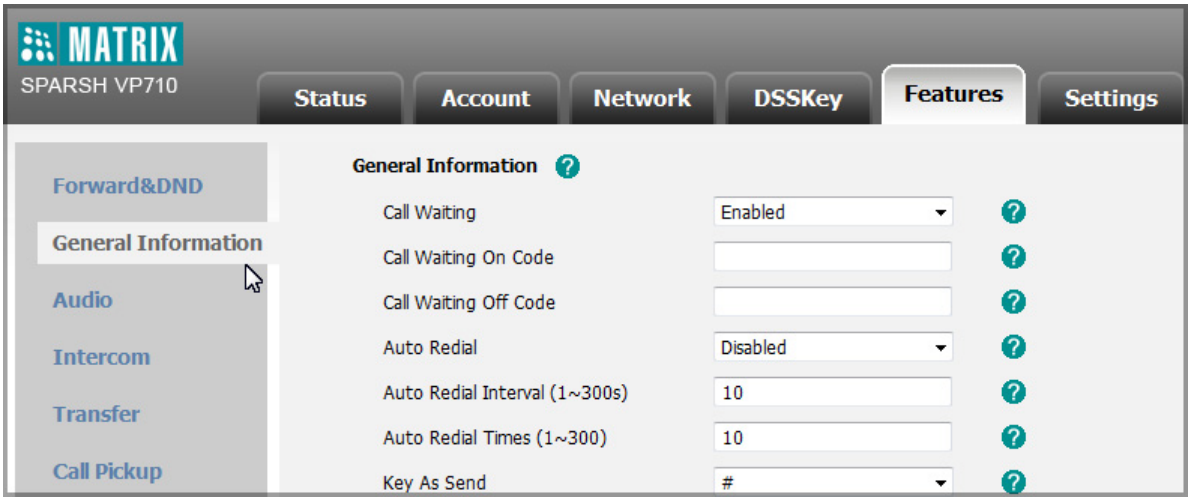
Procedure

Headset prior can be configured using the configuration files or locally.

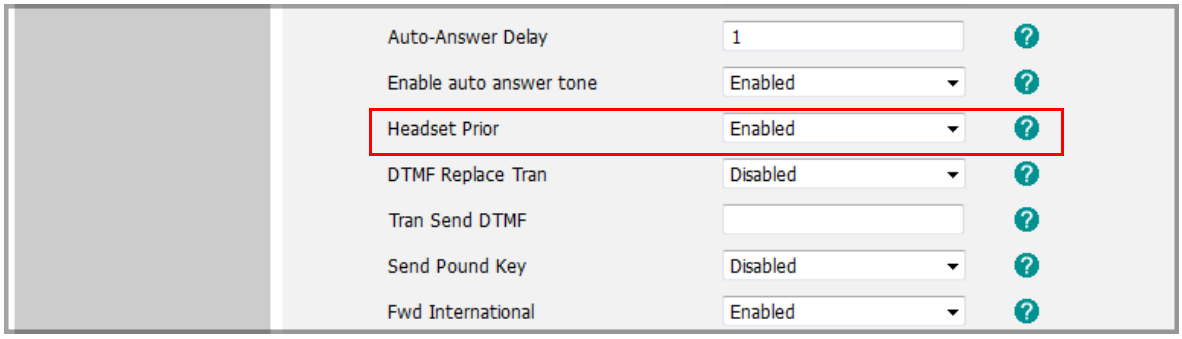
Configuration File	<MAC>.cfg	Configure headset prior. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure headset prior. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

To configure headset prior via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Headset Prior** list.



- Click **Confirm** to save the change.

To use headset prior, you should activate the headset mode in advance:

- Physically connect the headset.
- Press  to activate the headset mode.

- *If headset prior is enabled, the headset mode will not be deactivated until you press the Headset key again.*
- *If headset prior is disabled, the headset mode can be deactivated by pressing the Speakerphone key or the HEADSET key.*
- *Headset prior is configurable via web user interface only.*

Dual Headset

Dual headset allows you to use two headsets on an IP phone. To use this feature, you need to physically connect two headsets to the headset jack and handset jack respectively. Once the IP phone connects to a call, the headset connected to the headset jack has full-duplex capabilities, while the one connected to the handset jack is only able to listen.

Procedure

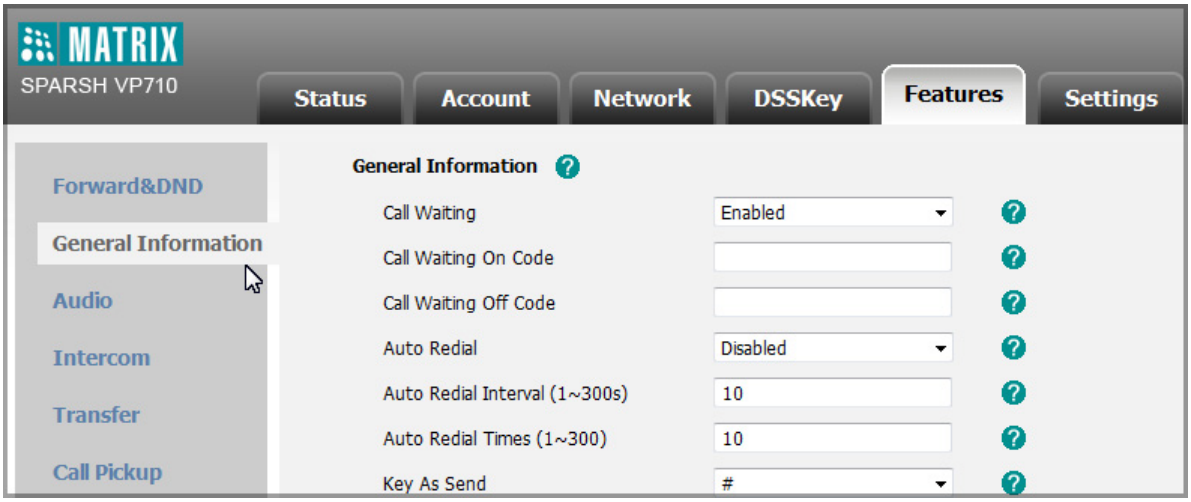
Dual Headset can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure dual headset. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure dual headset. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=features-general&q=load

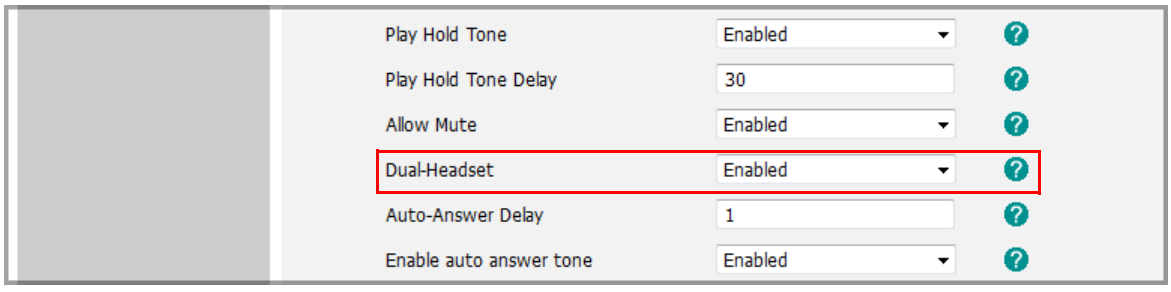
- *Dual headset is not applicable to Bluetooth headset and USB headset.*
- *Dual headset is configurable via web user interface only.*

To configure dual headset via web user interface:

- Click on **Features->General Information**.



- Select the desired value from the **Dual-Headset** list.



- Click **Confirm** to save the change.

Bluetooth Headset

If you want to use a Bluetooth headset, activate the Bluetooth mode and then pair and connect the Bluetooth headset with your phone. For more information on using a Bluetooth headset, refer “Bluetooth”.

Audio Codecs

CODEC is an abbreviation of COmpress-DECompress, capable of coding or decoding a digital data stream or signal by implementing an algorithm. The objective of the algorithm is to represent the high-fidelity audio signal with minimum number of bits while retaining the quality. This can effectively reduce the frame size and the bandwidth required for audio transmission.

The audio codec that the IP phone uses to establish a call should be supported by the SIP server.When placing a call, the IP phone offers the enabled audio codec list to the server and then use the audio codec negotiated with the called party according to the priority.

Supported Audio Codecs

The audio codecs supported by the IP phone are summarized in the following table:

Phone Model	Supported Audio Codecs	Default Audio Codecs
SPARSH VP710	G.722.1c(48kb/s), G.722.1c(32kb/s) ,G.722.1c(24kb/s),G.722.1(24kb/s), G722, PCMU, PCMA, G729, G726-40,G726-32,G726-24,G726-16, iLBC,G723_53, G723_63, Opus	G.722.1c(48kb/s), G.722.1c(32kb/s), G.722.1c(24kb/s), G.722.1(24kb/s), G722, PCMU, PCMA, G729

The Opus codec supports various audio bandwidths, defined as follows:

Abbreviation	Audio Bandwidth	Sample Rate (Effective)
NB (narrowband)	4kHz	8 kHz
MB (medium-band)	6kHz	12 kHz
WB (wideband)	8kHz	16 kHz
SWB (super-wideband)	12kHz	24 kHz
FB (fullband)	20kHz	48 kHz

The following table summarizes the supported audio codecs on IP phone:

Codec	Algorithm	Reference	Bit Rate	Sample Rate	Packetization Time
G.722.1c	G.722.1	RFC 5577	48 Kbps	32 Ksps	20ms
G.722.1c		RFC 5577	32 Kbps	32 Ksps	20ms
G.722.1c		RFC 5577	24 Kbps	32 Ksps	20ms
G.722.1	G.722.1	RFC 5577	24 Kbps	16 Ksps	20ms
G722	G.722	RFC 3551	64 Kbps	16 Ksps	20ms

PCMA	G.711 a-law	RFC 3551	64 Kbps	8 Ksps	20ms
PCMU	G.711 u-law	RFC 3551	64 Kbps	8 Ksps	20ms
G729	G.729	RFC 3551	8 Kbps	8 Ksps	20ms
G726-16	G.726	RFC 3551	16 Kbps	8 Ksps	20ms
G726-24	G.726	RFC 3551	24 Kbps	8 Ksps	20ms
G726-32	G.726	RFC 3551	32 Kbps	8 Ksps	20ms
G726-40	G.726	RFC 3551	40 Kbps	8 Ksps	20ms
G723_53/ G723_63	G.723.1	RFC 3551	5.3 Kbps 6.3 Kbps	8 Ksps	30ms
iLBC	iLBC	RFC 3952	15.2 Kbps 13.33 Kbps	8 Ksps	20ms 30ms
Opus	Opus	RFC 6716	8-12 Kbps 16-20 Kbps 28-40 Kbps 48-64 Kbps 64-128 Kbps	8 Ksps 12 Ksps 16 Ksps 24 Ksps 48 Ksps	20ms

The network bandwidth necessary to send the encoded audio is typically 5~10% higher than the bit rate due to packetization overhead. For example, a two-way G.722 audio-only call at 64 Kbps consumes about 135 Kbps of network bandwidth.

Codecs and priorities of these codecs are configurable on a per-line basis. The attribute “rtpmap” is used to define a mapping from RTP payload codes to a codec, clock rate and other encoding parameters.

.The corresponding attributes of the codec for SPARSH VP710 are listed as follows:

Codec	Configuration	Priority	RTPmap
G.722.1c(48kb/s)	Configuration Files Web User Interface	1	121
G.722.1c(32kb/s)	Configuration Files Web User Interface	2	122
G.722.1c(24kb/s)	Configuration Files Web User Interface	3	123
G.722.1(24kb/s)	Configuration Files Web User Interface	4	124
G722	Configuration Files Web User Interface	5	9

PCMU	Configuration Files Web User Interface	6	0
PCMA	Configuration Files Web User Interface	7	8
G729	Configuration Files Web User Interface	8	18
Opus	Configuration Files Web User Interface	0	107
G726-40	Configuration Files Web User Interface	0	105
G726-32	Configuration Files Web User Interface	0	102
G726-24	Configuration Files Web User Interface	0	104
G726-16	Configuration Files Web User Interface	0	103
iLBC	Configuration Files Web User Interface	0	106
G723_53	Configuration Files Web User Interface	0	4
G723_63	Configuration Files Web User Interface	0	4

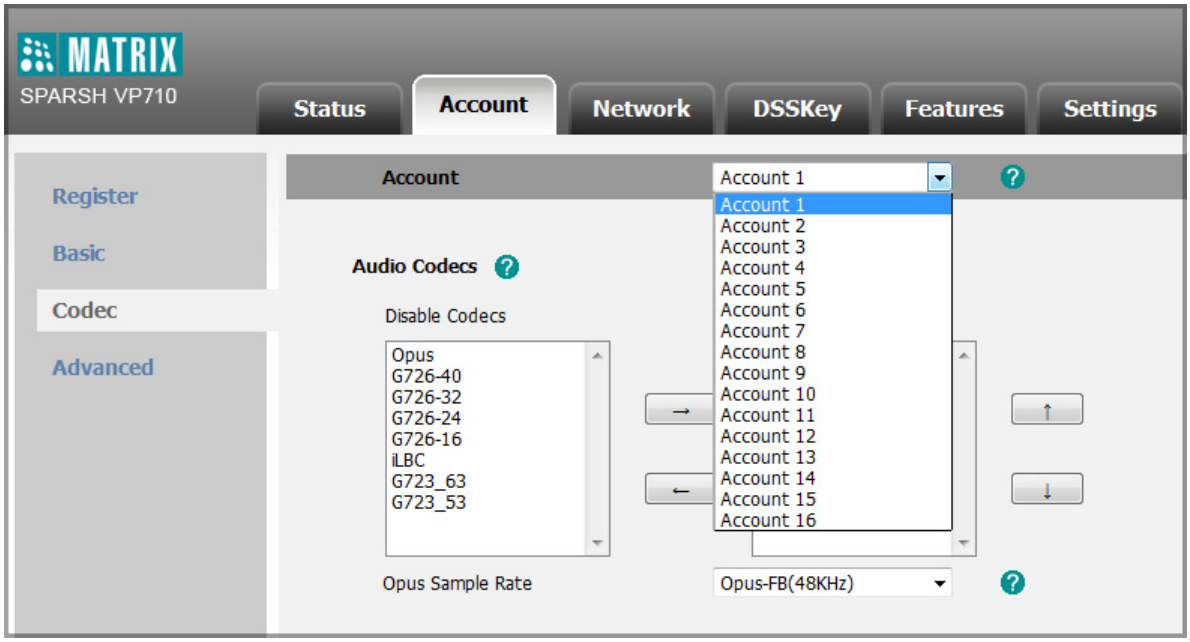
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the codecs to use on a per-line basis. Configure the priority and rtpmap for the enabled codec. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the codecs to use on a per-line basis. Configure the priority and rtpmap for the enabled codec. **Navigate to:** http://<phoneIPAddress>/servlet?p=account-codec&q=load&acc=0

To configure the codecs to use and adjust the priority of the enabled codecs via web user interface:

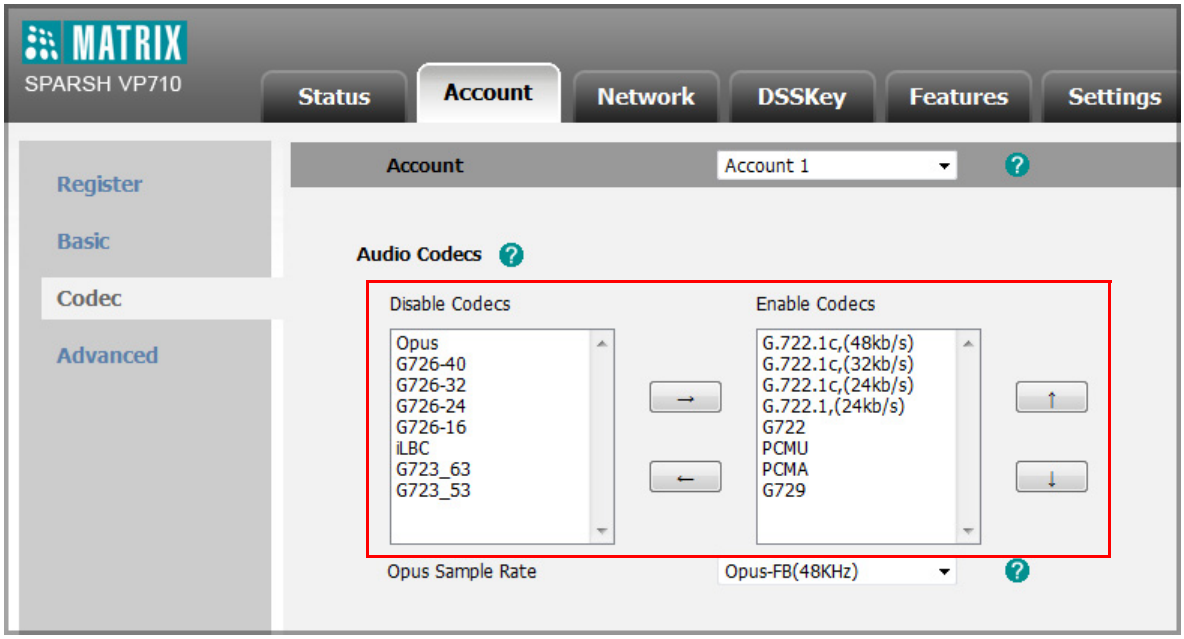
- Click on **Account ->Codec**.
- Select the desired account from the **Account** list.



- Select the desired codec from the **Disable Codecs** column and then click →.

  The selected codec appears in the **Enable Codecs** column.

- Repeat the above step to add more codecs to the **Enable Codecs** column.
- To remove the codec from the **Enable Codecs** column, select the desired codec and then click ←.
- To adjust the priority of codecs, select the desired codec and then click ↑ or ↓.



- Click **Confirm** to save the change.

Packetization Time

Ptime (Packetization Time) is a measurement of the duration (in milliseconds) of the audio data in each RTP packet sent to the destination, and defines how much network bandwidth is used for the RTP stream transfer. Before establishing a conversation, codec and ptime are negotiated through SIP signaling. The valid values of ptime range from 10 to 60, in increments of 10 milliseconds. The default ptime is 20ms. You can also disable the ptime negotiation.

The following table summarizes the valid values of ptime for each audio codec:

Codec	Packetization Time (Minimun)	Packetization Time (Maximun)
G.722.1c(48kb/s)	20ms	60ms
G.722.1c(32kb/s)	20ms	60ms
G.722.1c(24kb/s)	20ms	60ms
G.722.1(24kb/s)	20ms	60ms
G722	10ms	40ms
PCMA	10ms	40ms

PCMU	10ms	40ms
G729	10ms	80ms
G726-16	10ms	30ms
G726-24	10ms	30ms
G726-32	10ms	30ms
G726-40	10ms	30ms
G723_53/G723_63	30ms	60ms
iLBC	20ms	30ms
Opus	10ms	20ms

Procedure

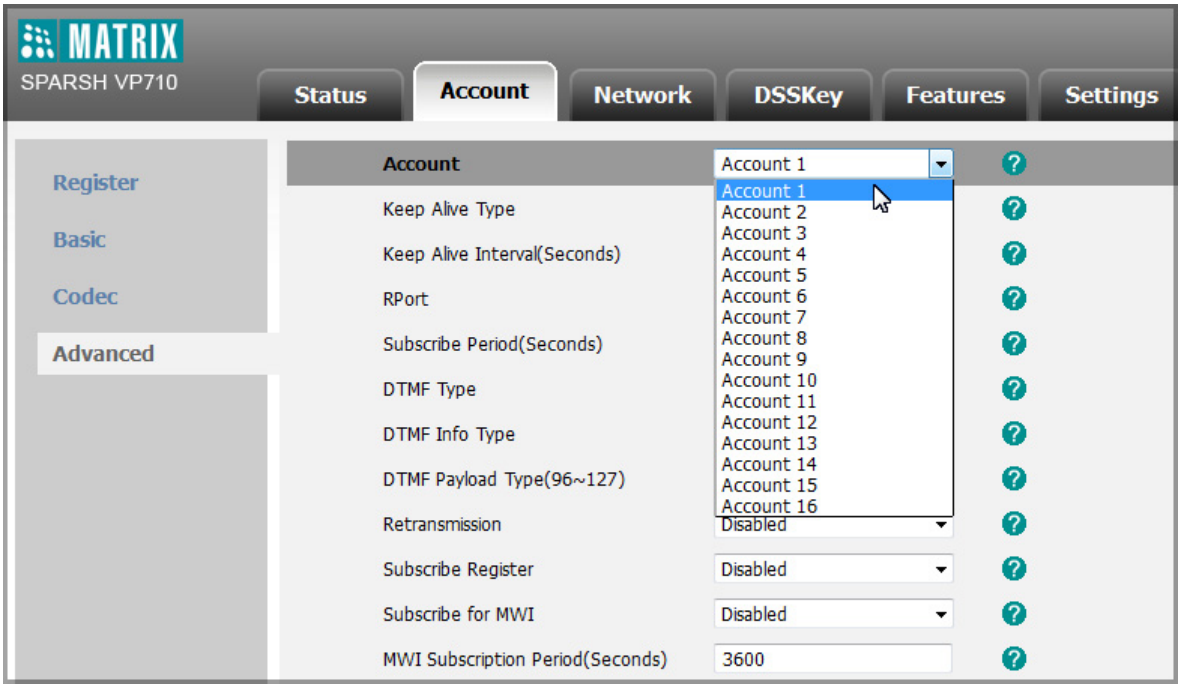
Packetization time can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the ptime. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the ptime. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

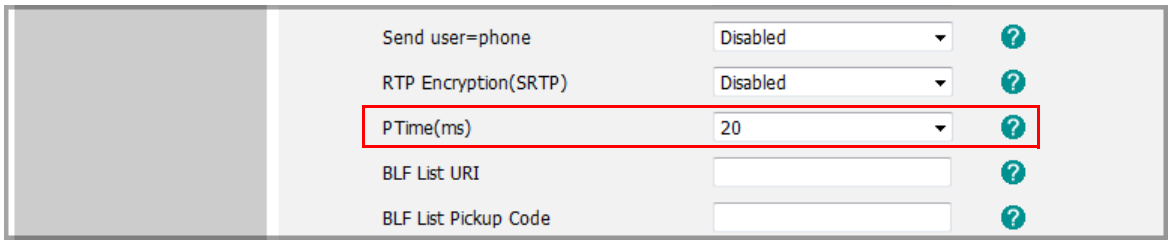
To configure the ptime for the account via web user interface:

- Click on **Account ->Advanced**.

- Select the desired account from the **Account** list.



- Select the desired value from the **PTime (ms)** list.



- Click **Confirm** to save the change.

Opus Sample Rate

You can configure the following types of sample rate for Opus audio codec:

- Opus-FB(48KHz)
- Opus-SWB(24KHz)
- Opus-WB(16KHz)
- Opus-MB(12KHz)
- Opus-NB(8KHz)

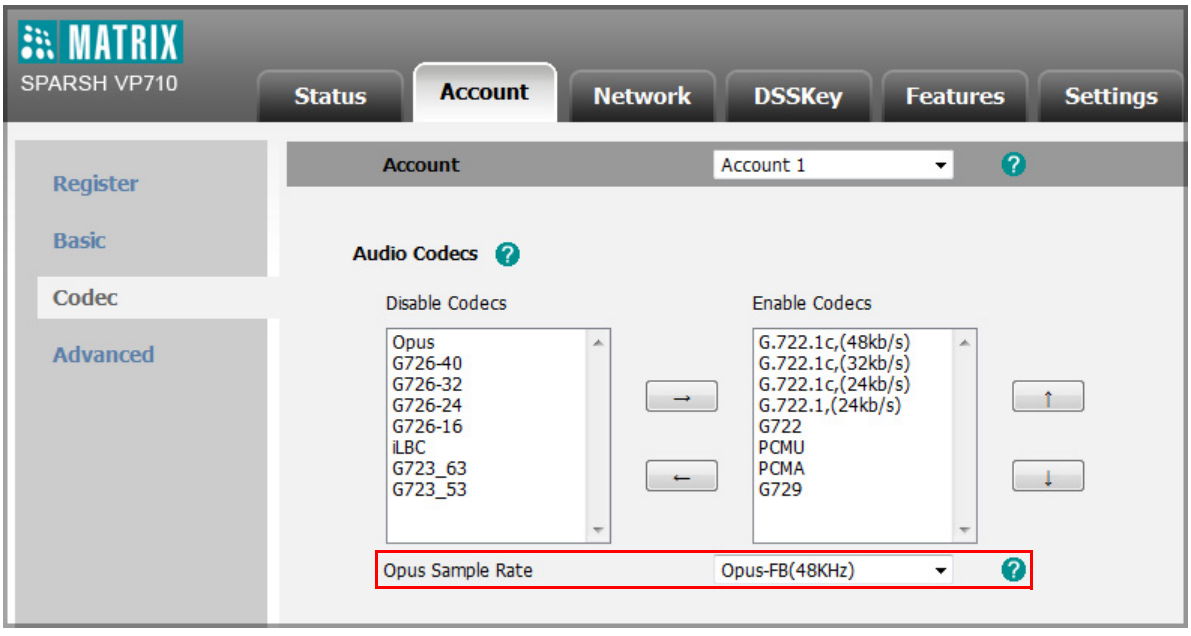
Procedure

Opus sample rate can be only configured via web user interface.

Configuration File	<MAC>.cfg	Configure the Opus sample rate. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the Opus sample rate. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-codec&q=load&acc=0

To configure the opus sample rate via web user interface:

- Click on **Account->Codec**.
- Select the desired value from the **Opus Sample Rate** list.



- Click **Confirm** to save the change.

Acoustic Clarity Technology

Acoustic Clarity Technology involves the following:

- Acoustic Echo Cancellation
- Background Noise Suppression
- Automatic Gain Control
- Voice Activity Detection
- Comfort Noise Generation
- Jitter Buffer

Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) is used to reduce acoustic echo from a voice communication in order to improve the voice quality. It also increases the capacity achieved through silence suppression by preventing echo from traveling across the network.

The IP phone employs advanced acoustic echo cancellation for hands-free operation. Acoustic Echo Cancellation is normally not required for calls via the handset.In certain situations, where echo is experienced by the remote party, AEC may be used to reduce/avoid echo when the user uses the handset.



Utilizing acoustic echo cancellation will introduce a small delay increase into audio path which might cause a lower voice quality.

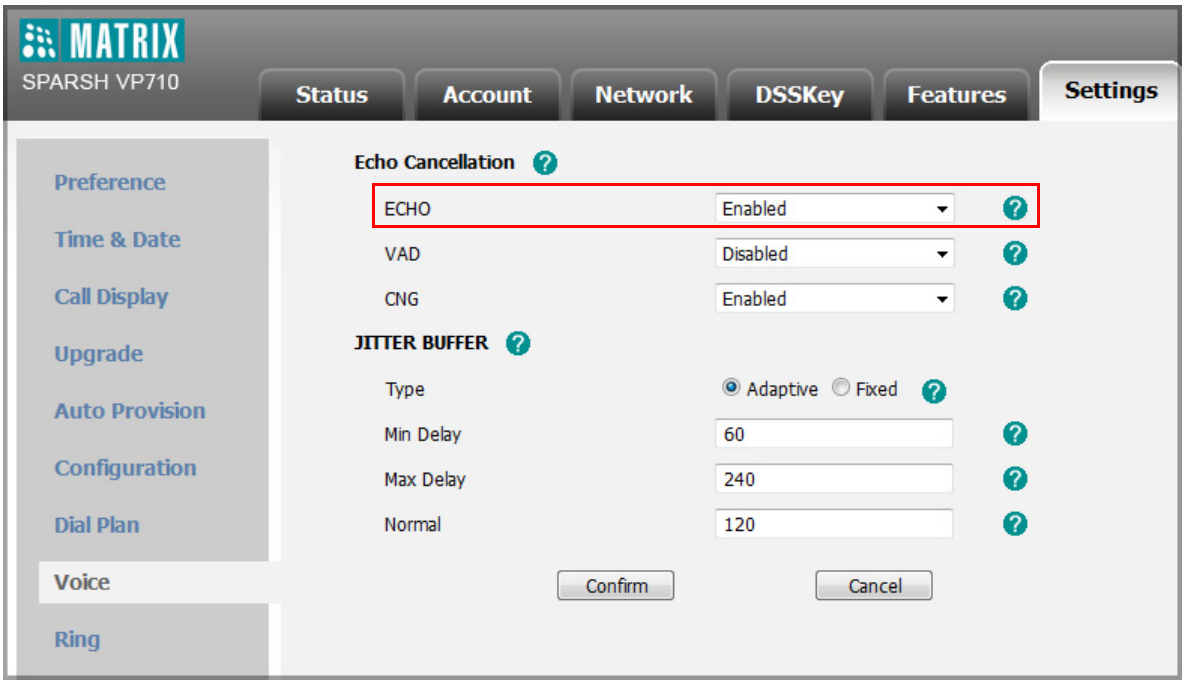
Procedure

AEC can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure acoustic echo cancellation. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure acoustic echo cancellation **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-voice&q=load

To configure AEC via web user interface:

- Click on **Settings->Voice**.
- Select the desired value from the **ECHO** list.



- Click **Confirm** to save the change.

Background Noise Suppression (BNS)

Background noise suppression (BNS) is designed primarily for hands-free operation and reduces background noise to enhance the communication in noisy environment.

Automatic Gain Control (AGC)

Automatic Gain Control (AGC) is applicable to hands-free operation and is used to keep the audio output at nearly a constant level by adjusting the gain of signals in certain circumstances. This increases the effective user-phone radius and helps with the intelligibility of soft-talkers.

Voice Activity Detection (VAD)

Voice Activity Detection (VAD) is used in speech processing to detect the presence or absence of human speech. When detecting period of “silence”, VAD replaces that silence efficiently with special packets that indicate silence is occurring. It facilitates speech processing and deactivate some processes during non-speech section of an audio session. VAD can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth.

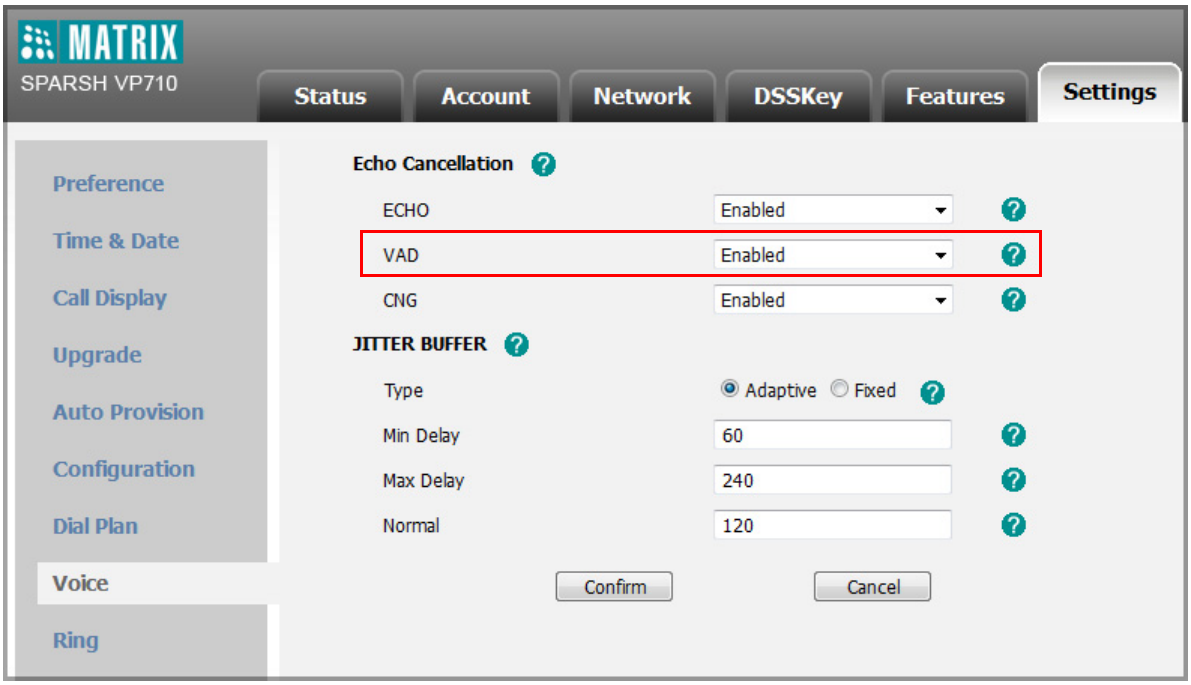
Procedure

VAD can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure VAD. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure Voice Activity Detection. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-voice&q=load

To configure VAD via web user interface:

- Click on **Settings->Voice**.
- Select the desired value from the **VAD** list.



- Click **Confirm** to save the change.

Comfort Noise Generation

Comfort Noise Generation (CNG) is used to generate background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that

background sound is consistent throughout the call and the listener does not think that the line has released. The purpose of VAD and CNG is to maintain an acceptable perceived QoS while simultaneously keeping the transmission costs and bandwidth usage as low as possible.



VAD is used to send CN packets when the phone detects a "silence" period.
CNG is used to generate comfortable noise when the phone receives CN packets from the other side.

For example:

- A is talking with B.

  A: VAD=1, CNG=1
  B: VAD=0, CNG=1
- If A mutes the call, since VAD=1, A will send CN packets to B. When receiving CN packets, B will generate comfortable noise.
- If B mutes the call, since VAD=0, B will not send CN packets to A. So even if CNG=1 (B), A will not hear comfortable noise.

Procedure

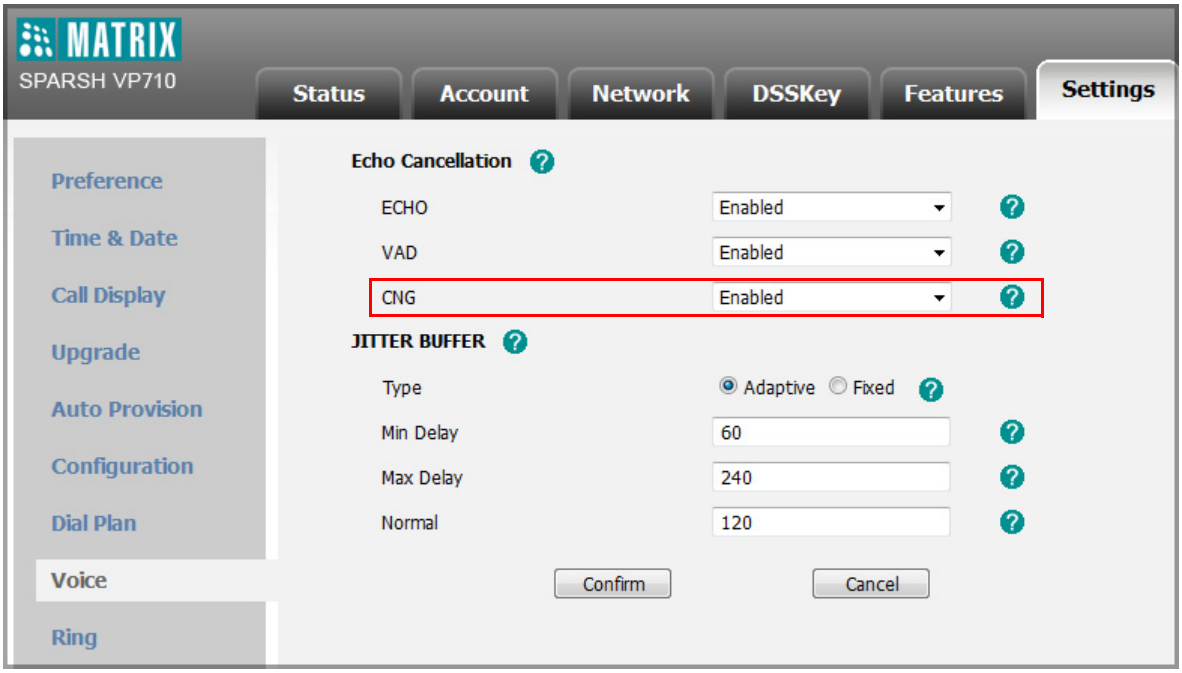
CNG can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure comfort noise generation. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure comfort noise generation. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-voice&q=load

To configure CNG via web user interface:

- Click on **Settings->Voice**.

- Select the desired value from the **CNG** list.



- Click **Confirm** to save the change.

Jitter Buffer

Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals.

Jitter indicates the variations in the packet arrival time which occurs due to network congestion, timing drift or changes in the route. The jitter buffer located at the receiving end of the voice connection intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion.

The IP phone supports two types of jitter buffers:

- Fixed
- Adaptive

Fixed: A fixed jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on the IP phone.

Adaptive: An adaptive jitter buffer is capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can also be configured on the IP phone.

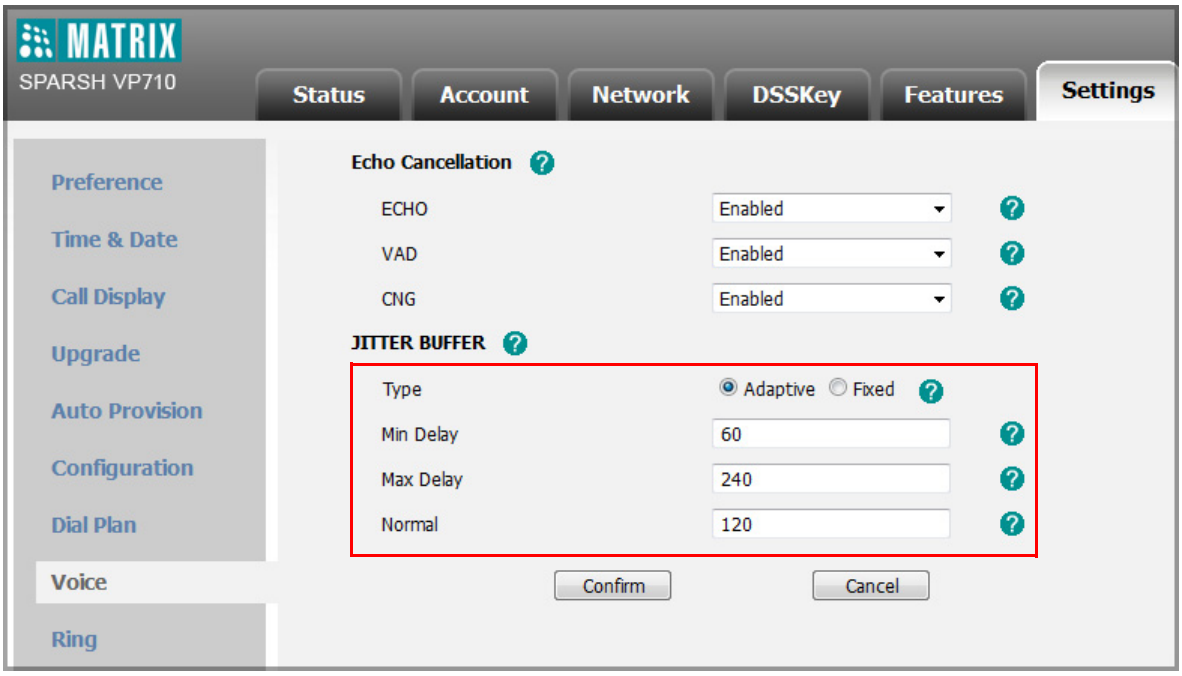
Procedure

Jitter buffer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the mode of jitter buffer and the delay time for jitter buffer. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the mode of jitter buffer and the delay time for jitter buffer in the wired/wireless network. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-voice&q=load

To configure Jitter Buffer in the wired network via web user interface:

- Click on **Settings->Voice**.
- Select the desired radio box in the **Type** field.
- Enter the minimum delay time for adaptive jitter buffer in the **Min Delay** field.
  Valid value ranges from 0 to 400.
- Enter the maximum delay time for adaptive jitter buffer in the **Max Delay** field.
  Valid value ranges from 0 to 400.
- Enter the fixed delay time for fixed jitter buffer in the **Normal** field.
  Valid value ranges from 0 to 400.



- Click **Confirm** to save the change.

CHAPTER 12

Configuring Video Features

The SPARSH VP710 supports transmission and reception of high quality video images. The video is compatible with:

RFC 3984 - RTP Payload Format for H.264 Video
RFC 7741 - RTP Payload Format for VP8 Video

This chapter provides information for making configuration changes for the following video-related features:

- Video Settings
- Video Codecs

Video Settings

The IP phone supports USB camera for point-to-point video calls. You can place and answer video calls.

The IP phone supports transmission and reception of high quality video images. You can adjust the camera flicker frequency according to the power source. The light is powered by 50Hz or 60Hz.

Toggle Between Audio-only or Video Calls

When you disable the video call feature, the calls are audio–only.By default, this feature is enabled.

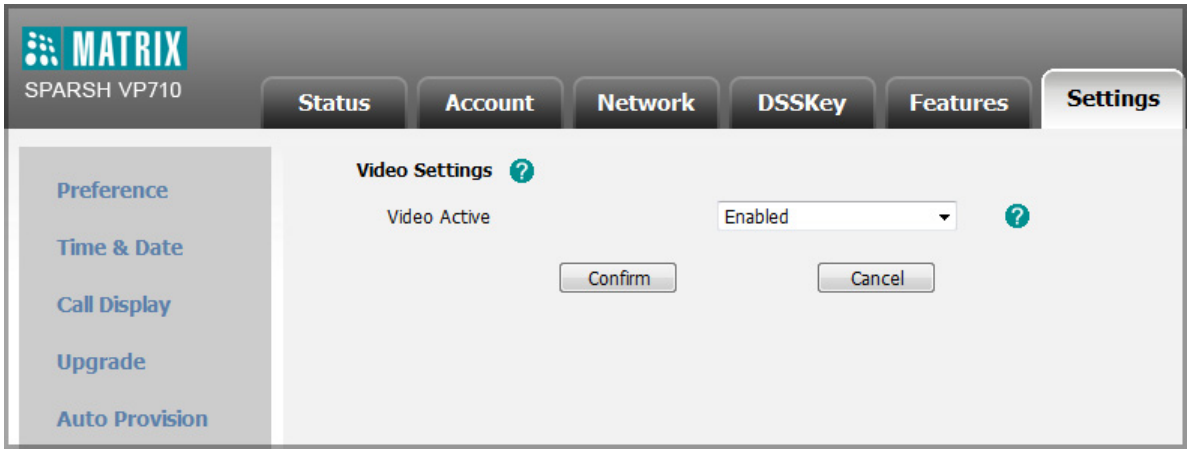
Procedure

Video settings can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure video settings. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the video settings. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-camera&q=load

To active the call video feature and configure the auto video feature via web user interface:

- Click on **Settings->Video.**
- Select the desired value from the **Video Active** list.



- Click **Confirm** to save the change.

Video Codecs

CODEC is an abbreviation of COmTap-DEComTap, capable of coding or decoding a digital data stream or signal by implementing an algorithm. The objective of the algorithm is to represent the high-fidelity video signal with minimum number of bits while retaining the quality.This effectively reduces the frame size and the bandwidth required for the video transmission.

The video codec that the IP phone uses to establish a call should be supported by the SIP server.When placing a call, the IP phone offers the enabled video codec list to the server and then use the video codec negotiated with the called party according to the priority.

RTPmap

Codecs and their priorities are configurable on a per-line basis. The attribute “rtpmap” is used to define a mapping from RTP payload codes to a codec, clock rate and other encoding parameters.

The following table lists the video codecs supported by IP phone:

Name	MIME Type	Bit Rate	Frame Rate	Frame Size
H.264 BP	H264/90000	90 kbps to 2048 kbps	5 fps to 30 fps	Tx: CIF, 360P, W448P, 720P Rx: Conventional Size Below 720P
H.264 HP	H264/90000			
VP8	VP8/90000	128kbps to 2048 kbps		

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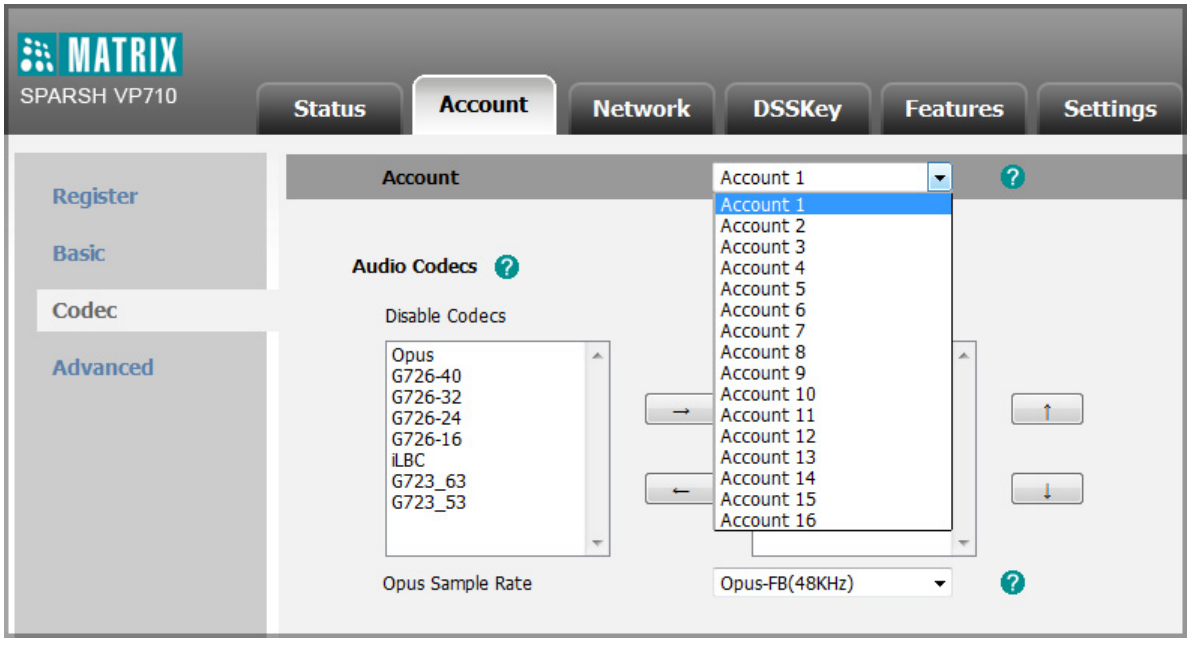
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the video codecs to use on a per-line basis. Configure the priority and rtpmap for the enabled video codec. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the video codecs to use on a per-line basis. Configure the priority for the enabled video codec. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-codec&q=load&acc=0

To configure the video codecs and adjust the priority of the enabled video codecs on a per-account basis via web user interface:

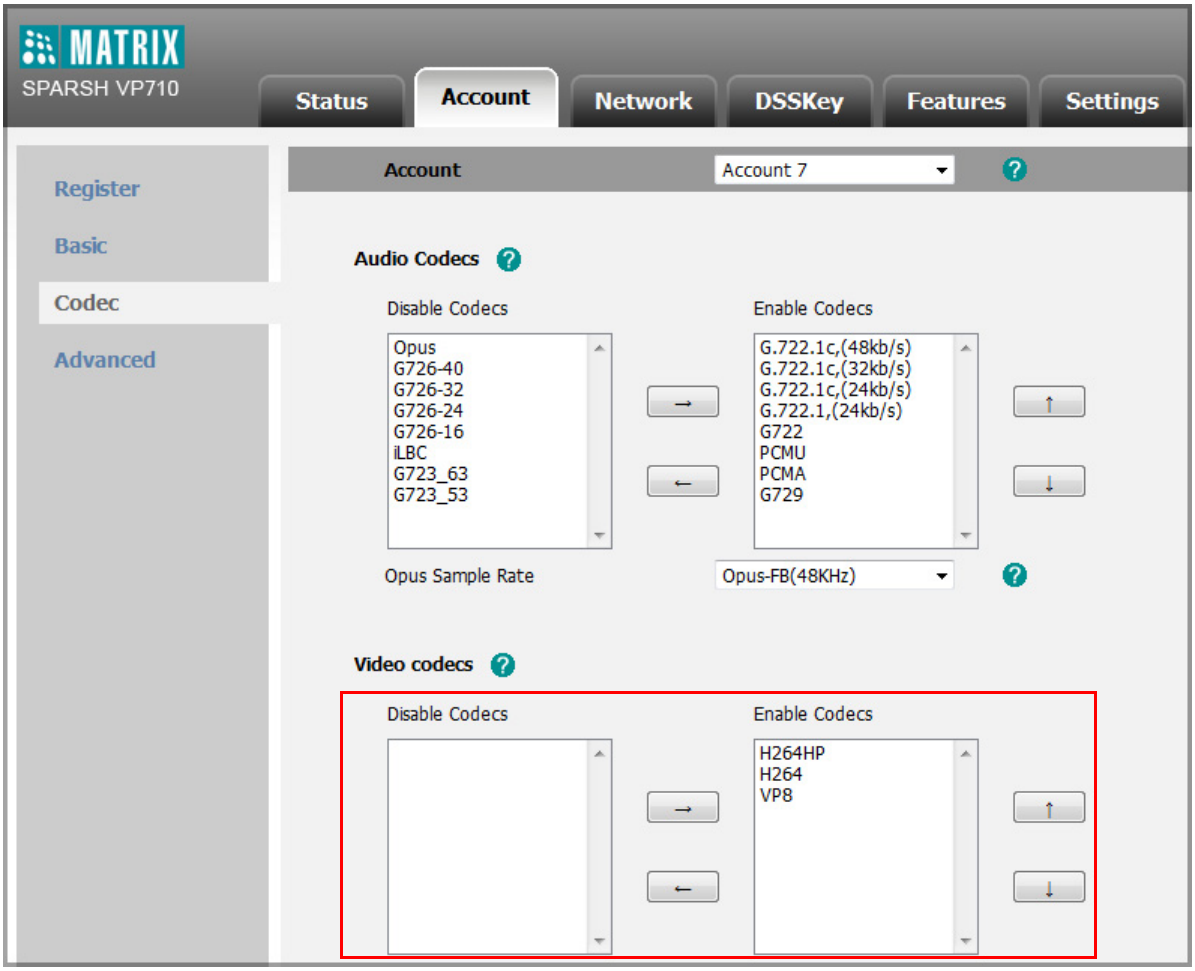
- Click on **Account->Codec**.
- Select the desired account from the **Account** list.



- In the **Video Codecs** field, select the desired codec from the **Disable Codecs** column and then click →

The selected codec appears in the **Enable Codecs** column.

- Repeat the above step to add more codecs to the **Enable Codecs** column.
- To remove the codec from the **Enable Codecs** column, select the desired codec and then click ←.
- To adjust the priority of codecs, select the desired codec and then click ↑ or ↓.



- .Click **Confirm** to save the change.

CHAPTER 13

Voice Quality Monitoring (VQM)

Voice quality monitoring allows the IP phone to generate various quality metrics for listening and conversational quality. These metrics can be sent between the phones in RTCP-XR packets. These metrics can also be sent in SIP PUBLISH messages to a central voice quality report collector.

Two mechanisms for voice quality monitoring supported by the IP phone:

- RTCP-XR
- VQ-RTCPXR

RTCP-XR

The RTCP-XR mechanism compliant with RFC 3611- RTP Control Extended Reports (RTCP XR) provides the metrics contained in RTCP-XR packets for monitoring the quality of the calls. These metrics include network packet loss, delay metrics, analog metrics and voice quality metrics.

Procedure

RTCP-XR can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure RTCP-XR. For more information, refer "Appendix D - Configuration Parameters".

VQ-RTCPXR

The VQ-RTCPXR mechanism compliant with RFC 6035 sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector.

Three types of quality reports can be enabled:

- **Session**: Generated at the end of a call.
- **Interval**: Generated during a call at a configurable period.
- **Alert**: Generated when the call quality degrades below a configurable threshold.

A wide range of performance metrics are generated in the following ways:

- Based on current values, such as jitter, jitter buffer max and round trip delay.
- Covers the time period from the beginning of the call until the report is sent such as network packet loss.
- Computed using other metrics as input, such as listening Mean Opinion Score (MOS-LQ) and conversational Mean Opinion Score (MOS-CQ).

To operate with central report collector, the IP phone must be configured to forward their voice quality reports to the specified report collector. The report collector can be specified on a per-line basis.You can check the voice quality data of the last call via web user interface or phone user interface.You can also specify the options of the RTP status to be displayed on the phone user interface where as it cannot be specified on the web user interface.

Procedure

VQ-RTCPXR can be performed using the configuration files or locally.

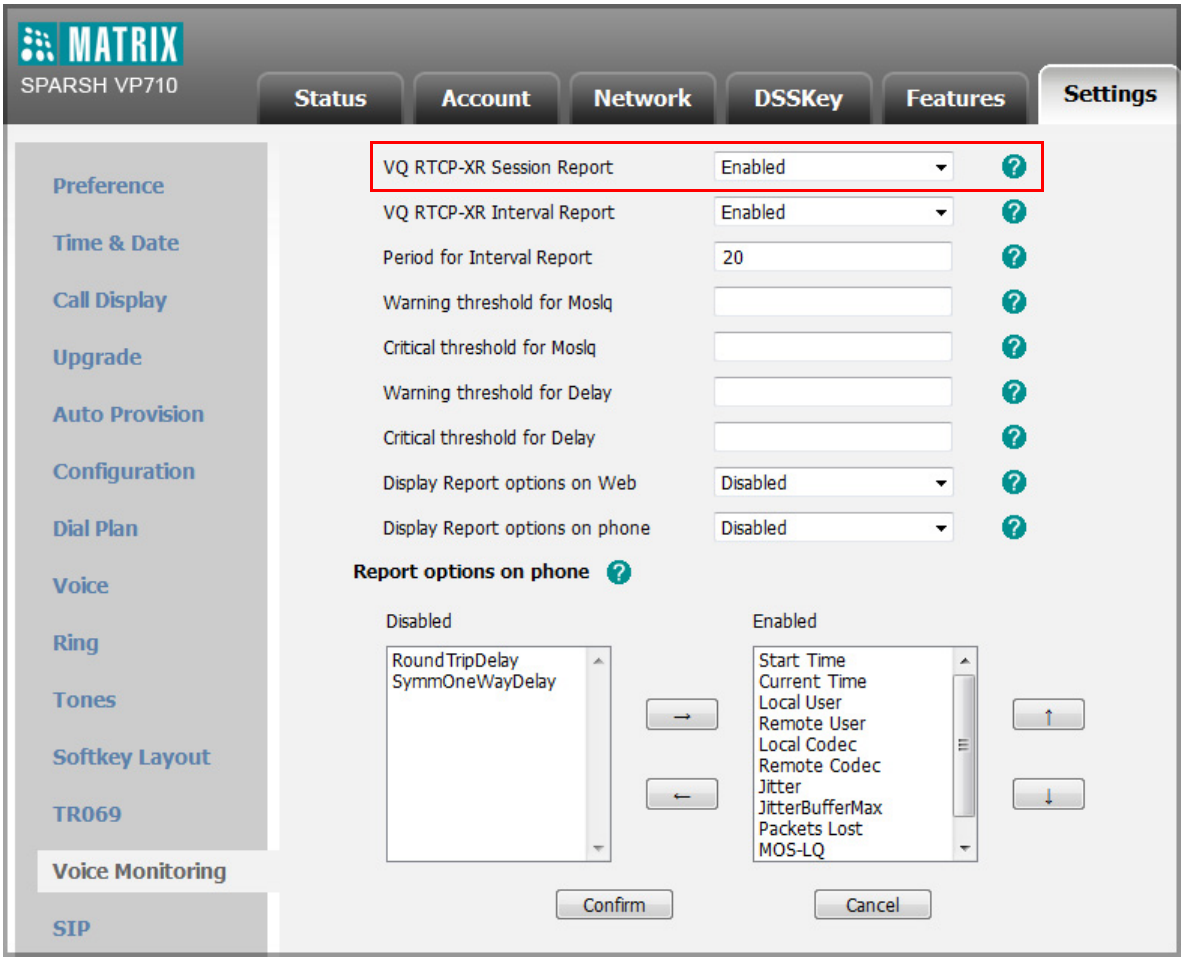
Configuration File	<MAC>.cfg	Configure the generation of session packets. Configure the generation of interval packets. Configure the generation of alert packets. Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface. Configure the phone to display RTP status showing the voice quality report of the last call or the current call on the phone user interface. Configure the options of the RTP status displayed on the phone user interface. Configure the central report collector. For more information, refer "Appendix D - Configuration Parameters".

	<MAC>.cfg	Configure the central report collector. For more information, refer "Appendix D - Configuration Parameters"
Local	Web User Interface	Configure VQ-RTCPXR. Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface. Configure the phone to display RTP status showing the voice quality report of the last call or the current call on the phone user interface. Configure the options of the RTP status displayed on the phone user interface. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-voicemonitoring&q=load Configure the central report collector. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure session report for VQ-RTCPXR via web user interface:

- Click on **Settings->Voice Monitoring**.

- Select the desired value from the **VQ RTCP-XR Session Report** list.



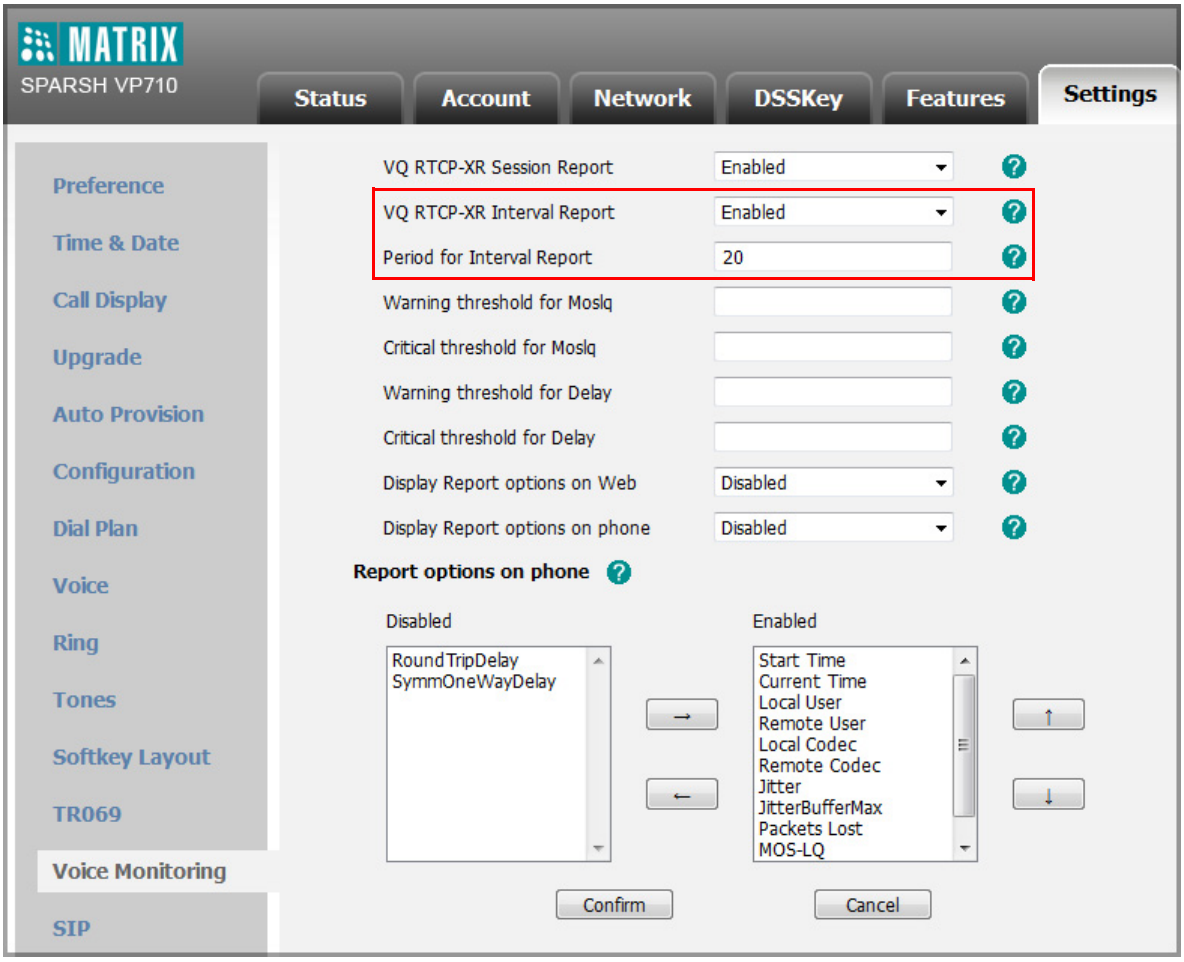
- Click **Confirm** to save the change.

.

To configure interval report for VQ-RTCPXR via web user interface:

- Click on **Settings->Voice Monitoring**.
- Select the desired value from the **VQ RTCP-XR Interval Report** list.

- Enter the desired value in the **Period for Interval Report** field.

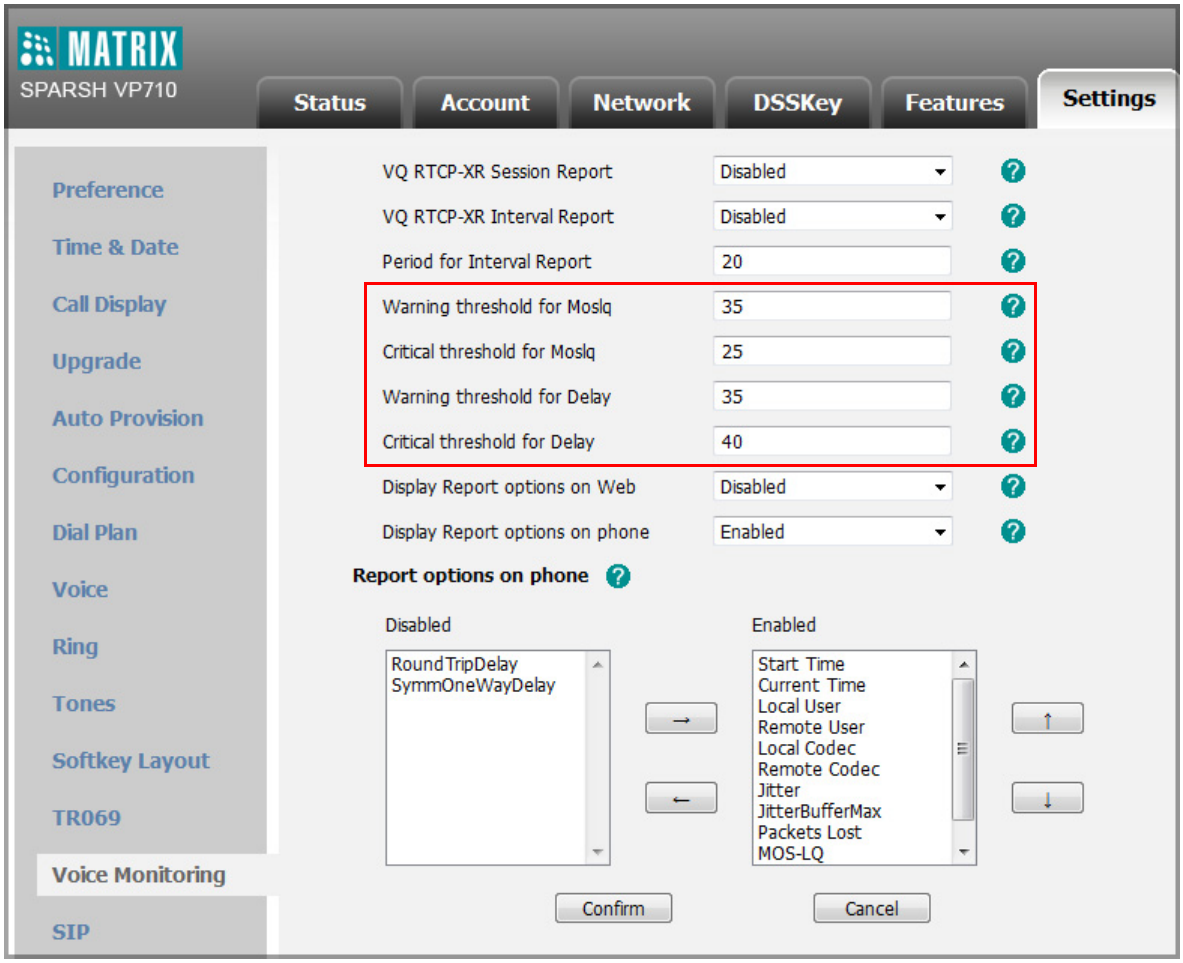


- .Click **Confirm** to save the change.

To configure alert report for VQ-RTCPXR via web user interface:

- Click on **Settings->Voice Monitoring.**
- Enter the desired value in the **Warning threshold for Moslq** field.
- Enter the desired value in the **Critical threshold for Moslq** field.
- Enter the desired value in the **Warning threshold for Delay** field.

- Enter the desired value in the **Critical threshold for Delay** field.

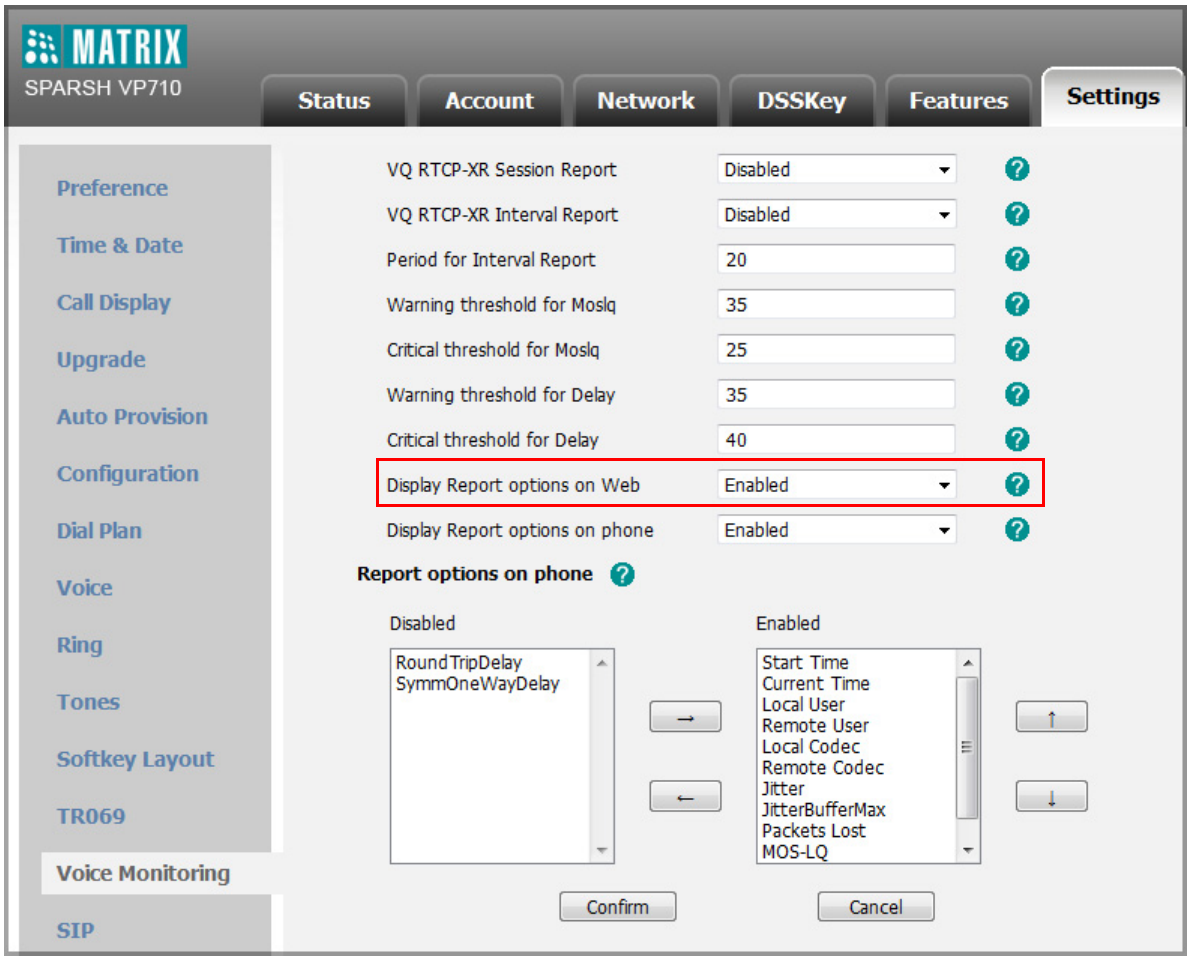


- Click **Confirm** to save the change.

To configure RTP status displayed on the web page via web user interface:

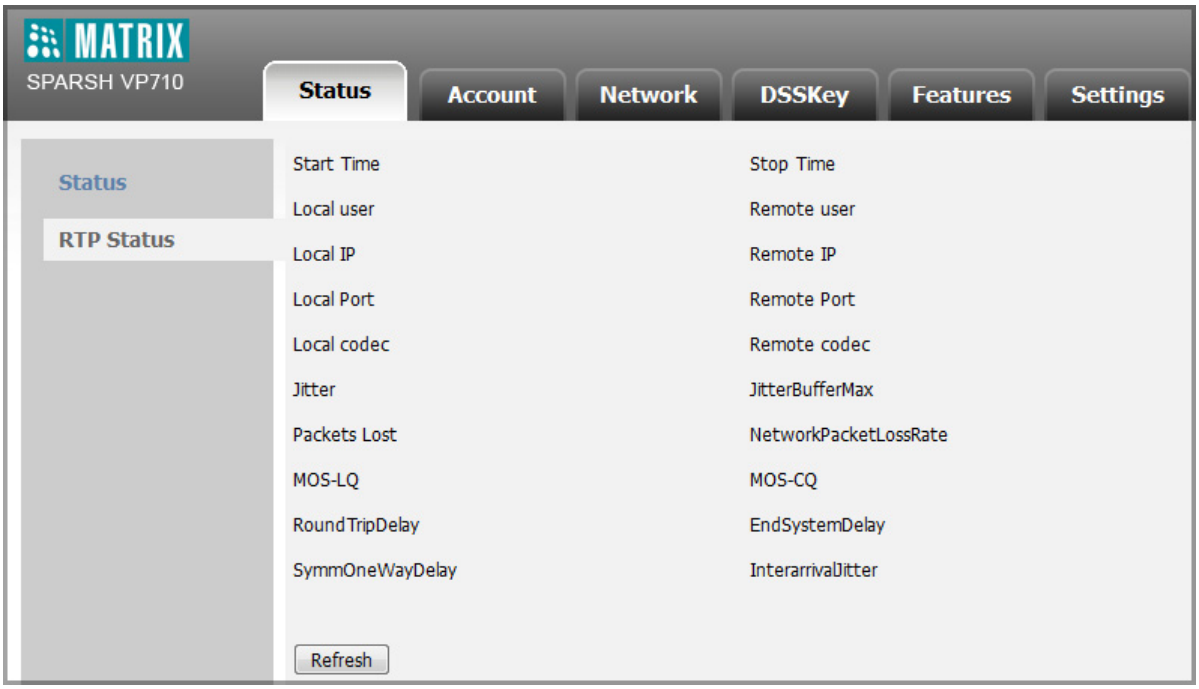
- Click on **Settings->Voice Monitoring**.

- Select the desired value from the **Display Report options on Web** list.



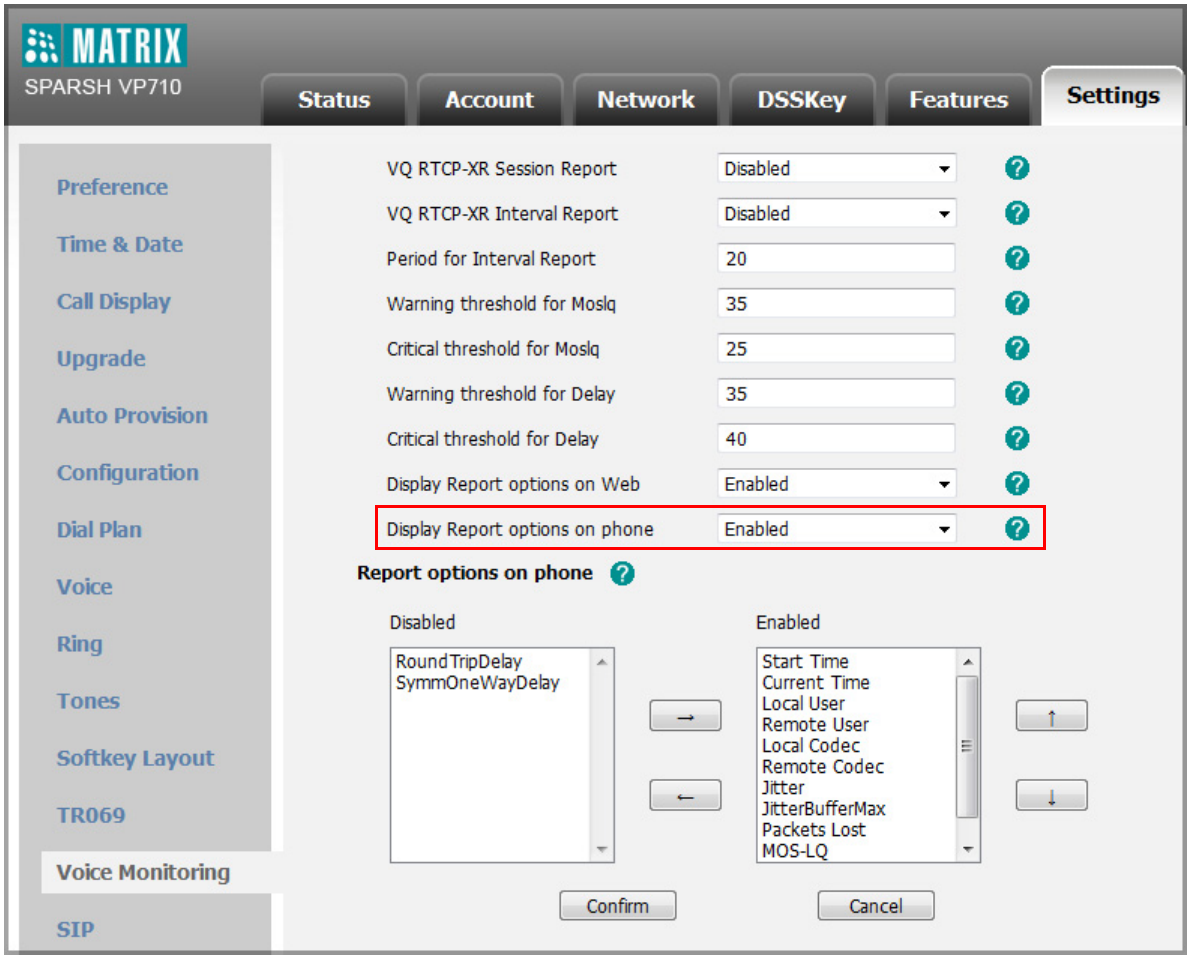
- Click **Confirm** to save the change.

  The RTP status will appear on the web user interface at the path: **Status->RTP Status**.



To configure RTP status displayed on the touch screen via web user interface:

- Click on **Settings->Voice Monitoring**.
- Select the desired value from the **Display Report options on phone** list.



- Click **Confirm** to save the change.

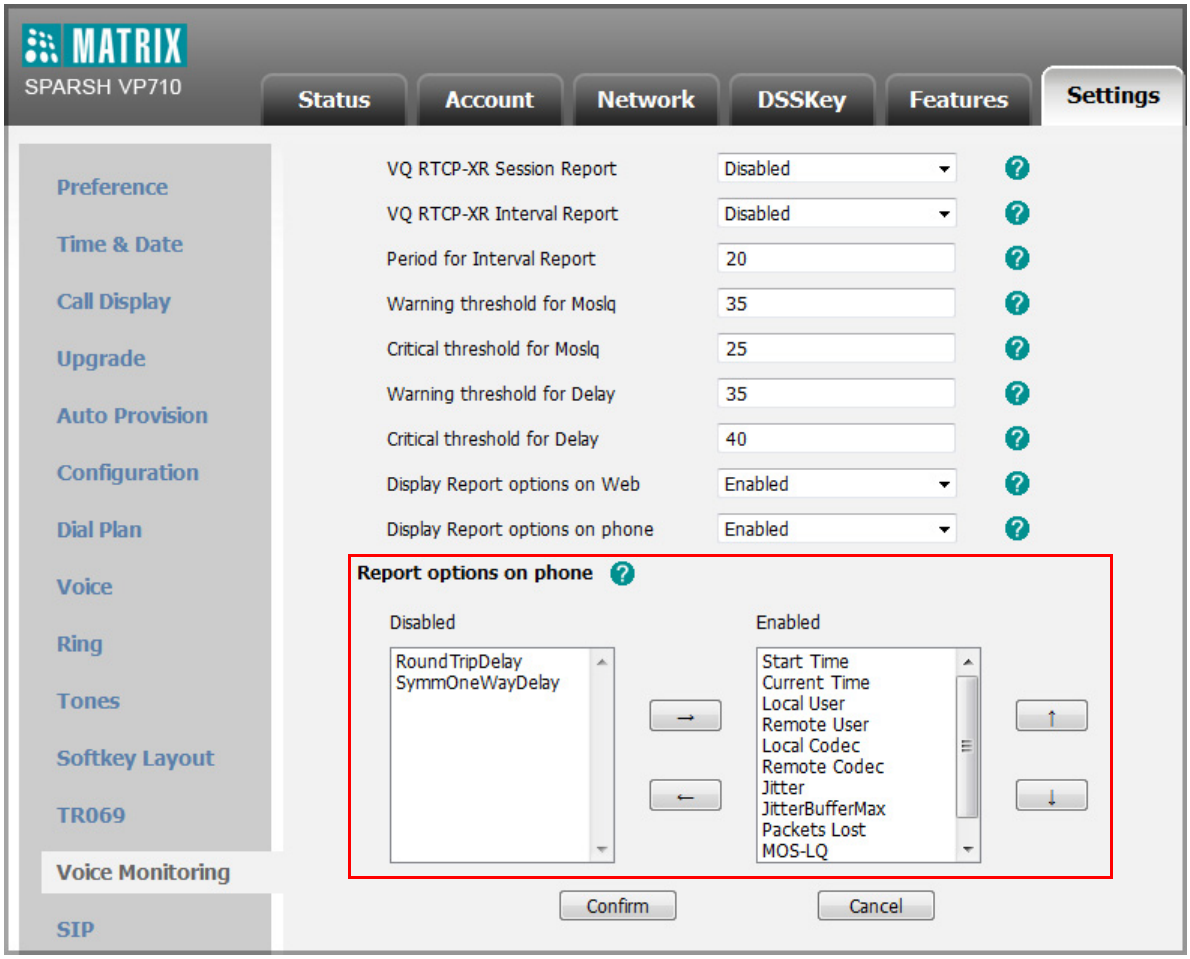
  The RTP status will appear on the phone user interface at the path: **Settings->Status->RTP Status.**

To configure the options of the RTP status displayed on the touch screen via web user interface:

- Click on **Settings->Voice Monitoring.**
- In the **Report options on phone** block, select the desired list from the **Disabled** column and then click

  .

The selected list appears in the **Enabled** column.



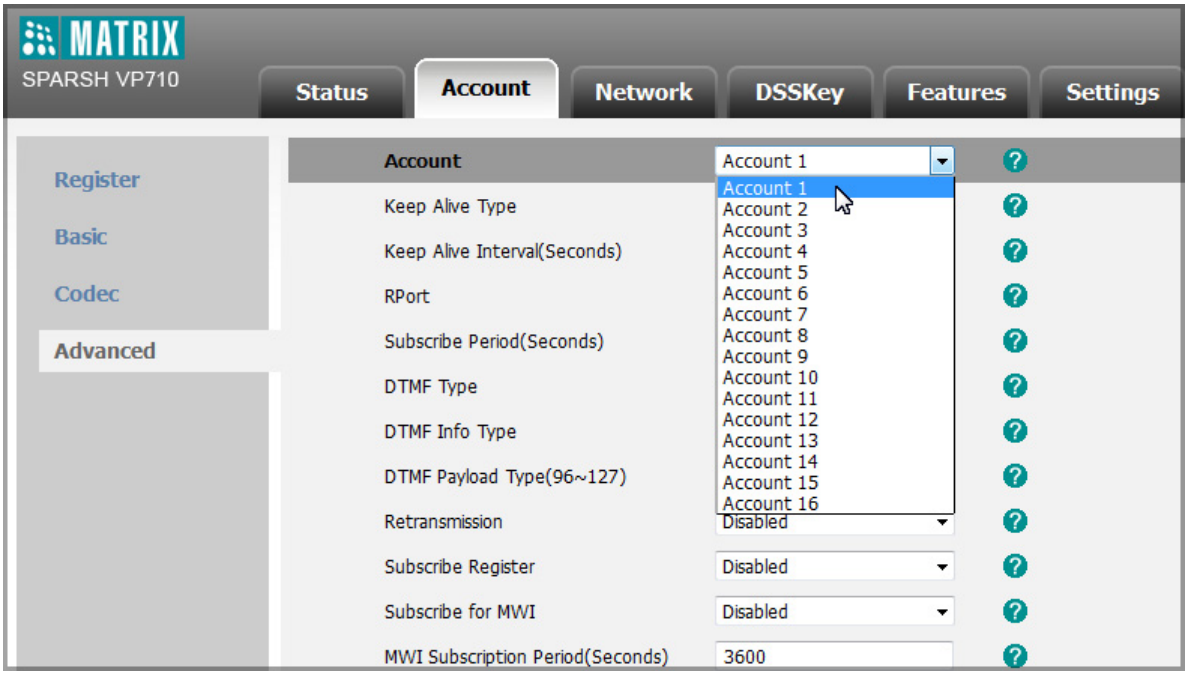
- Repeat the above step to add more items to the **Enabled** column.
- To remove an item from the **Enabled** column, select the desired item and then click ←.
- To adjust the display order of enabled items, select the desired item and then click ↑ or ↓.

  The touch screen will display the item(s) in the adjusted order.
- Click **Confirm** to save the change.

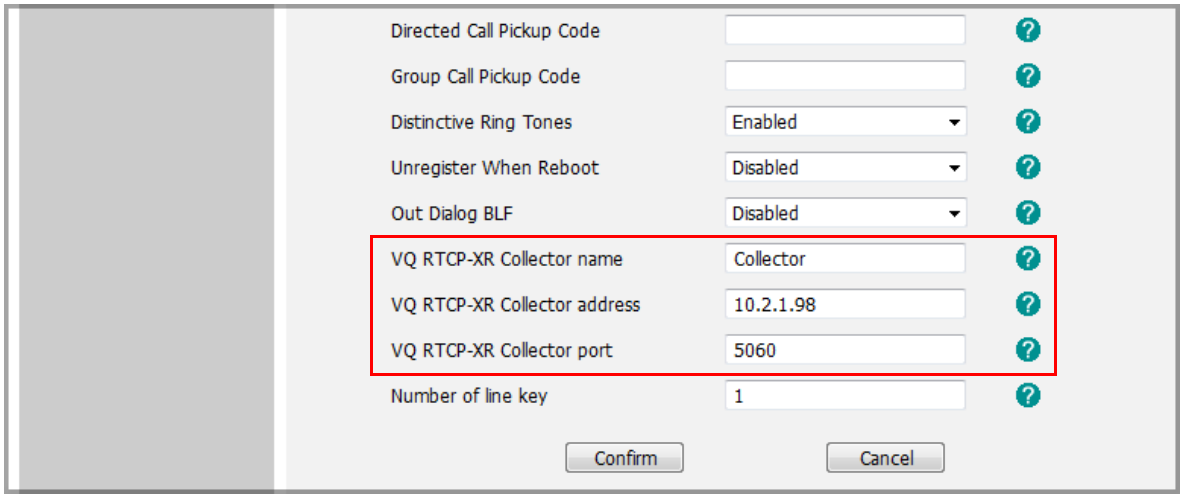
To configure the central report collector via web user interface:

- Click on **Account->Advanced.**

- Select the desired account from the **Account** list.



- Enter the host name of the central report collector in the **VQ RTCP-XR Collector name** field.
- Enter the IP address of the central report collector in the **VQ RTCP-XR Collector address** field.
- Enter the port of the central report collector in the **VQ RTCP-XR Collector port** field.



- Click **Confirm** to save the change.

CHAPTER 14

Configuring Security Features

This chapter provides information for making configuration changes for the following security-related features:

- Transport Layer Security
- Secure Real-Time Transport Protocol
- Encrypting Configuration Files
- "User Password" and "Administrator Password"
- "Auto-Logout Time"
- "Keypad Lock (Phone Lock)"

TLS

TLS is a commonly-used protocol for providing communications privacy and managing the security of message transmission, allowing IP phones to communicate with other remote parties and connect to the HTTPS URL for provisioning in a way that is designed to prevent eavesdropping and tampering.

TLS protocol is composed of two layers: TLS Record Protocol and TLS Handshake Protocol. The TLS Record Protocol completes the actual data transmission and ensures the integrity and privacy of the data. The TLS Handshake Protocol allows the server and client to authenticate each other and negotiate an encryption algorithm and cryptographic keys before data is exchanged.

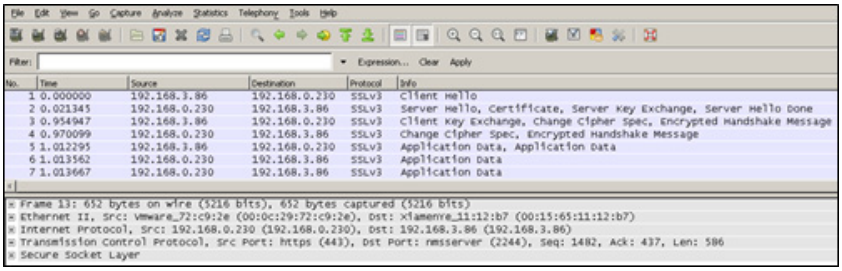
The TLS protocol uses asymmetric encryption for authentication of key exchange, symmetric encryption for confidentiality, and message authentication codes for integrity.

- **Symmetric encryption:** For symmetric encryption, the encryption key and the corresponding decryption key can be told by each other. In most cases, the encryption key is the same as the decryption key.
- **Asymmetric encryption:** For asymmetric encryption, each user has a pair of cryptographic keys – a public encryption key and a private decryption key. The information encrypted by the public key can only be decrypted by the corresponding private key and vice-versa. Usually, the receiver keeps its private key. The public key is known by the sender, so the sender sends the information encrypted by the known public key, and then the receiver uses the private key to decrypt it.

IP phones support TLS version 1.0. A cipher suite is a named combination of authentication, encryption, and message authentication code (MAC) algorithms used to negotiate the security settings for a network connection using the TLS/SSL network protocol. IP phones support the following cipher suites:

- DHE-RSA-AES256-SHA
- DHE-DSS-AES256-SHA
- AES256-SHA
- EDH-RSA-DES-CBC3-SHA
- EDH-DSS-DES-CBC3-SHA
- DES-CBC3-SHA
- DES-CBC3-MD5
- DHE-RSA-AES128-SHA
- DHE-DSS-AES128-SHA
- AES128-SHA
- RC2-CBC-MD5
- IDEA-CBC-SHA
- DHE-DSS-RC4-SHA
- RC4-SHA
- RC4-MD5
- RC4-64-MD5
- EXP1024-DHE-DSS-DES-CBC-SHA
- EXP1024-DES-CBC-SHA
- EDH-RSA-DES-CBC-SHA
- EDH-DSS-DES-CBC-SHA
- DES-CBC-SHA
- DES-CBC-MD5
- EXP1024-DHE-DSS-RC4-SHA
- EXP1024-RC4-SHA
- EXP1024-RC4-MD5
- EXP-EDH-RSA-DES-CBC-SHA
- EXP-EDH-DSS-DES-CBC-SHA
- EXP-DES-CBC-SHA
- EXP-RC2-CBC-MD5
- EXP-RC4-MD5

The following figure illustrates the TLS messages exchanged between the IP phone and TLS server to establish an encrypted communication channel:



Step1: IP phone sends “Client Hello” message proposing SSL options.

Step2: Server responds with “Server Hello” message selecting the SSL options, sends its public key information in “Server Key Exchange” message and concludes its part of the negotiation with “Server Hello Done” message.

Step3: IP phone sends session key information (encrypted by server’s public key) in the “Client Key Exchange” message.

Step4: Server sends “Change Cipher Spec” message to activate the negotiated options for all future messages it will send.
IP phone can encrypt SIP with TLS, which is called SIPS. When TLS is enabled for an account, the SIP message of this account will be encrypted, and a lock icon appears on the touch screen after the successful TLS negotiation.

Certificates

The IP phone can serve as a TLS client or a TLS server. The TLS requires the following security certificates to perform the TLS handshake:

- **Trusted Certificate:** When the IP phone requests a TLS connection with a server, the IP phone should verify the certificate sent by the server to decide whether it is trusted based on the trusted certificates list. The IP phone has 30 built-in trusted certificates. You can upload 10 custom certificates at most. The format of the trusted certificate files must be “*.pem”, “*.cer”, “*.crt” and “*.der” and the maximum file size is 5MB. For more information on 30 trusted certificates, refer “Appendix C - Trusted Certificates”.
- **Server Certificate:** When clients request a TLS connection with the IP phone, the IP phone sends the server certificate to the clients for authentication. The IP phone has two types of built-in server certificates: a unique server certificate and a generic server certificate. You can only upload one server certificate to the IP phone. The old server certificate will be overridden by the new one. The format of the server certificate files must be “*.pem” and “*.cer” and the maximum file size is 5MB.

The IP phone can authenticate the server certificate based on the trusted certificates list. The trusted certificates list and the server certificates list contain the default and custom certificates. You can specify the type of certificates the IP phone accepts: default certificates, custom certificates or all certificates.

Common Name Validation feature enables the IP phone to mandatorily validate the common name of the certificate sent by the connecting server.

- *In TLS feature, we use the terms trusted and server certificate. These are also known as CA and device certificates.*
- *Resetting the IP phone to factory defaults will delete custom certificates by default. But this feature is configurable using the configuration files. For more information on the configuration parameter, refer "Appendix D - Configuration Parameters".*

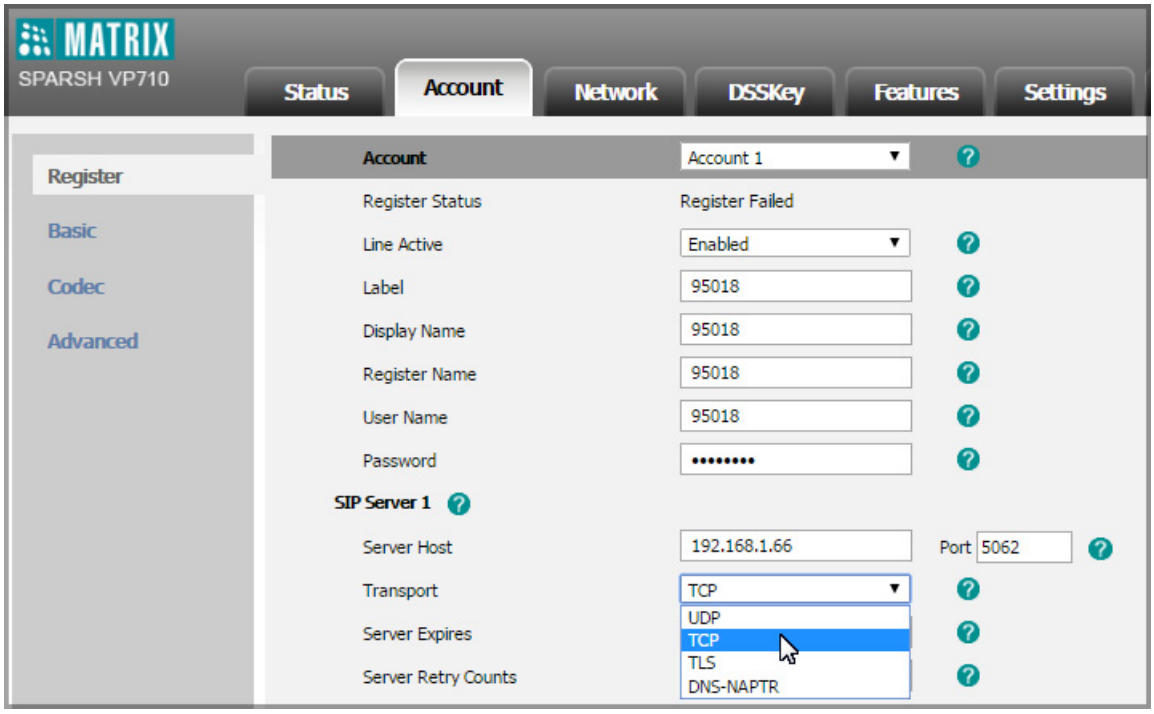
Procedure

Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure TLS on a per-line basis. Configure trusted certificates feature. Configure server certificates feature. Upload the trusted certificates. Upload the server certificates. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure TLS on a per-line basis. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-register&q=load& acc=0 Configure trusted certificates feature. Upload the trusted certificates. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=trusted-cert&q=load Configure server certificates feature. Upload the server certificates. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=server-cert&q=load

To configure TLS on a per-line basis via web user interface:

- Click on **Account->Register.**
- Select the desired account from the Account list.
- Select **TLS** from the **Transport** list.

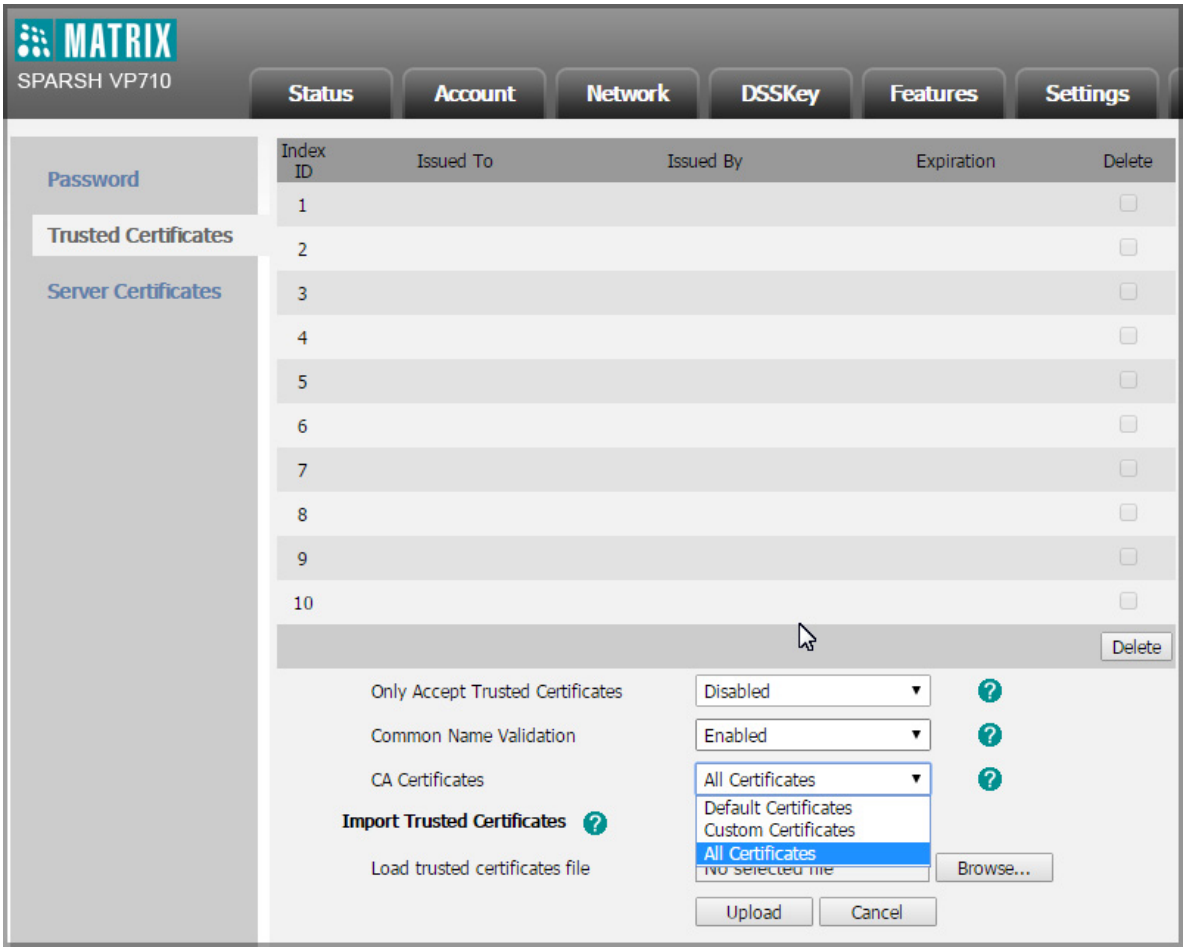


- Click **Confirm** to save the change.

To configure the trusted certificates via web user interface:

- Click on **Security->Trusted Certificates**.

- Select the desired values from the **Only Accept Trusted Certificates**, **Common Name Validation** and **CA Certificates** lists.

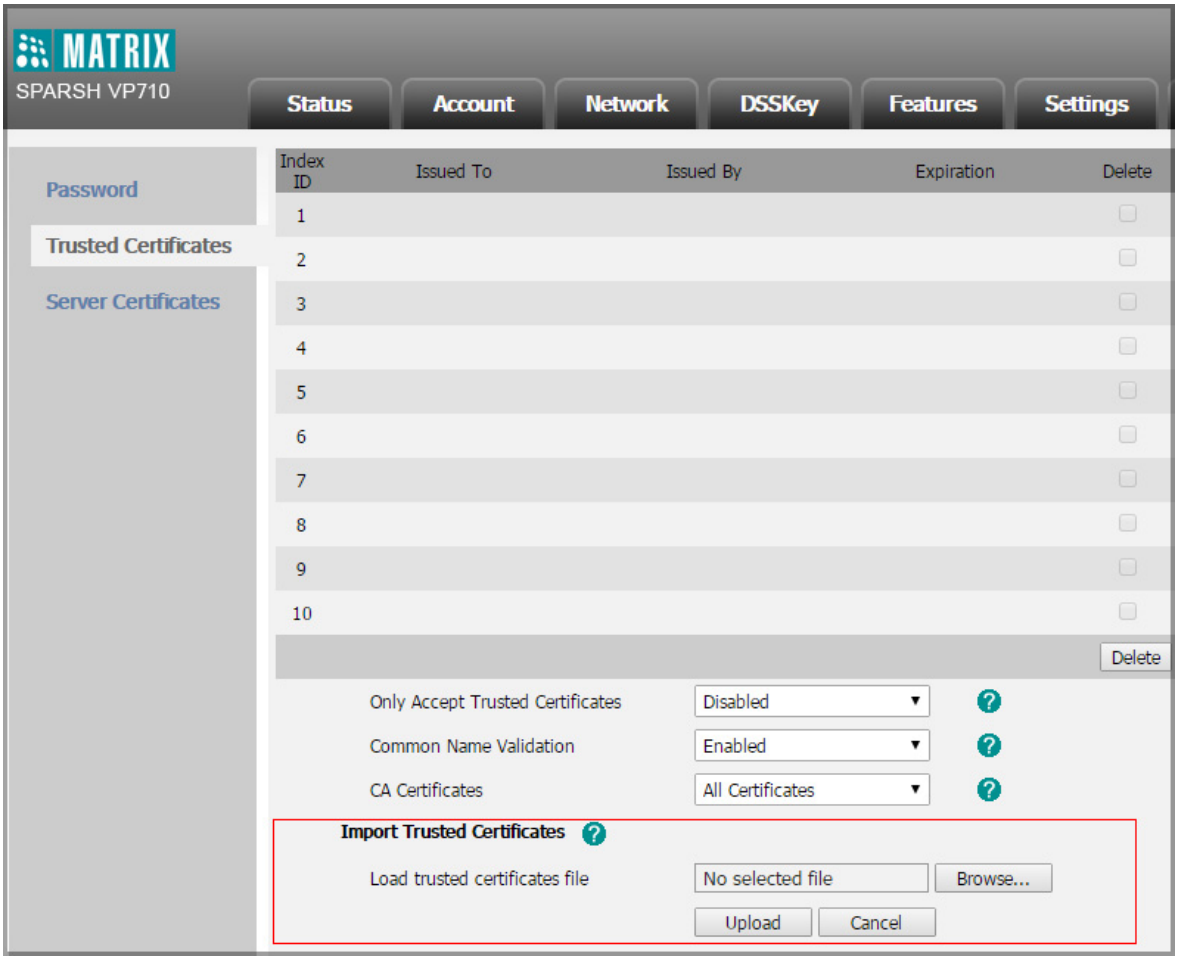


- Click **Confirm** to save the change.

To upload a trusted certificate via web user interface:

- Click on **Security->Trusted Certificates**.

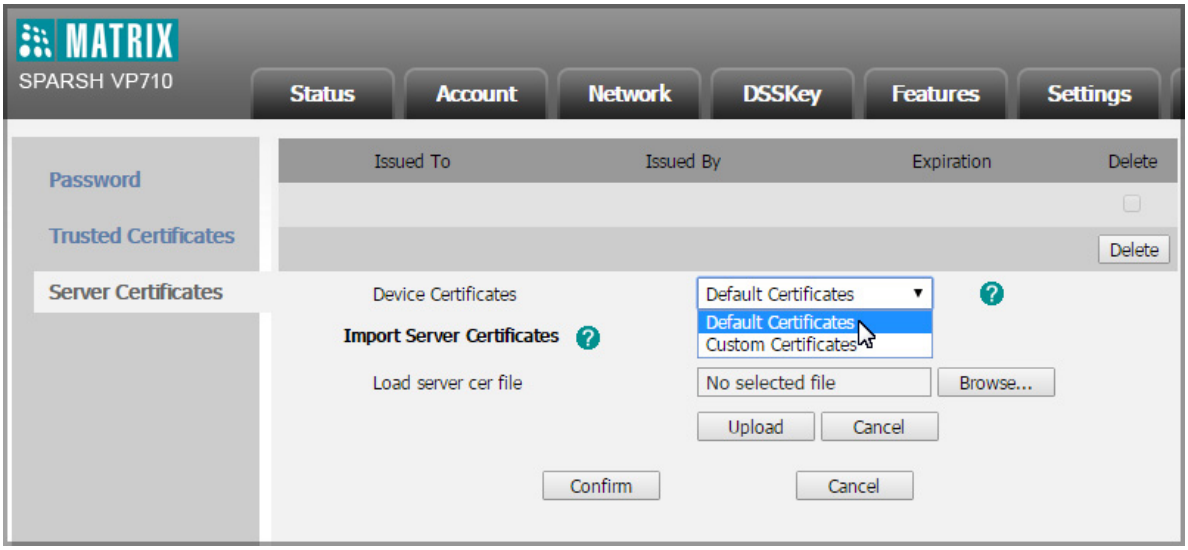
- Click **Browse** to select the certificate (*.pem, *.crt, *.cer or *.der) from your local system.



- Click **Upload** to upload the certificate.

To configure the server certificates via web user interface:

- Click on **Security->Server Certificates**.
- Select the desired value from the **Device Certificates** list.



- Click **Confirm** to save the change.

To upload a server certificate via web user interface:

- Click on **Security->Server Certificates**.
- Click **Browse** to select the certificate (*.pem and *.cer) from your local system.
- Click **Upload** to upload the certificate.

  A dialog box pops up to prompt “Success: The Server Certificate has been loaded! Rebooting, please wait…”.

SRTP

Secure Real-Time Transport Protocol (SRTP) encrypts the RTP streams during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call must enable SRTP feature simultaneously. When this feature is enabled on both phones, the type of encryption to utilize for the session is negotiated between the IP phones. This negotiation process is compliant with RFC 4568.

When a user places a call on the enabled SRTP phone, the IP phone sends an INVITE message with the RTP encryption algorithm to the destination phone.

Example of the RTP encryption algorithm carried in the SDP of the INVITE message:

```
m=audio 50020 RTP/AVP 118 9 0 8 18 101
a=rtpmap:118 opus/48000/2
a=fmtp:118 sprop-maxcapturerate=48000; maxaveragebitrate=40000;
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NWM5YmZkZDk5YjI1OTg2MDgwOTM5ZjIxNmJkNTY5
a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:N2M1NmMzZTI2MGFkZjY5YWYzMGE5MWViYTljNzg4
a=crypto:3 F8_128_HMAC_SHA1_80 inline:NjMxYmIzMDIxNGExM2JkYWQxZTMyODQ1MThkYzg0
m=video 50022 RTP/AVP 97 98 99 117
```

```
b=TIAS:2097152
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=64001f; packetization-mode=1
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42801f
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42801f; packetization-mode=1
a=rtpmap:117 YL-FPR/90000
a=fmtp:117 yl-capset=7;yl-ver=1;yl-ext=19
a=ptime:20
a=rtcp-fb:* ccm fir
a=sendrecv
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NDdmNGIxYjJiNjkwYmEwNDg4MDVlYmMyYTA0YThl
a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:IGE3NjY1MmZjYWEzOTQ4ZmU3ODEwZWQ3MWI3ZWYx
a=crypto:3 F8_128_HMAC_SHA1_80 inline:ZjlmZjg3Yjk5MmRkYTBjNDI0ZDRlNzFjZTc2MGIx
```

The callee receives the INVITE message with the RTP encryption algorithm, and then answers the call by responding with a 200 OK message which carries the negotiated RTP encryption algorithm.

Example of the RTP encryption algorithm carried in the SDP of the 200 OK message:

```
m=audio 50068 RTP/SAVP 118 9 0 8 18 101
a=rtpmap:118 opus/48000/2
a=fmtp:118 sprop-maxcapturerate=48000; maxaveragebitrate=40000;
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:ZDgyNzVlNzZkODQ5OTZhYmY4N2ZlNTI0ZjIyYTRl
a=sendrecv
m=video 50070 RTP/SAVP 97 98 99 117
b=TIAS:2097000
a=rtpmap:97 H264/90000
```

:

```
a=fmtp:97 profile-level-id=64001f; packetization-mode=1
a=rtpmap:98 H264/90000
a=fmtp:98 profile-level-id=42801f
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=42801f; packetization-mode=1
a=rtpmap:117 YL-FPR/90000
a=fmtp:117 yl-capset=7;yl-ver=1;yl-ext=19
a=ptime:20
a=rtcp-fb:* ccm fir
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NzVjNDQ1NTQgY2YzYmQ2YzJhZjc0MzZmNmJiMzM1
a=sendrecv
```

SRTP is configurable on a per-line basis. When SRTP is enabled on both IP phones, RTP streams will be encrypted, and a lock icon appears on the touch screen of each IP phone after successful negotiation.

If you enable SRTP, then you should also enable TLS. This ensures the security of SRTP encryption. For more information on TLS, refer "TLS".

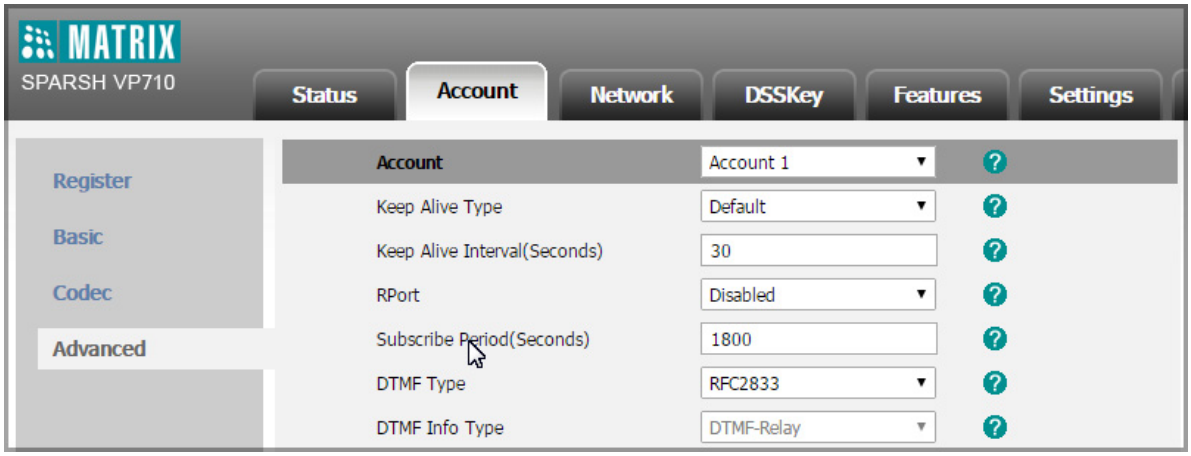
Procedure

SRTP can be configured using the configuration files or locally.

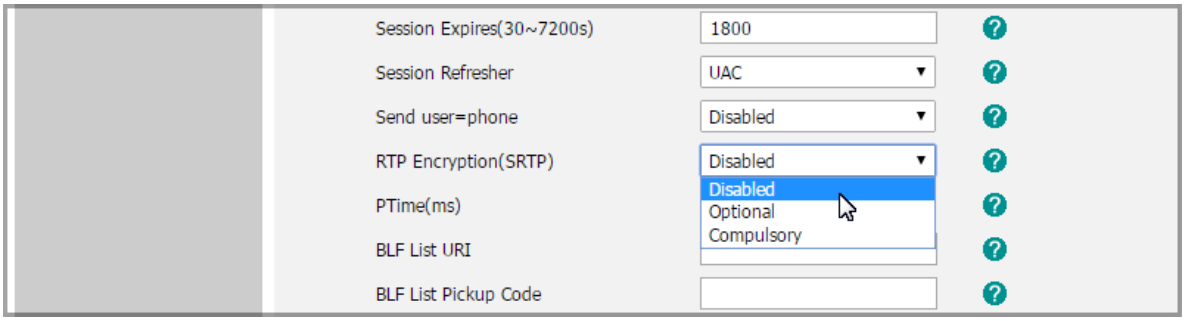
Configuration File	<MAC>.cfg	Configure SRTP feature on a per-line basis. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure SRTP feature on a per-line basis. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=account-adv&q=load&acc=0

To configure SRTP feature via web user interface:

- Click on **Account ->Advanced**.
- Select the desired account from **Account** list.



- Select the desired value from the **RTP Encryption(SRTP)** list.



- Click **Confirm** to save the change.

Encrypting Configuration Files

Encrypted configuration files can be downloaded from the provisioning server to protect against unauthorized access and tampering of sensitive information (e.g., login passwords or registration information).

If you are provisioning IP phones from a public server, you may encrypt your configuration files using the AES encryption method. It's very important that you encrypt the files with the same 16 character key as you have in the phone.

However, file names should be the same regardless of whether the file is encrypted or not.

A simple command line AES-encryption tool can be downloaded from the Public FTP server of Matrix. For details, contact Matrix Technical Support.

Procedure to Encrypt/Decrypt the Configuration Files (<MAC>.cfg)

- Unzip the AES encryption tool that you have downloaded.
- Keep the unzipped AES encryption tool (named as "**EncryptUtilityWindows.exe**") in your local drive; for example, in the 'D:' drive.
- Also keep the <MAC>.cfg file of Matrix SPARSH VP710 (for example, 001d29002794.cfg) in your local drive.
- Open the Command Prompt window and run the following commands:

  - To encrypt the <MAC>.cfg file, type-

    D:\>EncryptUtilityWindows.exe<space>**<AES Key>**<space>**E**<space>**<Destination File Path of MAC.cfg>**<space>**<Source File Path of MAC.cfg>**

    Or,

  - To decrypt the <MAC>.cfg file, type-

    D:\>EncryptUtilityWindows.exe<space>**<AES Key>**<space>**D**<space>**<Destination File Path of MAC.cfg>**<space>**<Source File Path of MAC.cfg>**

    Where,
    <AES Key> = It should be a 16 character key. Supported characters are 0 ~ 9, A ~ Z, a ~ z.
    E = Used to Encrypt
    D = Used to Decrypt
    <space> = Blank space



*Make sure that the ".exe" file is accessible from the path - **D:\>EncryptUtilityWindows.exe**. If it is kept at a different location, then specify the exact path in the Command Prompt window for successful encryption/ decryption of the <MAC>.cfg file.*

Example Case

For example, if you have kept the **EncryptUtilityWindows.exe** in your **"D:"** drive and the **001d29002794.cfg** file (<MAC>.cfg) in your **"F:"** drive, then:

To encrypt, run the following command,

D:\>EncryptUtilityWindows.exe 1234567890123456 E F:\001d29002794.cfg.en F:\001d29002794.cfg

On executing the above command, encrypted file **001d29002794.cfg.en** is created in the **"F:"** drive.

To decrypt, run the following command,

D:\>EncryptUtilityWindows.exe 1234567890123456 D F:\001d29002794.cfg.de F:\001d29002794.cfg.en

On executing the above command, decrypted file **001d29002794.cfg.de** is created in the **"F:"** drive.

Make sure to remove the ".en" suffix from the encrypted file before placing it on the provisioning server. This suffix is added just to identify that it is an encrypted file and distinguish it from the original <MAC>.cfg file.

For security reasons, you should upload the encrypted configuration file(s), <MAC>.cfg file (after removing the ".en" suffix) to the root directory of the provisioning server. During auto provisioning, the IP phone takes care of resolving the configuration files and updates configuration settings onto the system.

Procedure

Decryption method can be configured using the configuration files.

Configuration File	<MAC>.cfg	Configure whether to only download and resolve the encrypted files. Configure the decryption method. Configure AES keys. Specify if the MAC-local CFG file is encrypted when it is uploaded from the phone to the server For more information, refer "Appendix D - Configuration Parameters".

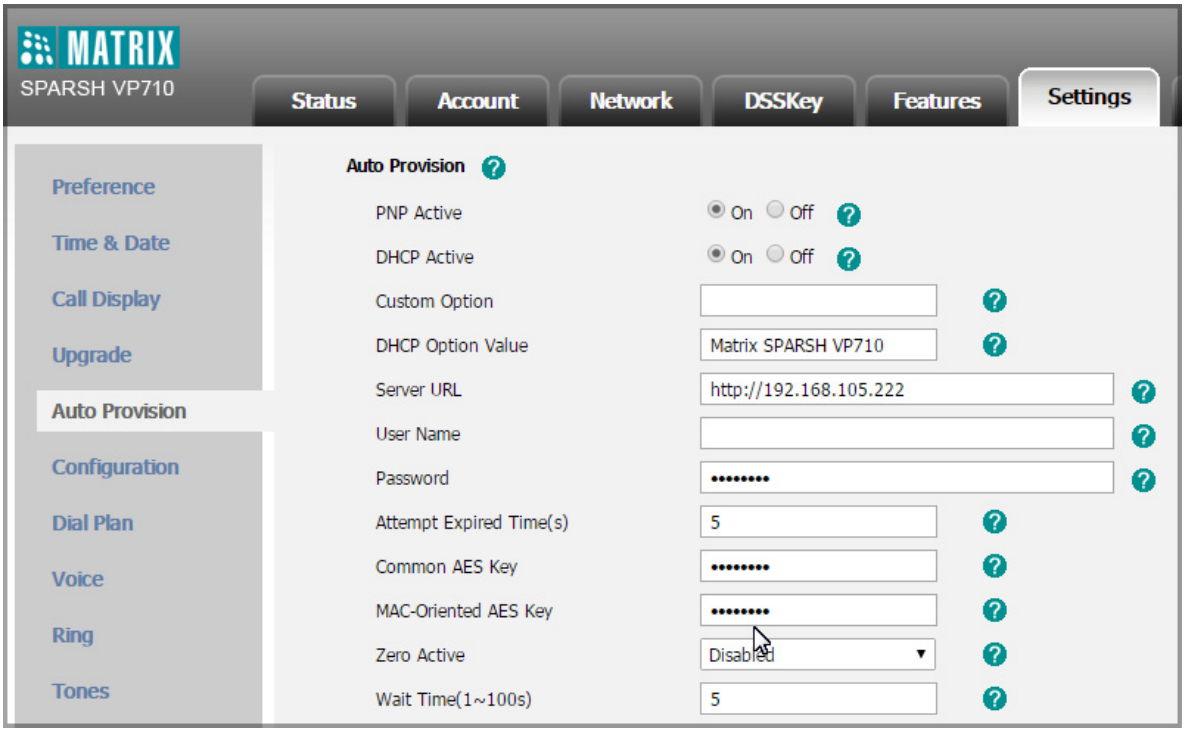
Local	Web User Interface	Configure AES keys. **Navigate to**: http://<phoneIPAddress>/servlet?p=settings-autop&q=load
	Phone User Interface	Configure AES keys.

To configure AES keys via web user interface:

- Click on **Settings->Auto Provision**.
- Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.

  AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z and the following special characters are also supported: # $ % * + , - . : = ? @ [ ] ^ _ { } ~.

***MAC-Oriented AES Key** must be same as the **<AES Key>** that you used to encrypt/decrypt the **<MAC>.cfg** file. For details, refer "Procedure to Encrypt/Decrypt the Configuration Files (<MAC>.cfg)" described above.*



- Click **Confirm** to save the change.

CHAPTER 15

Maintenance

This chapter describes the following features in details:

- Upgrading Firmware
- TR-069 Device Management
- Advanced Maintenance

Upgrading Firmware

This section provides information on upgrading the IP phone firmware. Two methods of firmware upgrade:

- Manually, from the local system for a single phone.
- Automatically, from the provisioning server for a mass of phones.

The following table lists the associated and latest firmware name for the IP phone model (X is replaced by the actual firmware version).

IP Phone Model	Associated Firmware Name	Firmware Name Example
SPARSH VP710	58.x.x.x.rom	58.80.0.10.rom

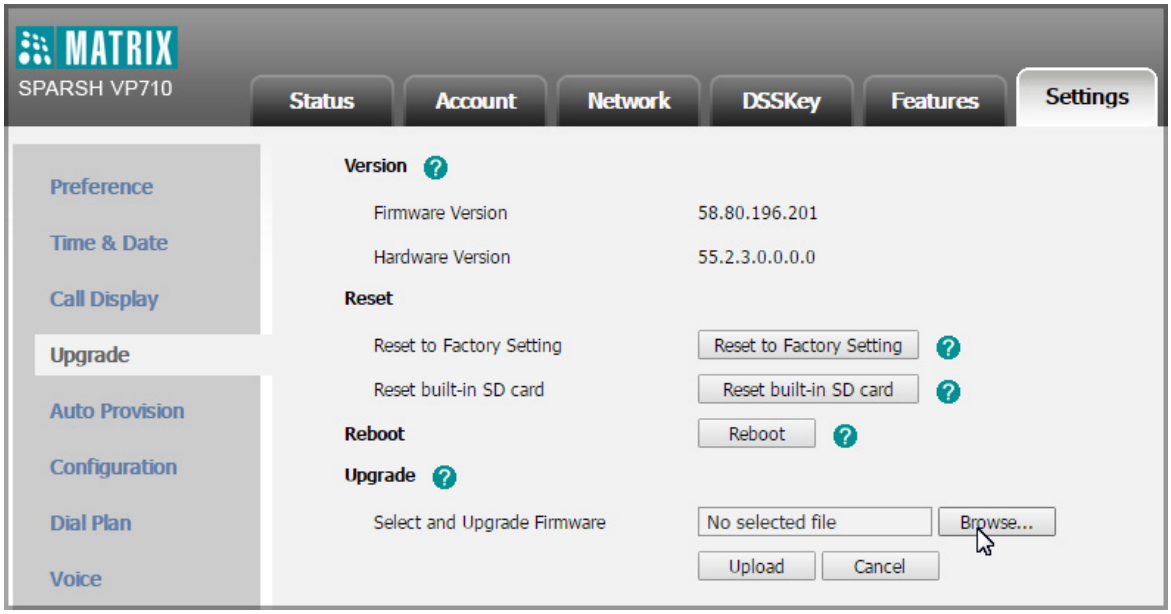
Upgrade via Web User Interface

To manually upgrade firmware via web user interface, you need to store firmware to your local system in advance.

To upgrade firmware manually via web user interface:

- Click on **Settings->Upgrade**.

- Click **Browse**.



- Select firmware from the local system.
- Click **Upload**.

  A dialog box pops up to prompt “Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!”

- Click **OK** to confirm the upgrade.

Do not close and refresh the browser when the IP phone is upgrading the firmware via web user interface.

Upgrade Firmware from the Provisioning Server

IP phones support using FTP, TFTP, HTTP and HTTPS protocols to download configuration files and firmware from the provisioning server, and then upgrade firmware automatically.

IP phones can download firmware stored on the provisioning server in one of two ways:

- Check for configuration files and then download firmware during start-up.
- Automatically check for configuration files and then download firmware at a fixed interval or specific time.

Method of checking for configuration files is configurable.

Procedure

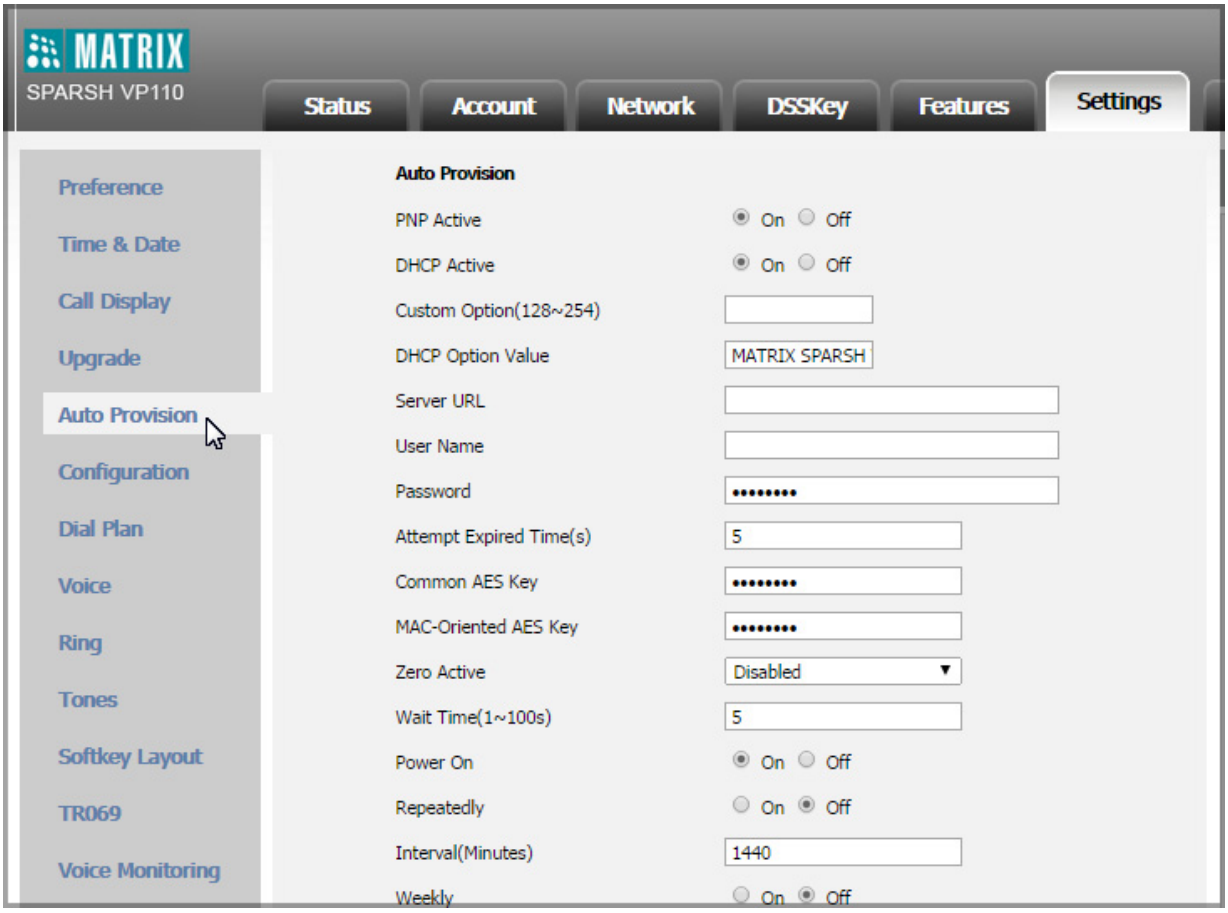
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the way for the IP phone to check for configuration files. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the way for the IP phone to check for configuration files. Upgrade firmware. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-autop&q=load

To configure the way for the IP phone to check for configuration files via web user interface:

- Click on **Settings->Auto Provision**.

- Make the desired change(s).



- Click **Confirm** to save the change.

  When the “Power On” is set to On, the IP phone will check configuration files stored on the provisioning server during start-up and then it will download firmware from the server.

TR-069 Device Management

TR-069 is a technical specification defined by the Broadband Forum, which defines a mechanism that encompasses secure auto-configuration of a CPE (Customer-Premises Equipment), and incorporates other CPE management functions into a common framework. TR-069 uses common transport mechanisms (HTTP and HTTPS) for communication between CPE and ACS (Auto Configuration Servers). The HTTP(S) messages contain XML-RPC methods defined in the standard for configuration and management of the CPE.

TR-069 is intended to support a variety of functionalities to manage a collection of CPEs, including the following primary capabilities:

- Auto-configuration and dynamic service provisioning
- Software or firmware image management
- Status and performance monitoring
- Diagnostics

The following table provides a description of RPC methods supported by IP phones.

RPC Method	Description
GetRPCMethods	This method is used to discover the set of methods supported by the CPE.
SetParameterValues	This method is used to modify the value of one or more CPE parameters.
GetParameterValues	This method is used to obtain the value of one or more CPE parameters.
GetParameterNames	This method is used to discover the parameters accessible on a particular CPE.
GetParameterAttributes	This method is used to read the attributes associated with one or more CPE parameters.
SetParameterAttributes	This method is used to modify attributes associated with one or more CPE parameters.
Reboot	This method causes the CPE to reboot.
Download	This method is used to cause the CPE to download a specified file from the designated location. File types supported by IP phones are: • Firmware Image • Configuration File
Upload	This method is used to cause the CPE to upload a specified file to the designated location. File types supported by IP phones are: • Configuration File • Log File
ScheduleInform	This method is used to request the CPE to schedule a one-time Inform method call (separate from its periodic Inform method calls) sometime in the future.
FactoryReset	This method resets the CPE to its factory default state.

TransferComplete	This method informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.
AddObject	This method is used to add a new instance of an object defined on the CPE.
DeleteObject	This method is used to remove a particular instance of an object.

Procedure

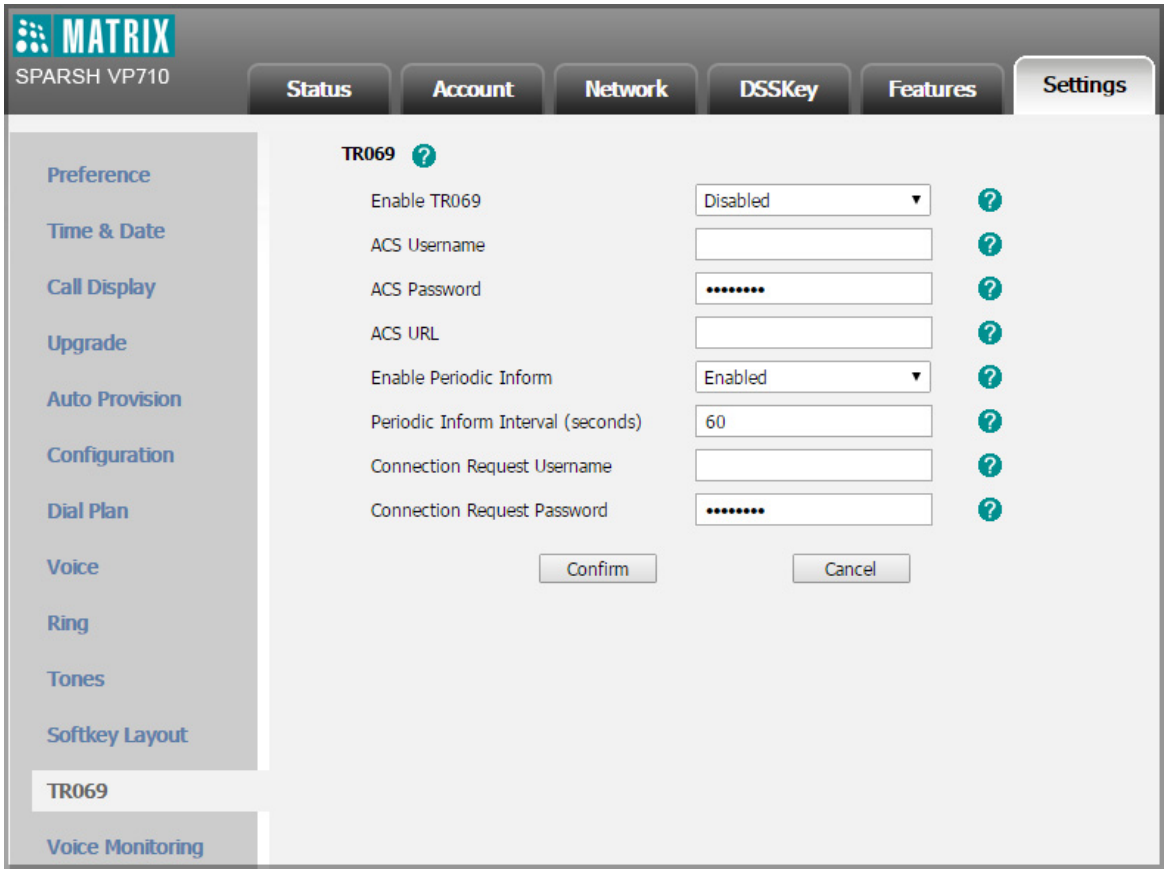
TR-069 can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure TR-069 feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure TR-069 feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-tr069&q=load

To configure TR-069 via web user interface:

- Click on **Settings->TR069**.
- Select **Enabled** from the **Enable TR069** list.
- Enter the username and password authenticated by the ACS in the **ACS Username** and **ACS Password** fields.
- Enter the URL of the ACS in the **ACS URL** field.
- Select the desired value from the **Enable Periodic Inform** list.
- Enter the desired time in the **Periodic Inform Interval (seconds)** field.

- Enter the username and password authenticated by the IP phone in the **Connection Request Username** and **Connection Request Password** fields.



- Click **Confirm** to save the change.

Advanced Maintenance

This chapter provides general information for maintenance and troubleshooting of some common problems using advanced features incorporated in the IP phone.

The IP phone can provide feedback in a variety of forms such as log files, packets, status indicators and so on, which can help you to easily find the system problem and fix it.

The following are helpful for better understanding and resolving the working status of the IP phone.

- Viewing Log Files
- Capturing Packets
- Enabling Watch Dog Feature
- Getting Information from Status Indicators



For more information about common problems and how to resolve them, refer "Troubleshooting".

Viewing Log Files

If your IP phone encounters some problems, commonly the log files are needed. You can export the log files to a syslog server or the local system.You can also specify the severity level of the log to be reported to a log file. The default system log level is 3 (Changes to this parameter via web user interface requires a reboot).

In the configuration files, you can use the following parameters to configure system log settings:

- static.syslog.log_level -- Specify the system log level. The following lists the log level of events you can log:
  - 0: system is unusable
  - 1: action must be taken immediately
  - 2: critical condition
  - 3: error conditions
  - 4: warning conditions
  - 5: normal but significant condition
  - 6: informational
- **static.syslog.mode** -- Specify the system log to be exported to the provisioning server, syslog server or local system.
- **static.syslog.server** -- Specify the IP address or domain name of the syslog server to which the log will be exported.
- **static.syslog.log_upload_period** -- Specify the period (in seconds) of the log uploads to the provisioning server.
- **static.syslog.ftp.post_mode** -- Specify whether the log files on the provisioning server are overwritten or appended.
- **static.syslog.ftp.max_logfile** -- Specify the maximum size of the log files can be stored on the provisioning server.

- **static.syslog.ftp.append_limit_mode** -- Specify the behavior when log file on the provisioning server reaches the max size.
- **static.syslog.bootlog_upload_wait_time** -- Specify the waiting time before the phone uploads the log file to the provisioning server.
- **static.auto_provision.server.url** -- Specify the access URL of the syslog server or provisioning server.

Configuring the Severity Level of the Log

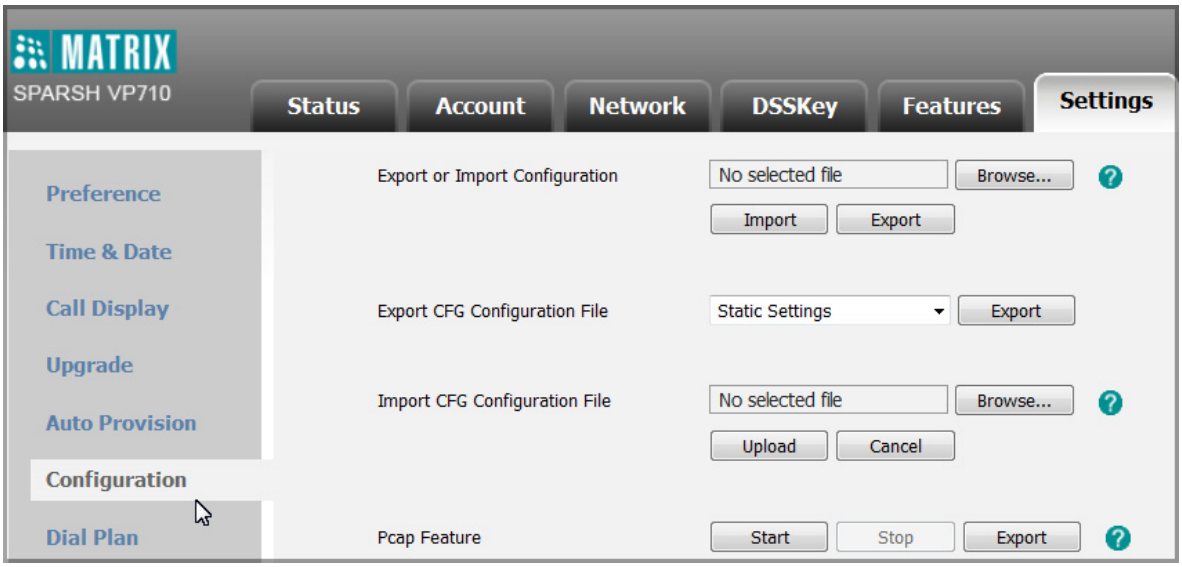
Procedure

Severity level can be configured using the configuration files or locally.

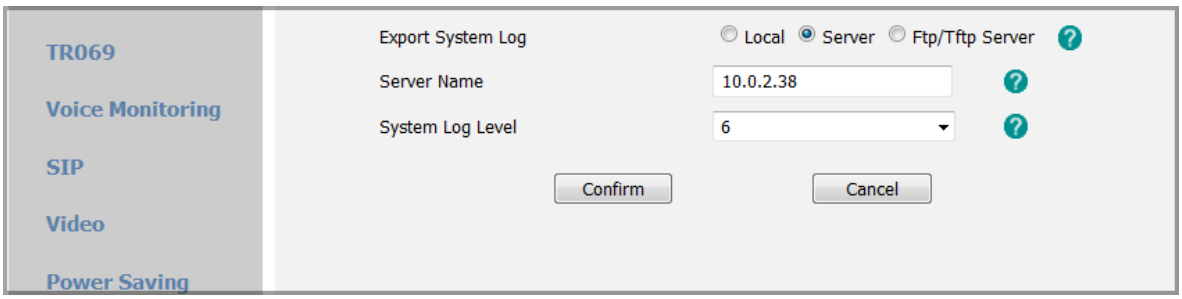
Configuration File	<MAC>.cfg	Configure the severity level of the logs to be reported to a log file. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the severity level of the logs to be reported to a log file. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-config&q=load

To configure the level of the system log via web user interface:

- Click on **Settings->Configuration**.



- Select the desired level from the **System Log Level** list.



- Click **Confirm** to save the change.

  The system log level is set as 6, the informational level.

Informational level may make some sensitive information accessible (e.g., password), we recommend that you reset the system log level to 3 after providing the syslog file.

Exporting the Log File to the Local System

Procedure

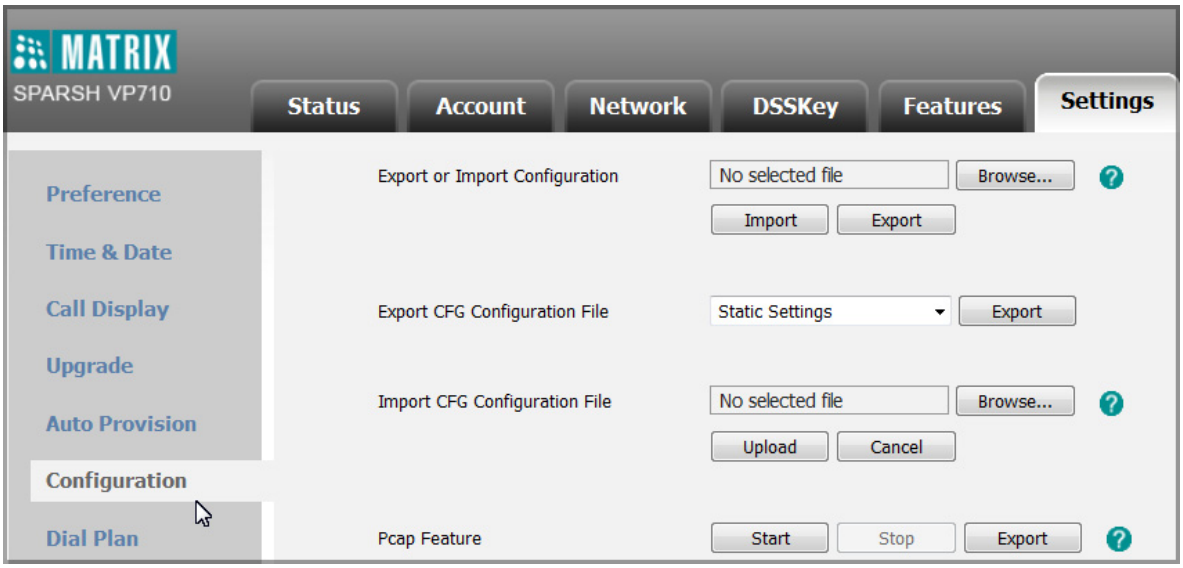
Log setting can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the syslog mode. For more information, refer "Appendix D - Configuration Parameters".

Local	Web User Interface	Configure the syslog mode. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-config&q=load

To export a log file to the local system via web user interface:

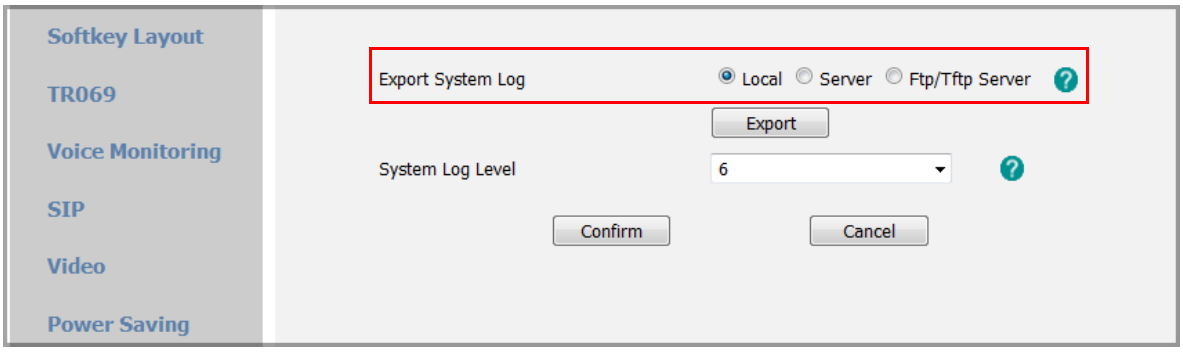
- Click on **Settings->Configuration**.



- Select the **Local** radio box in the **Export System Log**.

  A dialog box pops up to prompt “Warning: Some settings you changed take effect when you restart your machine! Do you want to reboot now?”. The configuration will take effect after a reboot.

- Click **OK** to reboot the phone.
- Reproduce the issue (e.g., account registration).
- Click **Export** to open file download window, and then save the file to your local system.



- A log file named **syslog.tar** is successfully exported to your local system.

Exporting the Log File to a Syslog Server

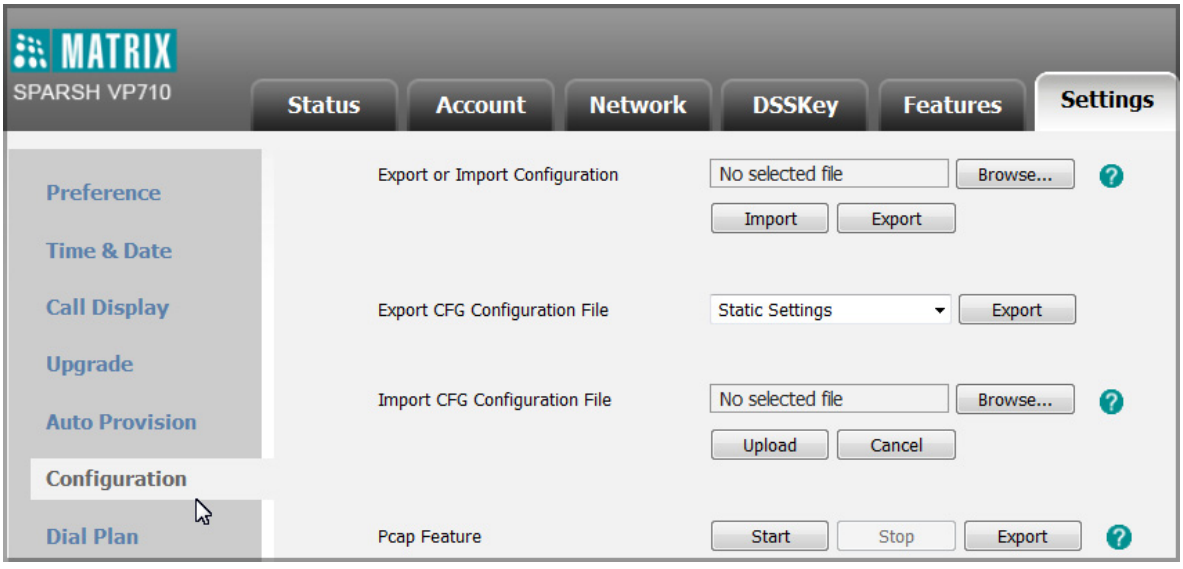
Procedure

Log setting can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the syslog mode. Configure the IP address or domain name of the syslog server where to export the log files. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure the syslog mode. Configure the IP address or domain name of the syslog server where to export the log files. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-config&q=load

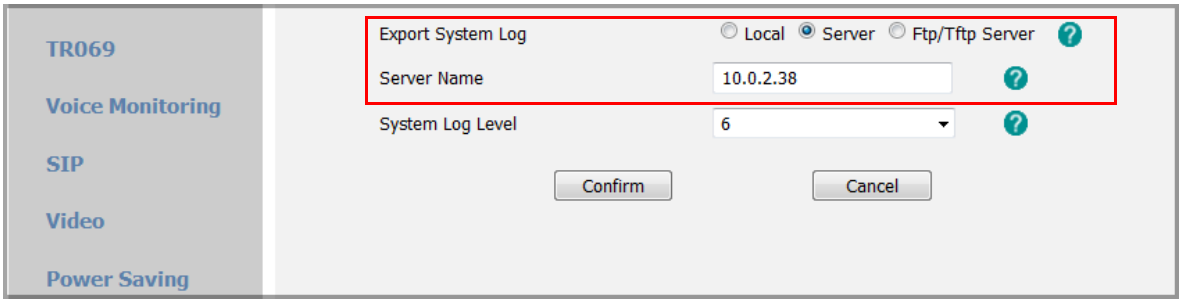
To configure the phone to export the system log to a syslog server via web user interface:

- Click on **Settings->Configuration**.



- Select the **Server** radio box in the **Export System Log**

- Enter the IP address or domain name of the syslog server in the **Server Name** field.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt “Warning: Some settings you changed take effect when you restart your machine! Do you want to reboot now?”. The configuration will take effect after a reboot.
- Click **OK** to reboot the IP phone.

  The system log will be exported successfully to the desired syslog server after a reboot.
- Reproduce the issue.

Exporting the Log File to a Provisioning Server (FTP/TFTP Server)

Procedure

Log setting can be configured using the configuration files or locally.

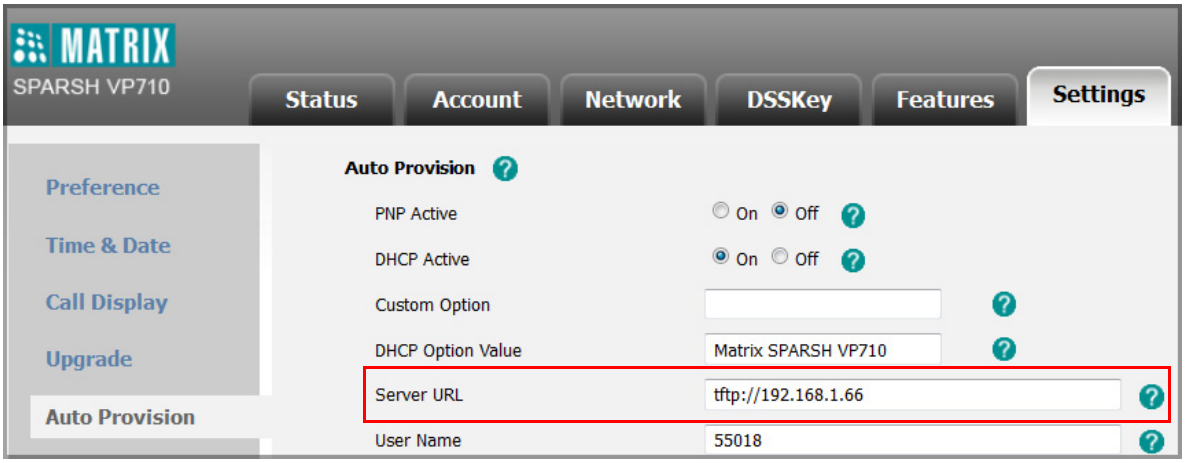
Configuration File	<MAC>.cfg	Configure the syslog mode. Configure the period (in seconds) of the log uploads to the provisioning server. Configure whether the log files on the provisioning server are overwritten or appended. Configure the maximum size of the log files can be stored on the provisioning server. Configure the behavior when log file on the provisioning server reaches the max size. Configure the waiting time before the phone uploads the log file to the provisioning server. Configure the access URL of the provisioning server. For more information, refer “Appendix D - Configuration Parameters”.

Local	Web User Interface	Configure the syslog mode. Configure the period (in seconds) of the log uploads to the provisioning server. Configure whether the log files on the provisioning server are overwritten or appended. Configure the maximum size of the log files on the provisioning server. Configure the behavior when log file on the provisioning server reaches the max size. Configure the waiting time before the phone uploads the log file to the provisioning server. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-config&q=load Configure the access URL of the provisioning server. **Navigate to:** http://<phoneIPAddress>/servlet?m=mod_data&p=settings-autop&q=load

To configure the URL of the provisioning server via web user interface:

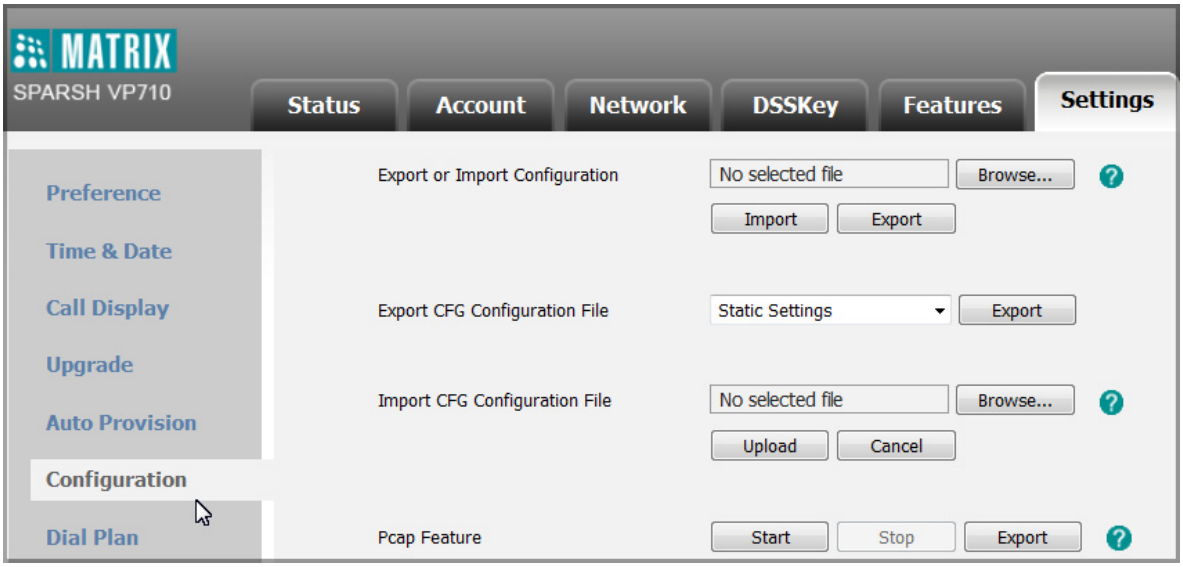
- Click on **Settings->Auto Provision.**
- Enter the URL of the FTP/TFTP server in the **Server URL** field.

  For example, if the IP address TFTP server is 192.168.1.66, then the URL “tftp://192.168.1.66/” is where the IP phone exports the system log.

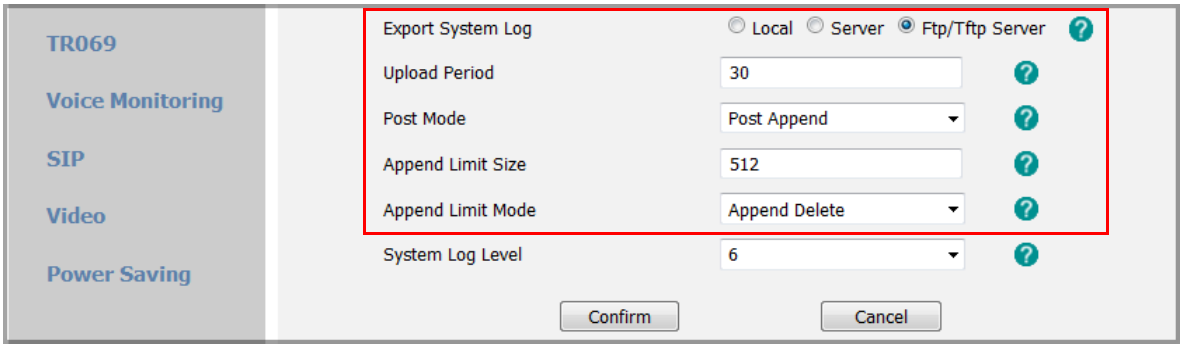


To configure the phone to export the system log to an FTP/TFTP server via web user interface:

- Click on **Settings->Configuration.**



- Mark the **Ftp/Tftp serve**r radio box in the **Export System Log** field.
- Enter the upload period of the log files in the **Upload Period** field.
- Select the desired post mode from the **Post Mode** list.
- Enter the limit size of the log files in the **Append Limit Size** field.
- Select the desired limit mode from the **Append Limit Mode** list.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt “Warning: Some settings you changed take effect when you restart your machine! Do you want to reboot now?”. The configuration will take effect after a reboot.
- Click **OK** to reboot the phone.

  The system log will be exported successfully to the desired FTP/TFTP server after a reboot.
- Reproduce the issue.

Capturing Packets

You can capture packet in two ways: capturing the packet via web user interface or using the Ethernet software. You can analyze the packet captured for troubleshooting purpose.

Capturing the Packets via Web User Interface

The IP phone supports exporting the packets file to the local system and analyze it. You can configure the maximum size and the filter type of the packets

Procedure

Pcap feature can be configured using the configuration files or locally.

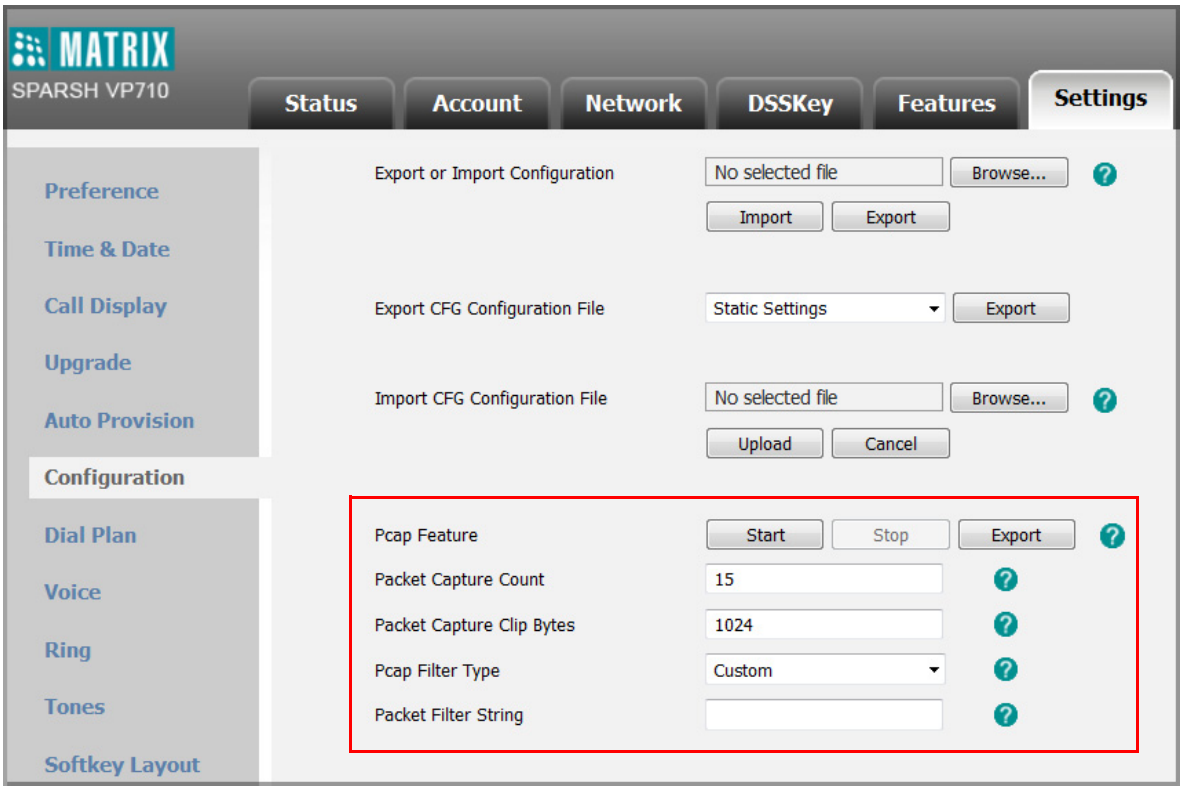
Configuration File	<MAC>.cfg	Configure Pcap feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure Pcap feature. **Navigate to**: http://<phoneIPAddress>/ servlet?m=mod_data&p=settings-config&q=load

To capture packets via web user interface:

- Click on **Settings->Configuration.**
- Enter the desired value in the **Packet Capture Count** field.
- Enter the desired value in the **Packet Capture Clip Bytes** field.
- Select the desired value from the **Pcap Filter Type** list.

  If **Custom** is selected, enter the desired packet filter string in the **Packet Filter String** field.
- Enter the desired value in the **Packet Filter String** field.
- Click **Start** to start capturing signal traffic.
- Reproduce the issue to get stack traces.
- Click **Stop** to stop capturing.

- Click **Export** to open the file download window, and then save the file to your local system.



Capturing packets using the Ethernet Software

Receiving data packets from the HUB

Connect the Internet port of the IP phone and the PC to the same HUB, and then use Sniffer, Ethereal or Wireshark software to capture the signal traffic.

Receiving data packets from PC port

Connect the Internet port of the IP phone to the Internet and the PC port of the IP phone to a PC. Before capturing the signal traffic, make sure the data packets can be received from the Internet port to the PC port.

Procedure

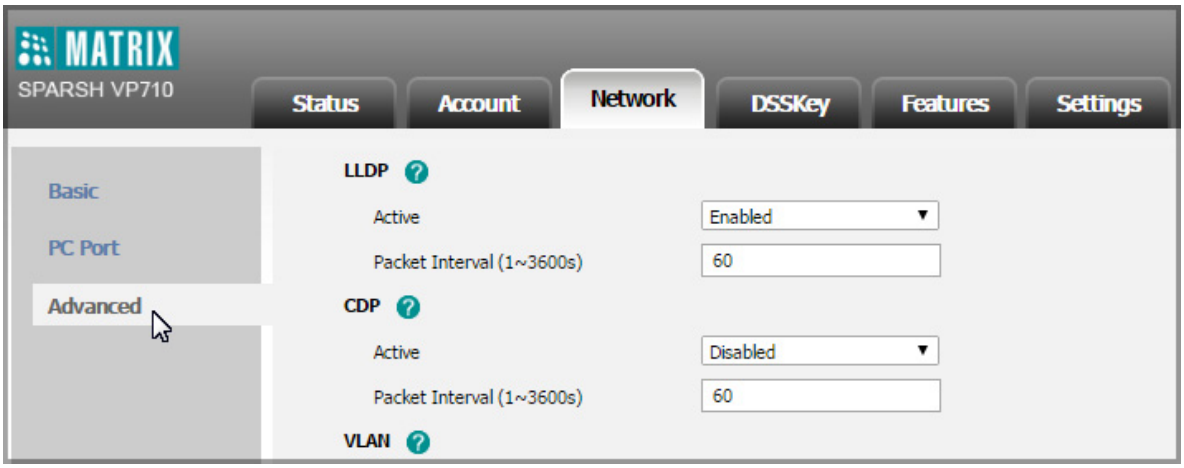
Span to PC port can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure span to PC port. For more information, refer "Appendix D - Configuration Parameters".

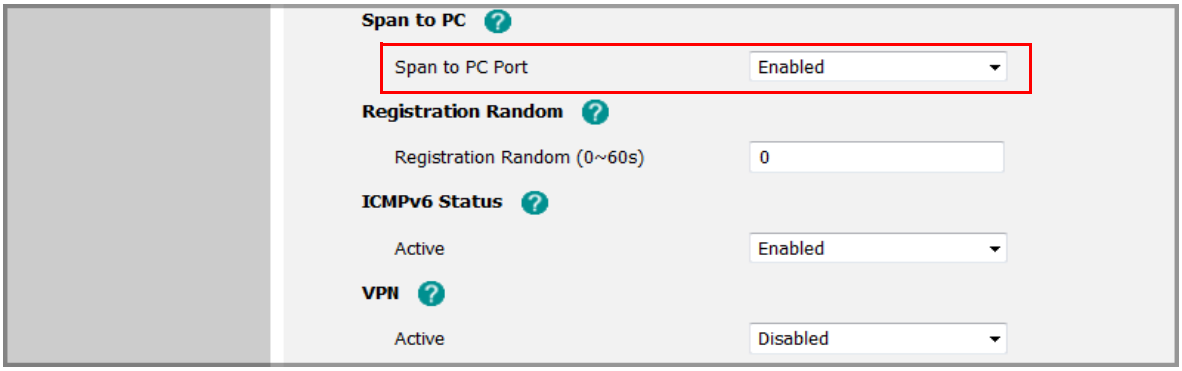
Local	Web User Interface	Configure span to PC port. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=network-adv&q=load

To enable span to PC port via web user interface:

- Click on **Network->Advanced.**



- Select **Enabled** from the **Span to PC Port** list.



- Click **Confirm** to save the change.

  A dialog box pops up to prompt that settings will take effect after a reboot.

- Click **OK** to reboot the phone.

  Then you can use Sniffer, Ethereal or Wireshark software to capture the signal traffic.

Enabling WatchDog Feature

The IP phone provides a troubleshooting feature called “WatchDog”, which helps you monitor the IP phone status and provides the ability to get stack traces from the last time the IP phone failed. If WatchDog feature is enabled, the IP phone will automatically reboot when it detects a fatal failure. This feature can be configured using the configuration files or via web user interface.

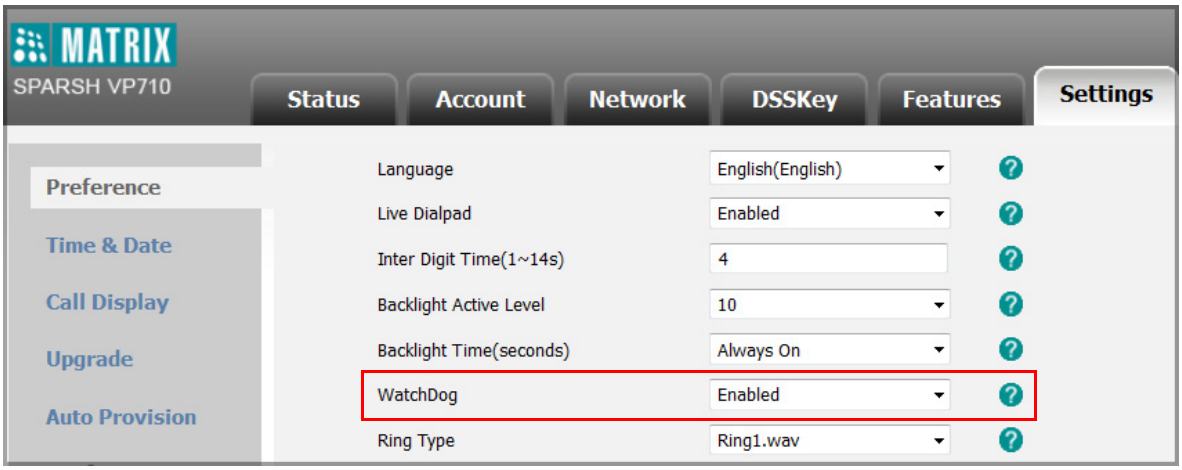
Procedure

Watch Dog can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure Watch Dog feature. For more information, refer "Appendix D - Configuration Parameters".
Local	Web User Interface	Configure Watch Dog feature. **Navigate to**: http://<phoneIPAddress>/servlet?m=mod_data&p=settings-preference&q=load

To configure watchdog feature via web user interface:

- Click on **Settings->Preference**.
- Select the desired value from the **WatchDog** list.



- Click **Confirm** to save the change.

Getting Information from Status Indicators

Status indicators may consist of the power LED, line key indicator, headset key indicator and the on-screen icon.

The following shows two examples of obtaining the IP phone information from status indicators on the IP phone:

- If a LINK failure of the IP phone is detected, a prompting message "Network unavailable", and the icon  will appear on the touch screen.

- If the headset mode is enabled, the headset key LED illuminates.

  For more information on the icons, refer “Icon Instructions”.

CHAPTER 16

Troubleshooting

This section describes solutions to common issues that may occur while using the IP phone. Upon encountering a scenario not listed in this section, contact your Matrix SPARSH VP710 reseller for further support.

Time and Date Issues

Why doesn¡¦t the IP phone display time and date correctly?
Check if the IP phone is configured to obtain the time and date from the NTP server automatically. If your phone is unable to access the NTP server, configure the time and date manually.

Display Issues

Why is the touch screen blank?
Do one of the following:

- Ensure that the IP phone is properly plugged into a functional AC outlet.
- Ensure that the IP phone is plugged into a socket controlled by a switch that is on.
- If the IP phone is plugged into a power strip, try plugging it directly into a wall outlet.
- If your phone is PoE powered, ensure that you are using a PoE-compliant switch or hub.

Why does the IP phone display “No Service”?
The touch screen prompts “No Service” message when there is no available SIP account on the IP phone.
Do one of the following:

- Ensure that an account is actively registered on the IP phone at the path Settings->Status->Accounts.
- Ensure that the SIP account parameters have been configured correctly.

Phone Book Issues

What is the difference between a remote phone book and a local phone book?
A remote phone book is placed on a server, while a local phone book is placed on the IP phone flash. A remote phone book can be used by everyone that can access the server, while a local phone book can only be used by a specific phone. A remote phone book is always used as a central phone book for a company; each employee can load it to obtain the real-time data from the same server.

Audio Issues

How to increase or decrease the volume?
Press the Volume key to increase or decrease the ringer volume when the IP phone is idle or when there is an incoming call arrives on the phone, to adjust the volume of engaged audio device (handset, speakerphone or headset) when there is an active call in progress, or to adjust the media volume when the phone is not on the idle screen.

Why do I get poor sound quality during a call?
If you have poor sound quality/acoustics like intermittent voice, low volume, echo or other noises, the possible reasons could be:

- Users are seated too far out of recommended microphone range and sound faint, or are seated too close to sensitive microphones and cause echo.
- Intermittent voice is mainly caused by packet loss, due to network congestion, and jitter, due to message recombination of transmission or receiving equipment (e.g., timeouthandling, retransmission mechanism, buffer under run).
- Noisy equipment, such as a computer or a fan, may cause voice interference. Turn off any noisy equipment.
- Line issues can also cause this problem; disconnect the old line and redial the call to ensure another line may provide better connection.

Why is there no sound when the other party picks up the call?
If the caller and receiver cannot hear anything - there is no sound at all when the other party picks up the call, the possible reason could be: the phone cannot send the real-time transport protocol (RTP) streams, in which audio data is transmitted, to the connected call. Try to disable the “180 Ring Workaround” feature.

Why does the IP phone play the local ringback tone instead of media when placing a long distance number without plus 0?
Ensure that the “180 Ring Workaround” feature is disabled.

Camera and Video Issues

Why is the video quality bad?

- Ensure that the display device has suitable resolution.
- Check whether the packet has been lost.
- Ensure that camera settings are configured correctly, such as brightness and white balance.
- Avoid high-intensity indoor light or direct sunlight on the camera.

Why can’t I preview local camera when the phone is idle?
If the camera is properly connected to the IP phone but there are no images on the screen when you launch Camera application or swipe down from the top of the screen and then tap Video, you may need to replace the camera.

Why is there some dazzle light on the images when previewing the local camera?
If the camera lens is oily or soiled, there may be some dazzle light on the images. Please try to clean it up.

Wi-Fi and Bluetooth Issues

Why is the wireless signal strength low?
Ensure the IP phone and your gateway/router are within the working range and there is no obvious interference (walls, doors, etc.) between them.

Why can't I connect the IP phone to the 2.4G wireless network?
If you successfully connect the IP phone to the 2.4G wireless network, but the video images is not smooth. Or, you cannot connect the IP phone to the 2.4G wireless network.

- Check if there are too many wireless devices connecting to the same 2.4G wireless network.
- Verify whether the distance between IP phone and the wireless router is too far.

Why can't I connect the Bluetooth device with the IP phone all the time?
Try to delete the registration information of the Bluetooth device on both IP phone and Bluetooth device, and then pair and connect it again.

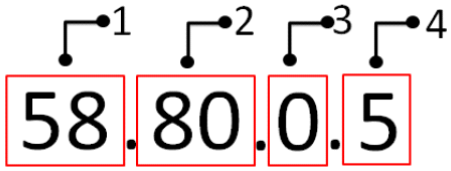
Why the Bluetooth headset affects IP phone's voice quality?
You may not experience the best voice quality if you use a Bluetooth headset while the 2.4 GHz band is enabled or while you are in an environment with many other Bluetooth devices. This possible loss in voice quality is due to inherent limitations with Bluetooth technology.

Firmware and Upgrading Issues

Why doesn't the IP phone upgrade firmware successfully?
Do one of the following:

- Ensure that the target firmware is not the same as the current firmware.
- Ensure that the target firmware is applicable to the IP phone model.
- Ensure that the current or the target firmware is not protected.
- Ensure that the power is on and the network is available in the process of upgrading.
- Ensure that the web browser is not closed or refreshed when upgrading firmware via web user interface.

How can I verify the firmware generation and version of the phone?
Tap **Settings->Status** when the IP phone is idle to check the firmware version. For example: 58.80.0.5.



	Item	Description
1	58	Firmware ID
2	80	Firmware generation. Note: The larger it is, the newer the firmware generation is.
3	0	A fixed number
4	5	Firmware version. Note: With the same firmware generation, the larger it is, the newer the firmware version is.

Why doesn¡¦t the IP phone update the configuration?
Do one of the following:

- Ensure that the configuration is set correctly.
- Reboot the phone. Some configurations require a reboot to take effect.
- Ensure that the configuration is applicable to the IP phone model.
- The configuration may depend on support from a server.

Provisioning Issues

What is auto provisioning?
Auto provisioning refers to the update of IP phones, including update on configuration parameters, local phone book, firmware and so on. You can use auto provisioning on a single phone, but it makes more sense in mass deployment.

What is PnP?
Plug and Play (PnP) is a method for IP phones to acquire the provisioning server address. With PnP enabled, the IP phone broadcasts the PnP SUBSCRIBE message to obtain a provisioning server address during startup. Any SIP server recognizing the message will respond with the preconfigured provisioning server address, so the IP phone will be able to download the CFG files from the provisioning server. PnP depends on support from a SIP server.

System Log Issues

Why can’t I export the system log to a provisioning server (FTP/TFTP server)?
Do one of the following:

- Ensure that the FTP/TFTP server is downloaded and installed on your local system.
- Ensure that you have configured the FTP/TFTP server address correctly via web user interface on your IP phone.
- Reboot the phone. The configurations require a reboot to take effect.

Why can’t I export the system log to a syslog server?
Do one of the following:

- Ensure that the syslog server supports saving the syslog files exported from IP phone.
- Ensure that you have configured the syslog server address correctly via web user interface on your IP phone.
- Reboot the phone. The configurations require a reboot to take effect.

Resetting Issues

Generally, some common issues may occur while using the IP phone. You can reset your phone to factory configurations after you have tried all troubleshooting suggestions but do not solve the problem. Resetting the phone to factory configurations clears the flash parameters, removes log files, user data, and cached data, and resets the administrator password to admin. All custom settings will be overwritten after resetting.

Six ways to reset the phone:

- Reset local settings: All configurations saved in the <MAC>-local.cfg file on the IP phone will be reset. Changes associated with non-static settings made via web user interface and phone user interface are saved in the <MAC>-local.cfg file.
- Reset non-static settings: All non-static settings on the phone will be reset. After resetting the non-static settings, the IP phone will perform the auto provisioning process immediately.
- Reset static settings: All static settings on the phone will be reset.
- Reset userdata & local config: All the local cache data (e.g., userdata, history, directory) will be cleared. And all configurations saved in the <MAC>-local.cfg configuration file on the IP phone will be reset.
- Reset to factory Setting: All configurations on the phone will be reset.
- Reset build-in SD card: All the files in the internal SD card will be cleared.

You can reset the IP phone to default factory configurations. The default factory configurations are the settings that reside on the IP phone after it has left the factory. You can also reset the IP phone to custom factory configurations if required. The custom factory configurations are the settings that defined by the user to keep some custom settings after resetting.



The Reset local settings/Reset non-static settings/Reset static settings/Reset userdata & local config option on the web user interface appears only if the value of the parameter static.auto_provision.custom.protect" is set to 1.

How to reset the IP phone to default factory configurations?

To reset the IP phone via web user interface:

- Click on **Settings->Upgrade.**

  The web user interface prompts the message "Do you want to reset to factory?".
- Click OK to confirm the resetting.

  The IP phone will be reset to factory successfully after startup.

How to reset internal SD card to factory?

To reset the internal SD card to factory:

- Click on **Settings->Upgrade.**
- Click **Reset build-in SD card** in the **Reset build-in SD card** field.

  The web user interface prompts the message "Reset build-in SD card to factory?".
- Click **OK** to confirm the resetting.

  And all the data (e.g., pictures, audio and video files) in the internal will be cleared if you reset the internal SD card.

Rebooting Issues

How to reboot the IP phone remotely?

IP phones support remote reboot by a SIP NOTIFY message with "Event: check-sync" header. Whether the IP phone reboots or not depends on the value of the parameter "sip.notify_reboot_enable". If the value is set to 1, or the value is set to 0 and the header of the SIP NOTIFY message contains an additional string "reboot=true", the IP phone will reboot immediately.

The NOTIFY message is formed as shown:

NOTIFY sip:<user>@<dsthost> SIP/2.0
To: sip:<user>@<dsthost>
From: sip:sipsak@<srchost>
CSeq: 10 NOTIFY
Call-ID: 1234@<srchost>
Event: check-sync;reboot=true

Procedure

Changes can only be configured using the configuration files..

Configuration File	<MAC>.cfg	Configure the IP phone behavior when receiving a SIP NOTIFY message which contains the header "Event: check-sync".

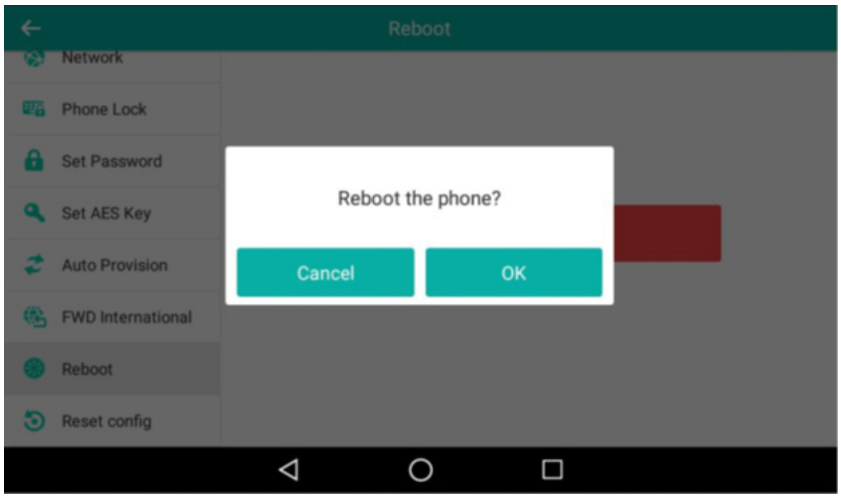
Parameter- sip.notify_reboot_enable	**Configuration File** <MAC>.cfg
Description	Configure the IP phone behavior when receiving a SIP NOTIFY message which contains the header "Event: check-sync".
Default Value	1
Range	0-The IP phone will reboot only if the SIP NOTIFY message contains an additional string "reboot=true". 1-The IP phone will be forced to reboot. 2-The IP phone will ignore the SIP NOTIFY message.

How to reboot the IP phone via web/phone user interface?
You can reboot your IP phone via web/phone user interface.

To reboot the phone via phone user interface:

- Tap **Settings->Advanced (default password: admin) ->Reboot.**
- Tap **Reboot**.

The touch screen prompts the following warning:



- Tap **OK** to reboot the phone.

  The phone begins rebooting. Any reboot of the phone may take a few minutes.

To reboot the phone via web user interface:

- Click on Settings->Upgrade.
- Click Reboot to reboot the IP phone.

  The phone begins rebooting. Any reboot of the phone may take a few minutes.

Password Issues

How to restore the administrator password?
Factory reset can restore the original password. All custom settings will be overwritten after reset.

Power and Startup Issues

What will happen if I connect both PoE cable and power adapter? Which has the higher priority?
The IP phone uses PoE preferentially.

Why does the IP phone have no power?
If no lights appear on the IP phone when it is powered up, do one of the following:

- Reboot your IP phone
- Replace the power adapter

Why is the touch screen black?
If the power indicator LED is on, the keypad is usable but the touch screen is black, please reboot your IP phone.

Why does the IP phone always display the MATRIX logo?
If your IP phone does not boot, check if the provisioning server is accessible on the network and a valid software firmware and valid configuration files are available. Try to use recovery mode to get your phone ready.

Why can't IP phone supply power for device using USB port?
The USB port of MATRIX IP phone has a limit current of 525 ~ 875mA. Make sure that the device is USB flash drive or mobile hard disk with low power.

Hardware Issues

Why is the sending/receiving volume of the speaker too low?

- If there is no volume sending from the speaker or sending volume is too low, the Hands-free MIC cable may not have been properly connected.
- If there is no volume receiving from the speaker or receiving volume is too low, the speaker cable may not have been properly connected.

Why is the sending/receiving volume of the headset or handset too low?
Ensure that the headset or handset is not damaged. If the headset or handset is usable, it may be the codec problem on the mainboard.

Why is there no response when pressing the keys on the keypad?
Do one of the following:

- Ensure that the keypad cables is properly connected and not damaged.
- Check if the keypad surface is clean.

Why is there no response when tapping the items on the touch screen?
Do one of the following:

- Ensure that the FPC of the touch screen is properly connected.
- Check if the touch screen is damaged.

Why is the LED off when pressing the hard key with LED indicator?
Make sure that the cable of keypad board is properly connected. If the cable is properly connected, it may be the LED on the board is damaged.

Other Issues

How do I find the basic information of the IP phone?
Tap **Settings->Status** when the IP phone is idle to check the basic information (e.g., IP address, MAC address and firmware version).

What is the difference among user name, register name and display name?
Both user name and register name are defined by the server. User name identifies the account, while register name matched with a password is for authentication purposes. Display name is the caller ID that will be displayed on the callee's phone touch screen. Server configurations may override the local ones.

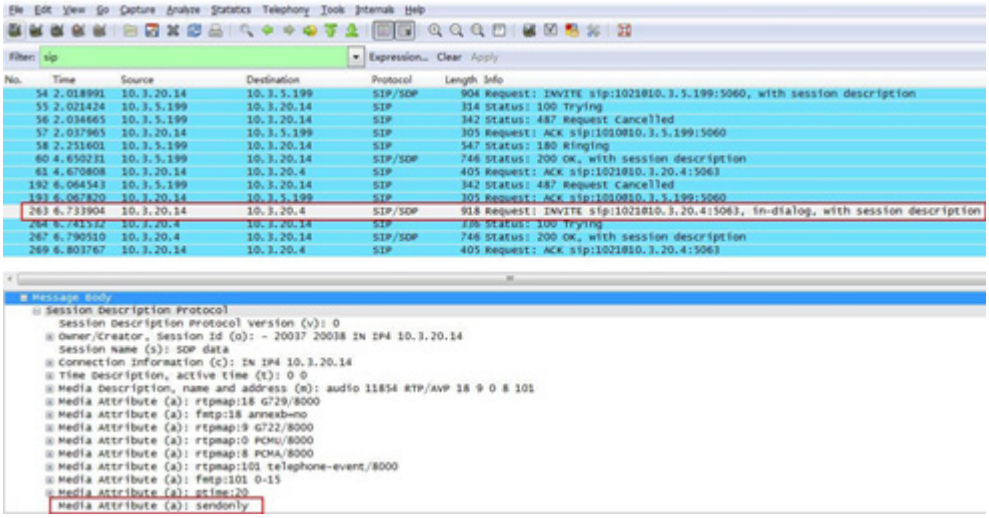
What do "on code" and "off code" mean?
These are the codes that the IP phone sends to the server when a certain action takes place. On code is used to activate a feature on the server side, while off code is used to deactivate a feature on the server side.

For example, if you set the Always Forward on code to be *78 (may vary on different servers), and the target number to be 201. When you enable Always Forward on the IP phone, the IP phone sends *78201 to the server,

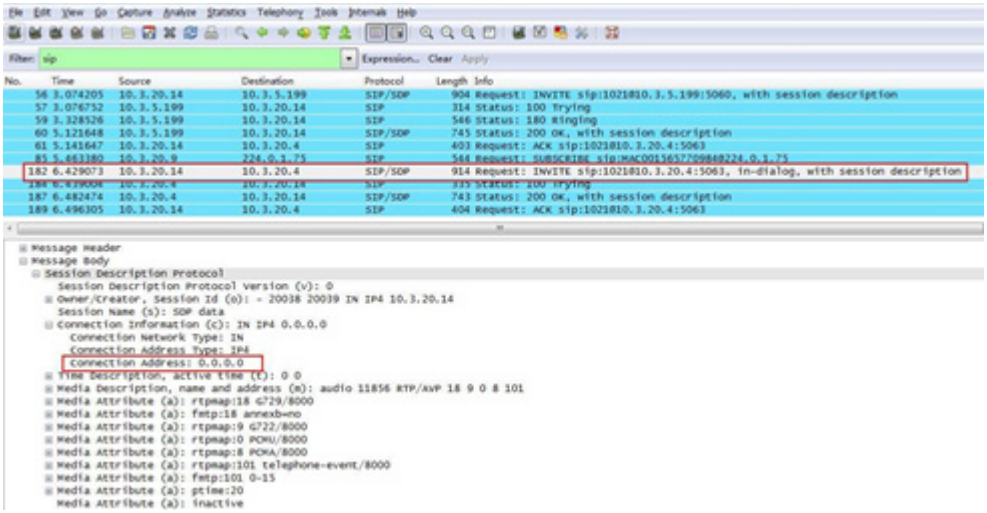
and then the server will enable Always Forward feature on the server side, hence being able to get the right status of the extension.

For anonymous call/anonymous call rejection feature, the phone will send either the on code or off code to the server according to the value of Send Anonymous Code/Send Rejection Code. For more information, refer to “Anonymous Call” and “Anonymous Call Rejection”.

What is the difference between enabling and disabling the RFC 2543 Hold feature?
Capturing packets after you enable the RFC 2543 Hold feature. SDP media direction attributes (such as a=sendonly) per RFC 2543 is used in the INVITE message when placing a call on hold.



Capturing packets after you disable the RFC 2543 Hold feature. SDP media connection address c=0.0.0.0 per RFC 3264 is used in the INVITE message when placing a call on hold.



For more information on RFC 2543 hold feature, refer to “Call Hold/Resume”.For more information on capturing packets, refer to “Capturing Packets”.

Appendix

Physical Features of the IP Phone

This section lists the available physical features of SPARSH VP710 IP phone.

- 7” 1024 x 600 pixel color touch screen with backlight
- Operating System: Android™ 5.1.1
- 16 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 20 dedicated hard keys, 3 dedicated soft Android keys for BACK, HOME and RECENT
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 4 LEDs: 1*power, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/2A output
- 1*USB2.0 port (on the top of the phone), support, USB camera.
- 1*USB2.0 port (on the rear of the phone), , USB flash drive or USB headset
- Built-in Wi-Fi, support 802.11b/g/n
- Built-in Bluetooth 4.0, support Bluetooth headset
- Power over Ethernet (IEEE 802.3af)
- Wall Mountable

Key Features of the IP Phone

In addition to physical features introduced above, IP phone also supports the following key features when running the latest firmware:

Phone Features

- **Call Options:** emergency call, call waiting, call hold, call mute, call forward, call transfer, call pickup, five-way audio-only conference, five-way audio-only and video mixed conference (up to three-way video conference.
- **Basic Features:** DND, auto redial, live dialpad, dial plan, hotline, caller identity, auto answer.
- **Advanced Features:** BLF, server redundancy, distinctive ring tones, remote phone book, LDAP.

Codecs and Voice Features

- Wideband codec: G.722, Opus
- Narrowband codec: G.711, G.726, G.729, iLBC, G.723

- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

Video Features

- Video codec: H264HP, H264, VP8
- Image codec: JPEG, PNG, BMP
- Adaptive bandwidth adjustment

Network Features

- SIP v1 (RFC 2543), v2 (RFC 3261)
- NAT Traversal: STUN mode
- DTMF: INBAND, RFC 2833, SIP INFO
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: Static/DHCP/PPPoE
- VLAN assignment: LLDP/Static/DHCP/CDP
- Bridge mode for PC port
- HTTP/HTTPS server
- DNS client
- NAT/DHCP server
- IPv6 support
- Wi-Fi

Management

- FTP/TFTP/HTTP/PnP auto-provision
- Configuration: browser/phone/auto-provision
- Direct IP call without SIP proxy
- Dial number via SIP server
- Dial URL via SIP server
- TR-069

Security

- HTTPS (server/client)
- SRTP (RFC 3711)
- Transport Layer Security (TLS)
- VLAN (802.1q), QoS
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES encryption
- Phone lock for personal privacy protection
- Admin/User configuration mode
- 802.1X authentication

Appendix A - Time Zone

Time Zone	Time Zone Name
-11	Samoa
-10	United States-Hawaii-Aleutian, United States-Alaska-Aleutian
-9:30	French Polynesia
-9	United States-Alaska Time
-8	Canada(Vancouver,Whitehorse), Mexico(Tijuana,Mexicali), United States-Pacific Time
-7	Canada(Edmonton,Calgary), Mexico(Mazatlan,Chihuahua), United States-MST no DST, United States-Mountain Time
-6	Canada-Manitoba(Winnipeg), Chile(Easter Islands), Mexico(Mexico City,Acapulco), United States-Central Time
-5	Bahamas(Nassau), Canada(Montreal,Ottawa,Quebec), Cuba(Havana), United States-Eastern Time
-4:30	Venezuela(Caracas)
-4	Canada(Halifax,Saint John), Chile(Santiago), Paraguay(Asuncion), United Kingdom-Bermuda(Bermuda), United Kingdom(Falkland Islands), Trinidad&Tobago
-3:30	Canada-New Foundland(St.Johns)
-3	Argentina(Buenos Aires), Brazil(DST), Brazil(no DST), Denmark-Greenland(Nuuk)
-2:30	Newfoundland and Labrador
-2	Brazil(no DST)
-1	Portugal(Azores)
0	Denmark-Faroe Islands(Torshavn), GMT, Greenland, Ireland(Dublin), Morocco, Portugal(Lisboa,Porto,Funchal), Spain-Canary Islands(Las Palmas), United Kingdom(London)
+1	Albania(Tirane), Austria(Vienna), Belgium(Brussels),Caicos, Chad, Croatia(Zagreb), Czech Republic(Prague), Denmark(Kopenhagen), France(Paris), Germany(Berlin), Hungary(Budapest), Italy(Rome), Luxembourg(Luxembourg), Macedonia(Skopje), Namibia(Windhoek), Netherlands(Amsterdam), Spain(Madrid)
+2	Estonia(Tallinn), Finland(Helsinki), Gaza Strip(Gaza), Greece(Athens), Israel(Tel Aviv), Jordan(Amman), Latvia(Riga), Lebanon(Beirut), Moldova(Kishinev), Romania(Bucharest), Russia(Kaliningrad), Syria(Damascus), Turkey(Ankara), Ukraine(Kyiv, Odessa)
+3	East Africa Time, Iraq(Baghdad), Russia(Moscow)
+3:30	Iran(Teheran)
+4	Armenia(Yerevan), Azerbaijan(Baku), Georgia(Tbilisi), Kazakhstan(Aktau), Russia(Samara)
+4:30	Afghanistan(Kabul)
+5	Kazakhstan(Aqtobe), Kyrgyzstan(Bishkek), Pakistan(Islamabad), Russia(Chelyabinsk)
+5:30	India(Calcutta)
+5:45	Nepal(Katmandu)
+6	Kazakhstan(Astana, Almaty), Russia(Novosibirsk,Omsk)
+6:30	Myanmar(Naypyitaw)
+7	Russia(Krasnoyarsk), Thailand(Bangkok)
+8	Australia(Perth), China(Beijing), Russia(Irkutsk, Ulan-Ude), Singapore(Singapore)
+8:45	Eucla
+9	Japan(Tokyo), Korea(Seoul), Russia(Yakutsk,Chita)
+9:30	Australia(Adelaide), Australia(Darwin)

+10	Australia(Brisbane), Australia(Hobart), Australia(Sydney,Melboume,Canberra), Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11	New Caledonia(Noumea), Russia(Srednekolymsk Time)
+11:30	Norfolk Island
+12	New Zealand(Wellington,Auckland), Russia(Kamchatka Time)
+12:45	New Zealand(Chatham Islands)
+13	Tonga(Nukualofa)
+13:30	Chatham Islands
+14	Kiribati

Appendix B - Glossary

- **802.1x:** An IEEE Standard for port-based Network Access Control (PNAC). It is a part of the IEEE 802.1group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.
- **ACS** (Auto Configuration Server)**:** It is responsible for auto-configuration of the Central Processing Element (CPE).
- **Cryptographic Key**: It is a piece of variable data that is fed as input into a cryptographic algorithm to perform operations such as encryption and decryption, or signing and verification.
- **DHCP** (Dynamic Host Configuration Protocol)**:** It is built on a client-server model, where designated DHCP server hosts allocate network addresses and deliver configuration parameters to dynamically configured hosts.
- **DHCP Option:** It can be configured for specific values and enabled for assignment and distribution to DHCP clients based on server, scope, class or client-specific levels.
- **DNS** (Domain Name System)**:** It is a hierarchical distributed naming system for computers, services, or any resource connected to the Internet or a private network.
- **EAP-MD5** (Extensible Authentication Protocol-Message Digest Algorithm 5)**:** It only provides authentication of the EAP peer to the EAP server but not mutual authentication.
- **EAP-TLS** (Extensible Authentication Protocol-Transport Layer Security)**:** It provides for mutual authentication, integrity-protected cipher suite negotiation between two endpoints.
- **PEAP-MSCHAPv2** (Protected Extensible Authentication Protocol-Microsoft Challenge Handshake Authentication Protocol version 2)**:** It provides for mutual authentication, but does not require a client certificate on the IP phone.
- **FAC** (Feature Access Code)**:** These special patterns of characters that are dialed from a phone keypad to invoke particular features.
- **HTTP** (Hypertext Transfer Protocol)**:** It is used to request and transmit data on the World Wide Web.
- **HTTPS** (Hypertext Transfer Protocol over Secure Socket Layer)**:** It is a widely-used communications protocol for secure communication over a network.
- **IEEE** (Institute of Electrical and Electronics Engineers)**:** It is a non-profit professional association headquartered in New York City that is dedicated to advancing technological innovation and excellence.
- **LAN** (Local Area Network)**:** It is used to interconnect network devices in a limited area such as a home, school, computer laboratory, or office building.
- **MIB** (Management Information Base)**:** It is a virtual database used for managing the entities in a communications network.
- **OID** (Object Identifier)**:** It is assigned to an individual object within a MIB.
- **PnP** (Plug and Play)**:** It is a term used to describe the characteristic of a computer bus, or device specification, which facilitates the discovery of a hardware component in a system, without the need for physical device configuration, or user intervention in resolving resource conflicts.
- **ROM** (Read-only Memory)**:** It is a class of storage medium used in computers and other electronic devices.

- **RTP** (Real-time Transport Protocol)**:** It provides end-to-end service for real-time data.
- **TCP** (Transmission Control Protocol)**:** It is a transport layer protocol used by applications that require guaranteed delivery.
- **UDP** (User Datagram Protocol): It is a protocol that offers non-guaranteed datagram delivery.
- **URI** (Uniform Resource Identifier)**:** It is a compact sequence of characters that identifies an abstract or physical resource.
- **URL** (Uniform Resource Locator): It specifies the address of an Internet resource.
- **VLAN** (Virtual LAN)**:** It is a group of hosts with a common set of requirements, which communicate as if they were attached to the same broadcast domain, regardless of their physical location.
- **VoIP** (Voice over Internet Protocol)**:** It is a family of technologies used for the delivery of voice communications and multimedia sessions over IP networks.
- **WLAN** (Wireless Local Area Network)**:** It is a type of local area network that uses high-frequency radio waves rather than wires to communicate between nodes.
- **XML-RPC** (Remote Procedure Call Protocol)**:** It uses XML to encode its calls and HTTP as a transport mechanism.

Appendix C - Trusted Certificates

SPARSH VP710 IP phones trust the following CAs by default:

- DigiCert High Assurance EV Root CA
- Deutsche Telekom AG Root CA-2
- Equifax Secure Certificate Authority
- Equifax Secure eBusiness CA-1
- Equifax Secure Global eBusiness CA-1
- GeoTrust Global CA
- GeoTrust Global CA2
- GeoTrust Primary CA
- GeoTrust Primary CA G2 ECC
- GeoTrust Universal CA
- GeoTrust Universal CA2
- Thawte Personal Freemail CA
- Thawte Premium Server CA
- Thawte Primary Root CA - G1 (EV)
- Thawte Primary Root CA - G2 (ECC)
- Thawte Primary Root CA - G3 (SHA256)
- Thawte Server CA
- VeriSign Class 1 Public Primary Certification Authority
- VeriSign Class 1 Public Primary Certification Authority - G2
- VeriSign Class 1 Public Primary Certification Authority - G3
- VeriSign Class 2 Public Primary Certification Authority - G2
- VeriSign Class 2 Public Primary Certification Authority - G3
- VeriSign Class 3 Public Primary Certification Authority
- VeriSign Class 3 Public Primary Certification Authority - G2
- VeriSign Class 3 Public Primary Certification Authority - G3
- VeriSign Class 3 Public Primary Certification Authority - G4
- VeriSign Class 3 Public Primary Certification Authority - G5
- VeriSign Class 4 Public Primary Certification Authority - G2
- VeriSign Class 4 Public Primary Certification Authority - G3
- VeriSign Universal Root Certification Authority
- ISRG Root X1 (Let’s Encrypt Authority X1 and Let’s Encrypt Authority X2 certificates are signed by the root certificate ISRG Root X1.)
- Baltimore CyberTrust Root
- DST Root CA X3
- Verizon Public SureServer CA G14-SHA2
- AddTrust External CA Root
- Go Daddy Class 2 Certification Authority

- Class 2 Primary CA
- Cybertrust Public SureServer SV CA
- DigiCert Assured ID Root G2
- DigiCert Assured ID Root G3
- DigiCert Assured ID Root CA
- DigiCert Global Root G2
- DigiCert Global Root G3
- DigiCert Global Root CA
- DigiCert Trusted Root G4
- Entrust Root Certification Authority
- Entrust Root Certification Authority - G2
- Entrust.net Certification Authority (2048)
- GeoTrust Primary Certification Authority - G3
- GlobalSign Root CA
- GlobalSign
- Starfield Root Certificate Authority - G2
- TC TrustCenter Class 2 CA II
- TC TrustCenter Class 3 CA II
- TC TrustCenter Class 4 CA II
- TC TrustCenter Universal CA I
- TC TrustCenter Universal CA III
- Thawte Universal CA Root
- VeriSign Class 3 Secure Server CA - G2
- VeriSign Class 3 Secure Server CA - G3
- Thawte SSL CA
- StartCom Certification Authority
- StartCom Certification Authority G2
- Starfield Services Root Certificate Authority - G2
- RapidSSL_CA_bundle
- Go Daddy Root Certificate Authority - G2
- Cybertrust Global Root
- COMODOSSLCA
- COMODO RSA Domain Validation Secure Server CA
- COMODO RSA Certification Authority
- AmazonRootCA4
- AmazonRootCA3
- AmazonRootCA2
- AmazonRootCA1



Matrix endeavors to maintain a built-in list of most common used CA Certificates. Due to memory constraints, we cannot ensure a complete set of certificates. If you are using a certificate from a commercial Certificate Authority not in the list above, you can send a request to your local distributor. At this point, you can upload your particular CA certificate into your phone. For more information on uploading custom CA certificate, refer "TLS".

Appendix D - Configuration Parameters

This appendix describes configuration parameters in the configuration files for each feature. The configuration file is <MAC>.cfg.

Setting Parameters in Configuration Files

You can set parameters in the configuration files to configure IP phones. The <MAC>.cfg file is stored on the provisioning server. The IP phone checks for configuration files and looks for resource files when restarting the IP phone. The <MAC>.cfg file stores configurations for a specific IP phone with that MAC address.

Basic and Advanced Parameters

DHCP

Parameter- static.network.internet_port.type	**Configuration File** <MAC>.cfg
Description	Configures the Internet port type for IPv4. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect. It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6).
Default Value	0
Range	**Valid values are:** **0**-DHCP **1**-PPPoE **2**-Static IP Address
Example	static.network.internet_port.type = 0

Static DNS

Parameter- static.network.static_dns_enable	**Configuration File** <MAC>.cfg
Description	Triggers the static DNS feature to on or off. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0

Range	**Valid values are:** **0**-Disabled **1**-Enabled If it is set to 0 (Off), the IP phone will use the IPv4 DNS obtained from DHCP. If it is set to 1 (On), the IP phone will use manually configured static IPv4 DNS. Note: It works only if the value of the parameter "static.network.internet_port.type" is set to 0 (DHCP). If you change this parameter, the IP phone will reboot to make the change take effect.
Example	static.network.static_dns_enable = 0

Parameter- static.network.primary_dns	**Configuration File** <MAC>.cfg
Description	Configures the primary IPv4 DNS server. **Note**: It works only if the value of the parameter "static.network.static_dns_enable" is set to 1(On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4 Address
Example	static.network.primary_dns = 202.101.103.55

Parameter- static.network.secondary_dns	**Configuration File** <MAC>.cfg
Description	Configures the secondary IPv4 DNS server. **Note**: It works only if the value of the parameter "static.network.static_dns_enable" is set to 1(On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4 Address
Example	static.network.secondary_dns = 202.101.103.54

Static Network Settings

Parameter- static.network.internet_port.type	**Configuration File** <MAC>.cfg
Description	Configures the Internet port type for IPv4. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect. It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6).
Default Value	0
Range	**Valid values are:** **0**-DHCP **1**-PPPoE **2**-Static IP Address
Example	static.network.internet_port.type = 2

Parameter- static.network.ip_address_mode	**Configuration File** <MAC>.cfg
Description	Configures the IP address mode. IP phones support using the IPv4 address only, the IPv6 address only or both IPv4 and IPv6 addresses. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	**Valid values are:** **0**-IPv4 **1**-IPv6 **2**-IPv4&IPv6
Example	static.network.ip_address_mode = 0

Parameter- static.network.internet_port.ip	**Configuration File** <MAC>.cfg
Description	Configures the IP address when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4Address
Example	static.network.internet_port.ip = 192.168.1.20

Parameter- static.network.internet_port.mask	Configuration File <MAC>.cfg
Description	Configures the subnet mask when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	Subnet Mask
Example	static.network.internet_port.mask = 255.255.255.0

Parameter- static.network.internet_port.gateway	Configuration File <MAC>.cfg
Description	Configures the default gateway when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4 Address
Example	static.network.internet_port.gateway = 192.168.1.254

Parameter- static.network.primary_dns	Configuration File <MAC>.cfg
Description	Configures the primary IPv4 DNS server. **Note**: It works only if the value of the parameter "static.network.static_dns_enable" is set to 1(On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4 Address
Example	static.network.primary_dns = 202.101.103.55

Parameter- static.network.secondary_dns	Configuration File <MAC>.cfg
Description	Configures the secondary IPv4 DNS server. **Note**: It works only if the value of the parameter "static.network.static_dns_enable" is set to 1(On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv4 Address
Example	static.network.secondary_dns = 202.101.103.54

PPPoE

Parameter- static.network.internet_port.type	Configuration File <MAC>.cfg
Description	Configures the Internet port type. **Note**: It works only if the value of the parameter "static.network.ip_address_mode" is set to 0 (IPv4) or 2 (IPv4 & IPv6). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	**Valid values are:** **0**-DHCP **1**-PPPoE **2**-Static IP Address
Example	static.network.internet_port.type = 1

Parameter- static.network.pppoe.user	Configuration File <MAC>.cfg
Description	Configures the PPPoE username when the Internet port type is configured as PPPoE and the IP address mode is configured as IPv4 or IPv4&IPv6. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	String within 32 characters
Example	static.network.pppoe.user = test.matrix

Parameter- static.network.pppoe.password	Configuration File <MAC>.cfg
Description	Configures the PPPoE password when the Internet port type is configured as PPPoE and the IP address mode is configured as IPv4 or IPv4&IPv6. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	String within 99 characters
Example	static.network.pppoe.password = matrix

Internet and PC Ports Transmission Methods

Internet Port Transmission Method

Parameter- static.network.internet_port.speed_duplex	Configuration File <MAC>.cfg
Description	Configures the transmission method of Internet port. **Note**: You can set the transmission speed to 1000Mbps/ Auto Negotiate to transmit in 1000Mbps if the IP phone is connected to the switch supports Gigabit Ethernet. We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	**Valid values are:** **0**-Auto negotiate **1**-Full duplex, 10Mbps **2**-Full duplex, 100Mbps **3**-Half duplex, 10Mbps **4**-Half duplex, 100Mbps **5**-Full Duplex 1000Mbps
Example	static.network.internet_port.speed_duplex = 0

PC Port Transmission Method

Parameter- static.network.pc_port.speed_duplex	Configuration File <MAC>.cfg

Description	Configures the transmission method of PC port. **Note**: It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiate). You can set the transmission speed to 1000Mbps/Auto Negotiate to transmit in 1000Mbps if the IP phone is connected to the switch supports Gigabit Ethernet. We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	**Valid values are:** **0**-Auto negotiate **1**-Full duplex, 10Mbps **2**-Full duplex, 100Mbps **3**-Half duplex, 10Mbps **4**-Half duplex, 100Mbps **5**-Full Duplex 1000Mbps
Example	static.network.pc_port.speed_duplex = 0

PC Port Mode

Parameter- static.network.pc_port.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the PC port. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	**Valid values are:** **0**-Disabled **1-**Auto Negotiation
Example	static.network.pc_port.enable = 1

Upgrading Firmware

Parameter- firmware.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of firmware. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	firmware.url = http://192.168.1.20/2.72.0.1.rom

Parameter- auto_provision.power_on	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to perform an auto provisioning process when powered on.
Format	Boolean
Default Value	1
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	auto_provision.power_on = 1

Parameter- auto_provision.repeat.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to check new configuration repeatedly.
Format	Boolean
Default Value	0
Range	**Valid values are:** **0**-Disabled **1-**Enabled
Example	auto_provision.repeat.enable = 0

Parameter- auto_provision.repeat.minutes	**Configuration File** <MAC>.cfg
Description	Configures the interval (in minutes) for the IP phone to check new configuration. **Note**: It works only if the parameter "auto_provision.repeat.enable" is set to 1 (Enabled).
Format	Integer
Default Value	1440
Range	1 to 43200
Example	auto_provision.repeat.minutes = 1000

Parameter- auto_provision.weekly.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to check new configuration weekly.
Format	Boolean
Default Value	0

Range	**Valid values are:** **0**-Disabled **1-**Enabled
Example	auto_provision.weekly.enable =0

Parameter- auto_provision.weekly.end_time	**Configuration File** <MAC>.cfg
Description	Configures the end time of the day for the IP phone to check new configuration weekly. **Note**: It works only if the parameter "auto_provision.weekly.enable" is set to 1 (Enabled).
Format	Time
Default Value	00:00
Range	00:00 to 23:59
Example	auto_provision.weekly.end_time = 21:30

Parameter- auto_provision.weekly.begin_time	**Configuration File** <MAC>.cfg
Description	Configures the begin time of the day for the IP phone to check new configuration weekly. **Note**: It works only if the parameter "auto_provision.weekly.enable" is set to 1(Enabled).
Format	Time
Default Value	00:00
Range	00:00 to 23:59
Example	auto_provision.weekly.begin_time = 01:30

Parameter- auto_provision.weekly.dayofweek	**Configuration File** <MAC>.cfg
Description	Configures the days of the week for the IP phone to check new configuration weekly.
Format	Integer
Default Value	0123456

Range	**Valid values are:** **0**-Sunday **1**-Monday **2**-Tuesday **3-**Wednesday **4**-Thursday **5**-Friday **6**-Saturday
Example	auto_provision.weekly.dayofweek = 0123456

Power Indicator LED

Parameter- phone_setting.common_power_led_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to be turned on.
Format	Boolean
Default Value	0
Range	**Valid values are**: **0**-Disabled (power indicator LED is off) **1**-Enabled (power indicator LED is solid green)
Example	phone_setting.common_power_led_enable = 1

Parameter- phone_setting.ring_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when the phone receives an incoming call. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Default Value	1
Range	**Valid values are**: **0**-Disabled (power indicator LED does not flash) **1**-Enabled (power indicator LED fast flashes (300ms) green)
Example	phone_setting.ring_power_led_flash_enable = 1

Parameter- phone_setting.mail_power_led_flash_enable	**Configuration File** <MAC>.cfg

Description	Enables or disables the power indicator LED to flash when the phone receives a voice mail or a text message. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Default Value	1
Range	**Valid values are**: **0**-Disabled (power indicator LED does not flash) **1**-Enabled (power indicator LED slow flashes (1000ms) green)
Example	phone_setting.mail_power_led_flash_enable = 0

Parameter- phone_setting.mute_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when a call is mute. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Format	Boolean
Default Value	0
Range	**Valid values are**: **0**-Disabled (power indicator LED does not flash) **1**-Enabled (power indicator LED fast flashes (300ms) green)
Example	phone_setting.mute_power_led_flash_enable = 1

Parameter- phone_setting.hold_and_held_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when a call is placed on hold or is held. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Format	Boolean
Default Value	0
Range	**Valid values are**: **0**-Disabled (power indicator LED does not flash) **1**-Enabled (power indicator LED fast flashes (500ms) green)

Example	phone_setting.hold_and_held_power_led_flash_enable = 0

Parameter- phone_setting.talk_and_dial_power_led_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to be turned on when the phone is busy. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Format	Boolean
Default Value	0
Range	**Valid values are**: **0**-Disabled (power indicator LED is off) **1**-Enabled (power indicator LED is solid green)
Example	phone_setting.talk_and_dial_power_led_enable = 1

Web Server Type

Parameter- static.wui.http_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the user to access web user interface of the IP phone using the HTTP protocol. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	static.wui.http_enable = 1

Parameter- static.network.port.http	**Configuration File** <MAC>.cfg

Description	Configures the HTTP port for the user to access web user interface of the IP phone using the HTTP protocol. **Note:** Please take care when choosing an alternate port. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	80
Range	1 to 65535
Example	static.network.port.http= 80

Parameter- static.wui.https_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the user to access web user interface of the IP phone using the HTTPS protocol. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	static.wui.https_enable= 1

Parameter- static.network.port.https	**Configuration File** <MAC>.cfg
Description	Configures the HTTPS port for the user to access web user interface of the IP phone using the HTTPS protocol. **Note:** Please take care when choosing an alternate port. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	443
Range	1 to 65535
Example	static.network.port.https= 443

Call Log

Parameter- features.save_call_history	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to save call log. If it is set to 0 (Disabled), the IP phone cannot log the placed calls, received calls, missed calls and the forwarded calls in the call log lists.
Format	Boolean
Default Value	1
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	features.save_call_history = 1

Call Waiting

Parameter- call_waiting.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables call waiting feature. If it is set to 0 (Disabled), a new incoming call is automatically rejected by the IP phone with a busy signal (configured by the parameter "features.normal_refuse_code") while during a call. If it is set to 1 (Enabled), the touch screen will present a new incoming call while during a call. In both cases, users can put an active call on hold to make outgoing calls.
Default Value	1
Range	**0 or 1** **0** - Disabled **1** - Enabled
Example	call_waiting.enable = 1

Parameter- call_waiting.tone	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to play the call waiting tone when the IP phone receives an incoming call during a call. If it is set to 1 (Enabled), the IP phone will perform an audible indicator when receiving a new incoming call during a call. **Note**: It works only if the value of the parameter "call_waiting.enable" is set to 1 (Enabled).
Default Value	1
Range	**0 or 1** **0**-Disabled **1**-Enabled
Example	call_waiting.tone = 1

Parameter- call_waiting.on_code	**Configuration File** <MAC>.cfg
Description	Configures the call waiting on code to activate the server-side call waiting feature. The IP phone will send the call waiting on code to the server when you activate call waiting feature on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.on_code = *71

Parameter- call_waiting.off_code	**Configuration File** <MAC>.cfg
Description	Configures the call waiting off code to deactivate the server-side call waiting feature. The IP phone will send the call waiting off code to the server when you deactivate call waiting feature on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.off_code = *72

Auto Redial

Parameter- auto_redial.enable	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to automatically redial the dialed number when the callee is temporarily unavailable. If it is set to 1 (Enabled), the IP phone will dial the previous dialed out number automatically when the dialed number is temporarily unavailable.
Default Value	0
Range	**0** -Disabled **1** -Enabled
Example	auto_redial.enable = 1

Parameter- auto_redial.interval	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to wait between redials. The IP phone redials the dialed number at regular intervals till the callee answers the call. **Note:** It works only if the value of the parameter "auto_redial.enable" is set to 1 (Enabled).
Default Value	10
Range	1 to 300
Example	auto_redial.interval = 10

Parameter- auto_redial.times	**Configuration File** <MAC>.cfg
Description	Configures the auto redial times when the callee is temporarily unavailable. The IP phone tries to redial the dialed number as many times as configured till the callee answers the call. **Note:** It works only if the value of the parameter "auto_redial.enable" is set to 1 (Enabled).
Default Value	10
Range	1 to 300
Example	auto_redial.times = 10

Auto Answer

Parameter- account.X.auto_answer	**Configuration File** <MAC>.cfg
Description	Enables or disables auto answer feature for account X. If it is set to 1 (Enabled), the IP phone can automatically answer an incoming call. X ranges from 1 to 16. **Note**: The IP phone cannot automatically answer the incoming call during a call even if auto answer is enabled.
Default Value	0
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	account.1.auto_answer = 1

Parameter- features.auto_answer_delay	**Configuration File** <MAC>.cfg
Description	Configures the delay time (in seconds) before the IP phone automatically answers an incoming call. **Note:** It works only if the value of the parameter "account.X.auto_answer" is set to 1 (Enabled).
Default Value	1
Range	1 to 4
Example	features.auto_answer_delay = 1

Parameter- features.auto_answer_tone.enable	**Configuration File** <MAC>.cfg

Description	Configure auto answer tone. Enables or disables the phone to play a warning tone when an incoming call is automatically answered. **Note:** For the call coming from a SIP account, it works only if the value of the parameter "account.X.auto_answer" is set to 1 (Enabled). It is also applicable to IP calls.
Default Value	1
Range	0 or 1
Example	features.auto_answer_tone.enable = 1

Call Completion

Parameter- features.call_completion_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables call completion feature. If a user places a call and the callee is temporarily unavailable to answer the call, call completion feature allows notifying the user when the callee becomes available to receive a call. If it is set to 1 (Enabled), the caller is notified when the callee becomes available to receive a call.
Default Value	0
Range	0 or 1 0 - Disabled 1 - Enabled
Example	features.call_completion_enable = 1

Anonymous Call

Parameter- account.X.anonymous_call (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Triggers the anonymous call feature to on or off for account X. If it is set to 1 (On), the IP phone will block its identity from showing up to the callee when placing a call. The callee's phone touch screen presents anonymous instead of the caller's identity.
Default Value	0

Range	0 or 1 **0**-Off **1**-On
Example	account.1.anonymous_call = 1

Parameter- account.X.send_anonymous_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the IP phone to send anonymous on/off code to activate/deactivate the server-side anonymous call feature for account X. If it is set to 0 (Off Code), the IP phone will send anonymous off code to the server when you deactivate the anonymous call feature. If it is set to 1 (On Code), the IP phone will send anonymous on code to the server when you activate the anonymous call feature.
Default Value	0
Range	0 or 1 **0**-Off Code **1**-On Code
Example	account.1.send_anonymous_code = 0

Parameter- account.X.anonymous_call_oncode (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the anonymous call on code to activate the server-side anonymous call feature for account X. **Note:** It works only if the value of the parameter "account.X.send_anonymous_code" is set to 1 (On Code).
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_oncode = *72

Parameter- account.X.anonymous_call_offcode (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the anonymous call off code to deactivate the server-side anonymous call feature for account X. **Note:** It works only if the value of the parameter "account.X.send_anonymous_code" is set to 0 (Off Code).
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_offcode = *73

Anonymous Call Rejection

Parameter- account.X.reject_anonymous_call (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Triggers the anonymous call rejection feature to on or off for account X. If it is set to 1 (On), the IP phone will automatically reject incoming calls from users enabled anonymous call feature. The anonymous user's phone touch screen presents "Anonymity Disallowed".
Default Value	0
Range	**0 or 1** **0** - Off **1** - On
Example	account.1.reject_anonymous_call = 1

Parameter- account.X.send_anonymous_rejection_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the IP phone to send anonymous rejection on/off code to activate/deactivate the server-side anonymous call rejection feature for account X. If it is set to 0 (Off Code), the IP phone will send anonymous rejection off code to the server when you deactivate the anonymous call rejection feature. If it is set to 1 (On Code), the IP phone will send anonymous rejection on code to the server when you activate the anonymous call rejection feature.
Default Value	0
Range	0 0- Off code 1- On code
Example	account.X.send_anonymous_rejection_code=0

Parameter- account.X.anonymous_reject_oncode (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the anonymous call rejection on code to activate the server-side anonymous call rejection feature for account X. **Note:** It works only if the value of the parameter "account.X.send_anonymous_rejection_code" is set to 1 (On Code).
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_oncode = *74

Busy Tone Delay

Parameter- account.X.anonymous_reject_offcode (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the anonymous call rejection off code to deactivate the server-side anonymous call rejection feature for account X. **Note:** It works only if the value of the parameter "account.X.send_anonymous_rejection_code" is set to 0 (Off Code).
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_offcode = *75

Parameter- features.busy_tone_delay	**Configuration File** <MAC>.cfg
Description	Configures the duration time (in seconds) for the busy tone. When one party releases the call, a busy tone is audible to the other party indicating that the call connection breaks. 0-0s 3-3s 5-5s If it is set to 3 (3s), a busy tone is audible for 3 seconds on the IP phone.
Default Value	0
Range	**0, 3 or 5** **0**-0s **3**-3s **5**-5s
Example	features.busy_tone_delay = 0

Return Code When Refuse

Parameter- features.normal_refuse_code	**Configuration File** <MAC>.cfg
Description	Configures a return code and reason of SIP response messages when the IP phone rejects an incoming call. A specific reason is displayed on the caller's phone touch screen. If it is set to 486 (Busy Here), the caller's phone touch screen will display the message "Busy Here" when the callee rejects the incoming call.

Default Value	486
Range	**404**-No Found **480**-Temporarily not available **486**-Busy here **603**-Decline
Example	features.normal_refuse_code = 486

180 Ring Workaround

Parameter- phone_setting.is_deal180	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message. If it is set to 1 (Enabled), the IP phone will resume and play the local ring-back tone upon a subsequent 180 message received.
Default Value	1
Range	0 or 1 **0** - Disabled **1** - Enabled
Example	phone_setting.is_deal180 = 1

Use Outbound Proxy in Dialog

Parameter- sip.use_out_bound_in_dialog	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send all SIP requests to the outbound proxy server forcibly in a dialog. If it is set to 0 (Disabled), only the new SIP request messages from the IP phone will be sent to the outbound proxy server in a dialog. If it is set to 1 (Enabled), all the SIP request messages from the IP phone will be forced to send to the outbound proxy server in a dialog. **Note:** It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled) and the outbound server address has been correctly configured on the phone.
Default Value	1

Range	0 or 1 **0** - Disabled **1** - Enabled
Example	sip.use_out_bound_in_dialog = 1

SIP Session Timer

Parameter- sip.timer_t1	**Configuration File** <MAC>.cfg
Description	Configures the SIP session timer T1 (in seconds). T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server.
Default Value	0.5
Range	0.5 to 10
Example	sip.timer_t1=0.5

Parameter- sip.timer_t2	**Configuration File** <MAC>.cfg
Description	Configures the SIP session timer T2 (in seconds). Timer T2 represents the maximum retransmitting time of any SIP request message.
Default Value	4
Range	2 to 40
Example	sip.timer_t2= 4

Parameter- sip.timer_t4	**Configuration File** <MAC>.cfg
Description	Configures the SIP session timer of T4 (in seconds). T4 represents the maximum duration a message will remain in the network.
Default Value	5
Range	2.5 to 60

Example	sip.timer_t4 = 5

Session Timer

Parameter- account.X.session_timer.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the session timer for account X. If it is set to 1 (Enabled), IP phone will send periodic UPDATE requests to refresh the session during a call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	account.1.session_timer.enable= 0

Parameter- account.X.session_timer.expires (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for refreshing the SIP session during a call for account X. For example, an UPDATE will be sent after 50% of its value has elapsed. If it is set to 1800 (1800s), the IP phone will refresh the session during a call before 900 seconds. **Note:** It works only if the value of the parameter "account.X.session_timer.enable" is set to 1 (Enabled).
Default Value	1800
Range	Integer from 30 to 7200
Example	account.1.session_timer.expires= 1800

Parameter- account.X.session_timer.refresher (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the function of the endpoint who initiates the SIP request for account X. **Note:** It works only if the value of the parameter "account.X.session_timer.enable" is set to 1 (Enabled).
Default Value	0
Range	0-UAC 1-UAS
Example	account.1.session_timer.refresher = 0

Call Hold

Parameter- features.play_hold_tone.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play a warning tone when there is a call on hold.
Default Value	1
Range	0 or 1 0 - Disabled 1 - Enabled
Example	features.play_hold_tone.enable = 1

Parameter- features.play_hold_tone.delay	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) at which the IP phone play a warning tone when there is a call on hold. If it is set to 30 (30s), the IP phone will play a warning tone every 30 seconds when there is a call on hold. **Note:** It works only if the value of the parameter "features.play_hold_tone.enable" is set to 1 (Enabled).
Default Value	30
Range	3 to 3600
Example	features.play_hold_tone.delay = 30

Parameter- sip.rfc2543_hold	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to use RFC 2543 (c=0.0.0.0) outgoing hold signaling. If it is set to 0 (Disabled), SDP media direction attributes (such as a=sendonly) per RFC 3264 is used when placing a call on hold. If it is set to 1 (Enabled), SDP media connection address c=0.0.0.0 per RFC 2543 is used when placing a call on hold.
Default Value	0
Range	0 or 1 0 - Disabled 1 - Enabled
Example	sip.rfc2543_hold = 0

Call Forward

Parameter- features.fwd_mode	**Configuration File** <MAC>.cfg
Description	Configures the call forward mode for the IP phone. If it is set to 0 (Phone), call forward feature is effective for the IP phone. If it is set to 1 (Custom), you can configure call forward feature for each or all accounts.
Default Value	0
Range	0 - Phone 1 - Custom
Example	features.fwd_mode = 0

Parameter- account.X.always_fwd.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Triggers always forward feature to on or off for account X. 0 - Off 1 - On If it is set to 1 (On), incoming calls to the account X are forwarded to the destination number (configured by the parameter “account.X.always_fwd.target”) immediately. **Note:** It works only if the value of the parameter “features.fwd_mode” is set to 1 (Custom).

Default Value	0
Range	0 - Off 1- On
Example	account.X.always_fwd.enable = 0

Parameter- account.X.always_fwd.target (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the destination number of the always forward for account X. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.always_fwd.target = 1003

Parameter- account.X.always_fwd.on_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the always forward on code to activate the server-side always forward feature for account X. The IP phone will send the always forward on code and the pre-configured destination number (configured by the parameter "account.X.always_fwd.target") to the server when you activate always forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.always_fwd.on_code = *72

Parameter- account.X.always_fwd.off_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the always forward off code to deactivate the server-side always forward feature for account X. The IP phone will send the always forward off code to the server when you deactivate always forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.always_fwd.off_code = *73

Parameter- account.X.busy_fwd.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Triggers busy forward feature to on or off for account X. 0-Off 1-On If it is set to 1 (On), incoming calls to the account X are forwarded to the destination number (configured by the parameter "account.X.busy_fwd.target") when the callee is busy. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	0
Range	0 or 1 0 - Off 1 - On
Example	account.X.busy_fwd.enable= 1

Parameter- account.X.busy_fwd.target (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the destination number of the busy forward for account X. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.target = 3602

Parameter- account.X.busy_fwd.on_code (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the busy forward on code to activate the server-side busy forward feature for account X. The IP phone will send the busy forward on code and the pre-configured destination number (configured by the parameter "account.X.busy_fwd.target") to the server when you activate busy forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.on_code = *74

Parameter- account.X.busy_fwd.off_code (X ranges from 1 to 16)	Configuration File <MAC>.cfg

Description	Configures the busy forward off code to deactivate the server-side busy forward feature for account X. The IP phone will send the busy forward off code to the server when you deactivate busy forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.off_code = *75

Parameter- account.X.timeout_fwd.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Triggers no answer forward feature to on or off for account X. 0-Off 1-On If it is set to 1 (On), incoming calls to the account X are forwarded to the destination number (configured by the parameter "account.X.timeout_fwd.target") after a period of ring time. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	0
Range	0 or 1 0 - Off 1 - On
Example	account.X.timeout_fwd.enable=1

Parameter- account.X.timeout_fwd.target (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the destination number of the no answer forward for account X. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.timeout_fwd.target = 3603

Parameter- account.X.timeout_fwd.timeout (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures ring times (N) to wait before forwarding incoming calls for account X. Incoming calls will be forwarded when not answered after N*6 seconds. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	2
Range	0 to 20
Example	account.X.timeout_fwd.timeout=2

Parameter- account.X.timeout_fwd.on_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the no answer forward on code to activate the server-side no answer forward feature for account X. The IP phone will send the no answer forward on code and the pre-configured destination number (configured by the parameter "account.X.timeout_fwd.target") to the server when you activate no answer forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank

Range	String within 32 characters
Example	account.X.timeout_fwd.on_code =*76

Parameter- account.X.timeout_fwd.off_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the no answer forward off code to deactivate the server-side no answer forward feature for account X. The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature for account X on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 1 (Custom).
Default Value	Blank
Range	String within 32 characters
Example	account.1.timeout_fwd.off_code = *77

Parameter- features.forward.emergency.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the incoming calls from some authorized numbers not to be forwarded when the call forward feature is enabled.
Default Value	0
Range	0 or 1 0-Disabled 1-Enabled
Example	features.forward.emergency.enable= 1

Parameter- features.forward.emergency.authorized_number	**Configuration File** <MAC>.cfg

Description	Configures the authorized numbers not to be forwarded even if call forward feature is enabled. Multiple numbers are separated by commas. **Note:** It works only if the value of the parameter "features.forward.emergency.enable" is set to 1 (Enabled).
Default Value	Blank
Range	String within 511 characters
Example	features.forward.emergency.authorized_number = 123,124

Parameter- forward.always.enable	**Configuration File** <MAC>.cfg
Description	Triggers the always forward feature to on or off for the IP phone. If it is set to 1 (On), incoming calls are forwarded to the destination number (configured by the parameter "forward.always.target") immediately. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone)..
Default Value	0
Range	0 or 1 0-Off 1-On
Example	forward.always.enable=1

Parameter- forward.always.target	**Configuration File** <MAC>.cfg
Description	Configures the destination number of the always forward for the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.target = 3601

Parameter- forward.always.on_code	**Configuration File** <MAC>.cfg
Description	Configures the always forward on code to activate the server-side always forward feature. The IP phone will send the always forward on code and the pre-configured destination number (configured by the parameter "forward.always.target") to the server when you activate always forward feature on the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.always.on_code = *72

Parameter- forward.always.off_code	**Configuration File** <MAC>.cfg
Description	Configures the always forward off code to deactivate the server-side always forward feature. The IP phone will send the always forward off code to the server when you deactivate always forward feature on the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.always.off_code = *73

Parameter- forward.busy.enable	**Configuration File** <MAC>.cfg

Description	Triggers the busy forward feature to on or off for the IP phone. If it is set to 1 (On), incoming calls are forwarded to the destination number (configured by the parameter "forward.busy.target") when the callee is busy. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	0
Range	0 or 1 0 - Off 1 - On
Example	forward.busy.enable = 1

Parameter- forward.busy.target	**Configuration File** <MAC>.cfg
Description	Configures the destination number of the busy forward for the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.busy.target = 3602

Parameter- forward.busy.on_code	**Configuration File** <MAC>.cfg
Description	Configures the busy forward on code to activate the server-side busy forward feature. The IP phone will send the busy forward on code and the pre-configured destination number (configured by the parameter "forward.busy.target") to the server when you activate busy forward feature on the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters

Example	forward.busy.on_code = *74

Parameter- forward.busy.off_code	**Configuration File** <MAC>.cfg
Description	Configures the busy forward off code to deactivate the server-side busy forward feature. The IP phone will send the busy forward off code to the server when you deactivate busy forward feature on the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.busy.off_code = *75

Parameter- forward.no_answer.enable	**Configuration File** <MAC>.cfg
Description	Triggers the no answer forward feature to on or off for the IP phone. If it is set to 1 (On), incoming calls are forwarded to the destination number (configured by the parameter "forward.no_answer.target") after a period of ring time. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	0
Range	0 or 1 0 - Off 1 - On
Example	forward.no_answer.enable =1

Parameter- forward.no_answer.target	**Configuration File** <MAC>.cfg

Description	Configures the destination number of the no answer forward for the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.target = 3603

Parameter- forward.no_answer.timeout	**Configuration File** <MAC>.cfg
Description	Configures ring times (N) to wait before forwarding incoming calls. Incoming calls will be forwarded when not answered after N*6 seconds. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	2
Range	0 to 20
Example	forward.no_answer.timeout= 2

Parameter- forward.no_answer.on_code	**Configuration File** <MAC>.cfg
Description	Configures the no answer forward on code to activate the server-side no answer forward feature. The IP phone will send the no answer forward on code and the pre-configured destination number (configured by the parameter "forward.no_answer.target") to the server when you activate no answer forward feature on the IP phone. **Note:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.on_code = *76

Parameter- forward.no_answer.off_code	**Configuration File** <MAC>.cfg
Description	Configures the no answer forward off code to deactivate the server-side no answer forward feature. The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature on the IP phone. N**ote:** It works only if the value of the parameter "features.fwd_mode" is set to 0 (Phone).
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.off_code = *77

Parameter- features.fwd_diversion_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to present the diversion information when an incoming call is forwarded to your IP phone.
Default Value	1
Range	0 or 1 0 - Disabled 1 - Enabled
Example	features.fwd_diversion_enable= 1

Parameter- forward.international.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to forward incoming calls to international numbers (the prefix is 00).
Default Value	1
Range	0 or 1 0 - Disabled 1 - Enabled
Example	forward.international.enable=1

Call Transfer

Parameter- transfer.blind_tran_on_hook_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to complete the blind transfer through on-hook besides tapping the Transfer soft key or TRANSFER/TRAN key. **Note:** Blind transfer means transfer a call directly to another party without consulting.
Default Value	1
Range	0-Disabled 1-Enabled
Example	transfer.blind_tran_on_hook_enable= 1

Parameter- transfer.on_hook_trans_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to complete the semi-attended/attended transfer through on-hook besides tapping the Transfer soft key or TRANSFER/ TRAN key. **Note:** Semi-attended transfer means transfer a call after hearing the ringback tone; Attended transfer means transfer a call with prior consulting.
Default Value	1
Range	0-Disabled 1-Enabled
Example	transfer.on_hook_trans_enable = 1

Parameter- transfer.semi_attend_tran_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the transfer-to party's phone not to prompt a missed call on the touch screen before displaying the caller ID when completing a semi-attended transfer. **Note:** Semi-attended transfer means transfer a call after hearing the ringback tone.
Default Value	1

Range	0-Disabled 1-Enabled
Example	transfer.semi_attend_tran_enable= 1

Local Conference

Parameter- account.X.conf_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the network conference type for SIP account X. If it is set to 0 (Local Conference), conferences are set up on the IP phone locally. If it is set to 2 (Network Conference), conferences are set up by the server.
Default Value	0
Range	0-Local Conference 2-Network Conference
Example	account.1.conf_type= 0

Network Conference

Parameter- account.X.conf_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the network conference type for account X. If it is set to 0 (Local Conference), conferences are set up on the IP phone locally. If it is set to 2 (Network Conference), conferences are set up by the server.
Default Value	0
Range	0-Local Conference 2-Network Conference
Example	account.1.conf_type = 0

Parameter- account.X.conf_uri (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the network conference URI for account X. **Note:** It works only if the value of the parameter "account.X.conf_type" is set to 2 (Network Conference).
Default Value	Blank
Range	SIP URI within 511 characters
Example	account.1.conf_uri = conference@example.com

Transfer on Conference Hang Up

Parameter- transfer.tran_others_after_conf_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to transfer the local conference call to the other two parties after the conference initiator drops the local conference call. If it is set to 0 (Disabled), all parties are disconnected when the conference initiator drops the conference call. If it is set to 1 (Enabled), the other two parties remain connected when the conference initiator drops the conference call. **Note:** It works only if the value of parameter "account.X.conf_type" is set to 0 (Local Conference).
Default Value	0
Range	0-Disabled 1-Enabled
Example	transfer.tran_others_after_conf_enable=1

Directed Call Pickup

Parameter- account.X.direct_pickup_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the directed call pickup code for account X. **Note:** The directed call pickup code configured on a per-line basis ("account.X.direct_pickup_code") takes precedence over that configured on a phone basis ("features.pickup.direct_pickup_code").
Default Value	Blank
Range	String within 32 characters
Example	account.1.direct_pickup_code = *68

Parameter- features.pickup.direct_pickup_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display the **Dpickup** key when the IP phone is on the pre-dialing screen.
Default Value	0
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	features.pickup.direct_pickup_enable= 0

Parameter- features.pickup.direct_pickup_code	**Configuration File** <MAC>.cfg
Description	Configures the directed call pickup code on a phone basis. **Note:** The directed call pickup code configured on a per-line basis ("account.X.direct_pickup_code") takes precedence over that configured on a phone basis ("features.pickup.direct_pickup_code").
Default Value	Blank
Range	String within 32 characters
Example	features.pickup.direct_pickup_code = *97

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a directed call pickup key on the IP phone. The digit 9 stands for the key type Direct Pickup. **For line keys:** X ranges from 1 to 27. **For programable keys:** X ranges from 12 to 14.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	9
Example	linekey.2.type = 9

Parameter- linekey.X.line/ programablekey.X.line	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the directed call pickup key. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	1-16 for lines 1-16, 1 for programable keys
Range	1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Example	llinekey.2.line = 1

Parameter- linekey.X.value/programablekey.X.value	Configuration File <MAC>.cfg
Description	Configures the directed call pickup feature code followed by the monitored extension. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.value = *971008

Parameter- linekey.X.label	Configuration File <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.label = *971008

Group Call Pickup

Parameter- features.pickup.group_pickup_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to display the **GPickup** key when the IP phone is on the pre-dialing screen.
Default Value	0
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	features.pickup.group_pickup_enable= 0

Parameter- account.X.group_pickup_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the group pickup code for account X. **Note:** The group call pickup code configured on a per-line basis (configured by the parameter "account.X.group_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.group_pickup_code").
Default Value	Blank
Range	String within 32 characters
Example	account.1.group_pickup_code = *69

Parameter- features.pickup.group_pickup_code	**Configuration File** <MAC>.cfg
Description	Configures the group call pickup code on a phone basis. **Note:** The group call pickup code configured on a per-line basis (configured by the parameter "account.X.group_pickup_code") takes precedence over that configured on a phone basis (configured by the parameter "features.pickup.group_pickup_code").
Default Value	Blank
Range	String within 32 characters
Example	features.pickup.group_pickup_code = *98

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg

Description	Configures a DSS key as a group call pickup key on the IP phone. The digit 23 stands for the key type Group Pickup. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	23
Example	linekey.2.type = 23

Parameter- linekey.X.line/ programablekey.X.line	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the group call pickup key. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	1-16 for lines 1-16, 1 for programable keys
Range	1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Example	linekey.2.line = 1

Parameter- linekey.X.value/ programablekey.X.value	Configuration File <MAC>.cfg
Description	Configures the group call pickup feature code. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.value = *98

Parameter- linekey.X.label	Configuration File <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key.(Optional) For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.label = *98

Dialog Info Call Pickup

Parameter- account.X.dialoginfo_callpickup (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to pick up a call according to the Replaces header of the INVITE message for account X. If it is set to 1 (Enabled), call pickup is implemented through SIP signals.
Default Value	0
Range	0 or 1 0 - Disabled 1 - Enabled
Example	account.X.dialoginfo_callpickup=0

Calling Line Identification Presentation (CLIP)

Parameter- account.X.cid_source (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the presentation of the caller identity when receiving an incoming call for account X. 0 - FROM 1 - PAI 2 - PAI-FROM 3 - RPID-PAI-FROM 4 - PAI-RPID-FROM 5 - RPID-FROM
Default Value	0
Range	0 to 5
Example	account.1.cid_source = 0

Parameter- account.X.cid_source_privacy (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to process Privacy header field in the SIP message for account X. If it is set to 0 (Disabled), the IP phone doesn't process Privacy header. If it is set to 1 (Enabled), the caller identification information will be hidden and the IP phone touch screen presents anonymous if there is a Privacy: id in the INVITE request.
Default Value	1
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	account.X.cid_source_privacy = 1

Parameter- account.X.cid_source_ppi (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to process the P-Preferred-Identity (PPI) header for caller identity presentation when receiving an incoming call for account X. If it is set to 0 (Disabled), the IP phone doesn't process P-Preferred-Identity (PPI) header. If it is set to 1 (Enabled), the IP phone presents the caller identification from the P-Asserted-Identity (PPI) header.
Default Value	1
Range	**0 or 1** 0 - Disabled 1 - Enabled
Example	account.X.cid_source_ppi = 1

Connected Line Identification Presentation

Parameter- account.X.cp_source (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the presentation of the callee's identity for account X. 0 - PAI-RPID 1 - Dialed Digits 2 - RFC 4916 When the RFC 4916 is enabled on the IP phone, the caller sends the SIP request message which contains the from-change tag in the Supported header. The caller then receives an UPDATE message from the callee, and displays the identity in the "From" header.
Default Value	0
Range	0 to 2
Example	account.1.cp_source = 0

Parameter- account.X.cid_source_privacy (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to process Privacy header field in the SIP message for account X. If it is set to 0 (Disabled), the IP phone doesn't process Privacy header. If it is set to 1 (Enabled), the caller identification information will be hidden and the IP phone touch screen presents anonymous if there is a Privacy: id in the INVITE request.
Default Value	1
Range	0 or 1 0 - Disabled 1 - Enabled
Example	account.X.cid_source_privacy = 1

DTMF

Parameter- account.X.dtmf.type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the DTMF type for account X. 0-INBAND 1-RFC 2833 2-SIP INFO 3-RFC2833 + SIP INFO If it is set to 0 (INBAND), DTMF digits are transmitted in the voice band. If it is set to 1 (RFC 2833), DTMF digits are transmitted by RTP Events compliant to RFC 2833. If it is set to 2 (SIP INFO), DTMF digits are transmitted by the SIP INFO messages. If it is set to 3 (RFC2833 + SIP INFO), DTMF digits are transmitted by RTP Events compliant to RFC 2833 and the SIP INFO messages.
Default Value	1
Range	0 to 3
Example	account.1.dtmf.type = 1

Parameter- account.X.dtmf.dtmf_payload (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the value of DTMF payload for account X. Note: It works only if the value of parameter "account.X.dtmf.type" is set to 1 (RFC2833) or 3 (RFC2833 + SIP INFO).
Default Value	101
Range	96 to 127
Example	account.1.dtmf.dtmf_payload = 101

Parameter- account.X.dtmf.info_type (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the DTMF info type. 1-DTMF-Relay 2-DTMF 3-Telephone-Event **Note:** It works only if the value of parameter "account.X.dtmf.type" is set to 2 (SIP INFO) or 3 (RFC2833 + SIP INFO).
Default Value	1
Range	1 to 3
Example	account.1.dtmf.info_type = 0

Parameter- features.dtmf.repetition	Configuration File <MAC>.cfg
Description	Configures the repetition times for the IP phone to send the end RTP Event packet during an active call.
Default Value	3
Range	1 to 3
Example	features.dtmf.repetition = 3

Parameter- features.dtmf.duration	Configuration File <MAC>.cfg
Description	Configures the duration time (in milliseconds) for DTMF. **Note:** If the time interval to between two DTMF digits is less than this value, two or more same DTMF digits could be identified as one DTMF digit. This may cause the loss of one or more DTMF digits. For example, 2662 may be identified as 262. If so, you can modify the value of this parameter to a little lower than the default value.
Default Value	100
Range	0 to 300
Example	features.dtmf.repetition = 100

Parameter- features.dtmf.volume	Configuration File <MAC>.cfg
Description	Configures the frequency level of DTMF digits (in db).
Default Value	-10
Range	-33 to 0
Example	features.dtmf.volume= -10

Suppress DTMF Display

Parameter- features.dtmf.hide	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to suppress the display of DTMF digits during an active call. If it is set to 1 (Enabled), the DTMF digits are displayed as asterisks.
Default Value	0

Range	0 or 1 0 - Disabled 1 - Enabled
Example	features.dtmf.hide = 0

Parameter- features.dtmf.hide_delay	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display the DTMF digits for a short period before displaying asterisks during an active call. **Note:** It works only if the value of the parameter "features.dtmf.hide" is set to 1 (Enabled).
Default Value	0
Range	0 or 1 **0** - Disabled **1** - Enabled
Example	features.dtmf.hide_delay = 0

Transfer via DTMF

Parameter- features.dtmf.replace_tran	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send DTMF sequences for transfer function when tapping the Transfer soft key or pressing TRANSFER/TRAN key. If it is set to 0 (Disabled), the IP phone will perform the transfer as normal when tapping the Transfer soft key or pressing TRANSFER/TRAN key during a call. If it is set to 1 (Enabled), the IP phone will transmit the designated DTMF digits to the server for performing call transfer when tapping the Transfer soft key or pressing TRANSFER/TRAN key during a call.
Default Value	0
Range	0 or 1 0 - Disabled 1 - Enabled

Example	features.dtmf.replace_tran = 0

Parameter- features.dtmf.transfer	**Configuration File** <MAC>.cfg
Description	Configures the DTMF digits to be transmitted to perform call transfer. Valid values are: 0-9, *, # and A-D. **Note:** It works only if the value of the parameter "features.dtmf.replace_tran" is set to 1 (Enabled).
Default Value	Blank
Range	String within 32 characters
Example	features.dtmf.transfer = 123

Play Local DTMF Tone

Parameter- features.play_local_dtmf_tone_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play a local DTMF tone. If it is set to 1 (Enabled), you can hear the DTMF tone when pressing the IP phone's keypad during a call.
Default Value	1
Range	0 or 1 0-Disabled 1-Enabled
Example	features.play_local_dtmf_tone_enable = 1

Incoming Intercom calls

Parameter- features.intercom.allow	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to answer an incoming intercom call. If it is set to 0 (Disabled), the IP phone will reject incoming intercom calls and sends a busy signal (configured by the parameter“features.normal_refuse_code”) to the caller. If it is set to 1 (Enabled), the IP phone will automatically answer an incoming intercom call.
Default Value	1
Range	0-Disabled 1-Enabled
Example	features.intercom.allow = 1

Parameter- features.intercom.mute	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to mute the microphone when answering an intercom call. If it is set to 1 (Enabled), the microphone is muted for intercom calls, and then the other party cannot hear you. **Note:** It works only if the value of the parameter “features.intercom.allow” is set to 1(Enabled).
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.intercom.mute=0

Parameter- features.intercom.tone	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play a warning tone when answering an intercom call. **Note:** It works only if the value of the parameter “features.intercom.allow” is set to 1(Enabled).
Default Value	1
Range	**0**-Disabled **1**-Enabled

Example	features.intercom.tone = 1

Parameter- features.intercom.barge	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to answer an incoming intercom call while there is already an active call on the IP phone. If it is set to 0 (Disabled), the IP phone will handle an incoming intercom call like a waiting call while there is already an active call on the IP phone. If it is set to 1 (Enabled), the IP phone will automatically answer the intercom call while there is already an active call on the IP phone and place the active call on hold. **Note:** It works only if the values of parameters "features.intercom.allow" and "call_waiting.enable" are set to 1 (Enabled).
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.intercom.barge = 0

Outgoing Intercom Calls

Parameter- features.intercom.subscribe.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables intercom subscription for the IP phone.
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.intercom.subscribe.enable = 0

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg

Description	Configures a DSS key as an intercom key. The digit **14** stands for the key type Intercom. For line keys: X ranges from 1 to 27.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Range	**14**
Example	linekey.2.type = 14

Parameter- inekey.X.line	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the intercom key. For line keys: X ranges from 1 to 27.
Default Value	1-16 for lines 1-16
Range	1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Example	linekey.2.line = 1

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the intercom key. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.value = 1008

Distinctive Ring Tones

Parameter- account.X.alert_info_url_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to download the ring tone from the URL contained in the Alert-Info header for account X.
Default Value	1
Range	**0 or 1** **0** - Disabled **1** - Enabled
Example	account.1.alert_info_url_enable= 1

Parameter- features.alert_info_tone	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to map the keywords in the Alert-info header to the specified Bellcore ring tones.
Default Value	0
Range	0 or 1 **0** - Disabled **1** - Enabled
Example	features.alert_info_tone= 0

Parameter- distinctive_ring_tones.alert_info.X.text (X ranges from 1 to 10)	**Configuration File** <MAC>.cfg
Description	Configures the internal ringer text to map the keywords contained in the Alert-Info header.
Default Value	Blank
Range	String within 32 characters
Example	distinctive_ring_tones.alert_info.1.text = Family

Parameter- distinctive_ring_tones.alert_info.X.ringer (X ranges from 1 to 10)	Configuration File <MAC>.cfg
Description	Configures the desired ring tones for each internal ringer text. The value ranges from 1 to 10, the digit stands for the appropriate ring tone. 1-Ring1.wav 2-Ring2.wav 3-Ring3.wav 4-Ring4.wav 5-Ring5.wav 6-Ring6.wav 7-Ring7.wav 8-Ring8.wav 9-Silent.wav 10-Splash.wav
Default Value	1
Range	1 to 10
Example	distinctive_ring_tones.alert_info.1.ringer = 1

Tones

Parameter- voice.tone.country	Configuration File <MAC>.cfg
Description	Configures the country tone for the IP phone.
Default Value	Custom

Range	• Custom • Australia • Austria • Brazil • Belgium • Chile • China • Czech • Czech ETSI • Denmark • Finland • France • Germany • Great Britain • Greece • Hungary • Lithuania • India • Italy • Japan • Mexico • New • Zealand • Netherlands • Norway • Portugal • Spain • Switzerland • Sweden • Russia • United States.
Example	voice.tone.country = Custom

Parameter- voice.tone.dial	Configuration File <MAC>.cfg
Description	Customizes the dial tone. tonelist = element[,element] [,element]… Where element = [!]Freq1[+Freq2][+Freq3][+Freq4] / Duration **Freq:** the frequency of the tone (ranges from 200 to 4000Hz). If it is set to 0Hz, it means the tone is not played. A tone is comprised of at most four different frequencies. **Duration:** the duration (in milliseconds) of the dial tone, ranges from 0 to 30000ms. You can configure at most eight different tones for one condition, and separate them by commas. (e.g., 250/200,0/1000,200+300/500,200+500+800+1500/1000). If you want the IP phone to play tones once, add an exclamation mark "!" before tones (e.g., !250/200,0/1000,200+300/500,200+500+800+1500/1000). **Note:** It works only if the value of the parameter "voice.tone.country" is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.secondary_dial	Configuration File <MAC>.cfg
Description	Customizes the secondary dial tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". **Note:** It works only if the value of the parameter "voice.tone.country" is set to Custom. If you want to disable this warning tone, set it to 0.
Default Value	350+440/3000
Range	String

Example	voice.tone.secondary_dial = 350+440/3000

Parameter- voice.tone.ring	**Configuration File** <MAC>.cfg
Description	Customizes the ringback tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.busy	**Configuration File** <MAC>.cfg
Description	Customizes the tone when the callee is busy. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.congestion	**Configuration File** <MAC>.cfg

Description	Customizes the tone when the network is congested. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.callwaiting	**Configuration File** <MAC>.cfg
Description	Customizes the call waiting tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.dialrecall	**Configuration File** <MAC>.cfg
Description	Customizes the call back tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.info	**Configuration File** <MAC>.cfg
Description	Customizes the info tone. The phone will play the info tone with the special information, for example, the number you are calling is not in service. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.stutter	**Configuration File** <MAC>.cfg
Description	Customizes the tone when the IP phone receives a voice mail. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Parameter- voice.tone.autoanswer	**Configuration File** <MAC>.cfg

Description	Customizes the warning tone for auto answer. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”. **Note:** It works only if the value of the parameter “voice.tone.country” is set to Custom.
Default Value	Blank
Range	String
Example	NA

Remote Phone Book

Parameter- remote_phonebook.data.X.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the remote XML phonebook. X ranges from 1 to 5.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml

Parameter- remote_phonebook.data.X.name	**Configuration File** <MAC>.cfg
Description	Configures the display name of the remote phone book item.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.data.1.name = yl01

Parameter- features.remote_phonebook.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to perform a remote phone book search for outgoing/incoming calls.
Format	Boolean
Default Value	0

Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	features.remote_phonebook.enable = 1

Parameter- features.remote_phonebook.flash_time	**Configuration File** <MAC>.cfg
Description	Configures how often to refresh the local cache of the remote phonebook. If it is set to 3600, the IP phone will refresh the local cache of the remote phone book every 3600 seconds.
Format	Integer
Default Value	21600
Range	120 to 2592000
Example	features.remote_phonebook.flash_time = 21600

Message Waiting Indicator

Parameter- account.X.subscribe_mwi (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to subscribe the message waiting indicator for account X. If it is set to 1 (Enabled), the IP phone will send a SUBSCRIBE message to the server for message-summary updates. If it is set to 0 (Disabled), the server automatically sends a message-summary NOTIFY in a new dialog each time the MWI status changes. (This requires server support)
Default Value	0
Value	**0**-Disabled **1**-Enabled
Example	account.1.subscribe_mwi=0

Parameter- account.X.subscribe_mwi_expires (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures MWI subscribe expiry time (in seconds) for account X. The IP phone is able to successfully refresh the SUBSCRIBE for message-summary events before expiration of the subscription dialog. **Note:** It works only if the value of the parameter "account.X.subscribe_mwi" is set to 1 (Enabled).
Default Value	3600
Value	Integer from 0 to 84600
Example	account.1.subscribe_mwi_expires= 3600

Parameter- account.X.subscribe_mwi_to_vm (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to subscribe the message waiting indicator to the voice mail number for account X. If it is set to 0 (Disabled), the IP phone will subscribe the message waiting indicator to the account X. **Note:** It works only if the value of the parameter "account.X.subscribe_mwi" is set to 1(Enabled) and "voice_mail.number.X" is configured.
Default Value	0
Value	0-Disabled 1-Enabled
Example	account.1.subscribe_mwi_to_vm= 0

Parameter- voice_mail.number.X (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the voice mail number for account X.
Default Value	Blank
Value	String within 99 characters
Example	voice_mail.number.1= 1234

Parameter- account.X.display_mwi.enable (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to present audio and visual MWI when receiving new voice messages. **Note:** It always works at the time of Unsolicited MWI; at the time of solicited MWI, MWI subscription feature should be configured in advance. To present audio MWI, you also need to set the value of the parameter "features.voice_mail_tone_enable" to 1 (Enabled) in advance.
Default Value	1
Value	0-Disabled 1-Enabled
Example	account.1.display_mwi.enable= 1

Multicast Paging

Sending RTP Stream

Parameter- multicast.codec	Configuration File <MAC>.cfg
Description	Configures the codec of multicast paging.
Default Value	G722
Range	PCMU PCMA G729 G722
Example	multicast.codec = G722

Parameter- multicast.paging_address.X.ip_address (X ranges from 1 to 31)	Configuration File <MAC>.cfg

Description	Configures the IP address and port number of the multicast paging group in the paging list. It will be displayed on the touch screen when placing the multicast paging call. **Note:** The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Default Value	Blank
Range	String
Example	multicast.paging_address.1.ip_address = 224.5.6.20:10008 multicast.paging_address.2.ip_address = 224.1.6.25:1001

Parameter- multicast.paging_address.X.label (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg
Description	Configures the name of the multicast paging group to be displayed in the paging list. It will be displayed on the touch screen when placing the multicast paging calls.
Default Value	Blank
Range	String
Example	multicast.paging_address.1.label = Product multicast.paging_address.2.label = Sales

Parameter- multicast.paging_address.X.channel (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg

Description	Configures the channel of the multicast paging group in the paging list. If it is set to 0, all the Matrix IP phones running old firmware version (old paging mechanism) or prior or Matrix IP phones listens to channel 0 or third-party available devices (e.g., Cisco IP phones) in the paging group can receive the RTP stream. If it is set to 1 to 25, the Polycom or Matrix IP phones preconfigured to listen to the channel can receive the RTP stream. It it is set to 26 to 30, the matrix IP phones preconfigured to listen to the channel can receive the RTP stream.
Default Value	0
Range	Integer from 0 to 30
Example	multicast.paging_address.1.channel = 3 multicast.paging_address.2.channel = 5

Receiving RTP Stream

Parameter- multicast.listen_address.X.ip_address (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg
Description	Configures the multicast address and port number that the IP phone listens to. **Note:** The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Default Value	Blank
Range	IP address: port
Example	multicast.listen_address.1.ip_address = 224.5.6.20:10008

Parameter- multicast.listen_address.X.label (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg

Description	Configures the label to be displayed on the touch screen when receiving the multicast paging calls (Optional).
Default Value	Blank
Range	String within 99 characters
Example	multicast.listen_address.1.label = Paging1

Parameter- multicast.listen_address.X.channel (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg
Description	Configures the channel that the IP phone listens to. If it is set to 0, the IP phone can receive an RTP stream of the pre-configured multicast address from the IP phones running old firmware version (old paging mechanism), from the IP phones listen to the channel 0, or from the available third-party devices (e.g., Cisco IP phones). If it is set to 1 to 25, the IP phone can receive an RTP stream of the pre-configured multicast address on the channel 1 to 25 respectively from Matrix or Polycom IP phones. It it is set to 26 to 30, the IP phone can receive the RTP stream of the pre-configured multicast address on the channel 26 to 30 respectively from Matrix IP phones.
Default Value	0
Range	Integer from 0 to 30
Example	multicast.listen_address.1.channel = 2

Parameter- multicast.listen_address.X.volume (X ranges from 1 to 31)	**Configuration File** <MAC>.cfg

Description	Configures the volume of the speaker when receiving the multicast paging calls. If it is set to 0, the current volume of the speaker takes effect. The volume of the speaker can be adjusted by pressing the Volume key in advance when the phone is during a call. You can also adjust the volume of the speaker during the paging call. If it is set to 1 to 15, the configured volume takes effect and the current volume of the speaker will be ignored. You are not allowed to adjust the volume of the speaker during the paging call.
Default Value	0
Range	Integer from 0 to 15
Example	multicast.listen_address.1.volume = 1

Parameter- multicast.receive.use_speaker	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to always use the speaker as the audio device when receiving the multicast paging calls. If it is set to 0 (Disabled), the engaged audio device will be used when receiving the multicast paging calls.
Default Value	0
Range	0-Disabled 1-Enabled
Example	multicast.receive.use_speaker= 0

Parameter- multicast.receive_priority.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to handle the incoming multicast paging calls when there is an active multicast paging call on the IP phone. If it is set to 0 (Disabled), the IP phone will ignore the incoming multicast paging calls when there is an active multicast paging call on the IP phone. If it is set to 1 (Enabled), the IP phone will receive the incoming multicast paging call with a higher or equal priority and ignore that with a lower priority.
Default Value	1

Range	0-Disabled 1-Enabled
Example	multicast.receive_priority.enable= 1

Parameter- multicast.receive_priority.priority	**Configuration File** <MAC>.cfg
Description	Configures the priority of the voice/video call (a normal phone call rather than a multicast paging call) in progress. 1 is the highest priority, 31 is the lowest priority. 0-Disabled 1-1 2-2 3-3 … 31-31 If it is set to 0 (Disabled), all incoming multicast paging calls will be automatically ignored when a voice/video call is in progress. If it is not set to 0 (Disabled), the IP phone will receive the incoming multicast paging call with a higher or same priority than this value and ignore that with a lower priority than this value when a voice/video call is in progress.
Default Value	31
Range	Integer from 0 to 31
Example	multicast.receive_priority.priority=31

Action URL

Parameter- action_url.setup_completed	**Configuration File** <MAC>.cfg

Description	Configures the action URL the IP phone sends after startup. The value format is: http(s)://<serverIPAddress>/help.xml? variable name=variable value. **Valid variable values are:** • $mac • $ip • $model • $firmware • $active_url • $active_user • $active_host • $local • $remote • $display_local • $display_remote • $call_id • $callerID • $calledNumber
Default Value	Blank
Range	URL within 511 characters
Example	action_url.setup_completed = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.registered	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends after an account is registered.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.registered = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.unregistered	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends after an account is unregistered.

Default Value	Blank
Range	URL within 511 characters
Example	action_url.unregistered = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.register_failed	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends after a register failed.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.register_failed = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.off_hook	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when off hook.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.off_hook = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.on_hook	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when on hook.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.on_hook = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.incoming_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when receiving an incoming call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.incoming_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.outgoing_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when placing a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.outgoing_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.call_established	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when establishing a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.call_established = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.dnd_on	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when DND feature is enabled.
Default Value	Blank
Range	URL within 511 characters

Example	action_url.dnd_on = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.dnd_off	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when DND feature is disabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.dnd_off = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.always_fwd_on	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when always forward feature is enabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.always_fwd_on = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.always_fwd_off	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when always forward feature is disabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.always_fwd_off = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.busy_fwd_on	**Configuration File** <MAC>.cfg

Description	Configures the action URL the IP phone sends when busy forward feature is enabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.busy_fwd_on = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.busy_fwd_off	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when busy forward feature is disabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.busy_fwd_off = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.no_answer_fwd_on	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when no answer forward feature is enabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.no_answer_fwd_on = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.no_answer_fwd_off	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when no answer forward feature is disabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.no_answer_fwd_off = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.transfer_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when performing a transfer.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.transfer_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.blind_transfer_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when performing a blind transfer..
Default Value	Blank
Range	URL within 511 characters
Example	action_url.blind_transfer_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.attended_transfer_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when performing an attended/semi-attended transfer.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.attended_transfer_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.hold	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when placing a call on hold.

Default Value	Blank
Range	URL within 511 characters
Example	action_url.hold = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.unhold	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when resuming a hold call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.unhold = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.held	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when a call is held.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.held = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.unheld	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when a call being held is resumed.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.unheld = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.mute	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when muting a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.mute = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.unmute	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when un-muting a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.unmute = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.missed_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when missing a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.missed_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.call_terminated	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when terminating a call.
Default Value	Blank
Range	URL within 511 characters

Example	action_url.call_terminated = http:// 192.168.0.20/help.xml?IP=$ip

Parameter- action_url.busy_to_idle	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when changing the state of the IP phone from busy to idle.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.busy_to_idle = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.idle_to_busy	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when changing the state of the IP phone from idle to busy.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.idle_to_busy = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.ip_change	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when changing the IP address of the IP phone.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.ip_change = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.forward_incoming_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when forwarding an incoming call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.forward_incoming_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.reject_incoming_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when rejecting an incoming call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.reject_incoming_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.answer_new_incoming_call	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when answering a new incoming call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.answer_new_incoming_call = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.transfer_finished	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when completing a call transfer.
Default Value	Blank
Range	URL within 511 characters

Example	action_url.transfer_finished = http:// 192.168.0.20/help.xml?IP=$ip

Parameter- action_url.transfer_failed	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when failing to transfer a call.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.transfer_failed = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.setup_autop_finish	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when completing auto provisioning via power on.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.setup_autop_finish = http:// 192.168.0.20/help.xml?IP=$ip

Parameter- action_url.call_waiting_on	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when call waiting feature is enabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.call_waiting_on = http:// 192.168.0.20/help.xml?IP=$ip

Parameter- action_url.call_waiting_off	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when call waiting feature is disabled.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.call_waiting_off = http:// 192.168.0.20/help.xml?IP=$ip

Parameter- action_url.headset	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when pressing the HEADSET key.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.headset = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.handfree	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when pressing the Speakerphone key.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.handfree = http://192.168.0.20/ help.xml?IP=$ip

Parameter- action_url.cancel_callout	Configuration File <MAC>.cfg
Description	Configures the action URL the IP phone sends when cancelling the outgoing call in the ring-back state.
Default Value	Blank

Range	URL within 511 characters
Example	action_url.cancel_callout = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.remote_busy	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when the outgoing call is rejected.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.remote_busy = http://192.168.0.20/help.xml?IP=$ip

Parameter- action_url.call_remote_canceled	**Configuration File** <MAC>.cfg
Description	Configures the action URL the IP phone sends when the remote party cancels the outgoing call in the ringing state.
Default Value	Blank
Range	URL within 511 characters
Example	action_url.call_remote_canceled = http://192.168.0.20/help.xml?IP=$ip

Action URI

Parameter- features.action_uri.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to receive the action URI requests.
Default Value	1
Range	0 - Disabled 1 - Enabled
Example	features.action_uri.enable= 1

Parameter- features.show_action_uri_option	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to pop up the Allow Remote Control prompt. If it is set to 0 (Disabled), the phone will not pop up the Allow Remote Control prompt when receiving an HTTP or HTTPS GET request, or receiving a SIP NOTIFY message with the "Event: ACTION-URI" header. The phone will directly perform the specified action. **Note:** It works only if the value of the parameter "features.action_uri.enable" is set to 1(Enabled).
Default Value	1
Range	0 - Disabled 1 - Enabled
Example	features.show_action_uri_option= 1

Parameter- features.action_uri_limit_ip	**Configuration File** <MAC>.cfg
Description	Configures the IP address of the server from which the IP phone receives the action URI requests. For discontinuous IP addresses, multiple IP addresses are separated by commas. For continuous IP addresses, the format likes *.*.*.* and the "*" stands for the values 0~255. For example: 10.10.*.* stands for the IP addresses that range from 10.10.0.0 to 10.10.255.255. If left blank, the IP phone will reject any HTTP GET request. If it is set to "any", the IP phone will accept and handle HTTP GET requests from any IP address. **Note:** It works only if the value of the parameter "features.action_uri.enable" is set to 1 (Enabled).
Default Value	Blank
Range	IP address or any
Example	features.action_uri_limit_ip = any

Server Redundancy

Parameter- account.X.sip_server.Y.address (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the IP address or domain name of the SIP server Y that accepts registrations for account X.
Default Value	Blank
Range	String within 256 characters
Example	account.1.sip_server.1.address = matrix.pbx.com

Parameter- account.X.sip_server.Y.port (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the port of the SIP server Y that specifies registrations for account X. **Note:** If the value of this parameter is set to 0, the port used depends on the value specified by the parameter "account.X.sip_server.Y.transport_type".
Default Value	5060
Range	Integer from 0 to 65535
Example	account.1.sip_server.1.port = 5060

Parameter- account.X.sip_server.Y.expires (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the registration expiration time (in seconds) of the SIP server Y for account X.
Default Value	3600
Range	Integer from 30 to 2147483647
Example	account.1.sip_server.1.expires = 3600

Parameter- account.X.sip_server.Y.retry_counts (X ranges from 1 to 16, Y ranges from 1 to 2)	Configuration File <MAC>.cfg
Description	Configures the retry times for the IP phone to resend requests when the SIP server Y is unavailable or there is no response from the SIP server Y for account X.
Default Value	3
Range	Integer from 0 to 20
Example	account.X.sip_server.Y.retry_counts = 3

Parameter- account.X.sip_server.Y.retry_counts (X ranges from 1 to 16, Y ranges from 1 to 2)	Configuration File <MAC>.cfg
Description	Configures the retry times for the IP phone to resend requests when the SIP server Y is unavailable or there is no response from the SIP server Y for account X.
Default Value	3
Range	Integer from 0 to 20
Example	account.X.sip_server.Y.retry_counts = 3

Parameter- account.X.sip_server.Y.register_on_enable (X ranges from 1 to 16, Y ranges from 1 to 2)	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to register to the secondary server before sending requests to it for account X when encountering a failover. If it is set to 0 (Disabled), the IP phone will directly send the requests to the secondary server. If it is set to 1 (Enabled), the IP phone will register to the secondary server first, and then send the requests to it.
Default Value	0

Range	**0 - Disabled** **1 - Enabled**
Example	account.X.sip_server.Y.register_on_enable= 0

Parameter- account.X.outbound_proxy_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send requests to the outbound proxy server for account X.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	account.X.outbound_proxy_enable = 0

Parameter- account.X.outbound_proxy.Y.address (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the IP address or domain name of the outbound proxy server Y for account X. **Note:** It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).
Default Value	Blank
Range	IP address or domain name.
Example	account.1.outbound_proxy.1.address = 10.1.8.11

Parameter- account.X.outbound_proxy.Y.port (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg

Description	Configures the port of the outbound proxy server Y for account X. **Note:** It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).
Default Value	5060
Range	Integer from 0 to 65535
Example	account.1.outbound_proxy.1.port = 5060

Parameter- account.X.fallback.redundancy_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the registration mode for the IP phone in fallback mode. **Note:** It is not applicable to outbound proxy servers.
Default Value	0
Range	0-Concurrent Registration 1-Successive Registration
Example	account.1.fallback.redundancy_type= 0

Parameter- account.X.fallback.timeout (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the time interval (in seconds) for the IP phone to detect whether the working server is available by sending the registration request for account X after the fallback server takes over call control. **Note:** It works only if the value of the parameter "account.X.fallback.redundancy_type" is set to 1 (Successive Registration). It is not applicable to outbound proxy servers.
Default Value	120
Range	Integer from 10 to 2147483647
Example	account.X.fallback.timeout= 120

Parameter- account.X.outbound_proxy_fallback_interval (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the time interval (in seconds) for the IP phone to detect whether the working outbound proxy server is available by sending the registration request after the fallback server takes over call control. **Note:** It is only applicable to outbound proxy servers.
Default Value	3600
Range	Integer from 0 to 65535
Example	account.1.outbound_proxy_fallback_interval = 3600

Parameter- account.X.sip_server.Y.failback_mode (X ranges from 1 to 16, Y ranges from 1 to 2)	Configuration File <MAC>.cfg

Description	Configures the failback mode for the IP phone to retry the primary server in failover for account X. 0 - newRequests: all requests are sent to the primary server first, regardless of the last server that was used. If the primary server does not respond correctly, the IP phone will try to send requests to the secondary server. 1 - DNSTTL: the IP phone will send requests to the last registered server first. If the TTL for the DNS A records on the registered server expires, the phone will retry to send requests to the primary server. 2 - Registration: the IP phone will send requests to the last registered server first. If the registration expires, the phone will retry to send requests to the primary server. 3 - duration: the IP phone will send requests to the last registered server first. If the time defined by the parameter "account.X.sip_server.Y.failback_timeout" expires, the phone will retry to send requests to the primary server. **Note:** DNSTTL, Registration and duration mode can only be processed when the IP phone is idle (that is, no incoming/outbound calls, no active calls or meetings, etc.).
Default Value	0
Range	0,1,2 or 3
Example	account.X.sip_server.Y.failback_mode = 0

Parameter-	**Configuration File**
account.X.sip_server.Y.failback_timeout (X ranges from 1 to 16, Y ranges from 1 to 2)	<MAC>.cfg

Description	Configures the time (in seconds) for the phone to retry to send requests to the primary server after failing over to the current working server for account X. If you set the parameter to 0, the IP phone will not send requests to the primary server until a failover event occurs with the current working server. If you set the parameter from 1 to 59, the timeout will be 60 seconds. **Note:** It works only if the value of the parameter "account.X.sip_server.Y.failback_mode" is set to 3 (duration).
Default Value	3600
Range	0, Integer from 60 to 65535
Example	account.X.sip_server.Y.failback_timeout = 3600

SIP Server Domain Name Resolution

Parameter- account.X.sip_server.Y.transport_type (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the transport method the IP phone uses to communicate with the SIP server for account X. 0-UDP 1-TCP 2-TLS 3-DNS-NAPTR If the value of this parameter is set to 3 (DNS-NAPTR), the value of the parameter "account.X.sip_server.Y.address" is set to a host name and the value of the parameter "account.X.sip_server.Y.port" is set to 0, the IP phone will perform the DNS NAPTR and SRV queries for the transport protocol, ports and servers. If the value of this parameter is set to 3 (DNS-NAPTR), the value of the parameter "account.X.sip_server.Y.address" is set to an IP address and the value of the parameter "account.X.sip_server.Y.port" is set to an explicit port (except 0), then UDP is used.

Default Value	0
Range	**Valid values are:** **0**-UDP **1**-TCP **2**-TLS **3**-DNS-NAPTR
Example	account.1.sip_server.1.transport_type = 0

Parameter- account.X.naptr_build (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the way of SRV query for the IP phone to be performed when no result is returned from NAPTR query for account X. 0 - SRV query using UDP only 1 - SRV query using UDP, TCP and TLS1.
Default Value	0
Range	0 - SRV query using UDP only 1 - SRV query using UDP, TCP and TLS1.
Example	account.1.naptr_build= 0

Allow IP call

Parameter- features.direct_ip_call_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables allow IP address call. **Note:** If you want to receive an IP address call, make sure the value of the parameter "sip.trust_ctrl" is set to 0 (Disabled).
Default Value	1
Range	**0** - Disabled **1** - Enabled
Example	features.direct_ip_call_enable= 1

LLDP

Parameter- static.network.lldp.enable	Configuration File <MAC>.cfg
Description	Enables or disables the LLDP (Linker Layer Discovery Protocol) feature on the IP phone. If it is set to 1 (Enabled), the IP phone will attempt to determine its VLAN ID through LLDP. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	0 - Disabled 1 - Enabled
Example	static.network.lldp.enable = 1

Parameter- static.network.lldp.packet_interval	Configuration File <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to send the LLDP (Linker Layer Discovery Protocol) request. **Note**: It works only if the value of the parameter "static.network.lldp.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	60
Range	Integer from 1 to 3600
Example	static.network.lldp.packet_interval = 60

VLAN

VLAN assignment method

Parameter- static.network.vlan.vlan_change.enable	Configuration File <MAC>.cfg

Description	Enables or disables the IP phone to obtain VLAN ID using lower priority of VLAN assignment method or disable VLAN feature when the IP phone cannot obtain VLAN IDusing the current VLAN assignment method. The priority of each method is: LLDP>CDP>Manual>DHCP VLAN. If it is set to 1 (Enabled), the IP phone will attempt to use the lower priority of VLAN assignment method when failing to obtain the VLAN ID using higher priority of VLAN assignment method. If all the methods are attempted, the phone will disable VLAN feature. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	static.network.vlan.vlan_change.enable= 0

Internet Port

Parameter- static.network.vlan.internet_port_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables VLAN for the Internet port. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	static.network.vlan.internet_port_enable= 0

Parameter- static.network.vlan.internet_port_vid	Configuration File <MAC>.cfg
Description	Configures VLAN ID for the Internet port. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	Integer from 1 to 4094
Example	static.network.vlan.internet_port_vid= 1

Parameter- static.network.vlan.internet_port_priority	Configuration File <MAC>.cfg
Description	Configures VLAN priority for the Internet port. 7 is the highest priority, 0 is the lowest priority. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	Integer from 0 to 7
Example	static.network.vlan.internet_port_priority= 0

PC Port

Parameter- static.network.vlan.pc_port_enable	Configuration File <MAC>.cfg
Description	Enables or disables VLAN for the PC port. **Note:** It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiate). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0 or 1 0 - Disabled 1 - Enabled
Example	static.network.vlan.pc_port_enable= 0

Parameter- static.network.vlan.pc_port_vid	Configuration File <MAC>.cfg
Description	Configures VLAN ID for the PC port. **Note:** It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiate). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	Integer from 1 to 4094
Example	static.network.vlan.pc_port_vid= 1

Parameter- static.network.vlan.pc_port_priority	Configuration File <MAC>.cfg
Description	Configures VLAN priority for the PC port. 7 is the highest priority, 0 is the lowest priority. **Note:** It works only if the value of the parameter "static.network.pc_port.enable" is set to 1 (Auto Negotiate). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	Integer from 0 to 7
Example	static.network.vlan.pc_port_priority= 0

DHCP VLAN

Parameter- static.network.vlan.dhcp_enable	Configuration File <MAC>.cfg
Description	Enables or disables DHCP VLAN discovery feature on the IP phone. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1

Range	0 - Disabled 1 - Enabled
Example	static.network.vlan.dhcp_enable = 1

Parameter- static.network.vlan.dhcp_option	**Configuration File** <MAC>.cfg
Description	Configures the DHCP option from which the IP phone will obtain the VLAN settings. You can configure at most five DHCP options and separate them by commas. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	132
Range	Integer from 1 to 255
Example	static.network.vlan.dhcp_option= 132

QoS

Parameter- static.network.qos.signaltos	**Configuration File** <MAC>.cfg
Description	Configures the DSCP (Differentiated Services Code Point) for SIP packets. The default DSCP value for SIP packets is 26 (Assured Forwarding). **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	26
Range	Integer from 0 to 63
Example	static.network.qos.signaltos = 26

Parameter- static.network.qos.audiotos	**Configuration File** <MAC>.cfg

Description	Configures the DSCP (Differentiated Services Code Point) for voice packets. The default DSCP value for RTP packets is 46 (Expedited Forwarding). **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	46
Range	Integer from 0 to 63
Example	static.network.qos.audiotos = 46

Parameter- static.network.qos.videotos	**Configuration File** <MAC>.cfg
Description	Configures the DSCP (Differentiated Services Code Point) for video packets. The default DSCP value for H264 packets is 34 (Assured Forwarding). **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	34
Range	Integer from 0 to 63
Example	static.network.qos.videotos = 34

Parameter- static.wifi.802_11e.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the WMM feature (Wi-Fi MultiMedia) in the wireless network. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	1
Range	0 or 1
Example	static.wifi.802_11e.enable = 1

Network Address Translation

Parameter- sip.nat_stun.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the STUN (Simple Traversal of UDP over NATs) feature on the IP phone. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0-Disabled 1-Enabled
Example	sip.nat_stun.enable= 0

Parameter- sip.nat_stun.server	**Configuration File** <MAC>.cfg
Description	Configures the IP address or the domain name of the STUN (Simple Traversal of UDP over NATs) server. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IP Address or Domain Name
Example	sip.nat_stun.server = 218.107.220.201

Parameter- sip.nat_stun.port	**Configuration File** <MAC>.cfg
Description	Configures the port of the STUN (Simple Traversal of UDP over NATs) server. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	3478
Range	Integer from 1024 to 65000
Example	sip.nat_stun.port = 3478

Parameter- account.X.nat.nat_traversal (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the NAT traversal for account X. **Note:** It works only if the value of the parameter "sip.nat_stun.enable" is set to 1 (Enabled).
Default Value	0
Range	0-Disabled 1-Enabled
Example	account.X.nat.nat_traversal = 0

Parameter- sip.listen_port	**Configuration File** <MAC>.cfg
Description	Configures the local SIP port.
Default Value	5060
Range	Integer from 1024 to 65535
Example	sip.listen_port = 5060

Parameter- sip.tls_listen_port	**Configuration File** <MAC>.cfg
Description	Configures the local TLS listen port.
Default Value	5061
Range	Integer from 1024 to 65535
Example	sip.listen_port = 5061

802.1X authentication

Parameter- static.network.802_1x.mode	**Configuration File** <MAC>.cfg
Description	Configures the 802.1x authentication method. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0-Disabled 1-EAP-MD5 2-EAP-TLS 3-EAP-PEAP/MSCHAPv2 4-EAP-TTLS/EAP-MSCHAPv2 5-EAP-PEAP/GTC 6-EAP-TTLS/EAP-GTC 7-EAP-FAST
Example	static.network.802_1x.mode = 0

Parameter- static.network.802_1x.identity	**Configuration File** <MAC>.cfg
Description	Configures the identity (or user name) for 802.1x authentication. **Note:** It works only if the value of the parameter "static.network.802_1x.mode" is set to 1, 2, 3, 4, 5, 6 or 7. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	String within 32 characters
Example	static.network.802_1x.identity = admin

Parameter- static.network.802_1x.md5_password	**Configuration File** <MAC>.cfg

Description	Configures the password for 802.1x authentication. **Note:** It works only if the value of the parameter "static.network.802_1x.mode" is set to 1,3, 4, 5, 6 or 7. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	String within 32 characters
Example	static.network.802_1x.md5_password = admin123

Parameter- static.network.802_1x.root_cert_url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the CA certificate. **Note:** It works only if the value of the parameter "static.network.802_1x.mode" is set to 2,3, 4, 5, 6 or 7. The format of the CA certificate must be *.pem, *.crt, *.cer or *.der.
Default Value	Blank
Range	URL within 511 characters
Example	static.network.802_1x.root_cert_url = http://192.168.1.10/ca.pem

Parameter- network.802_1x.client_cert_url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the device certificate. **Note:** It works only if the value of the parameter "static.network.802_1x.mode" is set to 2(EAP-TLS). The format of the device certificate must be *.pem.
Default Value	Blank
Range	URL within 511 characters
Example	static.network.802_1x.client_cert_url = http://192.168.1.10/client.pem

TR-069

Parameter- managementserver.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables TR-069 feature on the IP phone.
Format	Integer
Default Value	0
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	managementserver.enable = 1

Parameter- managementserver.username	**Configuration File** <MAC>.cfg
Description	Configures the username to authenticate with the ACS. This string is set to the empty string if no authentication is required.
Format	String
Default Value	Blank
Range	String within 128 characters
Example	managementserver.username = user1

Parameter- managementserver.password	**Configuration File** <MAC>.cfg
Description	Configures the password to authenticate with the ACS. This string is set to the empty string if no authentication is required.
Format	String
Default Value	Blank
Range	String within 64 characters
Example	managementserver.password = pwd123

Parameter- managementserver.url	**Configuration File** <MAC>.cfg
Description	Configures the URL of the ACS.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	managementserver.url = http://192.168.1.20/acs/

Parameter- managementserver.connection_request_username	Configuration File <MAC>.cfg
Description	Configures the user name for the IP phone to authenticate the incoming connection requests.
Format	String
Default Value	Blank
Range	String within 128 characters
Example	managementserver.connection_request_username = acsuser

Parameter- managementserver.connection_request_password	Configuration File <MAC>.cfg
Description	Configures the password for the IP phone to authenticate the incoming connection requests.
Format	String
Default Value	Blank
Range	String within 64 characters
Example	managementserver.connection_request_password = acspwd

Parameter- managementserver.periodic_inform_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to periodically report its configuration information to the ACS.
Format	Boolean
Default Value	1
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	managementserver.periodic_inform_enable = 1

Parameter- managementserver.periodic_inform_interval	Configuration File <MAC>.cfg
Description	Configures the interval (in seconds) to report its configuration information to the ACS.
Format	Integer
Default Value	60
Range	5 to 4294967295
Example	managementserver.periodic_inform_interval = 60

IPv6

Parameter- static.network.ip_address_mode	**Configuration File** <MAC>.cfg
Description	Configures the IP address mode. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0-IPv4 1-IPv6 2-IPv4 & IPv6
Example	static.network.ip_address_mode= 0

Parameter- static.network.ipv6_internet_port.type	**Configuration File** <MAC>.cfg
Description	Configures the Internet port type for IPv6. **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0-DHCP 1-Static IP Address
Example	static.network.ipv6_internet_port.type= 0

Parameter- static.network.ipv6_static_dns_enable	**Configuration File** <MAC>.cfg

Description	Triggers the static IPv6 DNS feature to on or off. If it is set to 0 (Off), the IP phone will use the IPv6 DNS obtained from DHCP. If it is set to 1 (On), the IP phone will use manually configured static IPv6 DNS. **Note:** It works only if the value of the parameter "static.network.ipv6_internet_port.type" is set to 0 (DHCP). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0-Off 1-On
Example	static.network.ipv6_static_dns_enable= 0

Parameter- static.network.ipv6_internet_port.ip	**Configuration File** <MAC>.cfg
Description	Configures the IPv6 address. **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv6 address
Example	static.network.ipv6_internet_port.ip = 2026:1234:1:1:215:65ff:fe1f:caa

Parameter- static.network.ipv6_prefix	**Configuration File** <MAC>.cfg

Description	Configures the IPv6 prefix. **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	64
Range	Integer from 0 to 128
Example	static.network.ipv6_prefix= 64

Parameter- static.network.ipv6_internet_port.gateway	**Configuration File** <MAC>.cfg
Description	Configures the IPv6 default gateway. **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6), and "static.network.ipv6_internet_port.type" is set to 1 (Static IP Address).If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv6 Address
Example	static.network.ipv6_internet_port.gateway = 3036:1:1:c3c7:c11c:5447:23a6:255

Parameter- static.network.ipv6_primary_dns	**Configuration File** <MAC>.cfg

Description	Configures the primary IPv6 DNS server. **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6). In DHCP environment, you also need to make sure the value of the parameter "static.network.ipv6_static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv6 Address
Example	static.network.ipv6_primary_dns = 3036:1:1:c3c7:c11c:5447:23a6:256

Parameter- static.network.ipv6_secondary_dns	**Configuration File** <MAC>.cfg
Description	Configures the secondary IPv6 DNS server. Example: **Note:** It works only if the value of the parameter "static.network.ip_address_mode" is set to 1 (IPv6) or 2 (IPv4 & IPv6). In DHCP environment, you also need to make sure the value of the parameter "static.network.ipv6_static_dns_enable" is set to 1 (On). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IPv6 Address
Example	static.network.ipv6_secondary_dns = 2026:1234:1:1:c3c7:c11c:5447:23a6

Parameter- static.network.ipv6_icmp_v6.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to obtain IPv6 network settings via SLAAC (Stateless Address Autoconfiguration) method. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.

Default Value	1
Range	0-Disabled 1-Enabled
Example	static.network.ipv6_icmp_v6.enable = 1

Multicast Paging Key

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a multicast paging key on the IP phone. The digit 24 stands for the key type Multicast Paging. For line keys: X ranges from 1 to 27. **Default:** The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Range	24 (Refer the above content)
Example	linekey.2.type = 24

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the multicast IP address and port number. For line keys: X ranges from 1 to 27. **Note:** The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Range	String within 99 characters
Value	Blank
Example	linekey.2.value = 224.5.5.6:10008

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg

Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Example	NA

Parameter- linekey.X.extension	**Configuration File** <MAC>.cfg
Description	Configures the channel of multicast paging group. For line keys: X ranges from 1 to 27. **Note:** This parameter only applies to multicast paging feature.
Range	String within 256 characters
Example	linekey.2.extension = 2

Paging List Key

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a paging list key on the IP phone. The digit 66 stands for the key type Paging List. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14. For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	66
Default	Refer the above content

Example	linekey.2.type = 66

Parameter- llinekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default	Blank
Example	NA

Recent Call In Dialing

Parameter- super_search.recent_call	**Configuration File** <MAC>.cfg
Description	Enables or disables recent call in dialing feature. If it is set to 1 (Enabled), you can see the placed calls list when the IP phone is on the pre-dialing screen.
Value	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	super_search.recent_call = 0

Notification Popups

Parameter- features.voice_mail_popup.enable	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to display the pop-up message box when it receives a new voice mail. **Note:** If the voice mail pop-up message box disappears, it won't pop up again unless the user receives a new voice mail or the user re-registers the account that has unread voice mail(s).
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	1
Example	features.voice_mail_popup.enable = 1

Parameter- features.missed_call_popup.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display the pop-up message box when it misses a call.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	1
Example	features.missed_call_popup.enable= 1

Parameter- features.forward_call_popup.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display the pop-up message box when it forwards an incoming call to other party.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	1
Example	features.forward_call_popup.enable= 1

IP Direct Auto Answer

Parameter- features.ip_call.auto_answer.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the auto answer feature for IP call. **Note:** It works only if the value of the parameter "features.direct_ip_call_enable" is set to 1(Enabled). The IP phone cannot automatically answer the incoming IP call during a call even if IP call auto answer is enabled.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	features.ip_call.auto_answer.enable= 1

Accept SIP Trust Server Only

Parameter- sip.trust_ctrl	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to only accept the SIP message from the SIP server and outbound proxy server.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	sip.trust_ctrl= 1

WiFi

Parameter- static.wifi.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the Wi-Fi feature.
Range	0 or 1 0-Disabled 1-Enabled
Default Value	0

Example	static.wifi.enable= 1

Wallpaper

Parameter- phone_setting.backgrounds	**Configuration File** <MAC>.cfg
Description	Configures the wallpaper displayed on the IP phone idle screen.
Range	Default.jpg, 01.jpg, 02.jpg, 03.jpg, 04.jpg, 05.jpg, 06.jpg, 07.jpg, 08.jpg, 09.jpg or 10.jpg or custom wallpaper name (e.g., wallpaper.jpg)
Default Value	Default.jpg
Example	phone_setting.backgrounds = Default.jpg

Parameter- wallpaper_upload.url	**Configuration File** <MAC>.cfg
Description	Configures the wallpaper displayed on the IP phone idle screen. **Note:** It works only if the value of the parameter "features.direct_ip_call_enable" is set to 1(Enabled). The IP phone cannot automatically answer the incoming IP call during a call even if IP call auto answer is enabled.
Range	0 or 1
Default Value	Blank
Example	wallpaper_upload.url = http://192.168.10.25/ wallpaper.jpg

Allow Mute

Parameter- features.allow_mute	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to mute an active call.

Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	1
Example	features.allow_mute=1

Keep Mute

Parameter- features.keep_mute.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the keep mute feature for the IP phone. If it is set to 0 (Disabled), mute feature is automatically deactivated when the active call ends. If it is set to 1 (Enabled), the mute state can be kept across calls after the mute feature is activated until you manually deactivate the mute feature or the phone restarts. **Note:** If it is set to 1 (Enabled), you can not customize the Mute key. It works only if the value of the parameter "features.allow_mute" is set to 1 (Enabled).
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	features.keep_mute.enable=1

Call Number Filter

Parameter- features.call_num_filter	**Configuration File** <MAC>.cfg

Description	Configures the characters the IP phone filters when dialing. If the dialed number contains configured characters, the IP phone will automatically filter these characters when dialing. If you dial 3-61, the IP phone will filter the characters - and 1, and then dial out 36. **Note:** If it is left blank, the IP phone will not automatically filter any characters when dialing. If you want to filter just a space, you have to set the value to " ," (a space first followed by a comma).
Range	String within 99 characters
Default Value	?,-( )
Example	features.call_num_filter = ,-12

Power Saving

Parameter- features.power_saving.intelligent_mode	**Configuration File** <MAC>.cfg
Description	Enables or disables the power saving intelligent mode.
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone stays in power-saving mode even if the office hour arrives the next day. If it is set to 1 (Enabled), the IP phone will automatically identify the office hour and exit power-saving mode once the office hour arrives the next day.
Default Value	1
Example	features.power_saving.intelligent_mode = 1

Parameter- features.power_saving.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power saving feature.
Range	0-Disabled 1-Enabled
Default Value	1

Example	features.power_saving.enable = 1

Parameter- features.power_saving.office_hour.idle_timeout	**Configuration File** <MAC>.cfg
Description	Configures the time (in minutes) to wait in the idle state before the IP phone enters power-saving mode during the office hours.
Range	1 to 960
Default Value	960
Example	features.power_saving.office_hour.idle_timeout = 960 The IP phone will enter power-saving mode when it has been inactivated for 960 minutes (16 hour) during the office hours.

Parameter- features.power_saving.off_hour.idle_timeout	**Configuration File** <MAC>.cfg
Description	Configures the time (in minutes) to wait in the idle state before the IP phone enters power-saving mode during the non-office hours.
Range	1 to 10
Default Value	10
Example	features.power_saving.off_hour.idle_timeout = 5 The IP phone will enter power-saving mode when it has been inactivated for 5 minutes during the non-office hours.

Parameter- features.power_saving.user_input_ext.idle_timeout	**Configuration File** <MAC>.cfg
Description	Configures the minimum time (in minutes) to wait in the idle state - after using the phone - before the IP phone enters power-saving mode.

Range	1 to 30
Default Value	10
Example	features.power_saving.user_input_ext.idle_timeout = 5

Parameter- features.power_saving.office_hour.monday features.power_saving.office_hour.tuesday features.power_saving.office_hour.wednesday features.power_saving.office_hour.thursday features.power_saving.office_hour.friday features.power_saving.office_hour.saturday features.power_saving.office_hour.sunday	**Configuration File** <MAC>.cfg
Description	Configures the starting time and ending time of the day's office hour. Starting time and ending time are separated by a comma.
Range	0 to 23
Default Value	7,19 7,19 7,19 7,19 7,19 7,7 7,7
Example	features.power_saving.office_hour.monday = 7,19

Backlight

Parameter- phone_setting.active_backlight_level	**Configuration File** <MAC>.cfg

Description	Configures the intensity of the touch screen when the phone is active. 10 is the highest intensity. It configures the LCD's intensity of the IP phone.
Range	1 to 10
Default Value	8
Example	phone_setting.active_backlight_level = 8

Parameter- phone_setting.backlight_time	**Configuration File** <MAC>.cfg
Description	Configures the delay time (in seconds) before the backlight is turned off when the IP phone is inactive.
Range	0, 15, 30, 60, 120, 300, 600 or 1800 If it is set to 0 (Always On), the backlight will not be turned off when the IP phone is inactive. If it is set to 60 (60), the backlight will be turned off when the IP phone is inactivated for 60 seconds.
Default Value	0
Example	phone_setting.backlight_time = 0

Bluetooth

Parameter- features.bluetooth_enable	**Configuration File** <MAC>.cfg
Description	Triggers Bluetooth mode to on or off.
Range	0-Off 1-On **Note:** To use a Bluetooth headset or connect a Bluetooth device, you must trigger Bluetooth mode to on.
Default Value	0
Example	features.bluetooth_enable = 0

Parameter- features.bluetooth_adapter_name	**Configuration File** <MAC>.cfg

Description	Configures the Bluetooth device name.
Range	64 characters **Note:** It works only if the value of the parameter "features.bluetooth_enable" is set to 1 (On).
Default Value	The default value is Matrix SPARSH VP710
Example	features.bluetooth_adapter_name = Matrix SPARSH VP710

Parameter- phone_setting.bluetooth_talk.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the user to have the permission to use the Bluetooth feature during the call.
Range	0-Disabled 1-Enabled
Default Value	1
Example	phone_setting.bluetooth_talk.enable = 1

Call Park

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a call park key on the IP phone. The digit 10 stands for the key type Call Park. For line keys: X ranges from 1 to 27.
Range	10
Default Value	The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Example	linekey.2.type = 10

Parameter- linekey.X.line	**Configuration File** <MAC>.cfg

Description	Configures the desired line to apply the call park key. For line keys: X ranges from 1 to 27.
Range	1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Default Value	1-16 for lines 1-16
Example	linekey.2.line = 1

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the call park feature code. For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default Value	Blank
Example	linekey.2.value = *88

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key.(Optional) For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default Value	Blank
Example	linekey.X.label = *88

Call Timeout

Parameter- phone_setting.ringback_timeout	**Configuration File** <MAC>.cfg
Description	Configures the duration time (in seconds) in the ringback state. If it is set to 180, the phone will cancel the dialing if the call is not answered within 180 seconds.
Range	0 to 3600
Default Value	180
Example	phone_setting.ringback_timeout=180

Ringing timeout

Parameter- phone_setting.ringing_timeout	**Configuration File** <MAC>.cfg
Description	Configures the duration time (in seconds) in the ringing state. If it is set to 180, the phone will stop ringing if the call is not answered within 180 seconds.
Range	0 to 3600
Default Value	180
Example	phone_setting.ringing_timeout=180

Send user=phone

Parameter- account.X.enable_user_equal_phone (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to add “user=phone” to the SIP header of the INVITE message for account X.
Range	0 to 1 0 - Disabled 1 - Enabled
Default Value	0

Example	account.X.enable_user_equal_phone=0

SIP Send MAC

Parameter- account.X.register_mac (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to add MAC address to the SIP header of the REGISTER message for account X.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	account.1.register_mac=0

SIP Send Line

Parameter- account.X.register_line (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to add line number to the SIP header of the REGISTER message for account X.
Range	0 or 1 0 - Disabled 1 - Enabled
Default Value	0
Example	account.1.register_line=0

Reserve # in User Name

Parameter- sip.use_23_as_pound	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to reserve the pound sign (#) in the user name.
Range	0 or 1 0 - Disabled (convert the pound sign into “%23”) 1 - Enabled.
Default Value	1
Example	sip.use_23_as_pound=1

Password Dial

Parameter- features.password_dial.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables password dial feature for the IP phone.
Range	0 or 1 0 - Disabled 1 - Enabled.
Default Value	0
Example	features.password_dial.enable=0

Parameter- features.password_dial.prefix	**Configuration File** <MAC>.cfg
Description	Configures the prefix of the password dial number.
Range	String within 32 characters
Default Value	Blank
Example	features.password_dial.prefix = 12

Parameter- features.password_dial.length	**Configuration File** <MAC>.cfg

Description	Configures the number of digits to be hidden. The hidden digits are displayed as asterisks on the touch screen. If you set the prefix to 12 (configured by the parameter "features.password_dial.prefix") and the length to 3, when you want to dial the number 123456, the entered number is displayed as 12***6 on the touch screen.
Range	Integer from 0 to 99
Default Value	Blank
Example	features.password_dial.length = 3

Unregister When Reboot

Parameter- account.X.unregister_on_reboot (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to unregister first before re-registering account X when finishing a reboot.
Range	0 or 1 0 - Disabled 1 - Enabled.
Default Value	0
Example	account.1.unregister_on_reboot= 0

100 Reliable Retransmission

Parameter- account.X.100rel_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the 100 reliable retransmission feature for account X.
Range	0 or 1 0 - Disabled 1 - Enabled.
Default Value	0
Example	account.1.100rel_enable= 0

Reboot in Talking

Parameter- features.reboot_in_talk_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to reboot during a call when it receives a reboot request by action URI. **Note:** It works only if the value of the parameter "features.action_uri_limit_ip" is set to "any" or trusted IP address(es) and it is not the first time for the IP phone to receive HTTP/HTTPS GET request from the trusted IP address(es).
Range	0 or 1 0 - Disabled 1 - Enabled.
Default Value	0
Example	features.reboot_in_talk_enable= 0

Answer By Hand

Parameter- features.off_hook_answer.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to answer an incoming call by picking up the handset,pressing the Speakerphone key or pressing the HEADSET key directly. If it is set to 0 (Disabled), you need to tap the corresponding line key or the Answer soft key to answer an incoming call after picking up the handset, pressing the Speakerphone key or pressing the HEADSET key.
Range	0 or 1 0 - Disabled 1 - Enabled.
Default Value	1
Example	features.off_hook_answer.enable= 1

Music on Hold

Parameter- account.X.music_server_uri (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the SIP URI that provides the audio to play for the held party for account X. The URI points to a Music On Hold (MoH) server. Examples for valid values: <10.1.3.165>, 10.1.3.165, sip:moh@sip.com, <sip:moh@sip.com> **Note:** The DNS query in this parameter only supports A query.
Range	SIP URI within 256 characters
Default Value	Blank
Example	account.1.music_server_uri = sip:moh@sip.com

Parameter- account.X.music_on_hold_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the way to process Music On Hold when placing an active call on hold for account X.
Range	0 or 1 0 - Calling the Music On Hold server before holding 1 - Calling the Music On Hold server after holding
Default Value	0
Example	account.X.music_on_hold_type= 0

Redial Tone

Parameter- features.redial_tone	**Configuration File** <MAC>.cfg

Description	Configures the IP phone to continue to play the dial tone after inputting the preset numbers on the pre-dialing screen. The IP phone will continue to play the dial tone after inputting “123” on the pre-dialing screen. If it is left blank, the IP phone will not play the dial tone after inputting numbers on the pre-dialing screen.
Range	Integer within 6 digits
Default Value	Blank
Example	features.redial_tone = 123

Static DNS Cache

Parameter- dns_cache_naptr.X.name (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the domain name to which NAPTR record X refers.
Range	Domain name
Default Value	Blank
Example	dns_cache_naptr.1.name = matrix.pbx.com

Parameter- dns_cache_naptr.X.flags (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the flag of NAPTR record X. (Always “S” for SIP, which means to do an SRV lookup on whatever is in the replacement field). S-Do an SRV lookup next A-Do an A lookup next U-No need to do a DNS query next P-Service custom by the user **Note:** For more details of the permitted flags, refer to RFC 2915.
Range	S, A, U or P
Default Value	Blank

Example	dns_cache_naptr.1.flags = S

Parameter- dns_cache_naptr.X.order (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the order of NAPTR record X. NAPTR record with lower order is more preferred. For example, NAPTR record with the order 90 has the higher priority than that with the order 100 because 90 is lower than 100.
Range	Integer from 0 to 65535
Default Value	0
Example	dns_cache_naptr.1.order = 90

Parameter- dns_cache_naptr.X.preference (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the preference of NAPTR record X. NAPTR record with lower value is more preferred when the multiple NAPTR records have the same order value.
Range	Integer from 0 to 65535
Default Value	0
Example	dns_cache_naptr.1.preference = 50

Parameter- dns_cache_naptr.X.replace (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures a domain name to be used for the next SRV query in NAPTR record X.
Range	Domain name with SRV prefix
Default Value	Blank
Example	dns_cache_naptr.1.replace = _sip._tcp.matrix.pbx.com

Parameter- dns_cache_naptr.X.service (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the transport protocol available for the server in NAPTR record X. SIP+D2U: SIP over UDP SIP+D2T: SIP over TCP SIP+D2S: SIP over SCTP SIPS+D2T: SIPS over TCP **Note:** For more information, refer to RFC 2915.
Range	String within 32 characters
Default Value	Blank
Example	dns_cache_naptr.1.service = SIP+D2T

Parameter- dns_cache_naptr.X.ttl (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the time interval (in seconds) that NAPTR record X may be cached before the record should be consulted again.
Range	Integer from 30 to 2147483647
Default Value	300
Example	dns_cache_naptr.1.ttl = 3600

Parameter- dns_cache_srv.X.name (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the domain name in SRV record X.
Range	Domain name with SRV prefix
Default Value	Blank
Example	dns_cache_srv.1.name = _sip._tcp.matrix.pbx.com

Parameter- dns_cache_srv.X.port (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the port to be used in SRV record X. **Note:** For more information, refer to RFC 2782.
Range	Integer from 0 to 65535
Default Value	0
Example	dns_cache_srv.1.port = 5060

Parameter- dns_cache_srv.X.priority (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the priority for the target host in SRV record X. Lower priority is more preferred. For example, SRV record with the priority value 0 is more preferred than that with the priority value 1 because 0 is lower than 1. **Note:** For more information, refer to RFC 2782.
Range	Integer from 0 to 65535
Default Value	0
Example	dns_cache_srv.1.priority = 0

Parameter- dns_cache_srv.X.target (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the domain name of the target host for an A query in SRV record X. **Note:** For more information, refer to RFC 2782.
Range	Domain name
Default Value	Blank
Example	dns_cache_srv.1.target = server1.matrix.pbx.com

Parameter- dns_cache_srv.X.weight (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the weight of the target host in SRV record X. When priorities are equal, weight is used to differentiate the preference. Higher weight value is more preferred. **Note:** For more information, refer to RFC 2782.
Range	Integer from 0 to 65535
Default Value	0
Example	dns_cache_srv.1.weight = 1

Parameter- dns_cache_srv.X.ttl (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the time interval (in seconds) that SRV record X may be cached before the record should be consulted again.
Range	Integer from 30 to 2147483647
Default Value	300
Example	dns_cache_srv.1.ttl = 3600

Parameter- dns_cache_a.X.name (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the domain name in A record X.
Range	Domain name
Default Value	Blank
Example	dns_cache_a.1.name = matrix.pbx.com

Parameter- dns_cache_a.X.ip (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the IP address that the domain name in A record X maps to.
Range	IP address
Default Value	Blank
Example	dns_cache_a.1.ip = 192.168.1.13

Parameter- dns_cache_a.X.ttl (X ranges from 1 to 12)	**Configuration File** <MAC>.cfg
Description	Configures the time interval (in seconds) that A record X may be cached before the record should be consulted again.
Range	Integer from 30 to 2147483647
Default Value	300
Example	dns_cache_a.1.ttl = 3600

Parameter- account.X.dns_cache_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures whether the IP phone uses the DNS cache for domain name resolution of the server and caches the additional DNS records for account X. 0 - Perform real-time DNS query rather than using DNS cache. 1 - Use DNS cache, but do not cache the additional DNS records. 2 - Use DNS cache and cache the additional DNS records.
Range	0,1 or 2
Default Value	1
Example	account.1.dns_cache_type = 1

Parameter- account.X.static_cache_pri (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures whether preferentially to use the static DNS cache for domain name resolution of the server for account X. 0 - Use domain name resolution from the DNS server preferentially 1 - Use static DNS cache preferentially
Range	0 or 1
Default Value	0
Example	account.1.static_cache_pri = 1

ReCall

Parameter- linekey.X.type/ programablekey.X.type	Configuration File <MAC>.cfg
Description	Configures a DSS key as a recall key on the IP phone. The digit 7 stands for the key type ReCall. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.1 - Use static DNS cache preferentially
Range	7
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Example	linekey.2.type = 7

Parameter- linekey.X.label	Configuration File <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Example	NA

Softkey Layout

Parameter- phone_setting.custom_softkey_enable	Configuration File <MAC>.cfg
Description	Enables or disables custom soft keys layout feature.
Range	0-Disabled 1-Enabled
Default	0
Example	phone_setting.custom_softkey_enable=0

Call Recording

Parameter- features.call_recording.enable	Configuration File <MAC>.cfg
Description	Enables or disables the call recording feature for the IP phone. If it is set to 1 (Enabled), you can record the audio or video call by tapping the Record soft key during a call, and the recorded files will be saved according to the priority: USB flash drive>Internal SD card. **Note:** To save the recorded files to the USB flash drive, make sure the USB flash drive has been connected to the IP phone in advance.
Range	0-Disabled 1-Enabled
Default	1

Example	features.call_recording.enable=0

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a record key on the IP phone. The digit 25 stands for the key type Record. For line keys: X ranges from 1 to 27.
Range	25(refer the above text)
Default	The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Example	linekey.2.type = 25

Record Key

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default	Blank
Example	NA

URL Record Key

Parameter- inekey.X.type	**Configuration File** <MAC>.cfg

Description	Configures a DSS key as a URL record key on the IP phone. The digit 35 stands for the key type URL Record. For line keys: X ranges from 1 to 27. :
Range	35(refer the above text)
Default	The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Example	linekey.2.type = 35

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the URL to record a call. For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default	Blank
Example	linekey.2.value = http://10.3.5.97:8080/URLRecord/record.xml

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Range	String within 99 characters
Default	Blank
Example	NA

CDP

Parameter- static.network.cdp.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the CDP (Cisco Discovery Protocol) feature on the IP phone. If it is set to 1 (Enabled), the IP phone will attempt to determine its VLAN ID through CDP. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Range	0-Disabled 1-Enabled
Default	0
Example	static.network.cdp.enable = 0

Parameter- static.network.cdp.packet_interval	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to send the CDP (Cisco Discovery Protocol) request. **Note:** It works only if the value of the parameter "static.network.cdp.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.
Range	Integer from 1 to 3600
Default	60
Example	static.network.cdp.packet_interval = 60

VPN (Virtual Private network)

Parameter- static.network.vpn_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables OpenVPN feature on the IP phone. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.

Range	0-Disabled 1-Enabled
Default	0
Example	static.network.vpn_enable = 0

Parameter- static.openvpn.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the *.tar file for OpenVPN.
Range	URL within 511 characters
Default	Blank
Example	static.openvpn.url = http://192.168.10.25/OpenVPN.tar

Reserved Ports

Parameter- sip.min_udp_port	**Configuration File** <MAC>.cfg
Description	Configures the minimum UDP port. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Range	Integer from 1024 to 65535
Default	50000
Example	sip.min_udp_port= 50000

Parameter- sip.max_udp_port	**Configuration File** <MAC>.cfg
Description	Configures the maximum UDP port. **Note:** The value of the maximum UDP port cannot be less than that of the minimum UDP port (configured by the parameter “sip.min_udp_port”). If you change this parameter, the IP phone will reboot to make the change take effect.
Range	Integer from 1024 to 65535
Default	50249

Example	sip.max_udp_port= 50249

Parameter- sip.min_tcp_port	**Configuration File** <MAC>.cfg
Description	Configures the minimum TCP port. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Range	Integer from 1024 to 65535
Default	50000
Example	sip.min_tcp_port= 50000

Parameter- sip.max_tcp_port	**Configuration File** <MAC>.cfg
Description	Configures the maximum TCP port. **Note:** The value of the maximum TCP port cannot be less than that of the minimum TCP port (configured by the parameter “sip.min_tcp_port”). If you change this parameter, the IP phone will reboot to make the change take effect.
Range	Integer from 1024 to 65535
Default	50249
Example	sip.max_tcp_port= 50249

Keep Alive

Parameter- account.X.nat.udp_update_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the type of keep-alive packets sent by the IP phone to the NAT device to keep the communication port open so that NAT can continue to function for account X. 0-Disabled 1-Default (the IP phone sends UDP packets to the server) 2-Options (the IP phone sends SIP OPTIONS packets to the server) 3-Notify (the IP phone sends SIP NOTIFY packets to the server) If it is set to 0 (Disabled), the IP phone will not send keep-alive packets.
Range	0, 1, 2 or 3
Default	1
Example	account.X.nat.udp_update_enable= 1

Parameter- account.X.nat.udp_update_time (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the keep-alive interval (in seconds) for account X. **Note:** It works only if the value of the parameter "account.X.nat.udp_update_enable" is set to 1, 2 or 3.
Range	Integer from 15 to 2147483647
Default	30
Example	account.1.nat.udp_update_time = 60

Rport

Parameter- account.X.nat.rport (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables NAT Rport feature for account X. 0-Disabled 1-Enabled 2-enable direct process
Range	0, 1 or 2
Default	0

Example	account.1.nat.rport= 0

Feature Key Synchronization

Parameter- bw.feature_key_sync	**Configuration File** <MAC>.cfg
Description	Enables or disables feature key synchronization.
Range	0-Disabled 1-Enabled
Default	0
Example	bw.feature_key_sync = 0

Hide Feature Access Codes

Parameter- features.hide_feature_access_codes.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display feature name instead of the feature access code when dialing or in talk.
Range	0-Disabled 1-Enabled
Default	0
Example	features.hide_feature_access_codes.enable = 0

Security Feature Parameters

Uploading Certificates

Parameter- trusted_certificates.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the certificate used to authenticate the connecting server. **Note:** The certificate you want to upload must be in *.pem, *.crt, *.cer or *.der format.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	trusted_certificates.url = http://192.168.1.20/tc.crt

Parameter- server_certificates.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the certificate the IP phone sends for authentication. **Note**: The certificate you want to upload must be in *.pem or *.cer format.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	server_certificates.url = http://192.168.1.20/ca.pem

Language

Parameter- static.lang.gui	**Configuration File** <MAC>.cfg
Description	Configures the language used on the phone user interface.
Default Value	English

Range	English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish,Turkish, Russian or the custom language name.
Example	static.lang.gui = English If you want to use the custom language (e.g., Guilan) for the IP phone, configure the parameter "static.lang.gui = Guilan".

Parameter- static.lang.wui	**Configuration File** <MAC>.cfg
Description	Configures the language used on the web user interface.
Range	English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish, Turkish, Russian or the custom language name..
Default Value	English
Example	static.lang.wui = English If you want to use the custom language (e.g., Wuilan) for the IP phone, configure the parameter "static.lang.wui = Wuilan". **Note:** If the language of your browser is not supported by the IP phone, the web user interface will use English by default.

Time and Date

Parameter- local_time.manual_ntp_srv_prior	**Configuration File** <MAC>.cfg
Description	Configures the priority for the IP phone to use the NTP server address offered by the DHCP server.
Default Value	0
Range	0-High (use the NTP server address offered by the DHCP server preferentially) 1-Low (use the NTP server address configured manually preferentially)

Example	local_time.manual_ntp_srv_prior = 0

Parameter- local_time.dhcp_time	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to update time with the offset time offered by the DHCP server.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** It is only available to offset from Greenwich Mean Time (GMT).
Example	local_time.dhcp_time = 0

Parameter- local_time.ntp_server1	**Configuration File** <MAC>.cfg
Description	Configures the IP address or the domain name of the NTP server 1. The IP phone will obtain the current time and date from the NTP server 1.
Default Value	cn.pool.ntp.org
Range	IP Address or Domain Name
Example	local_time.ntp_server1 = 192.168.0.5

Parameter- local_time.ntp_server2	**Configuration File** <MAC>.cfg

Description	Configures the IP address or the domain name of the NTP server 2. If the NTP server 1 is not configured (configured by the parameter "local_time.ntp_server1") or cannot be accessed, the IP phone will request the time and date from the NTP server 2.
Default Value	pool.ntp.org
Range	IP Address or Domain Name
Example	local_time.ntp_server2 = 192.168.0.6

Parameter- local_time.interval	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) to update time and date from the NTP server.
Default Value	1000
Range	15 to 86400
Example	local_time.interval = 1000

Parameter- local_time.time_zone	**Configuration File** <MAC>.cfg
Description	Configures the time zone.
Default Value	+5.30
Range	-11 to +14
Example	local_time.time_zone = +5.30

Parameter- local_time.time_zone_name	**Configuration File** <MAC>.cfg

Description	Configures the time zone name. The available time zone names depend on the time zone configured by the parameter "local_time.time_zone".
Default Value	India (Calcutta)
Range	32 characters
Example	local_time.time_zone_name = India (Calcutta) **Note:** It works only if the value of the parameter "local_time.summer_time" is set to 2 (Automatic) and the parameter "local_time.time_zone" should be configured in advance.

Parameter- local_time.manual_time_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to obtain time and date from manual settings.
Default Value	0
Range	0-Disabled (obtain time and date from NTP server) 1-Enabled (obtain time and date from manual settings)
Example	local_time.manual_time_enable = 1

Parameter- local_time.time_format	**Configuration File** <MAC>.cfg
Description	Configures the time format.
Default Value	1
Range	0-Hour 12 1-Hour 24 If it is set to 0 (Hour 12), the time will be displayed in 12-hour format with AM or PM specified. If it is set to 1 (Hour 24), the time will be displayed in 24-hour format (e.g., 2:00 PM displays as 14:00).

Example	local_time.time_format = 1

DST

Parameter- local_time.date_format	**Configuration File** <MAC>.cfg
Description	Configures the date format for the date displayed in the control center.
Default Value	0
Range	0-WWW MMM DD 1-DD-MMM-YY 2-YYYY-MM-DD 3-DD/MM/YYYY 4-MM/DD/YY 5-DD MMM YYYY 6-WWW DD MMM **Note:** "WWW" represents the abbreviation of the week, "DD" represents a two-digit day, "MMM" represents the first three letters of the month, "YYYY" represents a four-digit year, and "YY" represents a two-digit year.
Example	NA

Parameter- local_time.summer_time	**Configuration File** <MAC>.cfg
Description	Configures Daylight Saving Time (DST) feature.
Default Value	2
Range	0-Disabled 1-Enabled 2-Automatic
Example	NA

Parameter- local_time.dst_time_type	**Configuration File** <MAC>.cfg

Description	Configures the Daylight Saving Time (DST) time type.
Default Value	0
Range	0-DST by Date 1-DST by Week Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).
Example	NA

Parameter- local_time.start_time	**Configuration File** <MAC>.cfg
Description	Configures the start time of the Daylight Saving Time (DST).
Default Value	1/1/0
Range	- Month/Day/Hour (for DST by Date) - Month/Week of Month/Day of Week/Hour of Day (for DST by Week) If "local_time.dst_time_type" is set to 0 (DST by Date), use the mapping: Month: 1=January, 2=February,.., 12=December Day: 1=the first day in a month,.., 31= the last day in a month Hour: 0=0am, 1=1am,..., 23=11pm
Example	Example: local_time.start_time = 1/1/2 If "local_time.dst_time_type" is set to 1 (DST by Week), use the mapping: Month: 1=January, 2=February,…, 12=December Week of Month: 1=the first week in a month,…, 5=the last week in a month Day of Week: 1=Monday, 2=Tuesday,…, 7=Sunday Hour of Day: 0=0am, 1=1am,…, 23=11pm Example: local_time.start_time = 1/1/7/0 Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).

Parameter- local_time.end_time	**Configuration File** <MAC>.cfg
Description	Configures the end time of the Daylight Saving Time (DST).
Default Value	12/31/23
Range	- Month/Day/Hour (for DST by Date) - Month/Week of Month/Day of Week/Hour of Day (for DST by Week) If “local_time.dst_time_type¨ is set to 0 (DST by Date), use the mapping: Month: 1=January, 2=February,.., 12=December Day: 1=the first day in a month,..., 31= the last day in a month Hour: 0=0am, 1=1am,..., 23=11pm
Example	Example: local_time.start_time = 12/12/22 If “local_time.dst_time_type” is set to 1 (DST by Week), use the mapping: Month: 1=January, 2=February,…, 12=December Week of Month: 1=the first week in a month,…, 5=the last week in a month Day of Week: 1=Monday, 2=Tuesday,…, 7=Sunday Hour of Day: 0=0am, 1=1am,…, 23=11pm Example: local_time.start_time = 4/3/2/3 Note: It works only if the value of the parameter “local_time.summer_time” is set to 1 (Enabled).

Parameter- local_time.offset_time	**Configuration File** <MAC>.cfg
Description	Configures the offset time (in minutes) of Daylight Saving Time (DST). **Note:** It works only if the value of the parameter “local_time.summer_time” is set to 1 (Enabled).
Default Value	Blank
Range	-300 to 300
Example	

Power Indicator LED

Parameter- phone_setting.common_power_led_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to be turned on.
Default Value	0
Range	0-Disabled (power indicator LED is off) 1-Enabled (power indicator LED is solid red)
Example	phone_setting.common_power_led_enable = 0

Parameter- phone_setting.ring_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when the IP phone receives an incoming call.
Default Value	1
Range	0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (300ms) red)
Example	phone_setting.ring_power_led_flash_enable = 1

Parameter- phone_setting.mail_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when the IP phone receives a voice mail.
Default Value	1
Range	0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED slowly flashes (1000ms) red)

Example	phone_setting.ring_power_led_flash_enable = 1

Parameter- phone_setting.mute_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when a call is muted.
Default Value	0
Range	0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (300ms) red)
Example	phone_setting.mute_power_led_flash_enable = 0

Parameter- phone_setting.hold_and_held_power_led_flash_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to flash when a call is placed on hold or is held.
Default Value	0
Range	0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (500ms) red)
Example	phone_setting.mute_power_led_flash_enable = 0

Parameter- phone_setting.talk_and_dial_power_led_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the power indicator LED to be turned on when the IP phone is busy.
Default Value	0

Range	0-Disabled (power indicator LED is off) 1-Enabled (power indicator LED is solid red)
Example	phone_setting.talk_and_dial_power_led_enable = 0

Administrative Password/User Password

Parameter- static.security.user_password	**Configuration File** <MAC>.cfg
Description	Configures the password of the user or administrator for phone's web user interface access. The IP phone uses "user" as the default user password and "admin" as the default administrator password. The valid value format is username:new password.
Default Value	user
Range	32 characters
Example	static.security.user_password = user:123 means setting the password of user (current user name is "user") to password 123. static.security.user_password = admin:456 means setting the password of administrator (current user name is "admin") to password 456. **Note:** IP phones support ASCII characters 32-126(0x20-0x7E) in passwords. You can set the password to be empty via web user interface only.

Key as Send

Parameter- features.key_as_send	**Configuration File** <MAC>.cfg
Description	Configures the "#" or "*" key as the send key.
Default Value	1

Range	0-Disabled 1-# key 2-* key If it is set to 0 (Disabled), neither “#” nor “*” can be used as the send key. If it is set to 1 (# key), the pound key is used as the send key. If it is set to 2 (* key), the asterisk key is used as the send key.
Example	features.key_as_send = 1

Parameter- features.key_tone	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play a key tone when a user presses any key on your phone keypad.
Default Value	1
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), the IP phone will play a key tone when a user presses any key on your phone keypad.
Example	features.key_tone = 1

Parameter- features.send_key_tone	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play a key tone when a user presses a send key.
Default Value	1
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), the IP phone will play a key tone when a user presses a send key. **Note:** It works only if the value of the parameter “features.key_tone” is set to 1 (Enabled).

Example	features.send_key_tone = 1

Parameter- features.send_pound_key	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone not to send any pound key when pressing double #.
Default Value	0
Range	Enables or disables the IP phone not to send any pound key when pressing double #. 0-Disabled (Send one pound key by pressing double #) 1-Enabled (Do not send any pound key when pressing double #) **Note:** It works only if the value of the parameter "features.key_as_send" is set to 1 (Enabled).
Example	features.send_pound_key = 0

Phone Lock

Parameter- phone_setting.phone_lock.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone not to send any pound key when pressing double #.
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.send_pound_key = 0

Parameter- phone_setting.phone_lock.unlock_pin	**Configuration File** <MAC>.cfg

Description	Configures the password for unlocking the phone.
Default Value	123
Range	Characters within 15 digits
Example	phone_setting.phone_lock.unlock_pin = 123

Parameter- phone_setting.phone_lock.lock_time_out	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) to automatically lock the phone. The default value is 0 (the phone is locked only by long pressing the pound key or pressing the phone lock key). **Note:** It works only if the value of the parameter "phone_setting.phone_lock.enable" is set to 1(Enabled).
Default Value	0
Range	0 to 3600
Example	phone_setting.phone_lock.lock_time_out = 0

Auto Logout Time

Parameter- features.relog_offtime	**Configuration File** <MAC>.cfg
Description	Configures the timeout interval (in minutes) for web access authentication.
Default Value	5
Range	1 to 1000
Example	features.relog_offtime = 5 If you log into the web user interface and leave it for 5 minutes, it will automatically log out.

Volume

Parameter- voice.handfree_send	**Configuration File** <MAC>.cfg
Description	Configures the sending volume of the speaker. **Note:** We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	-50 to 50
Example	voice.handfree_send = 0

Parameter- voice.handset_send	**Configuration File** <MAC>.cfg
Description	Configures the sending volume of the handset. **Note:** We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	-50 to 50
Example	voice.handset_send= 0

Parameter- voice.headset_send	**Configuration File** <MAC>.cfg
Description	Configures the sending volume of the headset. **Note:** We recommend that you modify this parameter cautiously. An unreasonable value may render the voice quality bad. If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0

Range	-50 to 50
Example	voice.headset_send = 0

Ring Tone

Parameter- phone_setting.ring_type	**Configuration File** <MAC>.cfg
Description	Configures a ring tone for the IP phone.
Default Value	Ring1.wav
Range	Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, Ring8.wav, Silent.wav, Splash.wav or custom ring tone name (e.g., Customring.wav).
Example	phone_setting.ring_type = Ring1.wav

Parameter- account.X.ringtone.ring_type (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures a ring tone for account X.
Default Value	Common
Range	Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, Ring8.wav, Silent.wav, Splash.wav or custom ring tone name (e.g., Customring.wav).
Example	account.1.ringtone.ring_type = Ring3.wav It means account1 will use the Ring3.wav as the ring tone. account.1.ringtone.ring_type = Common It means account1 will use the ring tone selected for the IP phone configured by the parameter "phone_setting.ring_type".

Parameter- ringtone.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom ring tone file.
Default Value	Blank
Range	URL within 511 characters
Example	ringtone.url = tftp://192.168.1.100/Customring.wav

Parameter- ringtone.delete	Configuration File <MAC>.cfg
Description	Delete all custom ring tone files.
Default Value	Blank
Range	http://localhost/all
Example	ringtone.delete = http://localhost/all

Key Tone

Parameter- features.key_tone	Configuration File <MAC>.cfg
Description	Configures a ring tone for the IP phone.
Default Value	1
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), the IP phone will play a key tone when a user presses any key on your phone keypad.
Example	features.key_tone = 1

Silent Mode

Parameter- phone_setting.permit_silent_mode.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the user to have the permission to use the silent mode feature.
Default Value	1
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the Silent Mode item will disappear from the phone user interface at the path: Settings->Basic->Sound. Users can neither enable the silent mode feature from the control center or via phone user interface, nor adjust the ringer volume to minimum.
Example	phone_setting.permit_silent_mode.enable = 1

Local Directory

Parameter- local_contact.data.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the local contact file (*.xml).
Default Value	Blank
Range	URL within 511 characters
Example	phone_setting.permit_silent_mode.enable = 1

Parameter- local_contact.photo.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the local contact file (*.xml).
Default Value	Blank

Range	URL within 511 characters The format of the contact avatar must be *.png, *.jpg, *.bmp, *.jpeg. The contact avatar file should be uploaded to the provisioning server in advance.
Example	local_contact.photo.url = tftp://192.168.10.25/ Photo.jpg

Parameter- local_contact.icon_image.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of a contact icon file.
Default Value	Blank
Range	URL within 511 characters The format of the contact avatar must be *.png, *.jpg, *.bmp, *.jpeg. The contact icon file should be uploaded to the provisioning server in advance.
Example	local_contact.icon_image.url = tftp://192.168.10.25/ Photo.jpg

Parameter- local_contact.image.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of a TAR contact avatar file.
Default Value	Blank
Range	URL within 511 characters The format of the contact avatar must be *.png, *.jpg, *.bmp, *.jpeg. The contact avatar file should be compressed as a TAR file in advance and then place it to the provisioning server.

Example	local_contact.image.url = tftp://192.168.10.25/photo.tar

Parameter- local_contact.data_photo_tar.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the compressed TAR file consisting of the avatars TAR file and contact XML file.
Default Value	Blank
Range	URL within 511 characters The format of the contact avatar must be *.png, *.jpg, *.bmp, *.jpeg. All avatars needed for contacts should be compressed as a TAR file in advance.
Example	local_contact.data_photo_tar.url = tftp://192.168.10.25/Contact.tar **Note:** The name of the avatars TAR file must be photo.tar (case-sensitive), and the name of the contact XML file must be ContactData.xml (case-sensitive).

Parameter- local_contact.icon.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of a TAR contact icon file.
Default Value	Blank
Range	URL within 511 characters The format of the contact icon must be *.png, *.jpg, *.bmp, *.jpeg. The contact icon file should be compressed as a TAR file in advance and then place it to the provisioning server.

Example	local_contact.icon.url = tftp://192.168.10.25/ photo2.tar

Remote Phone Book

Parameter- remote_phonebook.data.X.url (X ranges from 1 to 5)	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the remote phone book.
Default Value	Blank
Range	URL within 511 characters
Example	remote_phonebook.data.1.url = http://192.168.1.20/ phonebook.xml

Parameter- remote_phonebook.data.X.name (X ranges from 1 to 5)	**Configuration File** <MAC>.cfg
Description	Configures the display name of the remote phone book item.
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.data.1.name = Xmyl "Xmyl" will be displayed on the touch screen at the phone path ->Remote Phone Book. The name of Remote Phone Book can be configured by the parameter"remote_phonebook.display_name" introduced below.

Parameter- remote_phonebook.display_name	**Configuration File** <MAC>.cfg

Description	Configures the display name of the remote phone book.
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.display_name = Friends If it is left blank, Remote Phone Book will be the display name.

Parameter- features.remote_phonebook.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to perform a remote phone book search for an incoming or outgoing call and display the matched results on the touch screen.
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.remote_phonebook.enable = 0

Parameter- features.remote_phonebook.flash_time	**Configuration File** <MAC>.cfg
Description	Configures how often to refresh the local cache of the remote phone book. If it is set to 3600, the IP phone will refresh the local cache of the remote phone book every 3600 seconds (1 minute). If it is set to 0, the IP phone will refresh the local cache of the remote phone book aperiodically.
Default Value	21600

Range	0, Integer from 3600 to 1296000
Example	features.remote_phonebook.flash_time = 0

Parameter- features.remote_phonebook.enter_update_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to refresh the local cache of the remote phone book at a time when accessing the remote phone book.
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.remote_phonebook.enter_update_enable = 0

Parameter- features.remote_phonebook.enter_update_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to refresh the local cache of the remote phone book at a time when accessing the remote phone book.
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.remote_phonebook.enter_update_enable = 0

Call History Management

Parameter- features.save_call_history	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to save the call log.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone cannot log the missed calls, placed calls, received calls and forwarded calls in the call log lists.
Example	features.save_call_history = 0

Missed Call Log

Parameter- account.X.missed_calllog (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to indicate and record missed calls for account X.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone does not display a prompt message and an indicator icon on the idle screen and log the missed call in the Missed Calls list when missing calls. If it is set to 1 (Enabled), the IP phone displays a prompt message and an indicator icon on the idle screen and logs the missed call in the Missed Calls list when missing calls. **Note:** It works only if the value of the parameter "features.save_call_history" is set to 1 (Enabled). The prompt message displays only if the value of the parameter "features.missed_call_popup.enable" is set to 1 (Enabled).
Example	faccount.1.missed_calllog = 1

Replace Rule

Parameter- dialplan.replace.prefix.X (X ranges from 1 to 100)	**Configuration File** <MAC>.cfg
Description	Configures the entered number to be replaced.
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.prefix.1 = 1 Note: It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Parameter- dialplan.replace.replace.X (X ranges from 1 to 100)	**Configuration File** <MAC>.cfg
Description	Configures the alternate number to replace the entered number.
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.prefix.1 =1 and dialplan.replace.replace.1 = 254245 When you enter the number "1" and press the send key, the number "254245" will replace the entered number "1". **Note:** It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Parameter- dialplan.replace.line_id.X (X ranges from 1 to 100)	Configuration File <MAC>.cfg
Description	Configures the desired line to apply the replace rule. The digit 0 stands for all lines. If it is left blank, the replace rule will apply to all lines on the IP phone.
Default Value	Blank
Range	0, Integer from 1 to 16
Example	dialplan.replace.line_id.1 = 1,2 **Note:** Multiple line IDs are separated by commas. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Dial-now

Parameter- dialplan.dialnow.rule.X (X ranges from 1 to 100)	Configuration File <MAC>.cfg
Description	Configures the dial-now rule (the string used to match the numbers entered by the user). When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the numbers without pressing the send key.
Default Value	Blank
Range	String within 511 characters
Example	dialplan.dialnow.rule.1 = 123 Note: It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Parameter- dialplan.dialnow.line_id.X (X ranges from 1 to 100)	Configuration File <MAC>.cfg
Description	Configures the desired line to apply the dial-now rule. The digit 0 stands for all lines. If it is left blank, the dial-now rule will apply to all lines on the IP phone.
Default Value	Blank
Range	0, Integer from 1 to 16
Example	dialplan.dialnow.line_id.1 = 1,2 **Note:** Multiple line IDs are separated by commas. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Parameter- phone_setting.dialnow_delay	Configuration File <MAC>.cfg
Description	Configures the delay time (in seconds) for the dial-now rule. When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the entered number after the designated delay time. If it is set to 0, the IP phone will automatically dial out the entered number immediately. **Note:** It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).
Default Value	1
Range	Integer from 0 to 14
Example	phone_setting.dialnow_delay = 1

Area Code

Parameter- dialplan.area_code.code	Configuration File <MAC>.cfg
Description	Configures the area code to be added before the entered numbers when dialing out.
Default Value	Blank
Range	String within 16 characters
Example	dialplan.area_code.code = 0592 **Note:** The length of the entered number must be between the minimum length configured by the parameter “dialplan.area_code.min_len” and the maximum length configured by the parameter “dialplan.area_code.max_len”. It works only if the values of the parameters “dialplan.digitmap.enable” and “account.X.dialplan.digitmap.enable” are set to 0 (Disabled).

Parameter- dialplan.area_code.min_len	Configuration File <MAC>.cfg
Description	Configures the minimum length of the entered numbers.
Default Value	1
Range	1 to 15
Example	dialplan.area_code.min_len = 1 **Note:** It works only if the values of the parameters “dialplan.digitmap.enable” and “account.X.dialplan.digitmap.enable” are set to 0 (Disabled).

Parameter- dialplan.area_code.max_len	Configuration File <MAC>.cfg
Description	Configures the maximum length of the entered numbers.

Default Value	15
Range	1 to 15
Example	dialplan.area_code.max_len = 15 **Note:** The value must be larger than the minimum length. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Parameter- dialplan.area_code.line_id	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the area code rule. The digit 0 stands for all lines. If it is left blank, the area code rule will apply to all lines on the IP phone.
Default Value	Blank
Range	0, Integer from 1 to 16
Example	dialplan.area_code.line_id = 1 Note: Multiple line IDs are separated by commas. It works only if the values of the parameters "dialplan.digitmap.enable" and "account.X.dialplan.digitmap.enable" are set to 0 (Disabled).

Block Out

Parameter- dialplan.block_out.number.X (X ranges from 1 to 10)	**Configuration File** <MAC>.cfg
Description	Configures the block out numbers.
Default Value	Blank
Range	String within 16 characters

Example	dialplan.block_out.number.1 = 4321 When you dial the number “4321” on your phone, the dialing will fail and the touch screen will prompt "Forbidden Number". **Note:** It works only if the values of the parameters “dialplan.digitmap.enable” and “account.X.dialplan.digitmap.enable” are set to 0 (Disabled).

Parameter- dialplan.block_out.line_id.X (X ranges from 1 to 10)	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the block out rule. The digit 0 stands for all lines. If it is left blank, the block out rule will apply to all lines on the IP phone.
Default Value	Blank
Range	0, Integer from 1 to 16
Example	dialplan.block_out.line_id.1 = 1,2,3 **Note:** Multiple line IDs are separated by commas. It works only if the values of the parameters “dialplan.digitmap.enable” and “account.X.dialplan.digitmap.enable” are set to 0 (Disabled).

Live Dialpad

Parameter- phone_setting.predial_autodial	**Configuration File** <MAC>.cfg
Description	Enables or disables the live dialpad feature.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), the IP phone will automatically dial out the entered phone number on the dialing screen without pressing a send key.
Example	NA

Parameter- phone_setting.inter_digit_time	Configuration File <MAC>.cfg
Description	Configures the delay time (in seconds) for the IP phone to automatically dial out the entered digits without pressing a send key. **Note:** It works only if the value of the parameter "phone_setting.predial_autodial" is set to 1 (Enabled) and the value of the parameter "dialplan.digitmap.enable" is set to 0 (Disabled).
Default Value	4
Range	1 to 14
Example	NA

Hotline

Parameter- features.hotline_number	Configuration File <MAC>.cfg
Description	Configures the hotline number that the IP phone automatically dials out when you lift the handset, press the Speakerphone key or tap the line key. Leaving it blank disables hotline feature.
Default Value	Blank
Range	String 32 characters.
Example	features.hotline_number = 1234

Parameter- features.hotline_delay	Configuration File <MAC>.cfg

Description	Configures the waiting time (in seconds) for the IP phone to automatically dial out the hotline number. If it is set to 0 (0s), the IP phone will immediately dial out the preconfigured hotline number when you lift the handset, press the Speakerphone key or tap the line key. If it is set to a value greater than 0, the IP phone will wait the designated seconds before dialing out the predefined hotline number when you lift the handset, press the Speakerphone key or tap the line key.
Default Value	4
Range	0 to 10
Example	NA

Account Registration

Parameter- account.X.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the account X.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- account.X.label (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	(Optional.) Configures the label to be displayed on the touch screen for account X.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- account.X.display_name (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the display name to be displayed on the called party's touch screen for account X.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- account.X.auth_name (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the user name for register authentication for account X. **Note:** The user name for register authentication is provided by ITSP. It is always matched with a password (configured by the parameter "account.X.password") used for register authentication, if required by the server.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- account.X.user_name (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the register user name for account X. **Note:** The register user name is provided by ITSP. It is used to identify the account.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- account.X.password (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the password for register authentication for account X. **Note:** The password for register authentication is provided by ITSP.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- account.X.sip_server.Y.address (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the IP address or domain name of the SIP server Y that accepts registrations for account X.
Default Value	Blank
Range	String within 256 characters
Example	NA

Parameter- account.X.sip_server.Y.port (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the port of the SIP server Y that specifies registrations for account X.
Default Value	5060
Range	0 to 65535

Example	account.1.sip_server.1.port = 5060 **Note:** If the value of this parameter is set to 0, the port used depends on the value specified by the parameter "account.X.sip_server.Y.transport_type".

Parameter- account.X.outbound_proxy_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the port of the SIP server Y that specifies registrations for account X.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- account.X.outbound_proxy.Y.address (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the port of the SIP server Y that specifies registrations for account X.
Default Value	Blank
Range	IP address or domain name
Example	account.1.outbound_proxy.1.address = 10.1.8.11 Note: It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).

Parameter- account.X.outbound_proxy.Y.port (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg

Description	Configures the port of the outbound proxy server Y for account X.
Default Value	5060
Range	0 to 65535
Example	account.1.outbound_proxy.1.port = 5060 Note: It works only if the value of the parameter "account.X.outbound_proxy_enable" is set to 1 (Enabled).

Parameter- account.X.reg_fail_retry_interval (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to retry to re-register for account X when registration fails.
Default Value	30
Range	0 to 1800
Example	account.1.reg_fail_retry_interval = 30

Parameter- account.X.reg_fail_retry_interval (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to retry to re-register for account X when registration fails.
Default Value	30
Range	0 to 1800
Example	account.1.reg_fail_retry_interval = 30

Multiple Line Keys

Parameter- features.auto_linekeys.enable	**Configuration File** <MAC>.cfg

Description	Enables or disables the account X.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** The number of the DSS keys is determined by the value of the parameter "account.X.number_of_linekey".
Example	NA

Parameter- account.X.number_of_linekey (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the number of DSS keys to be assigned with Line type automatically from the first unused one (unused one means the DSS key is configured as N/A or Line). If a DSS key is used, the IP phone will skip to the next unused DSS key.
Default Value	1
Range	String within 32 characters
Example	account.1.number_of_linekey = 2 **Note:** It works only if the value of the parameter "features.auto_linekeys.enable" is set to 1 (Enabled).

DND

Parameter- features.dnd_mode	**Configuration File** <MAC>.cfg
Description	Configures the DND mode for the IP phone.
Default Value	0

Range	0-Phone 1-Custom If it is set to 0 (Phone), DND feature is effective for the IP phone. If it is set to 1 (Custom), you can configure DND feature for each or all accounts.
Example	features.dnd_mode = 0

Parameter- account.X.dnd.enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the DND mode for the IP phone.
Default Value	0
Range	0-Off 1-On If it is set to 1 (On), the IP phone will reject incoming calls with a busy signal (configured by the parameter "features.dnd_refuse_code") on account X. **Note:** It works only if the value of the parameter "features.dnd_mode" is set to 1 (Custom).
Example	NA

Parameter- account.X.dnd.on_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the DND on code to activate the server-side DND feature for account X. The IP phone will send the DND on code to the server when you activate DND feature for account X on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	account.1.dnd.on_code = *73 **Note:** It works only if the value of the parameter "features.dnd_mode" is set to 1 (Custom).

Parameter- account.X.dnd.off_code (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the DND off code to deactivate the server-side DND feature for account X. The IP phone will send the DND off code to the server when you deactivate DND feature for account X on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	account.1.dnd.off_code = *74 **Note:** It works only if the value of the parameter "features.dnd_mode" is set to 1 (Custom).

Parameter- features.dnd.enable	**Configuration File** <MAC>.cfg
Description	Triggers DND feature to on or off for the IP phone.
Default Value	0
Range	0-Off 1-On If it is set to 1 (On), the IP phone will reject incoming calls with a busy signal (configured by the parameter "features.dnd_refuse_code") on the IP phone. **Note:** It works only if the value of the parameter "features.dnd_model" is set to 0 (Phone).
Example	NA

Parameter- features.dnd.on_code	**Configuration File** <MAC>.cfg

Description	Configures the DND on code to activate the server-side DND feature. The IP phone will send the DND on code to the server when you activate DND feature on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	features.dnd.on_code = *71 **Note:** It works only if the value of the parameter "features.dnd_mode" is set to 0 (Phone).

Parameter- features.dnd.off_code	**Configuration File** <MAC>.cfg
Description	Configures the DND off code to deactivate the server-side DND feature. The IP phone will send the DND off code to the server when you deactivate DND feature on the IP phone.
Default Value	Blank
Range	String within 32 characters
Example	features.dnd.off_code = *72 **Note:** It works only if the value of the parameter "features.dnd_mode" is set to 0 (Phone).

Parameter- features.dnd.emergency_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to receive incoming calls from authorized numbers when DND feature is enabled.
Default Value	0

Range	0-Disabled 1-Enabled **Note:** The authorized numbers are configured by the parameter "features.dnd.emergency_authorized_number".
Example	NA

Parameter- features.dnd.emergency_authorized_number	**Configuration File** <MAC>.cfg
Description	Configures the authorized numbers the IP phone can receive incoming calls from even if DND feature is enabled. Multiple numbers are separated by commas
Default Value	Blank
Range	String within 511 characters
Example	features.dnd.emergency_authorized_number = 123,124 **Note:** It works only if the value of the parameter "features.dnd.emergency_enable" is set to 1 (Enabled).

Parameter- features.dnd_refuse_code	**Configuration File** <MAC>.cfg
Description	Configures a return code and reason of SIP response messages when rejecting an incoming call by DND. A specific reason is displayed on the caller's phone touch screen.
Default Value	480

Range	404, 480, 486 or 603 404-Not Found 480-Temporarily Unavailable 486-Busy Here 603-Decline If it is set to 486 (Busy Here), the caller's phone touch screen will display the reason "Busy Here" when the callee enables DND.
Example	NA

Parameter- linekey.X.type/ programablekey.X.type/ expansion_module.X.key.Y.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as a DND key on the IP phone. The digit 5 stands for the key type DND.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Example	linekey.2.type = 5

Headset Prior

Parameter- features.headset_prior	**Configuration File** <MAC>.cfg

Description	Enables or disables headset prior feature. You need to press the HEADSET key to activate the headset mode in advance.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the headset mode can be deactivated by pressing the Speakerphone key or the HEADSET key except off-hook. If it is set to 1 (Enabled), the headset mode will not be deactivated until the user presses the HEADSET key again.
Example	features.headset_prior = 0

Dual Headset

Parameter- features.headset_training	**Configuration File** <MAC>.cfg
Description	Enables or disables dual headset feature.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), users can use two headsets on one phone. When the IP phone joins in a call, the users with the headset connected to the headset jack have a full-duplex conversation, while the users with the headset connected to the handset jack are only allowed to listen to.
Example	features.headset_training = 0

Audio Codec

Parameter- account.X.codec.<payload_type>.enable (where <payload_type> should be replaced by the name of audio codec)	Configuration File <MAC>.cfg
Description	Enables or disables the specified codec for account X.
Default Value	When audio codec is G.722.1c(48kb/s), the default value is 1; When audio codec is G.722.1c(32kb/s), the default value is 1; When audio codec is G.722.1c(24kb/s), the default value is 1; When audio codec is G.722.1(24kb/s), the default value is 1; When audio codec is G722, the default value is 1; When audio codec is PCMU, the default value is 1; When audio codec is PCMA, the default value is 1; When audio codec is G729, the default value is 1; When audio codec is Opus, the default value is 0; When audio codec is G726-40, the default value is 0; When audio codec is G726-32, the default value is 0; When audio codec is G726-24, the default value is 0; When audio codec is G726-16, the default value is 0; When audio codec is iLBC, the default value is 0; When audio codec is G723_63, the default value is 0; When audio codec is G723_53, the default value is 0;
Range	0-Disabled 1-Enabled X ranges from 1 to 16 G722.1c(48kb/s), G722.1c(32kb/s), G722.1c(24kb/s), G722.1(24kb/s), G722, PCMU, PCMA, G729, Opus, G726-40, G726-32, G726-24, G726-16, iLBC, G723_63, G723_53

Example	account.1.codec.g722_1c_48kpbs.enable = 1 account.1.codec.g722_1c_32kpbs.enable = 1 account.1.codec.g722_1c_24kpbs.enable = 1 account.1.codec.g722_1_24kpbs.enable = 1 account.1.codec.g722.enable = 1 account.1.codec.pcmu.enable = 1 account.1.codec.pcma.enable = 1 account.1.codec.g729.enable = 1 account.1.codec.opus.enable = 0 account.1.codec.g726_40.enable = 0 account.1.codec.g726_32.enable = 0 account.1.codec.g726_24.enable = 0 account.1.codec.g726_16.enable = 0 account.1.codec.ilbc.enable = 0 account.1.codec.g723_63.enable = 0 account.1.codec.g723_53.enable = 0 It means that the codecs G.722.1c(48kb/s), G.722.1c(32kb/s), G.722.1c(24kb/s), G.722.1(24kb/s), G722, PCMU, PCMA and G729 are enabled on the account 1. **Note:** The name of audio codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.

Parameter- account.X.codec.<payload_type>.priority (where <payload_type> should be replaced by the name of audio codec)	**Configuration File** <MAC>.cfg
Description	Configures the priority of the enabled audio codec for account X. X ranges from 1 to 16

Default Value	When audio codec is G722.1c(48kb/s), the default value is 1; When audio codec is G722.1c(32kb/s), the default value is 2; When audio codec is G722.1c(24kb/s), the default value is 3; When audio codec is G722.1(24kb/s), the default value is 4; When audio codec is G722, the default value is 5; When audio codec is PCMU, the default value is 6; When audio codec is PCMA, the default value is 7; When audio codec is G729, the default value is 8; When audio codec is Opus, the default value is 0; When audio codec is G726_40, the default value is 0; When audio codec is G726_32, the default value is 0; When audio codec is G726_24, the default value is 0; When audio codec is G726_16, the default value is 0; When audio codec is iLBC, the default value is 0; When audio codec is G723_63, the default value is 0; When audio codec is G723_53, the default value is 0;
Range	G722.1c(48kb/s), G722.1c(32kb/s), G722.1c(24kb/s), G722.1(24kb/s), G722, PCMU, PCMA, G729, Opus, G726-40, G726-32, G726-24, G726-16, iLBC, G723_63, G723_53 0 to 8
Example	account.1.codec.g722_1c_48kpbs.priority = 1 account.1.codec.g722_1c_32kpbs.priority = 2 account.1.codec.g722_1c_24kpbs.priority = 3 account.1.codec.g722_1_24kpbs.priority = 4 account.1.codec.g722.priority = 5 account.1.codec.pcmu.priority = 6 account.1.codec.pcma.priority = 7 account.1.codec.g729.priority = 8 account.1.codec.opus.priority = 0 account.1.codec.g726_40.priority = 0 account.1.codec.g726_32.priority = 0 account.1.codec.g726_24.priority = 0 account.1.codec.g726_16.priority = 0 account.1.codec.ilbc.priority = 0 account.1.codec.g723_63.priority = 0 account.1.codec.g723_53.priority = 0 **Note:** The name of audio codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.

Parameter- account.X.codec.<payload_type>.rtpmap (where <payload_type> should be replaced by the name of audio codec)	Configuration File <MAC>.cfg
Description	Configures the rtpmap of the audio codec for account X. X ranges from 1 to 16
Default Value	When audio codec is G722.1c(48kb/s),the default value is 121; When audio codec is G722.1c(32kb/s),the default value is 122; When audio codec is G722.1c(24kb/s),the default value is 123; When audio codec is G722.1(24kb/s),the default value is 124; When audio codec is G726-40, the default value is 105; When audio codec is G726-32, the default value is 102; When audio codec is G726-24, the default value is 104; When audio codec is G726-16, the default value is 103; When audio codec is iLBC, the default value is 106; When audio codec is Opus, the default value is 107;
Range	G722.1c(48kb/s), G722.1c(32kb/s), G722.1c(24kb/s), G722.1(24kb/s), G726-16, G726-24, G726-32, G726-40, iLBC, Opus 0 to 127
Example	account.1.codec.g722_1c_48kpbs.rtpmap = 121 account.1.codec.g722_1c_32kpbs.rtpmap = 122 account.1.codec.g722_1c_24kpbs.rtpmap = 123 account.1.codec.g722_1_24kpbs.rtpmap = 124 account.1.codec.g726_40.rtpmap = 105 account.1.codec.g726_32.rtpmap = 102 account.1.codec.g726_24.rtpmap = 104 account.1.codec.g726_16.rtpmap = 103 account.1.codec.ilbc.rtpmap = 108 account.1.codec.opus.rtpmap = 107 **Note:** The name of audio codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.

Packetisation

Parameter- account.X.ptime (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the ptime (in milliseconds) for the codec for account X.
Default Value	20
Range	0-Disabled 10-10 20-20 30-30 40-40 50-50 60-60 If it is set to 0 (Disabled), the ptime negotiation is disabled.
Example	account.1.ptime = 20

Opus Sample Rate

Parameter- account.X.codec.opus.para (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the sample rate for the Opus codec.
Default Value	Opus-FB
Range	Opus-FB, Opus-SWB, Opus-WB, Opus-MB, Opus-NB
Example	account.1.codec.opus.para = Opus-FB

Acoustic Echo Cancellation

Parameter- voice.echo_cancellation	Configuration File <MAC>.cfg
Description	Configures the sample rate for the Opus codec.
Default Value	1
Range	0-Disabled 1-Enabled
Example	NA

VAD

Parameter- voice.vad	Configuration File <MAC>.cfg
Description	Enables or disables the VAD (Voice Activity Detection) feature on the IP phone.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Comfort Noise Generation

Parameter- voice.cng	Configuration File <MAC>.cfg
Description	Enables or disables the CNG (Comfortable Noise Generation) feature on the IP phone.
Default Value	1
Range	0-Disabled 1-Enabled
Example	NA

Jitter Buffer

Parameter- voice.jib.adaptive	**Configuration File** <MAC>.cfg
Description	Configures the type of jitter buffer in the wired network.
Default Value	1
Range	0-Fixed 1-Adaptive
Example	NA

Parameter- voice.jib.min	**Configuration File** <MAC>.cfg
Description	Configures the minimum delay time (in milliseconds) of jitter buffer in the wired network. **Note:** It works only if the value of the parameter "voice.jib.adaptive" is set to 1 (Adaptive)..
Default Value	60
Range	0 to 400
Example	NA

Parameter- voice.jib.normal	**Configuration File** <MAC>.cfg
Description	Configures the normal delay time (in milliseconds) of jitter buffer in the wired network. **Note:** It works only if the value of the parameter "voice.jib.adaptive" is set to 0 (Fixed).
Default Value	120
Range	0 to 400

Example	NA

Parameter- voice.jib.wifi.adaptive	**Configuration File** <MAC>.cfg
Description	Configures the normal delay time (in milliseconds) of jitter buffer in the wired network. **Note:** It works only if the value of the parameter "voice.jib.adaptive" is set to 0 (Fixed).
Default Value	1
Range	0-Fixed 1-Adaptive
Example	NA

Parameter- voice.jib.wifi.min	**Configuration File** <MAC>.cfg
Description	Configures the minimum delay time (in milliseconds) of jitter buffer in the wireless network. **Note:** It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 1 (Adaptive). The value of the minimum delay time should be less than or equal to that of the normal delay time (configured by the parameter "voice.jib.wifi.normal").
Default Value	60
Range	0 to 500
Example	NA

Parameter- voice.jib.wifi.max	**Configuration File** <MAC>.cfg

Description	Configures the maximum delay time (in milliseconds) of jitter buffer in the wireless network. **Note:** It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 1 (Adaptive).
Default Value	500
Range	0 to 500
Example	NA

Parameter- voice.jib.wifi.normal	**Configuration File** <MAC>.cfg
Description	Configures the normal delay time (in milliseconds) of jitter buffer in the wireless network. Note: It works only if the value of the parameter "voice.jib.wifi.adaptive" is set to 0 (Fixed). The value of the normal delay time should be less than or equal to that of the maximum delay time (configured by the parameter "voice.jib.wifi.max").
Default Value	240
Range	0 to 500
Example	NA

Video Settings

Parameter- video.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the video call feature for the IP phone.
Default Value	1

Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), video is not sent in outgoing calls and not received in incoming calls. All calls are audio-only. If it is set to 1 (Enabled), video is sent in outgoing calls and received in incoming calls.
Example	NA

Parameter- camera.flicker	**Configuration File** <MAC>.cfg
Description	Configures the value of camera flicker frequency (Hz).
Default Value	50
Range	50-50Hz 60-60Hz **Note:** Indoor lights powered by a 50Hz or 60Hz power source can produce a flicker. You can adjust the camera flicker frequency according to the power source the light is powered by.
Example	NA

Video Codecs

Parameter- account.X.video.<payload_type>.enable (where <payload_type> should be replaced by the name of video codec)	**Configuration File** <MAC>.cfg
Description	Enables or disables the specified video codec for account X. X ranges from 1 to 16

Default Value	When video codec is H264, the default value is 1; When video codec is H264HP, the default value is 1; When video codec is VP8, the default value is 1;
Range	0-Disabled 1-Enabled Video Codec: H264, H264HP, VP8
Example	account.1.video.h264.enable = 1 account.1.video.h264hp.enable = 1 account.1.video.vp8.enable = 1 It means that the codecs H264, H264HP, VP8 are enabled on the account 1. **Note:** The name of video codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.

Parameter- account.X.video.<payload_type>.priority (where <payload_type> should be replaced by the name of video codec)	**Configuration File** <MAC>.cfg
Description	Configures the priority of the enabled video codec for account X. X ranges from 1 to 16
Default Value	When video codec is H264, the default value is 2; When video codec is H264HP, the default value is 1; When video codec is VP8, the default value is 3;
Range	1, 2 or 3 Video Codec: H264, H264HP, VP8

Example	account.1.video.h264.priority = 2 account.1.video.h264hp.priority = 1 account.1.video.vp8.priority = 3 Note: The name of video codec in this parameter should be the correct one as listed in the above example, otherwise the corresponding configuration will not take effect.

RTCP-XR

Parameter- voice.rtcp_xr.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send RTCP-XR packets.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** If you change this parameter, the IP phone will reboot to make the change take effect. If it is set to 1 (Enabled), video is sent in outgoing calls and received in incoming calls.
Example	NA

Parameter- phone_setting.rtcp_xr_report.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to periodically (every 5 seconds) send RTCP-XR packets to another participating phone during a call for call quality monitoring and diagnosing.
Default Value	0

Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "voice.rtcp_xr.enable" is set to 1 (Enabled). If you change this parameter, the IP phone will reboot to make the change take effect.
Example	NA

VQ-RTCPXR

Parameter- phone_setting.vq_rtcpxr.session_report.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send a session quality report to the central report collector at the end of each call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- phone_setting.vq_rtcpxr.interval_report.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to send an interval quality report to the central report collector periodically throughout a call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- phone_setting.vq_rtcpxr_interval_period	Configuration File <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to send an interval quality report to the central report collector periodically throughout a call. **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.interval_report.enable" is set to 1 (Enabled).
Default Value	20
Range	5 to 20
Example	NA

Parameter- phone_setting.vq_rtcpxr_moslq_threshold_warning	Configuration File <MAC>.cfg
Description	Configures the threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a warning alert quality report to the central report collector. For example, a configured value of 35 corresponds to the MOS score 3.5. When the MOS-LQ value computed by the phone is less than or equal to 3.5, the phone will send a warning alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 3.5, the phone will not send a warning alert quality report to the central report collector. If it is set to blank, warning alerts are not generated due to MOS-LQ.
Default Value	Blank
Range	15 to 20
Example	NA

Parameter- phone_setting.vq_rtcpxr_moslq_threshold_critical	Configuration File <MAC>.cfg
Description	Configures the threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a critical alert quality report to the central report collector. For example, a configured value of 28 corresponds to the MOS score 2.8. When the MOS-LQ value computed by the phone is less than or equal to 2.8, the phone will send a critical alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 2.8, the phone will not send a critical alert quality report to the central report collector. If it is set to blank, critical alerts are not generated due to MOS-LQ.
Default Value	Blank
Range	15 to 40
Example	NA

Parameter- phone_setting.vq_rtcpxr_delay_threshold_warning	Configuration File <MAC>.cfg
Description	Configures the threshold value of one way delay (in milliseconds) that causes the phone to send a warning alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a warning alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a warning alert quality report to the central report collector. If it is set to blank, warning alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.
Default Value	Blank

Range	10 to 2000
Example	NA

Parameter- phone_setting.vq_rtcpxr_delay_threshold_critical	**Configuration File** <MAC>.cfg
Description	Configures the threshold value of one way delay (in milliseconds) that causes phone to send a critical alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a critical alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a critical alert quality report to the central report collector. If it is set to blank, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.
Default Value	Blank
Range	10 to 2000
Example	NA

Parameter- phone_setting.vq_rtcpxr.states_show_on_web.enable	**Configuration File** <MAC>.cfg

Description	Configures the threshold value of one way delay (in milliseconds) that causes phone to send a critical alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a critical alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a critical alert quality report to the central report collector. If it is set to blank, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- phone_setting.vq_rtcpxr.states_show_on_gui.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the voice quality data of the last call or current call to be displayed on the touch screen. You can view the voice quality data of the last call on the phone at the path Settings->Status->RTP Status. You can view the voice quality data of the current call by tapping the RTP Status soft key during a call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_start_time.enable	**Configuration File** <MAC>.cfg

Description	Enables or disables the phone to display Start Time on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_stop_time.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Current Time or Stop Time on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_local_call_id.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Local User on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to 1 (Enabled).

Example	NA

Parameter- phone_setting.vq_rtcpxr_display_remote_call_id.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Remote User on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_local_codec.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Local Codec on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_remote_codec.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Remote Codec on the touch screen.
Default Value	1

Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_jitter.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display Jitter on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_jitter_buffer_max.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display JitteBufferMax on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_packets_lost.enable	Configuration File <MAC>.cfg
Description	Enables or disables the phone to display Packets Lost on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_symm_oneway_delay.enable	Configuration File <MAC>.cfg
Description	Enables or disables the phone to display SymmOneWayDelay on the touch screen.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_round_trip_delay.enable	Configuration File <MAC>.cfg
Description	Enables or disables the phone to display RoundTripDelay on the touch screen.
Default Value	0

Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_moslq.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display MOS-LQ on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- phone_setting.vq_rtcpxr_display_moscq.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the phone to display MOS-CQ on the touch screen.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable " is set to 1 (Enabled).
Example	NA

Parameter- account.X.vq_rtcpxr.collector_name (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the host name of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X.
Default Value	Blank
Range	String within 32 characters
Example	NA

Parameter- account.X.vq_rtcpxr.collector_server_host (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the IP address of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X
Default Value	Blank
Range	IPv4 Address
Example	NA

Parameter- account.X.vq_rtcpxr.collector_server_port (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the port of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X.
Default Value	5060
Range	1 to 65535
Example	NA

TLS

Parameter- account.X.sip_server.Y.transport_type (X ranges from 1 to 16, Y ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the transport method the IP phone uses to communicate with the SIP server for account X.
Default Value	0
Range	0-UDP 1-TCP 2-TLS 3-DNS-NAPTR
Example	NA

Parameter- static.security.trust_certificates	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to only trust the server certificates in the Trusted Certificates list.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone will trust the server no matter whether the certificate sent by the server is valid or not. If it is set to 1 (Enabled), the IP phone will authenticate the server certificate based on the trusted certificates list. Only when the authentication succeeds, the IP phone will trust the server. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Example	NA

Parameter- static.security.ca_cert	**Configuration File** <MAC>.cfg
Description	Configures the type of certificates in the Trusted Certificates list for the IP phone to authenticate for TLS connection.
Default Value	2
Range	0-Default Certificates 1-Custom Certificates 2-All Certificates **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Example	NA

Parameter- static.security.cn_validation	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to mandatorily validate the CommonName or SubjectAltName of the certificate sent by the server.
Default Value	0
Range	0-Default Certificates 1-Custom Certificates 2-All Certificates **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Example	NA

Parameter- static.security.dev_cert	**Configuration File** <MAC>.cfg
Description	Configures the type of the device certificates for the IP phone to send for TLS authentication.
Default Value	0

Range	0-Default Certificates 1-Custom Certificates **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Example	NA

Parameter- static.trusted_certificates.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the custom trusted certificate used to authenticate the connecting server.
Default Value	Blank
Range	URL within 511 characters
Example	static.trusted_certificates.url = http://192.168.1.20/tc.crt **Note:** The certificate you want to upload must be in *.pem, *.crt, *.cer or *.der format.

Parameter- static.server_certificates.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the server certificate the IP phone sends for authentication.
Default Value	Blank
Range	URL within 511 characters
Example	static.server_certificates.url = http://192.168.1.20/ca.pem **Note:** The certificate you want to upload must be in *.pem or *.cer format.

Parameter- static.server_certificates.delete	Configuration File <MAC>.cfg
Description	Deletes all uploaded server certificates.
Default Value	Blank
Range	http://localhost/all
Example	static.server_certificates.delete = http://localhost/all

Parameter- static.phone_setting.reserve_certs_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to reserve custom certificates after it is reset to factory defaults.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

SRTP

Parameter- account.X.srtp_encryption (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures whether to use audio/video encryption service for account X.
Default Value	0

Range	0-Disabled 1-Optional 2-Compulsory If it is set to 0 (Disabled), the IP phone will not use audio/voice encryption service. If it is set to 1 (Optional), the IP phone will negotiate with the other IP phone what type of encryption to utilize for the session. If it is set to 2 (Compulsory), the IP phone is forced to use SRTP during a call.
Example	NA

Encrypting and Decrypting Files

Parameter- static.auto_provision.update_file_mode	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone only to download the encrypted files.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone will download the configuration files (e.g., sip.cfg, account.cfg, <MAC>-local.cfg) and <MAC>-contact.xml file from the server during auto provisioning no matter whether the files are encrypted or not. And then resolve these files and update settings onto the IP phone system. If it is set to 1 (Enabled), the IP phone will only download the encrypted configuration files (e.g., sip.cfg, account.cfg, <MAC>-local.cfg) or <MAC>-contact.xml file from the server during auto provisioning, and then resolve these files and update settings onto the IP phone system.
Example	NA

Parameter- static.auto_provision.aes_key_in_file	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone only to download the encrypted files.
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone will decrypt the encrypted configuration files using plain text AES keys configured on the IP phone. If it is set to 1 (Enabled), the IP phone will download <xx_Security>.enc files (e.g., <sip_Security>.enc, <account_Security>.enc) during auto provisioning, and then decrypts these files into the plaintext keys (e.g., key2, key3) respectively using the phone built-in key (e.g., key1). The IP phone then decrypts the encrypted configuration files using corresponding key (e.g., key2, key3).
Example	NA

Parameter- static.auto_provision.aes_key_16.com	**Configuration File** <MAC>.cfg
Description	Configures the plaintext AES key for encrypting/ decrypting the Common CFG/Custom CFG file.
Default Value	Blank
Range	16 characters The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # $ % * + , - . : = ? @ [ ] ^ _ { } ~.

Example	static.auto_provision.aes_key_16.com = 0123456789abcdef **Note:** For decrypting, it works only if the value of the parameter "static.auto_provision.aes_key_in_file" is set to 0 (Disabled). If the downloaded MAC-Oriented file is encrypted and the parameter "static.auto_provision.aes_key_16.mac" is left blank, the IP phone will try to encrypt/decrypt the MAC-Oriented file using the AES key configured by the parameter "static.auto_provision.aes_key_16.com".

Parameter- static.auto_provision.aes_key_16.mac	**Configuration File** <MAC>.cfg
Description	Configures the plaintext AES key for encrypting/ decrypting the MAC-Oriented files (<MAC>.cfg, <MAC>-local.cfg and <MAC>-contact.xml).
Default Value	Blank
Range	16 characters The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # $ % * + , - . : = ? @ [ ] ^ _ { } ~.
Example	static.auto_provision.aes_key_16.mac = 0123456789abmins **Note:** For decrypting, it works only if the value of the parameter "static.auto_provision.aes_key_in_file" is set to 0 (Disabled). If the downloaded MAC-Oriented file is encrypted and the parameter "static.auto_provision.aes_key_16.mac" is left blank, the IP phone will try to encrypt/decrypt the MAC-Oriented file using the AES key configured by the parameter "static.auto_provision.aes_key_16.com".

Parameter- static.auto_provision.encryption.config	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to encrypt <MAC>-local.cfg file using the plaintext AES key
Default Value	0
Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the MAC-local CFG file is uploaded unencrypted and replaces the one (encrypted or unencrypted) stored on the server if you have configured to back up the MAC-local CFG file to the server by the parameter "static.auto_provision.custom.sync". If it is set to 1 (Enabled), the MAC-local CFG file is uploaded encrypted and replaces the one (encrypted or unencrypted) stored on the server if you have configured to back up the MAC-local CFG file to the server by the parameter "static.auto_provision.custom.sync". The plaintext AES key is configured by the parameter "static.auto_provision.aes_key_16.mac".
Example	static.auto_provision.aes_key_16.mac = 0123456789abmins **Note:** For decrypting, it works only if the value of the parameter "static.auto_provision.aes_key_in_file" is set to 0 (Disabled). If the downloaded MAC-Oriented file is encrypted and the parameter "static.auto_provision.aes_key_16.mac" is left blank, the IP phone will try to encrypt/decrypt the MAC-Oriented file using the AES key configured by the parameter "static.auto_provision.aes_key_16.com".

Encrypting and Decrypting Files

Parameter- static.auto_provision.update_file_mode	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone only to download the encrypted files.
Default Value	0

Range	0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone will download the configuration files (e.g., sip.cfg, account.cfg, <MAC>-local.cfg) and <MAC>-contact.xml file from the server during auto provisioning no matter whether the files are encrypted or not. And then resolve these files and update settings onto the IP phone system. If it is set to 1 (Enabled), the IP phone will only download the encrypted configuration files (e.g., sip.cfg, account.cfg, <MAC>-local.cfg) or <MAC>-contact.xml file from the server during auto provisioning, and then resolve these files and update settings onto the IP phone system.
Example	NA

Auto-configuration

Parameter- static.auto_provision.power_on	**Configuration File** <MAC>.cfg
Description	Triggers the power on feature to on or off.
Default Value	1
Range	0-Off 1-On If it is set to 1 (On), the IP phone will perform an auto provisioning process when powered on.
Example	NA

Parameter- static.auto_provision.repeat.enable	**Configuration File** <MAC>.cfg
Description	Triggers the repeatedly feature to on or off.
Default Value	1

Range	0-Off 1-On If it is set to 1 (On), the IP phone will perform an auto provisioning process repeatedly.
Example	NA

Parameter- static.auto_provision.repeat.minutes	**Configuration File** <MAC>.cfg
Description	Triggers the repeatedly feature to on or off.
Default Value	1440
Range	1 to 43200
Example	NA

Parameter- static.auto_provision.weekly.enable	**Configuration File** <MAC>.cfg
Description	Triggers the weekly feature to on or off.
Default Value	1440
Range	0-Off 1-On If it is set to 1 (On), the IP phone will perform an auto provisioning process weekly.
Example	NA

Parameter- static.auto_provision.weekly.begin_time	**Configuration File** <MAC>.cfg
Description	Triggers the weekly feature to on or off.
Default Value	00:00

Range	00:00 to 23:59
Example	NA

Parameter- static.auto_provision.weekly.end_time	**Configuration File** <MAC>.cfg
Description	Configures the end time of the day for the IP phone to perform an auto provisioning process weekly. **Note:** It works only if the value of the parameter "static.auto_provision.weekly.enable" is set to 1 (On).
Default Value	00:00
Range	00:00 to 23:59
Example	NA

Parameter- static.auto_provision.weekly.dayofweek	**Configuration File** <MAC>.cfg
Description	Configures the days of the week for the IP phone to perform an auto provisioning process weekly.
Default Value	0123456
Range	0, 1, 2, 3, 4, 5, 6 or a combination of these digits 0-Sunday 1-Monday 2-Tuesday 3-Wednesday 4-Thursday 5-Friday 6-Saturday

Example	static.auto_provision.weekly.dayofweek = 01 It means the IP phone will perform an auto provisioning process every Sunday and Monday. **Note:** It works only if the value of the parameter “static.auto_provision.weekly.enable” is set to 1 (On).

Parameter- static.auto_provision.flexible.enable	**Configuration File** <MAC>.cfg
Description	Triggers the flexible feature to on or off.
Default Value	0
Range	0-Off 1-On If it is set to 1 (On), the IP phone will perform an auto provisioning process at random between a starting time configured by the parameter "static.auto_provision.flexible.begin_time" and an ending time configured by the parameter "static.auto_provision.flexible.end_time" on a random day within the period configured by the parameter "static.auto_provision.flexible.interval". **Note:** The day within the period is decided based upon the phone's MAC address and does not change with a reboot whereas the time within the start and end is calculated again with every reboot.
Example	NA

Parameter- static.auto_provision.flexible.interval	**Configuration File** <MAC>.cfg
Description	Configures the interval (in days) for the IP phone to perform an auto provisioning process. The auto provisioning occurs on a random day within this period based on the phone's MAC address.
Default Value	1
Range	1 to 1000

Example	static.auto_provision.flexible.interval = 30 The IP phone will perform an auto provisioning process on a random day (e.g., 18) based on the phone's MAC address. Note: It works only if the value of the parameter "static.auto_provision.flexible.enable" is set to 1 (On).

BLF

Parameter- features.pickup.blf_visual_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to display a visual alert when the monitored user receives an incoming call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- features.pickup.blf_audio_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to play an audio alert when the monitored user receives an incoming call.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- account.X.blf.subscribe_period (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg

Description	Configures the period (in seconds) of the BLF subscription for account X. The IP phone is able to successfully refresh the SUBSCRIBE before expiration of the SUBSCRIBE dialog.
Default Value	1800
Range	30 to 2147483647
Example	NA

Parameter- account.X.blf.subscribe_event (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the event of the BLF subscription for account X.
Default Value	0
Range	0-dialog 1-presence
Example	NA

Parameter- account.X.out_dialog_blf_enable (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to handle NOTIFY messages out of the BLF dialog for account X.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg

Description	Configures a DSS key as a BLF key on the IP phone. The digit 16 stands for the key type BLF. For line keys: X ranges from 1 to 27.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Range	16
Example	linekey.2.type = 16

Parameter- linekey.X.line	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the BLF key. For line keys: X ranges from 1 to 27.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.
Range	1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Example	linekey.2.line = 1

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg

Description	Configures the phone number or extension of the monitored user. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.value = 1008

Parameter- linekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the phone number or extension of the monitored user. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.value = 1008

Parameter- linekey.X.pickup_value	**Configuration File** <MAC>.cfg
Description	Configures the pickup code for BLF feature. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 256 characters

Example	line.2.pickup_value = *88 **Note:** This parameter only applies to BLF feature.

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	line.2.pickup_value = *88 **Note:** This parameter only applies to BLF feature.

Dial Plan using Digit Map String Rules

Parameter- dialplan.digitmap.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the digit map feature for the IP phone.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** The value configured by the parameter "account.X.dialplan.digitmap.enable" takes precedence over that configured by this parameter
Example	NA

Parameter- dialplan.digitmap.string	Configuration File <MAC>.cfg
Description	Configures digit map pattern used for the dial plan.
Default Value	[2-9]11 | 0T | 011xxx.T | [0-1][2-9]xx xxxxxxx | [2-9]xx xxxxxxx | [2-9]xxxT | **x.T | +x.T| 00x.T
Range	String within 2048 characters
Example	dialplan.digitmap.string = <[2-9]x:86>3.T|0x.!|1xxx Note: The string must be compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.string" takes precedence over that configured by this parameter.

Parameter- dialplan.digitmap.interdigit_long_timer	Configuration File <MAC>.cfg
Description	Configures the time (in seconds) for the IP phone to wait before dialing an entered number if it matches part of any string of the digit map. If it is set to 0, the IP phone will not dial the entered number if it only a partial match exists. The value of this parameter should be greater than that configured by the parameter "dialplan.digitmap.interdigit_short_timer".
Default Value	10
Range	0 to 255

Example	dialplan.digitmap.string = 1xxT|xxxxx<T1> dialplan.digitmap.interdigit_long_timer = 10 dialplan.digitmap.interdigit_short_timer = 5 When you enter 1, it matches part of two digit maps, the IP phone tries to wait 10 seconds and then dials out 1 if no numbers entered; When you enter 15, it also matches part of two digit maps, the IP phone tries to wait 10 seconds and then dials out 15 if no numbers entered; When you enter 153, it also matches part of two digit maps, the IP phone tries to wait 10 seconds. But after waiting for 5 seconds, it completely matches the first digit map and then immediately dials out 153. **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.interdigit_long_timer" takes precedence over that configured by this parameter.

Parameter- dialplan.digitmap.interdigit_short_timer	**Configuration File** <MAC>.cfg
Description	Configures the timeout interval (in seconds) for any string of digit map. The IP phone will wait this many seconds before matching the entered digits to the dial plan and placing the call.
Default Value	3
Range	Single configuration (configure a specific value for the timer letter "T" for all strings with "T" of the digit map)

Example	dialplan.digitmap.interdigit_short_timer = 5 If the value of the parameter 'dialplan.digitmap.string' is set to <[2-9]x:86>3.T|0T, the IP phone will wait 5 seconds before matching the entered digits to this dial plan and placing the call. Distribution configuration (configure a string of positive integers separated by "|" for each string of the digit map in the corresponding position) If there are more digit maps than timeout values, the last timeout is applied to the extra digit map. If there are more timeout values than digit maps, the extra timeout values are ignored. Example: dialplan.digitmap.interdigit_short_timer = 4|5|3|6|2|1 If the value of the parameter 'dialplan.digitmap.string' is set to <[2-9]x:86>3.T|2T|1xxT|0x.!|[2-9]11T, 4 is applied to the "<[2-9]x:86>3.T" digit map, 5 is applied to "2T" digit map, 3 is applied to "1xxT" digit map, 6 is applied to "0x.!" digit map, 2 is applied to the "[2-9]11" digit map, the last digit 1 is ignored. **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.interdigit_short_timer" takes precedence over that configured by this parameter.

Parameter- dialplan.digitmap.no_match_action	**Configuration File** <MAC>.cfg
Description	Configures the behavior when an impossible digit map match occurs.
Default Value	0

Range	0, 1 or 2 0-prevent users from entering a number and immediately dial out the entered numbers 1-the dialing will fail and the touch screen will prompt "Forbidden Number" 2-allow users to accumulate digits and dispatch call manually with the send key or automatically dial out the entered number after a certain period of time configured by the parameter "dialplan.digitmap.interdigit_long_timer" **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.no_match_action" takes precedence over that configured by this parameter.
Example	NA

Parameter- dialplan.digitmap.active.on_hook_dialing	**Configuration File** <MAC>.cfg
Description	Enables or disables the entered numbers to match the predefined string of the digit map in real time. It is only applicable to the on-hook dialing.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.active.on_hook_dialing" takes precedence over that configured by this parameter.
Example	NA

Parameter- dialplan.digitmap.apply_to.on_hook_dial	Configuration File <MAC>.cfg
Description	Enables or disables the entered number to match the predefined string of the digit map after pressing a send key when dialing on-hook or pressing the DSS key (e.g., speed dial, BLF or prefix key).
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.on_hook_dial" takes precedence over that configured by this parameter.
Example	NA

Parameter- dialplan.digitmap.apply_to.directory_dial	Configuration File <MAC>.cfg
Description	Enables or disables the digit map to be applied to the numbers dialed from the directory.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.directory_dial" takes precedence over that configured by this parameter.
Example	NA

Parameter- dialplan.digitmap.apply_to.forward	Configuration File <MAC>.cfg
Description	Enables or disables the digit map to be applied to the numbers that you want to forward to when performing call forward.
Default Value	1
Range	0-Disabled 1-Enabled If it is set to 1 (Enabled), the incoming calls will be forwarded to a desired destination number according to the string of the digit map. **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.forward" takes precedence over that configured by this parameter.
Example	NA

Parameter- dialplan.digitmap.apply_to.press_send	Configuration File <MAC>.cfg
Description	Enables or disables the entered number to match the predefined string of the digit map after pressing a send key. It is only applicable to off-hook dialing. The off-hook dialing includes: pick up the handset, press the Speakerphone key or press the line key when the phone is idle.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if the value of the parameter "dialplan.digitmap.enable" or "account.X.dialplan.digitmap.enable" is set to 1 (Enabled). The value configured by the parameter "account.X.dialplan.digitmap.apply_to.press_send" takes precedence over that configured by this parameter.

Example	NA

Dial Plan Digit Map - Per-Line Parameters

The parameters listed in the above table have a per-line equivalent that you can configure. All of the per-line parameters are listed in the following table. Note that the per-line parameters take precedence over the global parameters. For example, “account.X.dialplan.digitmap.enable” takes precedence over “dialplan.digitmap.enable”.

X stands for the serial number of the account.

X ranges from 1 to 16

Per-Line Parameters	Global Parameters
account.X.dialplan.digitmap.enable	dialplan.digitmap.enable
account.X.dialplan.digitmap.string	dialplan.digitmap.string
account.X.dialplan.digitmap.interdigit_long_timer	dialplan.digitmap.interdigit_long_timer
account.X.dialplan.digitmap.interdigit_short_timer	dialplan.digitmap.interdigit_short_timer
account.X.dialplan.digitmap.no_match_action	dialplan.digitmap.no_match_action
account.X.dialplan.digitmap.active.on_hook_dialing	dialplan.digitmap.active.on_hook_dialing
account.X.dialplan.digitmap.apply_to.on_hook_dial	dialplan.digitmap.apply_to.on_hook_dial
account.X.dialplan.digitmap.apply_to.directory_dial	dialplan.digitmap.apply_to.directory_dial
account.X.dialplan.digitmap.apply_to.forward	dialplan.digitmap.apply_to.forward
account.X.dialplan.digitmap.apply_to.press_send	dialplan.digitmap.apply_to.press_send

TR069 Device Management

Parameter- static.managementserver.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the TR-069 feature.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- static.managementserver.username	**Configuration File** <MAC>.cfg

Description	Configures the user name for the IP phone to authenticate with the ACS (Auto Configuration Servers). Leave it blank if no authentication is required.
Default Value	Blank
Range	String within 128 characters
Example	static.managementserver.username = tr69

Parameter- static.managementserver.password	**Configuration File** <MAC>.cfg
Description	Configures the password for the IP phone to authenticate with the ACS (Auto Configuration Servers). Leave it blank if no authentication is required.
Default Value	Blank
Range	String within 64 characters
Example	static.managementserver.password = tr69

Parameter- static.managementserver.url	**Configuration File** <MAC>.cfg
Description	Configures the access URL of the ACS (Auto Configuration Servers).
Default Value	Blank
Range	String within 511 characters
Example	static.managementserver.url = http://officetelprov.orangero.net:8080/ftacs-digest/ACS **Note:** IP phones also support obtaining the URL of the ACS by detecting DHCP option 43.

Parameter- static.managementserver.connection_request_username	Configuration File <MAC>.cfg
Description	Configures the user name for the IP phone to authenticate the incoming connection requests of the ACS (Auto Configuration Servers).
Default Value	Blank
Range	String within 128 characters
Example	static.managementserver.connection_request_username = accuser

Parameter- static.managementserver.connection_request_password	Configuration File <MAC>.cfg
Description	Configures the password for the IP phone to authenticate the incoming connection requests of the ACS (Auto Configuration Servers).
Default Value	Blank
Range	String within 64 characters
Example	static.managementserver.connection_request_password = acspwd

Parameter- static.managementserver.periodic_inform_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to periodically report its configuration information to the ACS (Auto Configuration Servers).
Default Value	1
Range	0-Disabled 1-Enabled

Example	NA

Parameter- static.managementserver.periodic_inform_interval	**Configuration File** <MAC>.cfg
Description	Configures the interval (in seconds) for the IP phone to report its configuration to the ACS (Auto Configuration Servers). Note: It works only if the value of the parameter "managementserver.periodic_inform_enable" is set to 1 (Enabled).
Default Value	60
Range	Integer from 5 to4294967295
Example	NA

Shared Call Appearance

Parameter- account.X.shared_line (X ranges from 1 to 16)	**Configuration File** <MAC>.cfg
Description	Configures the registration line type for account X.
Default Value	0
Range	0-Disabled 1-Shared Call Appearance If it is set to 0 (Disabled), the shared line feature is disabled.
Example	NA

Parameter- account.X.shared_line_callpull_code (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the registration line type for account X.
Default Value	Blank
Range	String within 32 characters
Example	NA

Parameter- account.X.shared_line_callpull_code (X ranges from 1 to 16)	Configuration File <MAC>.cfg
Description	Configures the call pull feature access code to retrieve an existing call from another shared phone that is in active or public hold status for account X. **Note:** It works only if the value of the parameter "account.X.shared_line" is set to 1 (Shared Call Appearance).
Default Value	Blank
Range	String within 32 characters
Example	NA

Private Hold Soft Key

Parameter- phone_setting.custom_softkey_enable	Configuration File <MAC>.cfg
Description	Enables or disables custom soft keys layout feature.
Default Value	0
Range	0-Disabled 1-Enabled

Example	NA

Parameter- custom_softkey_talking.url	**Configuration File** <MAC>.cfg
Description	Enables or disables custom soft keys layout feature.
Default Value	Blank
Range	URL within 511 characters
Example	NA

Parameter- custom_softkey_talking.url	**Configuration File** <MAC>.cfg
Description	Enables or disables custom soft keys layout feature.
Default Value	Blank
Range	URL within 511 characters
Example	NA

Private Hold Key

Parameter- linekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key to be a private hold key on the IP phone.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).

Range	20 The digit 20 stands for the key type Private Hold. For line keys: X ranges from 1 to 27.
Example	linekey.2.type = 20

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	(Optional.) Configures the label displayed on the touch screen for each DSS key. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.type = 20

Advance Maintenance

Parameter- static.syslog.log_level	**Configuration File** <MAC>.cfg
Description	Configures the detail level of syslog information to be exported. 0-system is unusable 1-action must be taken immediately 2-critical condition 3-error conditions 4-warning conditions 5-normal but significant condition 6-informational **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.

Default Value	3
Range	Integer from 0 to 6
Example	static.syslog.log_level = 3

Parameter- static.syslog.mode	**Configuration File** <MAC>.cfg
Description	Configures the IP phone to export log files to the local system, syslog server or an FTP/TFTP Server (provisioning server). 0-Local 1-Server 2-FTP/TFTP Server **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	0
Range	0,1 or 2
Example	static.syslog.mode = 0

Parameter- static.syslog.server	**Configuration File** <MAC>.cfg
Description	Configures the IP address or domain name of the syslog server when exporting log to the syslog server. **Note:** It works only if the value of the parameter "static.syslog.mode" is set to 1 (Server). If you change this parameter, the IP phone will reboot to make the change take effect.
Default Value	Blank
Range	IP address or domain name
Example	static.syslog.server = 192.168.1.100

LDAP

Parameter- ldap.enable	**Configuration File** <MAC>.cfg

Description	Enables or disables LDAP feature on the IP phone.
Default Value	0
Range	**0** - Disabled **1** - Enabled
Example	ldap.enable = 0

Parameter- ldap.name_filter	**Configuration File** <MAC>.cfg
Description	Configures the search criteria for LDAP contact names look up. The "*" symbol in the filter stands for any character. The "%" symbol in the filter stands for the name prefix entered by the user. Example: ldap.name_filter = (|(cn=%)(sn=%)) When the cn or sn of the LDAP contact starts with the entered prefix, the record will be displayed on the touch screen. ldap.name_filter = (&(cn=*)(sn=%)) When the cn of the LDAP contact is set and the sn of the LDAP contact start with the entered prefix, the records will be displayed on the phone touch screen. ldap.name_filter = (!(cn=%)) When the cn of the LDAP contact does not start with the entered prefix, the records will be displayed on the phone touch screen.
Default Value	Blank
Range	String within 99 characters
Example	ldap.name_filter = (|(cn=%)(sn=%))

Parameter- ldap.number_filter	**Configuration File** <MAC>.cfg

Description	Configures the search criteria for LDAP contact numbers look up. The “*” symbol in the filter stands for any number. The “%” symbol in the filter stands for the number prefix entered by the user. Example: ldap.number_filter = (|(telephoneNumber=%)(mobile=%)(ipPhone=%)) When the number prefix of the telephoneNumber, mobile or ipPhone of the contact record matches the search criteria, the record will be displayed on the touch screen. ldap.number_filter = (&(telephoneNumber=*)(mobile=%)) When the telephoneNumber of the LDAP contact is set and the mobile of the LDAP contact starts with the entered prefix, the record will be displayed on the phone touch screen.
Default Value	Blank
Range	String within 99 characters
Example	ldap.number_filter = (|(telephoneNumber=%)(mobile=%)(ipPhone=%))

Parameter- ldap.tls_mode	**Configuration File** <MAC>.cfg
Description	Configures the connection mode between the LDAP server and the IP phone. 0 - LDAP—Unencrypted connection between LDAP server and the IP phone (port 389 is used by default). 1 - LDAP TLS Start—TLS/SSL connection between LDAP server and the IP phone (port 389 is used by default). 2 - LDAPs—TLS/SSL connection between LDAP server and the IP phone (port 636 is used by default).
Default Value	0
Range	0,1 or 2
Example	ldap.tls_mode = 0

Parameter- ldap.host	**Configuration File** <MAC>.cfg

Description	Configures the IP address or domain name of the LDAP server.
Default Value	Blank
Range	IP address or domain name
Example	ldap.host = 192.168.1.20

Parameter- ldap.port	**Configuration File** <MAC>.cfg
Description	Configures the port of the LDAP server.
Default Value	389
Range	Integer from 1 to 65535
Example	ldap.port = 389

Parameter- ldap.base	**Configuration File** <MAC>.cfg
Description	Configures the LDAP search base which corresponds to the location of the LDAP phone book from which the LDAP search request begins. The search base narrows the search scope and decreases directory search time.
Default Value	Blank
Range	String within 99 characters
Example	ldap.base = dc=matrix,dc=cn

Parameter- ldap.user	**Configuration File** <MAC>.cfg
Description	Configures the user name used to login the LDAP server. This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the user name to login the LDAP server.
Default Value	Blank
Range	String within 99 characters

Example	ldap.user = cn=manager,dc=matrix,dc=cn

Parameter- ldap.password	**Configuration File** <MAC>.cfg
Description	Configures the password used to login the LDAP server. This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the password to login the LDAP server.
Default Value	Blank
Range	String within 99 characters
Example	ldap.password = secret

Parameter- ldap.max_hits	**Configuration File** <MAC>.cfg
Description	Configures the maximum number of search results to be returned by the LDAP server. If it is set to blank, the LDAP server will return all searched results. **Note:** A very large value of this parameter will slow down the LDAP search speed, therefore it should be configured according to the available bandwidth.
Default Value	50
Range	Integer from 1 to 32000
Example	ldap.max_hits = 50

Parameter- ldap.name_attr	**Configuration File** <MAC>.cfg

Description	Configures the name attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple name attributes separated by spaces. Example: ldap.name_attr = cn sn This requires the “cn” and “sn” attributes set for each contact record on the LDAP server.
Default Value	Blank
Range	String within 99 characters
Example	ldap.name_attr = cn sn

Parameter- ldap.numb_attr	**Configuration File** <MAC>.cfg
Description	Configures the number attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple number attributes separated by spaces. Example: ldap.numb_attr = mobile ipPhone This requires the “mobile” and “ipPhone” attributes set for each contact record on the LDAP server.
Default Value	Blank
Range	String within 99 characters
Example	ldap.numb_attr = mobile ipPhone

Parameter- ldap.display_name	**Configuration File** <MAC>.cfg
Description	Configures the display name of the contact record displayed on the touch screen. The value must start with “%” symbol. Example: ldap.display_name = %cn The cn of the contact record is displayed on the touch screen.
Default Value	Blank
Range	String within 99 characters

Example	ldap.display_name = %cn

Parameter- ldap.version	**Configuration File** <MAC>.cfg
Description	Configures the LDAP protocol version supported by the IP phone. Make sure the protocol value corresponds with the version assigned on the LDAP server.
Default Value	3
Range	2 or 3
Example	ldap.version = 3

Parameter- ldap.call_in_lookup	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to perform an LDAP search when receiving an incoming call.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	ldap.call_in_lookup = 0

Parameter- ldap.call_out_lookup	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to perform an LDAP search when placing a call.
Default Value	1
Range	0 - Disabled 1 - Enabled
Example	ldap.call_out_lookup = 1

Parameter- ldap.ldap_sort	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to sort the search results in alphabetical order or numerical order.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	ldap.ldap_sort = 0

Parameter- ldap.incoming_call_special_search.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to search the telephone numbers starting with ”+” symbol and “00” from the LDAP server if the incoming phone number starts with”+” or “00”. When completing the LDAP search, the all search results will be displayed on the touch screen. For example, If the phone receives an incoming call from the phone number 0044123456789, it will search 0044123456789 from the LDAP sever first, if no result found, it will search +44123456789 from the server again. The phone will display all the search results. **Note:** It works only if the value of the parameter “ldap.call_in_lookup” is set to 1 (Enabled). You may need to set the value of the parameter “ldap.name_filter” to be (|(cn=%)(sn=%)(telephoneNumber=%)(mobile=%)) for searching the telephone numbers starting with ”+” symbol.
Default Value	0
Range	0 - Disabled 1 - Enabled
Example	ldap.incoming_call_special_search.enable= 0

LDAP Key

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as an LDAP key on the IP phone. The digit 38 stands for the key type LDAP. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14. Example: linekey.2.type = 38 Default: For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Default Value	Refer to the above content
Range	Refer to the above content
Example	NA

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key (Optional). For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	NA

Door Phone

Parameter- features.doorphone.X.display_name (X ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the display name to be displayed on the IP phone's screen for door phone.
Default Value	Blank
Range	String within 99 characters
Example	NA

Parameter- features.doorphone.X.phone_number(X ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the phone number of the door phone.
Default Value	Blank
Range	String within 32 characters
Example	NA

Parameter- features.doorphone.X.unlock_pin (X ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Configures the unlock PIN of the door phone. The unlock PIN should be suffixed by "*"
Default Value	Blank
Range	String within 99 characters
Example	features.doorphone.1.unlock_pin=8888*

Parameter- features.doorphone.X.full_screen (X ranges from 1 to 2)	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to display the door phone call in full-screen mode automatically.
Default Value	0
Range	0 - Disabled 1 -Enabled
Example	features.doorphone.1.full_screen = 0

Parameter- features.doorphone.X.send_audio (X ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to transmit audio during a door phone call.
Default Value	1
Range	0 - Disabled 1 -Enabled
Example	features.doorphone.1.send_audio = 1

Parameter- features.doorphone.X.send_video (X ranges from 1 to 2)	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to transmit your video during a door phone call.
Default Value	1
Range	0 - Disabled 1 -Enabled
Example	features.doorphone.1.send_video = 1

Transfer mode via DssKey

Parameter- transfer.dsskey_deal_type	**Configuration File** <MAC>.cfg

Description	Configures the transfer mode when user presses the DSS key during an active call. 0-New Call 1-Attended Transfer 2-Blind Transfer **Note:** To use this feature, you need to configure the DSS key as a speed dial, transfer or BLF/BLF List in advance.
Default Value	2
Range	0-New Call 1-Attended Transfer 2-Blind Transfer
Example	transfer.dsskey_deal_type= 2

Emergency Dialplan

Parameter- dialplan.emergency.asserted_id_source	**Configuration File** <MAC>.cfg
Description	Configures the precedence of source of emergency outbound identities when placing an emergency call. If it is set to ELIN, the outbound identity used in the P-Asserted-Identity (PAI) header of the SIP INVITE request is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN). The custom outbound identity configured by "dialplan.emergency.custom_asserted_id" will be used if the phone fails to get the LLDP-MED ELIN value. If it is set to CUSTOM, the custom outbound identity configured by "dialplan.emergency.custom_asserted_id" will be used; if the value of the parameter "dialplan.emergency.custom_asserted_id" is left blank, the LLDP-MED ELIN value will be used. **Note:** If the obtained LLDP-MED ELIN value is blank and no custom outbound identity, the PAI header will not be included in the SIP INVITE request.
Default Value	ELIN
Range	ELIN or CUSTOM
Example	NA

Parameter- dialplan.emergency.custom_asserted_id	Configuration File <MAC>.cfg
Description	Configures the custom outbound identity when placing an emergency call. If using a TEL URI, for example, tel:+16045558000. The full URI is included in the P-Asserted-Identity (PAI) header (e.g., <tel:+16045558000>). If using a SIP URI, for example, sip:1234567890123@abc.com. The full URI is included in the P-Asserted-Identity (PAI) header and the address will be replaced by the emergency server (e.g., <sip:1234567890123@emergency.com>). If using a 10-25 digit number, for example, 1234567890. The SIP URI constructed from the number and SIP server (e.g., abc.com) is included in the P-Asserted-Identity (PAI) header (e.g., <sip:1234567890@abc.com>).
Default Value	Blank
Range	10-25 digits, SIP URI, or TEL URI
Example	NA

Parameter- dialplan.emergency.server.X.address (X ranges from 1 to 3)	Configuration File <MAC>.cfg
Description	Configures the IP address or domain name of the emergency server X to be used for routing calls. **Note:** If the account is registered successfully or failed (the account information has been configured), the emergency calls will be dialed using the following priority: SIP server>emergency server; if the account is not registered, the emergency server will be used.
Default Value	Blank
Range	IP address or domain name
Example	NA

Parameter- dialplan.emergency.server.X.port (X ranges from 1 to 3)	**Configuration File** <MAC>.cfg
Description	Configures the port of emergency server X to be used for routing calls.
Default Value	5060
Range	Integer from 1 to 65535
Example	NA

Parameter- dialplan.emergency.server.X.transport_type (X ranges from 1 to 3)	**Configuration File** <MAC>.cfg
Description	Configures the transport method the IP phone uses to communicate with the emergency server X.
Default Value	0
Range	0, 1, 2 or 3 0-UDP 1-TCP 2-TLS 3-DNS-NAPTR
Example	NA

Parameter- dialplan.emergency.X.value (X ranges from 1 to 255)	**Configuration File** <MAC>.cfg
Description	Configures the emergency number to use on your IP phone so a caller can contact emergency services in the local area when required.
Default Value	When X = 1, the default value is 911; When X = 2-255, the default value is Blank.

Range	number or SIP URI
Example	NA

Parameter- dialplan.emergency.X.server_priority (X ranges from 1 to 255)	**Configuration File** <MAC>.cfg
Description	Configures the priority for the emergency servers to be used. The digits are separated by commas. The servers to be used in the order listed (left to right). The IP phone tries to send the INVITE request to the emergency server with higher priority. If the emergency server with higher priority does not respond correctly to the INVITE, then the phone tries to make the call using the emergency server with lower priority, and so forth. The IP phone tries to send the INVITE request to each emergency server for three times.
Default Value	1, 2, 3
Range	a combination of digits 1, 2 and 3
Example	dialplan.emergency.1.server_priority = 2, 1, 3 It means the IP phone sends the INVITE request to the emergency server 2 first. If the emergency server 2 does not respond correctly to the INVITE, then tries to make the call using the emergency server 1. If the emergency server 1 does not respond correctly to the INVITE, then tries to make the call using the emergency server 3. The IP phone tries to send the INVITE request to each emergency server for three times. **Note:** If the IP address of the emergency server with higher priority has not been configured, the emergency server with lower priority will be used. If the account is registered successfully or failed (the account information has been configured), the emergency calls will be dialed using the following priority: SIP server>emergency server; if the account is not registered, the emergency server will be used.

Hot Desking

Parameter- hotdesking.dsskey_register_name_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of register name on the hot desking login wizard when tapping the Hot Desking key.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- hotdesking.dsskey_username_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of user name on the hot desking login wizard when tapping the Hot Desking key.
Default Value	1
Range	0-Disabled 1-Enabled
Example	NA

Parameter- hotdesking.dsskey_password_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of password on the hot desking login wizard when tapping the Hot Desking key.
Default Value	1
Range	0-Disabled 1-Enabled

Example	NA

Parameter- hotdesking.dsskey_sip_server_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of SIP server on the hot desking login wizard when tapping the Hot Desking key
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- hotdesking.dsskey_outbound_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of outbound server on the hot desking login wizard when tapping the Hot Desking key.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg

Description	Configures a DSS key as a hot desking key on the IP phone. The digit 34 stands for the key type Hot Desking. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14.
Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	34
Example	linekey.2.type = 34

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.type = 34

Logon Wizard

Parameter- phone_setting.logon_wizard	**Configuration File** <MAC>.cfg

Description	Enables or disables the IP phone to provide the logon wizard during the first startup.
Default Value	0
Range	0-Disabled 1-Enabled
Example	NA

Parameter- hotdesking.startup_register_name_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of register name on the logon wizard during the first startup.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** It works only if there is no registered account on the IP phone and the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).
Example	NA

Parameter- hotdesking.startup_username_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of user name on the logon wizard during the first startup.
Default Value	1

Range	0-Disabled 1-Enabled **Note:** It works only if there is no registered account on the IP phone and the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).
Example	NA

Parameter- hotdesking.startup_password_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of password on the logon wizard during the first startup.
Default Value	1
Range	0-Disabled 1-Enabled **Note:** It works only if there is no registered account on the IP phone and the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).
Example	NA

Parameter- hotdesking.startup_sip_server_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of SIP server on the logon wizard during the first startup.
Default Value	0

Range	0-Disabled 1-Enabled **Note:** It works only if there is no registered account on the IP phone and the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).
Example	NA

Parameter- hotdesking.startup_outbound_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the IP phone to provide input field of outbound server on the logon wizard during the first startup.
Default Value	0
Range	0-Disabled 1-Enabled **Note:** It works only if there is no registered account on the IP phone and the value of the parameter "phone_setting.logon_wizard" is set to 1 (Enabled).
Example	NA

Mobile Account

Parameter- linekey.X.type/ programablekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures a DSS key as an XML Browser key on the IP phone. The digit 77 stands for the key type Mobile Account

Default Value	For line keys: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For programable keys: When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).
Range	77
Example	linekey.2.type =77

Parameter- linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displayed on the touch screen for each DSS key. For line keys: X ranges from 1 to 27.
Default Value	Blank
Range	String within 99 characters
Example	linekey.2.type =77

Configuring DSS Key

This section provides the DSS key parameters you can configure on the IP phone. DSS key consists of the of line key, programable key.

The following table lists the number of DSS keys you can configure for the IP Phone:

Phone	Line Key	Programable Key
Matrix SPARSH VP710	27	3

The programable key takes effect only if the IP phone is idle.

The following tables list relationship between the values of X in the following parameters and programable keys for the IP phone.

 # X ranges from 12 to 14.

programablekey.X.type =

programablekey.X.line =

programablekey.X.value =

programablekey.X.xml_phonebook =

programablekey.X.history_type =

programablekey.X.pickup_value =

X \ IP Phone	SPARSH VP710
1	
2	
3	
4	
5	
6	
7	
8	
9	

10	
11	
12	Hold
13	Mute
14	Tran

DSS key can be assigned with various key features. The parameters of the DSS key are detailed in the following:

Parameter linekey.X.type **Parameter** programablekey.X.type	**Configuration File** <MAC>.cfg
Description	Configures key feature for the DSS key. For line keys: X ranges from 1 to 27. For programable keys: X ranges from 12 to 14. For line keys: Valid types are: 0-N/A 1-Conference 2-Forward 3-Transfer 4-Hold 5-DND 7-ReCall 9-Direct Pickup 10-Call Park 11-DTMF 12-Voice Mail 13-Speed Dial 14-Intercom 15-Line 16-BLF 17-URL 18-Group Listening 20-Private Hold 22-XML Group 23-Group Pickup 24-Multicast Paging 25-Record 34-Hot Desking 35-URL Record 38-LDAP 39-BLF List 40-Prefix 41-Zero Touch 45-Local Group 50-Phone Lock 61-Directory 66-Paging List 77-Mobile Account

	For programable keys: **Valid types are:** 0-N/A 2-Forward 5-DND 7-ReCall 9-Direct Pickup 13-Speed Dial 20-Private Hold 22-XML Group 23-Group Pickup 28-History 30-Menu 33-Status 34-Hot Desking 38-LDAP 41-Zero Touch 43-Local Directory 45-Local Group 47-XML Directory 51-Switch Account Up 52-Switch Account Down 61-Directory 66-Paging List 77-Mobile Account
Format	Integer
Default Value	**For line keys:** The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. **For programable keys:** When X=12, the default value is 0 (NA). When X=13, the default value is 0 (NA). When X=14, the default value is 2 (Forward).

Range	**Valid values are:** Valid values are: 0-N/A 1-Conference 2-Forward 3-Transfer 4-Hold 5-DND 7-ReCall 8-SMS 9-Direct Pickup 10-Call Park 11-DTMF 12-Voice Mail 13-Speed Dial 14-Intercom 15-Line 16-BLF 17-URL 18-Group Listening 20-Private Hold 22-XML Group 23-Group Pickup 24-Multicast Paging 25-Record 28-History 30-Menu 33-Status 34-Hot Desking 35-URL Record 38-LDAP 39-BLF List 40-Prefix 41-Zero Touch 43-Local Directory 45-Local Group 47-XML Directory 50-Phone Lock 51-Switch Account Up 52-Switch Account Down 61-Directory 66-Paging List 77-Mobile Account
Example	linekey.1.type = 8

Parameter linekey.X.line Parameter programablekey.X.line	**Configuration File** <MAC>.cfg
Description	Configures the desired line to apply the key feature. **For line keys:** X ranges from 1 to 27. **For programable keys:** X ranges from 12 to 14
Format	Integer
Default Value	For the programable key, the default value is not applicable. For the line key, when X=1, the default value is 1. When X=2, the default value is 2. When X=3 the default value is 3 … When X=16 the default value is 16.
Range	Permitted Values: 1 to 16 1-Line 1 2-Line 2 … 16-Line 16
Example	linekey.1.line = 2

Parameter linekey.X.value Parameter programablekey.X.value	**Configuration File** <MAC>.cfg
Description	Configures the value for some key features. **For line keys:** X ranges from 1 to 27. **For programable keys:** X ranges from 12 to 14.
Format	String
Default Value	Blank
Range	String within 99 characters

Example	When you assign the Speed Dial to the line key, this parameter is used to specify the number you want to dial out. linekey.1.value = 1001

Parameter linekey.X.label	**Configuration File** <MAC>.cfg
Description	Configures the label displaying on the touch screen for each line key and each soft key (Optional.). This is an optional configuration. **For line keys:** X ranges from 1 to 27.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	linekey.1.label = Dir

Parameter linekey.X.pickup_value	**Configuration File** <MAC>.cfg
Description	Configures the pickup code for BLF feature. This parameter is only applicable to BLF feature. **For line keys:** X ranges from 1 to 27.
Format	String
Default Value	Blank
Range	String within 256 characters
Example	linekey.1.pickup_value = *88

Parameter linekey.X.xml_phonebook	**Configuration File** <MAC>.cfg
Parameter programablekey.X.xml_phonebook	

Description	Configures the desired group or remote phone book when multiple groups or remote phone books are configured on the IP phone.This parameter is only applicable to Local Group/XML Group features. **For line keys:** X ranges from 1 to 27. **For programable keys:** X ranges from 12 to 14.
Format	Integer
Default Value	0
Range	0 to 48
Example	Configures the second remote phone book. linekey.1.xml_phonebook = 1

Parameter linekey.X.extension	**Configuration File** <MAC>.cfg
Description	Configures the channel of multicast paging group This parameter is only applicable to multicast paging features. **For line keys:** X ranges from 1 to 27.
Format	Integer
Default Value	0
Range	0 to 48
Example	Configures the second remote phone book. linekey.1.extension= 1

Appendix E - Acronyms

ACB	Auto Call Back
CLI	Caller Line Identification
CLIR	Calling Line Identity Restriction
CPTG	Call Progress Tone Generation
CWT	Call Waiting Tone
DHCP	Dynamic Host Configuration Protocol
DND	Do Not Disturb
DNS	Domain Name Service
DST	Daylight Saving Time
DTMF	Dual Tone Multi-Frequency
FTP	File Transfer Protocol
GMT	Greenwich Mean Time
IANA	Internet Assigned Numbers Authority
ICMP	Internet Control Message Protocol
IEEE	Institute of Electrical and Electronic Engineers
IP	Internet Protocol
ISP	Internet Service Provider
ITSP	Internet Telephony Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network
LCD	Liquid Crystal Display
LED	Light Emitting Diode
MAC	Media Access Control
ms	Mili seconds
NAT	Network Address Traversal
NTP	Network Time Protocol
P2P	Peer to Peer
PBX	Private Branch Exchange
PoE	Power Over Ethernet
PPPoE	Point to Point Protocol over Ethernet

QoS	Quality of Service
RBT	Ring Back Tone
RFC	Request for Comments
RTC	Real Time Clock
RTP	Real Time Transport Protocol
SIP	Session Initiation Protocol
SNTP	Simple Network Time Protocol
STUN	Simple Traversal of UDP over NAT
TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Universal Reference/Resource Locator
VLAN	Virtual Local Area Network
VM	Voice Mail
VoIP	Voice Over Internet Protocol

Disclaimer for Unauthorized Apps on MATRIX Android-based Phone

Matrix SPARSH VP710, Android-based Phone, is a business smart media phone, which runs on a deeply customized Android operating system. For system security, Matrix has streamlined the Android operating system for the IP phone and only the modules related to phone features is retained. Therefore, this Matrix streamlined Android operating system may not be completely compatible with all 3rd-party Android apps. Meanwhile, the available 3rd-party apps on the market are in varying qualities and some even contain malicious codes, which could compromise the system security. To avoid the bad performance on the phone and minimize the losses for both, You and Matrix, we, Matrix hereby declare:

- The 3rd-party apps on Matrix Android-based Phone should be installed via Matrix Auto-provisioning only, please contact Matrix Support or Field Application Engineer.
- Matrix makes long-term commitment to authorize some of Android apps officially for Matrix Android-based Phone and regularly releases the Compatibility List for World-ready Apps on Matrix Android-based Phone.
  - World-ready Apps are those that are developed by general developers and can be used in a worldwide.
  - The compatibility of 3rd-party apps with Matrix Android-based Phone depends on the firmware version of each Matrix Android-based Phone, meaning that different Android-based Phones with different firmware versions have different app compatibilities. Every time when Matrix released the new firmware version, the apps will be re-authorized by Matrix automatically.
  - The compatibility of 3rd-party apps with Matrix Android-based Phone depends on the software version of app itself, meaning that different software versions of one single app should be authorized by Matrix one by one. Each time the app is updated, you should ask Matrix to authorize this app for its new version.
- If it is essential for you to install and use the app which has not authorized by Matrix yet on Matrix Android-based Phone, you can submit your application to Matrix PMT Team for 3rd-party app authorization. Matrix will check both user information and app information and reserves the rights of whether to authorize or not.
  - If the app is officially authorized by Matrix, Matrix will issue a Certificate for this app.
  - If the app is not officially authorized by Matrix, Matrix Field Application Engineer (FAE) will explain the reasons.
- If you install and use the 3rd-party apps which is unauthorized by Matrix, any liability arising therefrom shall be assumed by YOU not Matrix, and Matrix will not provide any supports for this kind of issue.

For 3rd-party apps supported on Matrix Android-based Phone, refer “Application Compatibility List”.

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Application Compatibility List

APP Compatibility List				
Developer		APP	APP Version	MATRIX
Microsoft		Skype for Business	6.15.0.5	SPARSH VP710
		Skype	7.37.99.40	
ZOOM		Cloud Meeting	4.0.38982.0714	SPARSH VP710
Broadsoft		UC-ONE Communicator	22.1.2.5995	SPARSH VP710

Declaration of Conformity

Matrix SPARSH VP710 is manufactured by YEALINK NETWORK TECHNOLOGY CO. LTD., China and this phone is in conformity with the essential requirements and other relevant provisions of the CE, FCC-15 (Class-B), RCM and RoHS.

CE Mark Warning

This device is marked with the CE mark in compliance with EC Directives 2006/95/EC and 2004/108/EC.

Part 15 FCC Rules

This device is compliant with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interference that may cause undesired operation.

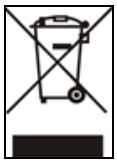
Class B Digital Device or Peripheral

Note: This device is tested and complies with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experience radio/TV technician for help.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.



GNU GPL Information

Matrix SPARSH VP710 is manufactured by YEALINK NETWORK TECHNOLOGY CO. LTD., China.

The firmware of this phone contains third-party software under the GNU General Public License (GPL). Yealink uses software under the specific terms of the GPL. Please refer to the GPL for the exact terms and conditions of the license.

The original GPL license, source code of components licensed under GPL and used in Yealink products can be downloaded online:

http://www.yealink.com/GPLOpenSource.aspx?BaseInfoCateId=293&NewsCateId=293&CateId=293.

Warranty Statement

Matrix warrants that its products will be free from defects in material and workmanship, under normal use and service for a period of twelve (12) months from the date of installation.

Matrix warranties the replacement or repair of any product or component(s) found to be defective during the applicable period and return the same, or grant a reimbursement credit with respect to the product or component. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only. In case of software program design defect(s) that prevents the program from performing the specified functionality, affecting service and beneficial use of the product, Matrix reserves the right to incorporate solutions in its new release of the software and make it available to the customer within a reasonable period of time. The above said with regard to the software design defect, constitutes the sole obligation of Matrix and its authorized installer with respect to the product.

Matrix does not, however, affirm or stand for that the functions or features contained in the system will satisfy its end-user's particular purpose and /or requirements or that the operation of the program will be uninterrupted or error free.

This warranty is voidable by Matrix:

1. If the product is used other than under normal use and is not properly serviced and maintained by qualified technicians.
2. If the product is not maintained under proper environmental conditions.
3. If the product is subjected to abuse, damage, misuse, neglect, fire, power flow, acts of God, accident.
4. If the product is installed or used in combination or in assembly with the products that are not supplied or authorized by Matrix or are of inferior quality or design than Matrix supplied products, which may cause reduction or degradation in functionality.
5. If the product is operated outside the product's specifications or used without designated protections.
6. If the completely filled warranty cards have not been received by Matrix within 15 days of the installation.

In no event will Matrix be liable for any damages, including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

Except for the obligations specifically set forth in this Warranty Policy Statement, in no event shall Matrix be liable for any direct, indirect, special, incidental or consequential damages, whether based on contract or any other legal theory, and where advised of the possibility of such damages.

Neither Matrix nor any of its channel partners makes any other warranty of any kind, whether expressed or implied, with respect to Matrix products. Matrix and its distributors, dealers or sub-dealers specifically disclaim the implied warranties of merchantability and fitness for a particular purpose.

This warranty is not transferable and applies only to the original user of the Product. All legal course of action subjected to Vadodara (Gujarat, India) jurisdiction only.



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