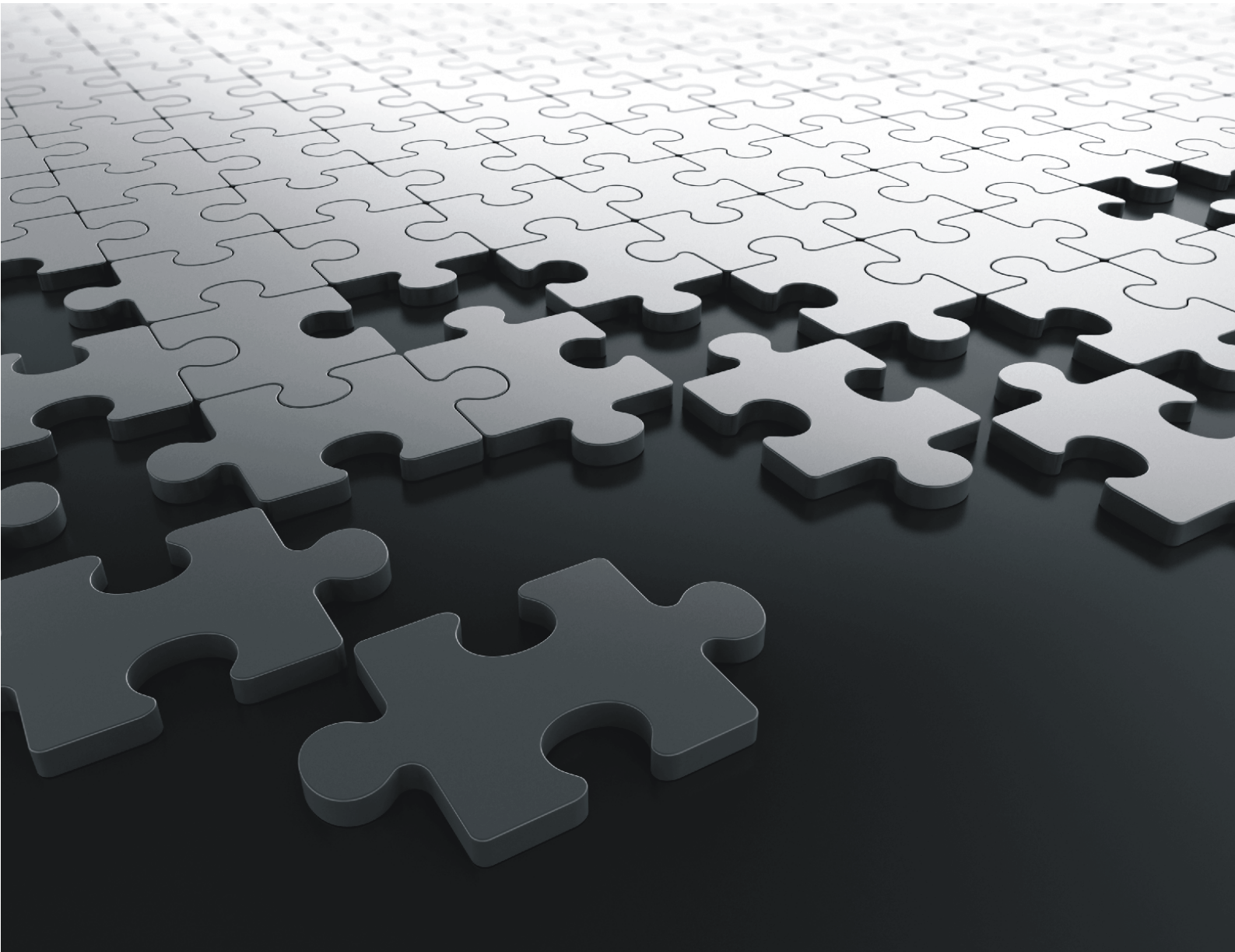


SPARSH VP110

User Guide



SPARSH VP110
The Business IP Phone

User Guide



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Welcome!

Thank you for choosing the Matrix SPARSH VP110! 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 VP110.

About this User Guide

This User Guide provides detailed information and instructions to instal and operate Matrix SPARSH VP110.

Reference Documents

The following reference documents are available for SPARSH VP110 IP phone.

Name	Contents	Where found	Language
Quick Installation Guide	Basic setup of the phone	In the package	English
User Card	Basic call features and phone customizations	In the package	English

The documentation can be found at <http://www.matrixtelesol.com/technical-document.html>

Intended Audience

This document is aimed at,

- **System Engineers** and **System Administrators**, who will install, maintain and support SPARSH VP110. 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 VP110 and troubleshoot the phone, incase it is malfunctioning.

- **End Users**, that is persons/organizations who will actually use SPARSH VP110. 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 VP110** - 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.
- **Customizing Your Phone** - It describes instructions regarding customizing the phone as per your requirement.
- **SIP Account Configuration** - It provides the information to configure the SIP account of the 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 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.
- **Resource Files** - It provides detailed information about typical resource files that can be downloaded by the IP phones and how to configure the templates.
- **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, the sample Configuration files and other information including the product disclaimers.

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 F - Acronyms”](#).

The words IP phone, phone, VoIP phone or SIP phone are used interchangeably and synonymously to mean Matrix SPARSH VP110. 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 VP110 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 VP110 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 to refer to Matrix SPARSH VP110.

Using this document, we hope, you will be able to install, operate and make optimum use of SPARSH VP110. If you encounter any technical problems, contact your dealer/reseller or you can also mail Matrix Technical Support at Support@MatrixComSec.com.

SPARSH VP110 is a new breed of entry-level SIP phone that redefines the desktop telephony experience and quality of business communications. It provides features and functions normally available in high-end phones such as large graphical LCD, 5-line display, context sensitive keys, full-duplex speakerphone, integrated PoE, auto provisioning and broad array of call management features. Perfect fit for everyday users with basic communication needs, SPARSH VP110 finds its applications in call centres, enterprises, small businesses and branch offices.

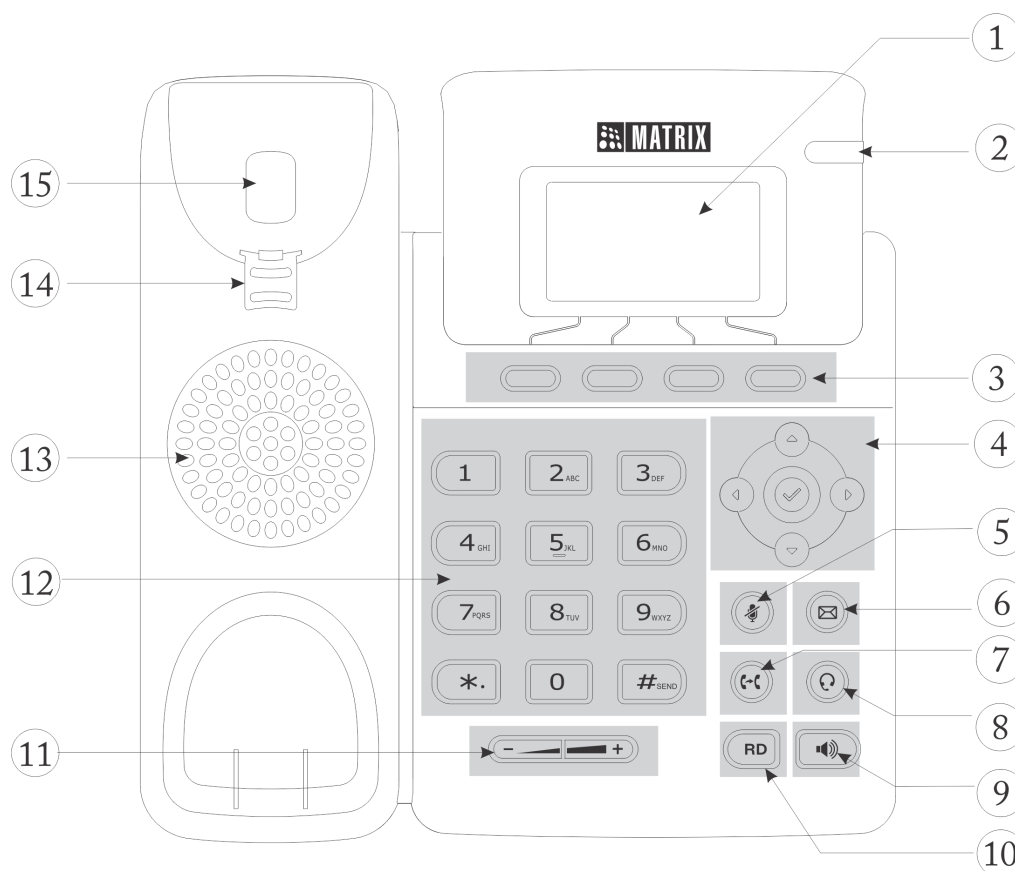


For a list of advanced technical information about the IP phone, refer [“Appendix”](#).

This chapter provides the overview of the SPARSH VP110 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 VP110 IP phone are the LCD screen and the keypad.





Hardware component instructions of the IP phone are:
























Key Label	Item	Description
1	LCD Screen	<p>Shows information about calls, messages, soft keys, time, date and other relevant data:</p> <ul style="list-style-type: none"> • Call information — caller ID, call duration • Icons (for example, DND) • Missed call text or second incoming caller information • Prompt text (for example, “Saving config file!”) • Time and date


2	Power Indicator LED	Indicates phone power status and phone status. <ul style="list-style-type: none"> Receives an incoming call - Fast flashing Receives a voice mail or text message - Slow flashing
3	Soft Keys	Label automatically to identify their context-sensitive features.
4	Navigation Keys 	Scroll through the displayed information.
	OK  Key	Confirms actions or answers incoming calls.
5	Mute Key	Mutes or un-mutes an active call.
6	Message Key	Accesses voice mails.
7	Transfer Key	Transfers a call to another party.
8	Headset Key	Toggles the headset mode.
9	Speakerphone Key	Toggles the hands-free speakerphone mode.
10	RD Key	Redials a previously dialed number.
11	Volume Key	Adjusts the volume of the handset, headset, speaker, and ringer.
12	Keypad	Provides the digits, letters and special characters in context-sensitive applications.
13	Speaker	Provides hands-free (speakerphone) audio output.
14	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 SPARSH VP110 Quick Start Guide.
15	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.

Icon Instructions

Icons appearing on the LCD screen are described in the following table:

Icon	Description
	Network is unavailable
	The private line registers successfully

	Register failed
	Registering
	The shared/bridged line registered successfully
	Hands-free speakerphone mode
	Handset mode
	Headset mode
	Multi-lingual lowercase letters input mode
	Multi-lingual uppercase letters input mode
	Alphanumeric
	Numeric input mode
	Multi-lingual uppercase and lowercase letters input mode
	Voice Mail
	Text Message
	Auto Answer
	Do Not Disturb
	Call Forwarded/Forwarded Calls
	Call Hold
	Call Mute
	Ringer volume is 0
	Phone Lock
	Received Calls
	Placed Calls
	Missed Calls

	The Contact icon
---	------------------

LED Instructions

Power Indicator LED

LED Status	Description
Solid green	The phone is initializing.
Fast flashing green (300ms)	The phone is ringing.
Slow flashing green (1s)	The phone receives a text message or 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.



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.

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.

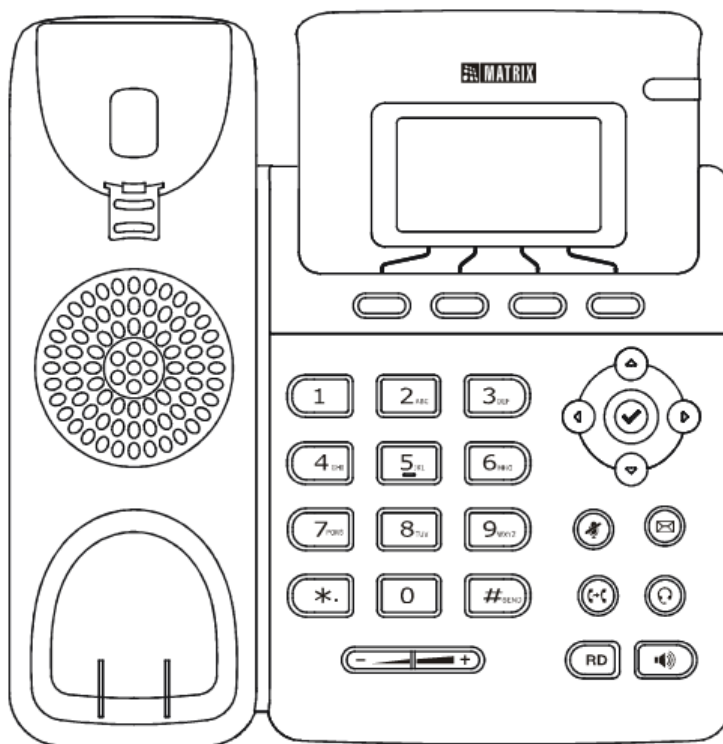
This chapter provides basic installation instructions and information for obtaining the best performance with the IP phone.

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

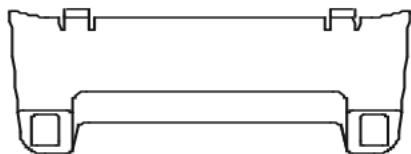
Packaging Contents

The following components are included in your IP phone package:

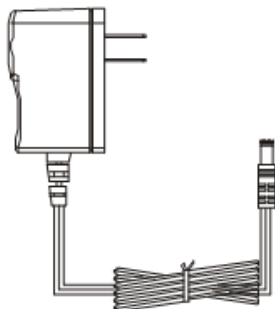
Matrix SPARSH VP110 IP Phone



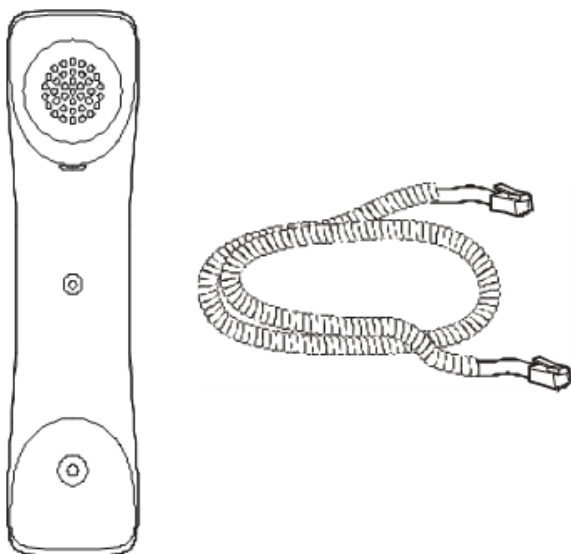
Phone Stand



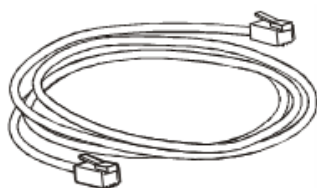
Power Adapter



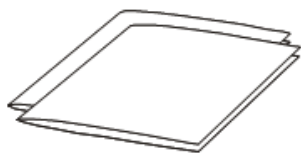
Handset & Handset Cord



Ethernet Cable



Quick Start Guide



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”](#).

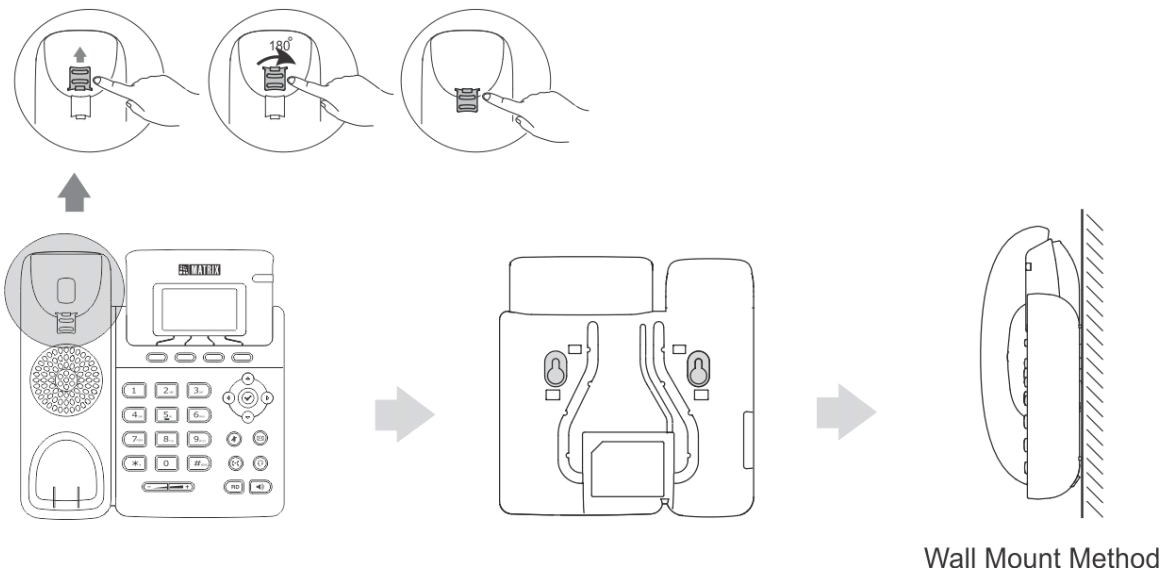
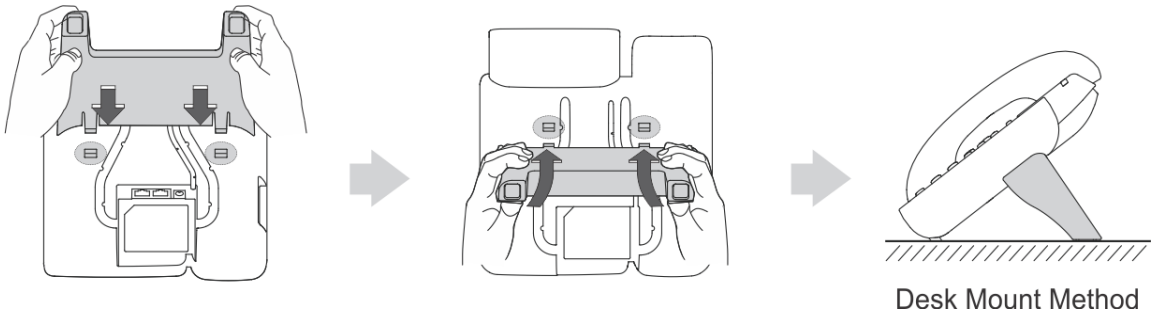
This section introduces how to install the phone:

1. Attach and detach the stand.
2. Connect the handset and optional headset.
3. Connect the network and power.

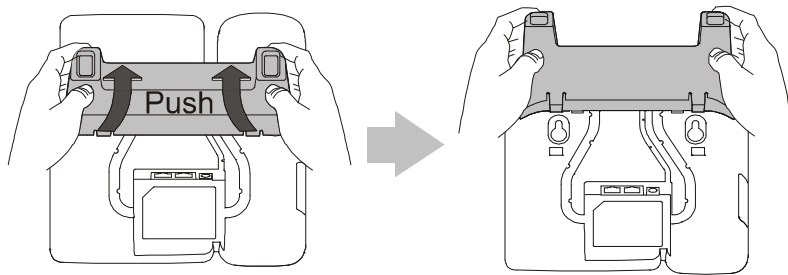
Refer to the following details for instructions.

1. Attach/Detach the stand.

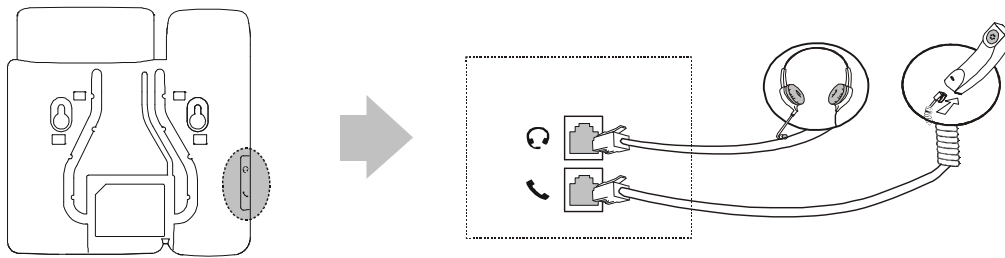
- Attach the stand



- Detach the stand



2. Connect the handset and optional headset.



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

3. Connect the network and power.

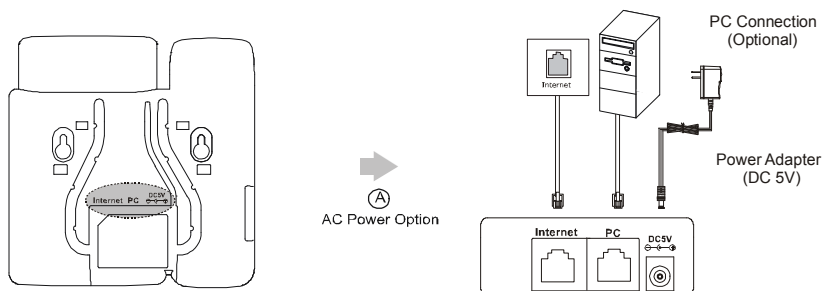
You have two options for power and network connections. Your network administrator will advise you for which one to use.

- 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.
- Connect the included or a standard Ethernet cable between the Internet port on the phone and the one on the wall or switch/hub device port.

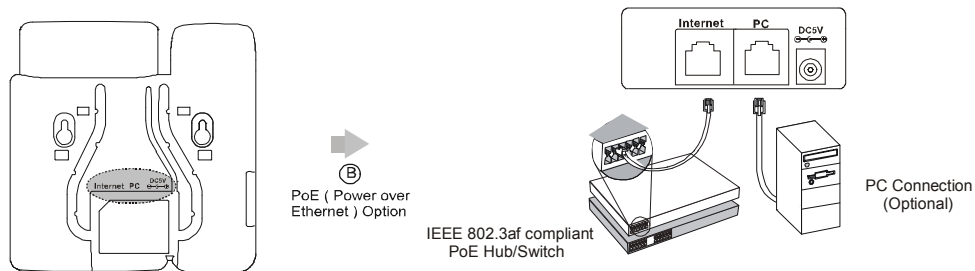


Power over Ethernet (PoE)

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 for the IP phone:

- 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.

The phone can also share the network with another network device such as a PC (personal computer). It is an optional connection.



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

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 LCD 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 in green.
2. The message "Welcome Initializing...please wait" appears on the LCD screen when the IP phone starts up.
3. The main LCD screen displays the following:
 - Time and date
 - Soft key labels
4. Press the **OK**  or press **Menu->Status** to check the IP phone status, the LCD 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 LCD 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 changes to most features.

The following section describes how to configure SPARSH VP110 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 customize your phone by pressing the **Menu** soft key to access the phone user interface. Access to specific features is restricted to the administrator. The default password is “1234”. 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.

You can configure the IP phone via web user interface by logging into either of the following two modes having different sets 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 32.0.3
- Internet Explorer - Version 11.0.9600.16428IS
- Google Chrome - Version 38.0.2125.104

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, press the **OK**  key on the phone. Enter the IP address (e.g., http://192.168.1.181) in the address bar of a web browser on your PC.

The default administrator username is “**admin**” (case-sensitive) and password is “**1234**”. 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, press the **OK**  key on the phone. Enter the IP address (e.g., http://192.168.1.181) 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 VP110 IP Phones can be centrally provisioned from a provisioning server using the configuration file (<MAC>.cfg). SPARSH VP110 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 VP110 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. For more information regarding the Configuration File Format, refer [“Appendix E - Sample Configuration File”](#).

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 Provisioning Server

SPARSH VP110 IP phones support obtaining the Provisioning Server Address in the following ways:

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

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)**.

The screenshot shows the 'MATRIX SPARSH VP110' web interface. The 'Settings' tab is selected. On the left sidebar, 'Auto Provision' is highlighted. The main content area is titled 'Auto Provision' and contains the following settings:

- PNP Active: ☒ On ☐ Off
- DHCP Active: ☒ On ☐ Off
- Custom Option(128~254):
- DHCP Option Value:
- Server URL:
- User Name:
- Password:
- Attempt Expired Time(s):
- Common AES Key:
- MAC-Oriented AES Key:
- Zero Active:
- Wait Time(1~100s):
- Power On: ☒ On ☐ Off
- Repeatedly: ☐ On ☒ Off
- Interval(Minutes):
- Weekly: ☐ On ☒ Off
- Time:
- Day of Week:
 - ☒ Sunday
 - ☒ Monday
 - ☒ Tuesday
 - ☒ Wednesday
 - ☒ Thursday
 - ☒ Friday
 - ☒ Saturday

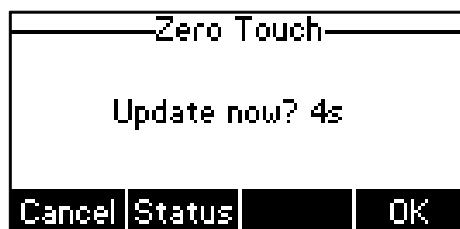
At the bottom of the settings area are 'Confirm' and 'Cancel' buttons. On the right, a 'NOTE' box contains the following text:

Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.

When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

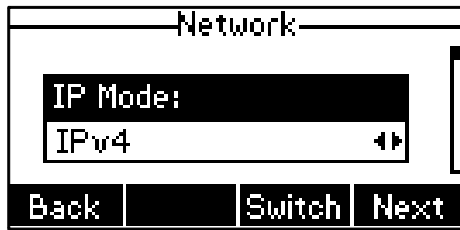
- Click **Confirm** to accept the change.

When Zero Touch is enabled, configuration wizard as shown below will appear on the phone user interface during start up:



- Press the **OK** soft key.

The network parameters are configurable via phone user interface:



- Press the **Next** soft key after finishing network setting.
- Configure the **Provisioning Server Address**, **Authentication User Name** (optional) and **Password** (optional) in the Auto Provision screen.



- Press the **OK** soft key.

Plug and Play (PnP) Server

SPARSH VP110 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 accept 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 VP110 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 VP110 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 VP110 will generate TFTP query
- If ftp://URL is configured then SPARSH VP110 will generate FTP query
- If http://URL is configured then SPARSH VP110 will generate HTTP query
- If https://URL is configured then SPARSH VP110 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 case 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 VP110**.

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

Phone Flash – Manual Entry

SPARSH VP110 IP phones support obtaining the provisioning server address from the phone flash. To obtain the provisioning server address by reading the phone flash, make sure the configuration is set properly. These parameters can be entered to the Phone by accessing the Phone using Phone User Interface or the Web User Interface of the Phone.

To configure the Phone Flash via web user interface:

- Click on **Settings->Auto Provision**.
- Enter the URL, user name and password of the provisioning server in the **Server URL**, **User Name** and **Password** fields (the user name and password are optional).

(e.g. you can program http://10.3.6.223 if you want to use HTTP server for Auto Configuration instead of TFTP).

The screenshot displays the Matrix SPARSH VP110 web user interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, Settings (selected), Directory, and Security. A left sidebar lists various configuration options: Preference, Time & Date, Call Display, Upgrade, Auto Provision (selected), Configuration, Dial Plan, Voice, Ring, Tones, Softkey Layout, TR069, Voice Monitoring, and SIP. The main content area is titled 'Auto Provision' and contains the following settings:

- PNP Active: ☒ On ☐ Off
- DHCP Active: ☒ On ☐ Off
- Custom Option(128~254):
- DHCP Option Value:
- Server URL:
- User Name:
- Password:
- Attempt Expired Time(s):
- Common AES Key:
- MAC-Oriented AES Key:
- Zero Active:
- Wait Time(1~100s):
- Power On: ☒ On ☐ Off
- Repeatedly: ☐ On ☒ Off
- Interval(Minutes):
- Weekly: ☐ On ☒ Off
- Time: : -- :
- Day of Week: ☒ Sunday, ☒ Monday, ☒ Tuesday, ☒ Wednesday, ☒ Thursday, ☒ Friday, ☒ Saturday

At the bottom of the settings area is a button labeled 'Autoprovision Now'. Below the settings area are 'Confirm' and 'Cancel' buttons. On the right side, a 'NOTE' box contains the following text:

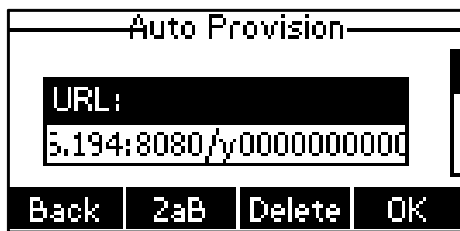
Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.

When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

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

To configure the Phone Flash via phone user interface:

- Go to the **Phone User Interface ->Menu ->Settings ->Advanced Settings ->Enter the Phone UI password ->Auto Provision ->URL/Username (Optional) /Password (Optional)**.



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 use. 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.

MATRIX SPARSH VP110 Log Out

Status Account Network DSSKey Features **Settings** Directory Security

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Auto Provision

PNP Active ☒ On ☐ Off

DHCP Active ☒ On ☐ Off

Custom Option(128~254)

DHCP Option Value

Server URL

User Name

Password

Attempt Expired Time(s)

Common AES Key

MAC-Oriented AES Key

Zero Active

Wait Time(1~100s)

Power On ☒ On ☐ Off

Repeatedly ☒ On ☐ Off

Interval(Minutes)

Weekly ☐ On ☒ Off

Time : -- :

Day of Week

☒ Sunday
☒ Monday
☒ Tuesday
☒ Wednesday
☒ Thursday
☒ Friday
☒ Saturday

NOTE

Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.
When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

- Click **Confirm** to accept 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.

MATRIX
SPARSH VP110

Log Out

Status Account Network DSSKey Features **Settings** Directory Security

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Auto Provision

PNP Active ☒ On ☐ Off

DHCP Active ☒ On ☐ Off

Custom Option(128~254)

DHCP Option Value

Server URL

User Name

Password

Attempt Expired Time(s)

Common AES Key

MAC-Oriented AES Key

Zero Active

Wait Time(1~100s)

Power On ☒ On ☐ Off

Repeatedly ☒ On ☐ Off

Interval(Minutes)

Weekly ☐ On ☒ Off

Time : -- :

Day of Week

☒ Sunday
☒ Monday
☒ Tuesday
☒ Wednesday
☒ Thursday
☒ Friday
☒ Saturday

NOTE

Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.

When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

- Click **Confirm** to accept 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.

MATRIX
SPARSH VP110

Log Out

Status Account Network DSSKey Features **Settings** Directory Security

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Auto Provision

PNP Active ☒ On ☐ Off

DHCP Active ☒ On ☐ Off

Custom Option(128~254)

DHCP Option Value

Server URL

User Name

Password

Attempt Expired Time(s)

Common AES Key

MAC-Oriented AES Key

Zero Active

Wait Time(1~100s)

Power On ☒ On ☐ Off

Repeatedly ☒ On ☐ Off

Interval(Minutes)

Weekly ☒ On ☐ Off

Time : -- :

Day of Week

☒ Sunday
☒ Monday
☒ Tuesday
☒ Wednesday
☒ Thursday
☒ Friday
☒ Saturday

NOTE

Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.

When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

- Click **Confirm** to accept 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**.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings' (selected), 'Directory', and 'Security'. A left sidebar lists various settings categories: Preference, Time & Date, Call Display, Upgrade, Auto Provision (selected), Configuration, Dial Plan, Voice, Ring, Tones, Softkey Layout, TR069, Voice Monitoring, and SIP. The main content area is titled 'Auto Provision' and contains the following settings:

- PNP Active: ☒ On ☐ Off
- DHCP Active: ☒ On ☐ Off
- Custom Option(128~254):
- DHCP Option Value:
- Server URL:
- User Name:
- Password:
- Attempt Expired Time(s):
- Common AES Key:
- MAC-Oriented AES Key:
- Zero Active:
- Wait Time(1~100s):
- Power On: ☒ On ☐ Off
- Repeatedly: ☒ On ☐ Off
- Interval(Minutes):
- Weekly: ☒ On ☐ Off
- Time: : : - : :
- Day of Week: ☒ Sunday, ☒ Monday, ☒ Tuesday, ☒ Wednesday, ☒ Thursday, ☒ Friday, ☒ Saturday

At the bottom of the settings area are 'Confirm' and 'Cancel' buttons. A 'NOTE' box on the right contains the following text:

Auto Provision
The IP phone can interoperate with provisioning server using auto provisioning for deploying the IP phones.
When the IP phone triggers to perform auto provisioning, it will request to download the configuration files from the provisioning server. During the auto provisioning process, the IP phone will download and update configuration files to the phone flash.

- Click **Autoprovision Now**.

The phone will perform the auto provisioning process immediately.

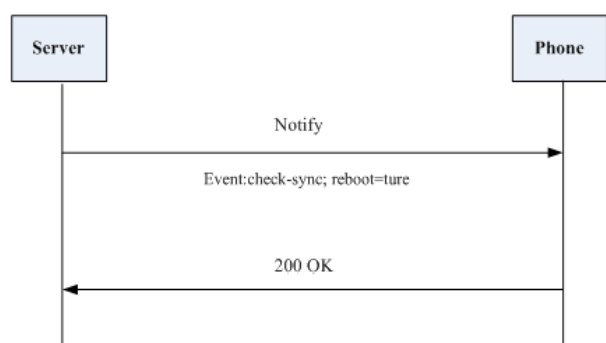
Multi-mode Mixed

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:

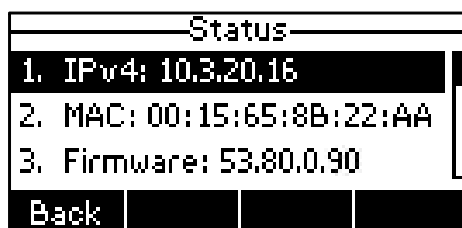
- Network status (for example, IPv4 status, IPv6 status, IP mode and MAC address).
- Phone status (for example, device model, hardware version, firmware version, product ID, MAC address and device certificate status).
- Account status (for example, the register status of SIP account).



You can view device certificate status via phone user interface only.

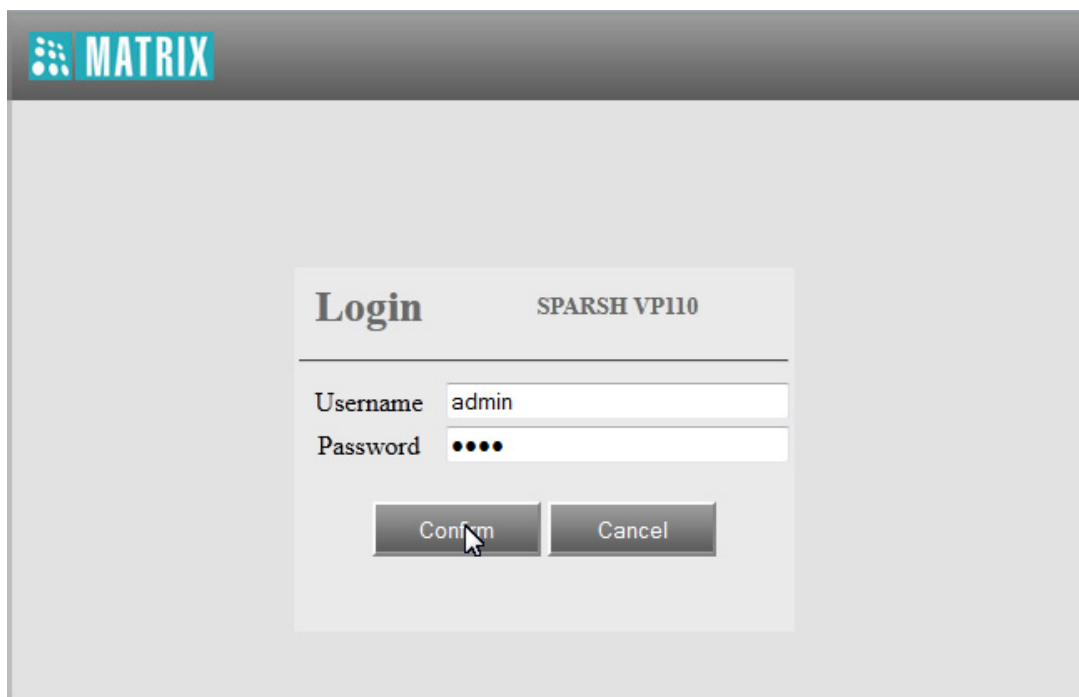
To view the phone status via phone user interface:

- Press **OK**  or press **Menu->Status**.
- Press  or  to scroll through the list and view the specific information.



To view the phone status via web user interface:

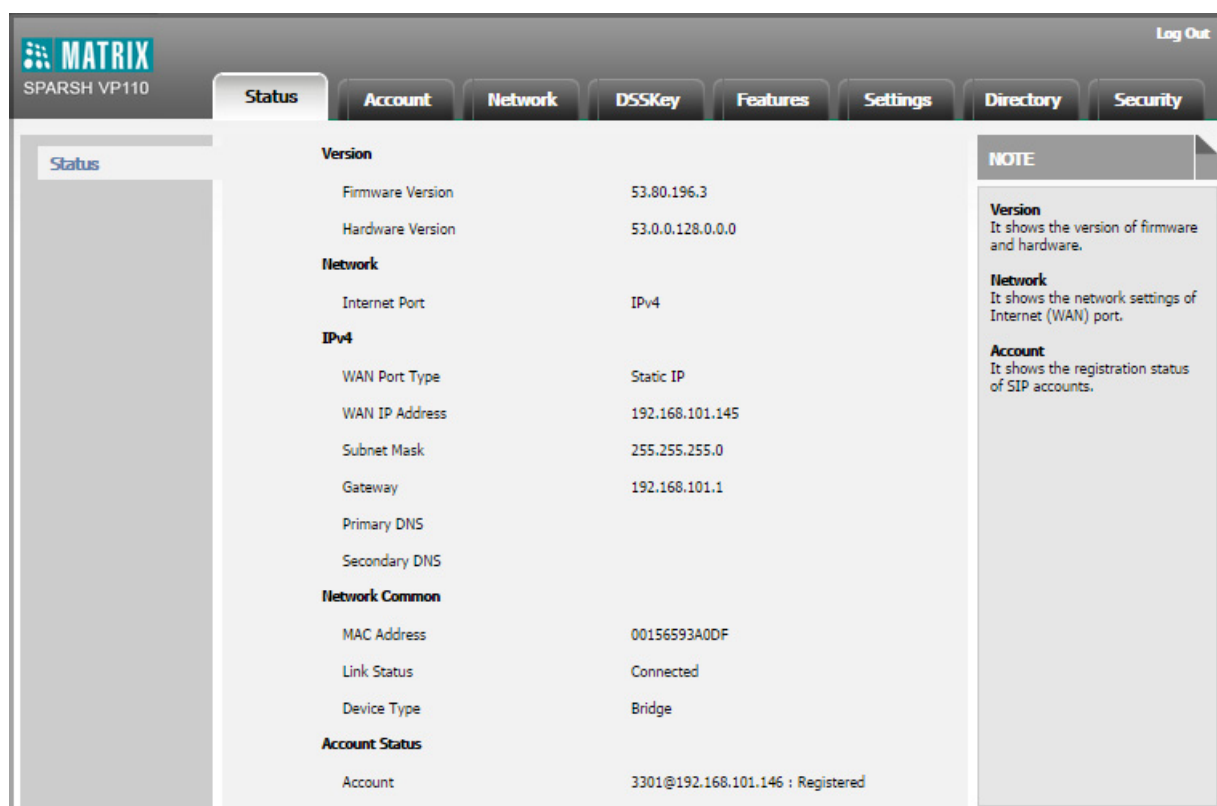
- Open the web browser of your computer.
- Enter the IP address in the browser's address bar, and then press **Enter**.
- Enter the **Username** (**admin**) and **Password** (default: **1234**) in the login page.



The login screen features the Matrix logo at the top left. The title 'Login' is on the left and 'SPARSH VP110' is on the right. Below the title bar, there are two input fields: 'Username' with the text 'admin' and 'Password' with four dots. At the bottom, there are two buttons: 'Confirm' and 'Cancel'.

- Click **Confirm** to login.

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



The status page has a top navigation bar with tabs: Status, Account, Network, DSSKey, Features, Settings, Directory, and Security. The 'Status' tab is active. On the left, there is a sidebar with the 'Status' link. The main content area displays the following information:

Version	
Firmware Version	53.80.196.3
Hardware Version	53.0.0.128.0.0.0

Network	
Internet Port	IPv4

IPv4	
WAN Port Type	Static IP
WAN IP Address	192.168.101.145
Subnet Mask	255.255.255.0
Gateway	192.168.101.1
Primary DNS	
Secondary DNS	

Network Common	
MAC Address	00156593A0DF
Link Status	Connected
Device Type	Bridge

Account Status	
Account	3301@192.168.101.146 : Registered

On the right side of the page, there is a 'NOTE' section with the following text:

Version
It shows the version of firmware and hardware.

Network
It shows the network settings of Internet (WAN) port.

Account
It shows the registration status of SIP accounts.

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 [“SIP Account Configuration”](#).

Idle Screen

If the phone has successfully started up, the idle screen will be shown as below:



The idle screen shows the label of current account, time and date, and four soft keys.

Basic Network Settings

In order to get your IP phones running, you must perform basic network setup, such as IP address and subnet mask configuration. This section describes how to configure basic network parameters for IP phones.

This section mainly introduces IPv4 network parameters. IP phones also support IPv6. For more information on IPv6, refer to [“IPv6 Support”](#).

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 when DHCP is used. 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: <a href="http://<phoneIPAddress>/servlet?p=network&q=load">http://<phoneIPAddress>/servlet?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**.

MATRIX
SPARSH VP110

Status Account **Network** DSSKey Features Settings

Basic
PC Port
Advanced

Internet Port

Mode(IPv4/IPv6) IPv4

IPv4 Config

☒ DHCP
☐ Static IP Address

IP Address 192.168.101.145

Subnet Mask 255.255.255.0

Gateway 192.168.101.1

Static DNS ☒ On ☐ Off

Primary DNS

Secondary DNS

- 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.




The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network' (selected), 'DSSKey', 'Features', and 'Settings'. On the left, a sidebar shows 'Basic' (selected), 'PC Port', and 'Advanced'. The main content area is titled 'Internet Port' and 'IPv4 Config'. Under 'IPv4 Config', 'DHCP' is selected with a radio button. Below it, 'Static IP Address' is an option. The 'Static DNS' section has 'On' selected with a radio button. The 'Primary DNS' field contains '192.168.101.5' and the 'Secondary DNS' field is empty. A mouse cursor is pointing at the 'Secondary DNS' 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 DHCP via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234)->**Network->WAN Port** and press the **Enter** soft key.
- Press  to select **IPv4** and press the **Enter** soft key.
- Press  or  to highlight the **DHCP IPv4 Client** field.

- Press the **Save** soft key to save the change.
- Reboot the IP phone to apply the changes.

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 IP phones to establish network connectivity:

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

Procedure

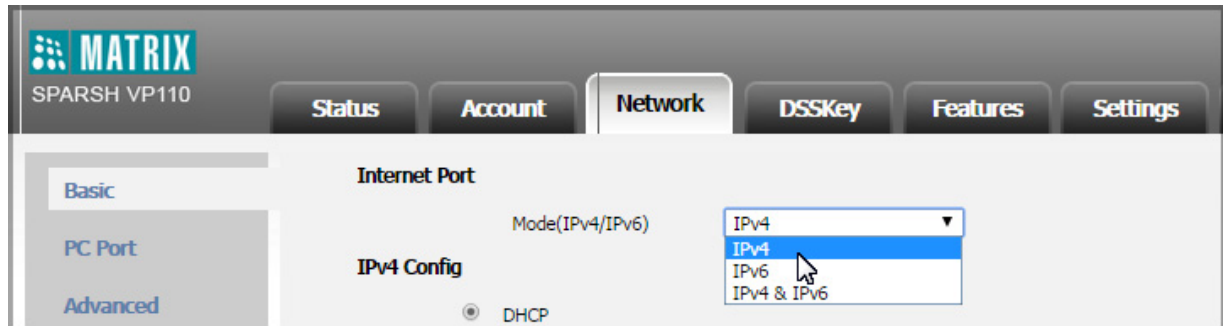
Network parameters can be configured manually using the configuration files or locally.

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?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.



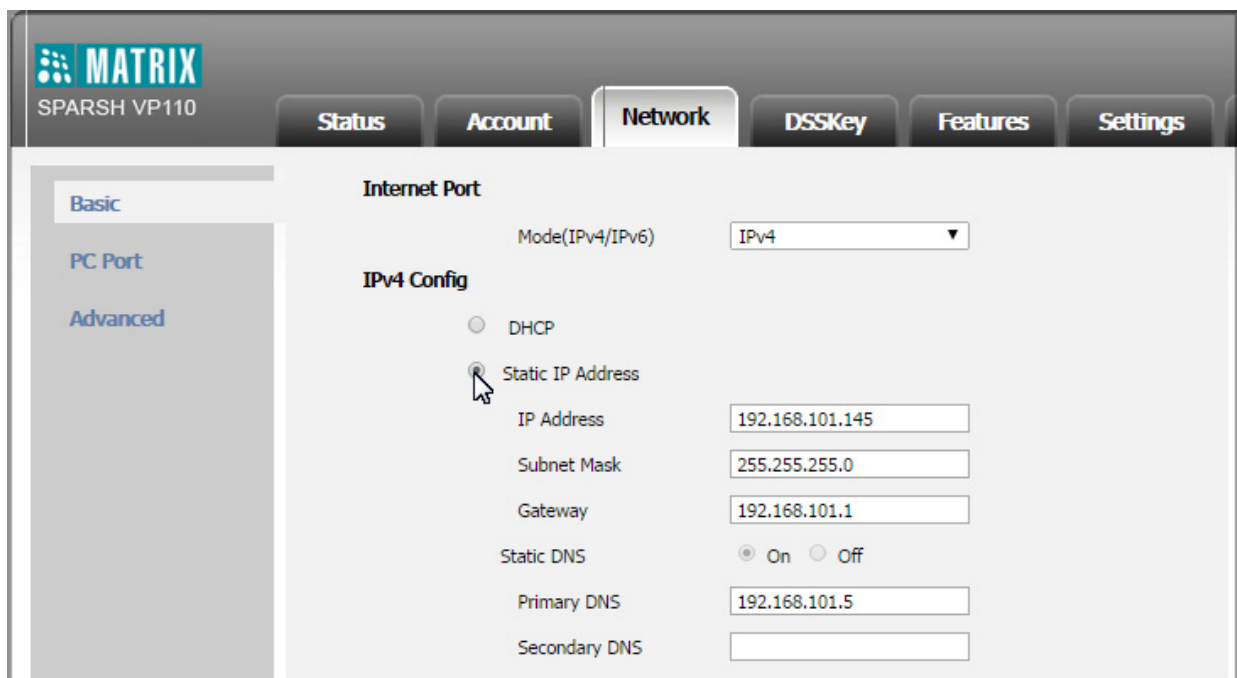
- Click **Confirm** to save the change.

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.



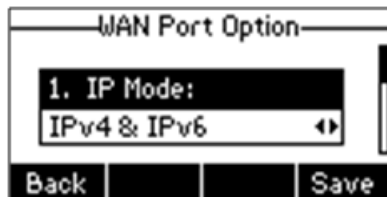
- Click **Confirm** to save the change.

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:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Network->WAN Port**.
- Press  or  to select **IPv4**, **IPv6** or **IPv4 & IPv6** from the **IP Mode** field.



- Press the **Save** soft key to save the change.
- Reboot the IP phone to apply the changes.

To configure a static IPv4 address via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234)**->Network->WAN Port** and press the **Enter** soft key.
- Press  to select **IPv4** and press the **Enter** soft key.
- Press  to select **Static IPv4 Client** and press the **Enter** soft key.
- Enter the desired values in the IP Address, Subnet Mask, Default Gateway, Pri DNS and Sec DNS fields.



- Press the **Save** soft key to save the changes.
- Reboot the IP phone to apply the changes.



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. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure PPPoE on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?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.

MATRIX
SPARSH VP110

Status Account **Network** DSSKey Features Settings

Basic
PC Port
Advanced

Internet Port

Mode(IPv4/IPv6) IPv4

IPv4 Config

☐ DHCP

☐ Static IP Address

IP Address 192.168.101.145

Subnet Mask 255.255.255.0

Gateway 192.168.101.1

Static DNS ☒ On ☐ Off

Primary DNS 192.168.101.5

Secondary DNS

☒ PPPoE

User Name



Password

- 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:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Network->WAN Port** and press the **Enter** soft key.
- Press  to select **IPv4** and press the **Enter** soft key.
- Press  to select **PPPoE IPv4 Client** and press the **Enter** soft key.
- Enter the **user name** and **password** in corresponding fields.
- Press the **Save** soft key to save the change.
- Reboot the IP phone to apply the changes.



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.

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

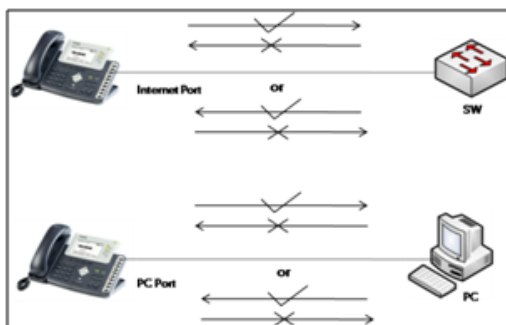
Auto-negotiation is configured for both Internet and PC ports on the IP phone by default.

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?p=network-adv&q=load

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

- Click on **Network->Advanced**.
- Select the desired value from the **WAN Port Link** list.
- Select the desired value from the **PC Port Link** list.

MATRIX SPARSH VP110 Log Out

Status Account **Network** DSSKey Features Settings Directory Security

Basic
PC Port
Advanced

LLDP
Active: Enabled
Packet Interval (1~3600s): 60

CDP
Active: Disabled
Packet Interval (1~3600s): 60

VLAN
WAN Port: Active: Disabled
VID (1~4094): 1
Priority: 0
PC Port: Active: Disabled
VID (1~4094): 1
Priority: 0
DHCP VLAN: Active: Enabled
Option (1~255): 132

NAT
Active: Disabled
STUN Server:
STUN Port(1024~65000): 3478

Port Link
WAN Port Link: Auto Negotiate
PC Port Link: Auto Negotiate (dropdown open: Auto Negotiate, Full Duplex 10Mbps, Full Duplex 100Mbps, Half Duplex 10Mbps, Half Duplex 100Mbps)

Voice QoS
Voice QoS (0~63):
SIP QoS (0~63): 26

NOTE
VLAN
It 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.

The priority of VLAN assignment method (from highest to lowest): LLDP/CDP->manual configuration->DHCP VLAN

NAT Traversal
It is a general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

You can configure NAT traversal for the IP phone.

Quality of Service (QoS)
It is the ability to provide different priorities for different packets in the network, allowing the transport of traffic with special requirements.

Web Server Type
It determines access protocol and port of the IP phone's web user interface.

802.1X Authentication
It offers an authentication mechanism for the IP phone to connect/link to a LAN or WLAN.

VPN
It provides remote offices or individual users with secure access to their organization's network.

- 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

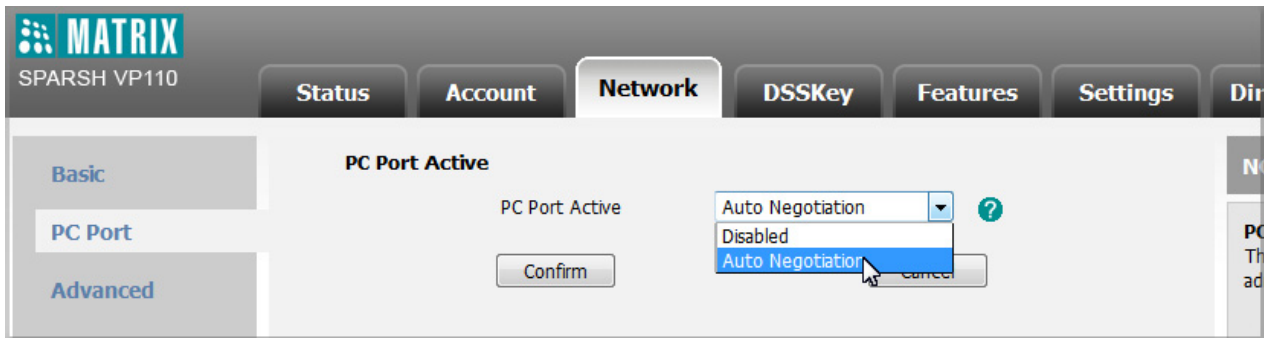
PC port mode can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the PC port mode. Disable the PC port. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Configure the PC port mode. Disable the PC port. Navigate to: http://<phoneIPAddress>/servlet?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.

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

Contrast

Contrast determines the readability of the texts displayed on the LCD screen. Adjusting the contrast to a comfortable level can optimize the screen viewing experience. When configured properly, contrast allows users to read the LCD's display with minimal eyestrain.

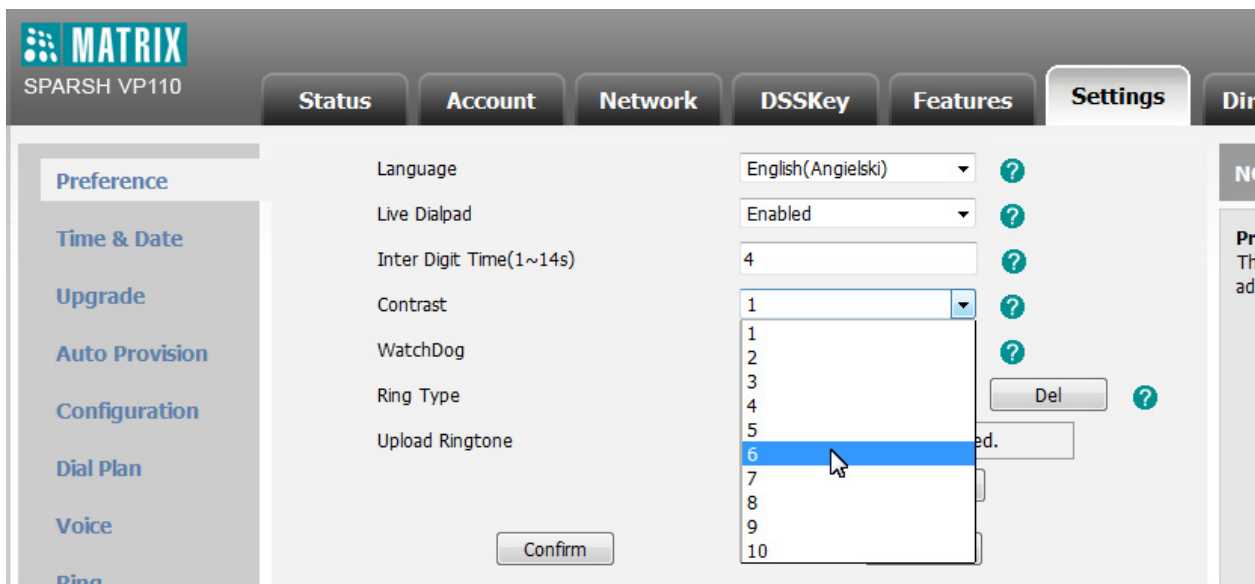
Procedure

Contrast can be configured using the configuration files or locally.

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

To configure contrast via web user interface:

- Click on **Settings->Preference**.
- Select the desired value from the **Contrast** list.



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

To configure the contrast via phone user interface:

- Press **Menu->Settings->Basic Settings->Display->Contrast**.
- Press or or the **Switch** soft key to increase or decrease the intensity of contrast.

The default contrast level is 6.



- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

Language

The default language of the phone user interface is English. The phone will detect and use the same language as which of your web browser for the web user interface. If the language of your web browser is not supported by the phone, the web user interface will use English. 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.

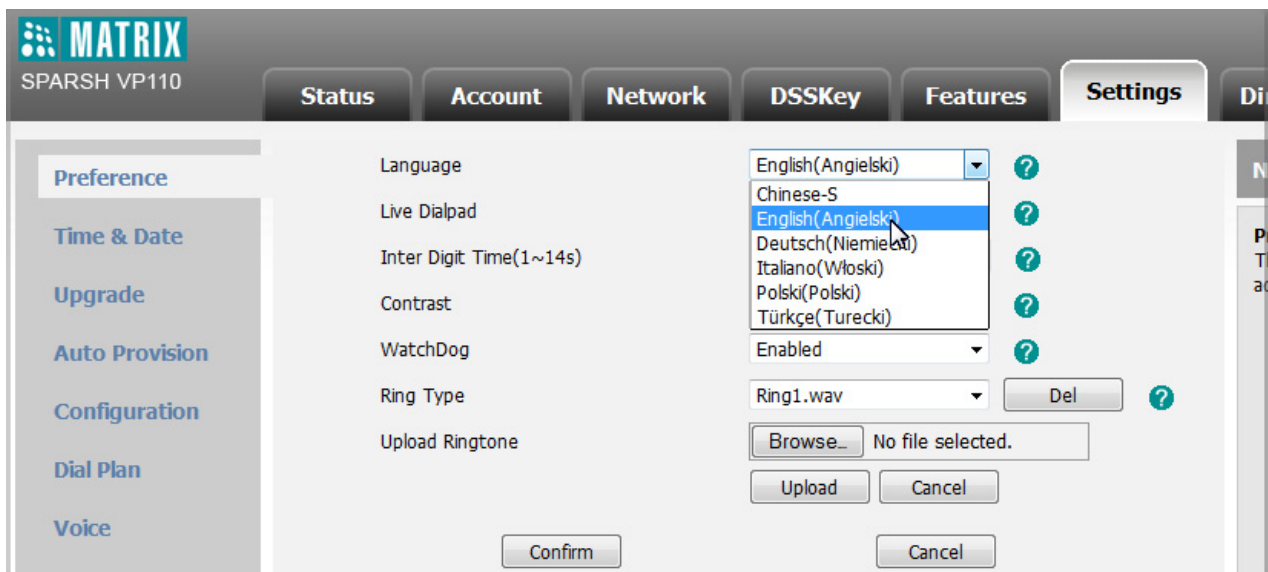
Procedure

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?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:

- Press **Menu->Settings->Basic Settings->Language**.
- Press  or  to select the desired language.



- Press the **Save** soft key to save the change.

Text displayed on the LCD screen will change to the selected language.

Time & Date

IP phones maintain a local clock and calendar. Time and date are displayed on the idle screen of IP phones. 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
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	<p>Configure NTP by DHCP priority feature.</p> <p>Configure the NTP server, time zone and DST.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure NTP by DHCP priority feature.</p> <p>Configure the NTP server, time zone and DST.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-datetime&q=load">http://<phoneIPAddress>/servlet?p=settings-datetime&q=load</p>
	Phone User Interface	<p>Configure the NTP server and time zone.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p>

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.

The screenshot shows the 'MATRIX SPARSH VP110' web interface. The 'Settings' tab is selected, and the 'Time & Date' sub-tab is active. The 'NTP by DHCP Priority' dropdown menu is open, with 'Low' selected. The 'NOTE' section on the right provides additional information about Time and Date, Time Zone, NTP Server, and Daylight Saving Time.

- 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 Server** and **Secondary 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.

MATRIX
SPARSH VP110

Log Out

Status Account Network DSSKey Features **Settings** Directory Security

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Time & Date

DHCP Time: Enabled

Time Zone: +5:30 India(Calcutta)

Daylight Saving Time: ☐ Automatic ☒ Enabled ☐ Disabled

Fixed Type: ☒ DST by Date ☐ DST by Week

Start Date: Month Day Hour

End Date: Month Day Hour

Offset(minutes):

NTP by DHCP Priority: Low

Primary Server: cn.pool.ntp.org

Secondary Server: cn.pool.ntp.org

Synchronism (15~86400s): 1000

Manual Time: Disabled

Time Format: Hour 24

Date Format: WWW MMM DD

Confirm Cancel

NOTE

Time and Date
It displays on the idle screen of IP phones.

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.

NTP Server
The IP phones synchronize the time and date automatically from the NTP time server by default.

Daylight Saving Time
It is the practice of temporary advancing clocks during the summer time 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.

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.

MATRIX
SPARSH VP110

Log Out

Status Account Network DSSKey Features **Settings** Directory Security

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Time & Date

DHCP Time: Enabled

Time Zone: +5:30 India(Calcutta)

Daylight Saving Time: ☐ Automatic ☒ Enabled ☐ Disabled

Fixed Type: ☐ DST by Date ☒ DST by Week

Start Date: January First in M Sunday 00:00

End Date: January First in M Sunday 00:00

Offset(minutes):

NTP by DHCP Priority: Low

Primary Server: cn.pool.ntp.org

Secondary Server: cn.pool.ntp.org

Synchronism (15~86400s): 1000

Manual Time: Disabled

Time Format: Hour 24

Date Format: WWW MMM DD

Confirm Cancel

NOTE

Time and Date
It displays on the idle screen of IP phones.

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.

NTP Server
The IP phones synchronize the time and date automatically from the NTP time server by default.

Daylight Saving Time
It is the practice of temporary advancing clocks during the summer time 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.

- 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.

The screenshot shows the 'MATRIX SPARSH VP110' interface with the 'Settings' tab selected. The 'Time & Date' section is active. The 'Manual Time' dropdown menu is open, showing 'Disabled', 'Disabled', and 'Enabled' (which is highlighted by the mouse cursor). Other settings include 'DHCP Time' (Enabled), 'Time Zone' (+5:30 India(Calcutta)), 'Daylight Saving Time' (Automatic, Enabled, Disabled), 'Fixed Type' (DST by Date, DST by Week), 'Start Date' and 'End Date' (Month, Day, Hour fields), 'Offset(minutes)', 'NTP by DHCP Priority' (Low), 'Primary Server' (cn.pool.ntp.org), 'Secondary Server' (cn.pool.ntp.org), 'Synchronism (15~86400s)' (1000), 'Time Format' (WWW MMM DD), and 'Date Format' (WWW MMM DD). A 'NOTE' section on the right provides information about Time and Date, Time Zone, NTP Server, and Daylight Saving Time.

- Enter the time and date in the corresponding fields.

The screenshot shows the 'MATRIX SPARSH VP110' interface with the 'Settings' tab selected. The 'Time & Date' section is active. The 'Manual Time' dropdown is set to 'Enabled'. The 'Date' field is filled with Year 2005, Month 3, and Day 17. The 'Time' field is filled with Hour 19, Minute 26, and Second 55. Other settings include 'DHCP Time' (Enabled), 'NTP by DHCP Priority' (Low), 'Primary Server' (cn.pool.ntp.org), 'Secondary Server' (cn.pool.ntp.org), 'Synchronism (15~86400s)' (1000), 'Time Format' (Hour 24), and 'Date Format' (WWW MMM DD). A 'NOTE' section on the right provides information about Time and Date, Time Zone, NTP Server, and Daylight Saving Time.

- 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 NTP server and time zone via phone user interface:

- Press **Menu->Settings->Basic Settings->Time & Date->SNTP Settings**.
- Press **▶** or **4**, or the **Switch** soft key to select the time zone that applies to your area from the **Time Zone** field.

The default time zone is "+5:30 India".

- Enter the domain names or IP addresses in the **NTP Server1** and **NTP Server2** fields respectively.
- Press the **Save** soft key to save the change.







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

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

- Press **Menu->Settings->Basic Settings->Time & Date->Manual Settings**.
- Enter the date in the **Date** field.
- Enter the time in the **Time** field.
- Press the **Save** soft key to save the change.

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

- Press **Menu->Settings->Basic Settings->Time & Date->Time & Date Format**.
- Press  or , or the **Switch** soft key to select the desired time format from the **Clock** field.
- Press  or , or the **Switch** soft key to select the desired date format from the **Date Format** field.
- Press the **Save** soft key to save the change.

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.

Ring 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.

The screenshot displays the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features (selected), Settings, Directory, and Security. A left sidebar lists various features: Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, SMS, Action URL, Power LED (selected), and Notification Popups. The main content area is titled 'Power LED' and contains six configuration items, each with a dropdown menu: Common Power Light On (Disabled), Ringing Power Light Flash (Enabled), Voice/Text Mail Power Light Flash (Enabled), Mute Power Light Flash (Disabled), Hold/Held Power Light Flash (Disabled), and Talk/Dial Power Light On (Disabled). At the bottom of this section are 'Confirm' and 'Cancel' buttons. On the right, a 'NOTE' box states: 'Power Indicator LED It indicates power status and phone status.'

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

Administrator Password

Advanced menu options are strictly used by administrators. Users can configure them only if they have administrator privileges. The administrator password can only be changed by an administrator. The default administrator password is "1234". For security reasons, the administrator should change the default administrator password as soon as possible.

Procedure

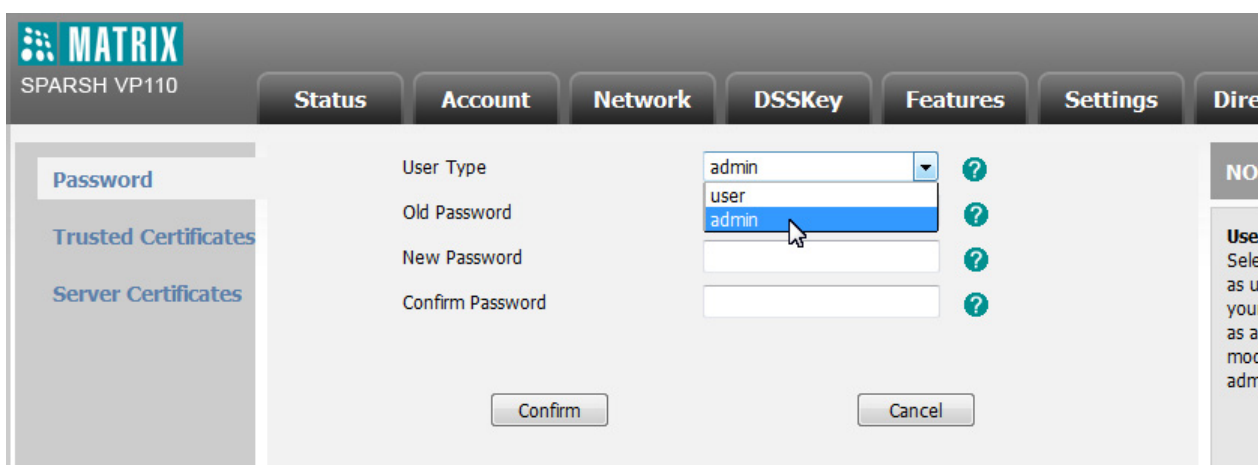
Administrator password can be changed using the configuration files or locally.

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?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).



The screenshot shows the 'MATRIX SPARSH VP110' web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', and 'Direct'. The left sidebar has 'Password', 'Trusted Certificates', and 'Server Certificates'. The main content area is titled 'Password' and contains a 'User Type' dropdown menu with 'admin', 'user', and 'admin' (highlighted). Below the dropdown are three text input fields: 'Old Password', 'New Password', and 'Confirm Password', each with a green question mark icon to its right. At the bottom are 'Confirm' and 'Cancel' buttons.

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

To change the administrator password via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Set Password**.
- Enter the current administrator password in the **Current PWD** field.

- Enter new password in the **New PWD** field and **Confirm PWD** field.

Valid characters are ASCII characters 32-126(0x20-0x7E).

- Press the **Save** soft key to save the change.

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 "1111". 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?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 star 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.

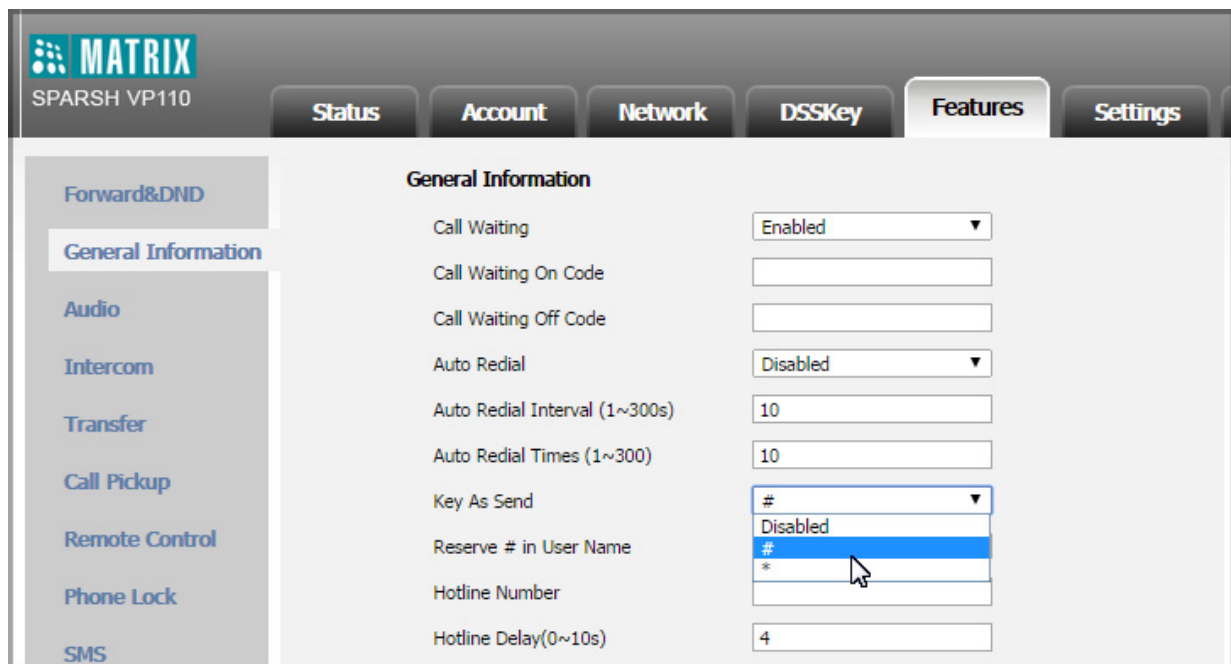
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load Configure a send sound and key tone. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-audio&q=load">http://<phoneIPAddress>/servlet?p=features-audio&q=load
	Phone User Interface	Configure the send key.

To configure a send key via web user interface:

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



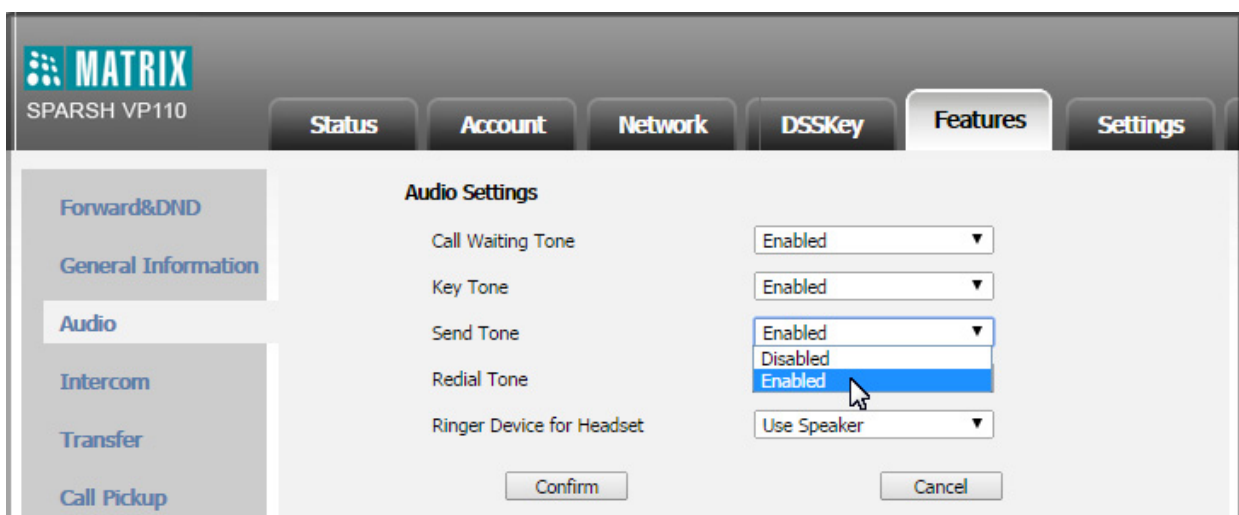
The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Features' tab is selected. On the left, a sidebar lists various features: Forward&DND, General Information (selected), Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, and SMS. The main content area is titled 'General Information' and contains several settings:

Setting	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	# (dropdown menu open showing options: #, Disabled, #, *)
Reserve # in User Name	
Hotline Number	
Hotline Delay(0~10s)	4

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

To configure a send sound 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 Sound** list.





The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Features' tab is selected. On the left, a sidebar lists various features: Forward&DND, General Information, Audio (selected), Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, and SMS. The main content area is titled 'Audio Settings' and contains several settings:

Setting	Value
Call Waiting Tone	Enabled
Key Tone	Enabled
Send Tone	Enabled (dropdown menu open showing options: Enabled, Disabled, Enabled)
Redial Tone	
Ringer Device for Headset	Use Speaker

At the bottom of the settings area, there are two buttons: **Confirm** and **Cancel**.



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

To configure send key via phone user interface:

- Press **Menu->Features->Key as Send**.
- Press  or , or the **Switch** soft key to select # or * from the **Key as Send** field, or select **Disable** to disable this feature.
- Press the **Save** soft key to save the change.

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. IP phones offer three types of phone lock:

- **Menu Key:** The Menu soft key is locked. You cannot access the menu of the phone until unlocked.
- **Function Keys:** The function keys are locked. You cannot use the Message, Mute, RD, Transfer, , navigation keys and soft keys until unlocked.
- **All Keys:** All keys are locked except the Volume key. You are only allowed to dial emergency numbers; answer incoming calls by lifting the handset, pressing the Speakerphone key, the Headset key or the  key, place an active call on hold by pressing the Hold soft key, resume the held call by pressing the Resume soft key, and end the call by hanging up the handset or pressing the Speakerphone key.



The emergency number setting, if desired, must be made before lock activation. For more information, refer [“Emergency Number”](#).

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	<p>Configure phone lock.</p> <p>Configure the type of phone lock.</p> <p>Change the unlock PIN.</p> <p>Configure the IP phone to automatically lock the keypad after a time interval.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p> <p>Assign a keypad lock key.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure phone lock.</p> <p>Configure the type of phone lock.</p> <p>Change the unlock PIN.</p> <p>Configure the IP phone to automatically lock the keypad after a time interval.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-phoneunlock&q=load">http://<phoneIPAddress>/servlet?p=features-phoneunlock&q=load</p> <p>Assign a keypad lock key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</p>
	Phone User Interface	Configure the type of phone lock.

To configure phone lock via web user interface:

- Click on **Features->Phone Lock**.
- Select **Enabled** from the **Phone Lock Enable** list.
- Select the desired type from the **Phone Lock Type** 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.

MATRIX SPARSH VP110

Log Out

Status Account Network **DSSKey** Features Settings Directory Security

Forward&DND
General Information
Audio
Intercom
Transfer
Call Pickup
Remote Control
Phone Lock

Phone Lock Enable: Disabled
Phone Lock Type: All Keys
Phone Unlock PIN(0~15 Digit): *****
Phone Lock Time Out(0~3600s): 0
Emergency: 112,911,110

Confirm Cancel

NOTE

Phone Lock
It 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.

IP phones offer three types of phone lock: Menu Key, Function Keys and All Keys.

The IP phone will not be locked immediately after the phone lock type is configured.

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

To configure a keypad lock key via web user interface:

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

MATRIX SPARSH VP110

Log Out

Status Account Network **DSSKey** Features Settings Directory Security

Programable Key

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Prefix	N/A			
SoftKey 3	Local Group	N/A			
SoftKey 4	XML Group	N/A			
Up	XML Browser	N/A			
Down	History	N/A			
Left	Menu	N/A			
Right	Forward	N/A			
OK	DND	Local History			
Mute	ReCall	N/A			
Tran	SMS	N/A			
	New SMS	N/A			
	XML Directory	N/A			
	Status	N/A			
	Multicast Paging	N/A			
	Local Directory	N/A			
	Hot Desking	N/A			
	Zero Touch	N/A			
	Phone Lock	N/A			
	Directory	N/A			
	Paging List	N/A			
	Tran	N/A			

Confirm Cancel Reset to default

NOTE

Programmable Keys
Customizes the soft keys, navigation keys and function keys.

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

To configure the type of phone lock via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Phone Lock**.
- Press **⏮** or **⏭**, or the **Switch** soft key to select Enabled from the **Lock Enable** field.

- Press ◀ or ▶, or the **Switch** soft key to select the desired type from the **Lock Type** field.
- Press the **Save** soft key to save the change.

To change the unlock PIN via phone user interface:

- Press **Menu->Settings->Basic Settings->Change PIN**.
- Enter the current unlock PIN in the **Current PIN** field.
- Enter the new unlock PIN in the **New PIN** field.
- Enter the new unlock PIN again in the **Confirm PIN** field.
- Press the **Save** soft key to save the change.

Audio Settings


Audio Setting allows you to configure the following:

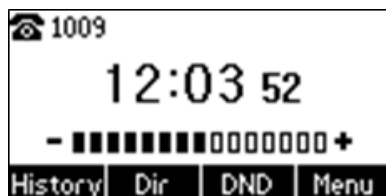
- Volume
- Ring Tones


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.


To adjust the volume when the phone is idle:

- Press  to adjust the ringer volume.



If the ringer volume is adjusted to minimum, the  icon will appear on the LCD screen.

To adjust the volume when the phone is during a call:

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



Ring Tones

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

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

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

The screenshot shows the MATRIX SPARSH VP110 web interface. The 'Account' tab is selected. Under the 'Basic' section, the 'Ring Type' dropdown menu is open, showing a list of ring tones: Common, Ring1.wav, Ring2.wav, Ring3.wav, Ring4.wav, and Ring5.wav. A mouse cursor is hovering over 'Ring1.wav'. Other settings like 'Proxy Require', 'Local Anonymous', 'Send Anonymous Code', 'On Code', 'Off Code', 'Anonymous Call Rejection', 'Missed Call Log', and 'Auto Answer' are also visible with their respective dropdown menus and a 'Confirm' button at the bottom.

- 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 phone via phone user interface:

- Press **Menu->Settings->Basic Settings-> Sound->Ring Tones**.
- Press or to select Common or the desired Account.
- Press or to select the desired ring tone.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

To upload a custom ring tone for your phone via web user interface:

- Click on **Settings->Preference**.
- Click **Browse** to locate a ring tone (file format must be *.wav) file from your local system.

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

MATRIX
SPARSH VP110

Status **Account** **Network** **DSSKey** **Features** **Settings**

Preference

Language: English(Angielski) ?

Live Dialpad: Enabled ?

Inter Digit Time(1~14s): 4 ?

Contrast: 1 ?

WatchDog: Enabled ?

Ring Type: Ring1.wav Del ?

Upload Ringtone: Browse No file selected. Upload No file selected.

Confirm Cancel



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 [“Adding Contacts”](#)). 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.

All custom ring tone files must be within 100KB. Uploading custom ring tones for your phone is configurable via web user interface only.

Contact Management

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

- Directory
- Local Directory
- Blacklist
- Remote Phone Book

Directory


Directory provides easy access to frequently used lists. The lists may contain Local Directory, History and Remote Phone Book. The desired lists can be added to Directory using a directory file. For more information on how to customize a directory file, refer "[Directory Template](#)".

Procedure


Directory can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify the access URL of the Directory file. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Configure the Directory. Navigate to: http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

To configure the directory via web user interface:

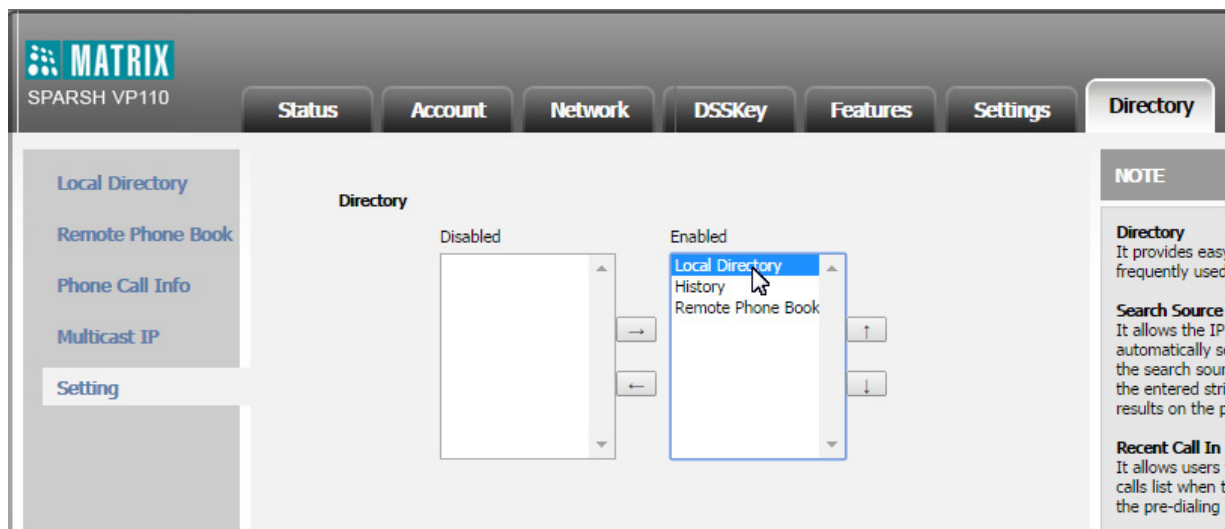
- Click on **Directory->Setting**.
- In the **Directory** block, select the desired list from the **Disabled** column and then click  .

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 enabled lists, select the desired list and then click  or  .

The LCD screen displays the list(s) in the adjusted order.



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

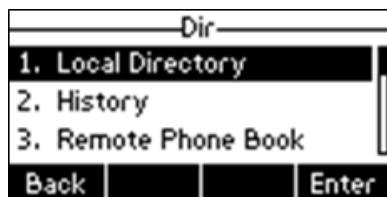


Directory is configurable via web user interface only.

To check the directory via phone user interface:

- Press the **Dir** soft key when the phone is idle.

The LCD screen displays the list(s) in the directory as per configurations done.



- If there is only one list in the directory, press the **Dir** soft key to enter this list directly.

Local Directory

The built-in phone directory can store contact names and numbers. You can store up to 1000 contacts and 5 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. For more information on how to customize a contact file, refer "[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. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Add a group and a contact to the local directory. Navigate to: <code>http://<phoneIPAddress>/servlet?p=contactsbasic&q=load&num=1&group=</code>
	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.

The screenshot displays the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The left sidebar shows a menu with options: Local Directory, Remote Phone Book, Phone Call Info, Multicast IP, and Setting. The main content area is divided into two sections. The top section, titled 'Local Directory', contains a table with columns: Index, Name, Office Number, Mobile Number, Other Number, and All Contacts. The table lists four contacts: Alice, Bob, JOHN DOE, and Ryan. The bottom section, titled 'Group Setting', contains a form with fields for Name, Office Number, Mobile Number, Other Number, Ring Tone (set to Auto), and Group (set to All Contacts). There are 'Add' and 'Edit' buttons for the Group field. To the right of the form, there are buttons for 'Delete All', 'Delete', 'Move To', and 'All Cont'. Below the form, there is a section titled 'Import Local Directory File' with buttons for 'Choose File', 'Import XML', 'Export XML', 'Import CSV', and 'Export CSV'. A 'Show Title' checkbox is also present.

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



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

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Press the **AddGrp** soft key.
- Enter the desired group name in the **Name** field.
- Press  to select the **Ring** and then press  or  to select the desired group ring tone.

If **Auto** is selected, this group will use the ring tone assigned to the account. For more information on ringtone for an account, refer ["Ring Tones"](#).

- Press the **Add** soft key to save the change or the **Back** soft key to cancel.

Editing Groups

To edit a group in the local directory via phone user interface:

- Press the **Dir** soft key.



The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

Select the desired group.

- Press the **Option** soft key, and then select **Detail** from the prompt list.

- Press  or  to scroll through the group information and then edit.
- Press the **Save** soft key to save change or the **Back** soft key to cancel.



Editing a group in the local directory via web user interface is similar to adding a group via web user interface, except that here the group is edited. For more information, you may refer [“Adding Groups”](#).

Deleting Groups

To delete a group from the local directory via phone user interface:

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Select the desired group.
- Press the **Option** soft key, and then select **Delete** from the prompt list.

The LCD screen prompts “Delete selected group?”

- Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.

You can also delete all groups by pressing the **Option** soft key and then select **Delete All**. For more information, refer above steps.



Deleting a group in the local directory via web user interface is similar to adding a group via web user interface, except that here the group gets deleted from the local directory. For more information, you may refer [“Adding Groups”](#).

Adding Contacts

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.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. On the left, a sidebar menu lists Local Directory, Remote Phone Book, Phone Call Info, Multicast IP, and Setting. The main content area displays a table of contacts under the 'Local Directory' section. The table has columns for Index, Name, Office Number, Mobile Number, Other Number, and a dropdown for 'All Contacts'. Below the table, there are pagination controls (Page 1, Prev, Next, Hang Up, Delete All, Delete, Move To, All Conts) and two main sections: 'Directory' and 'Group Setting'.

Index	Name	Office Number	Mobile Number	Other Number	All Contacts
1	Alice	26588856	989806336		All Contacts
2	Bob	96636665			All Contacts
3	JOHN DOE	6363446	9898905660		All Contacts
4	Ryan	789885656			All Contacts
5					
6					
7					
8					
9					
10					

Directory

Name:

Office Number:

Mobile Number:

Other Number:

Ring Tone:

Group:

Group Setting

Group:

Ring:

Import Local Directory File

No file chosen

No file chosen

☐ Show Title

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

You can add contacts to the local directory in the following ways via phone user interface:

- Manually
- From call history
- From remote phone book

Adding Contacts Manually

To add a contact to the local directory manually:

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Select the desired group and then press the **Enter** soft key.
- Press the **Add** soft key.
- Enter the name and the office, mobile or other numbers in the corresponding fields.



- Press ◀ or ▶, or the **Switch** soft key to select the desired ring tone from the **RingTones** field.

If **Auto** is selected, this contact will use the ring tone assigned to the group.

- Press the **Add** soft key to save the change or the **Back** soft key to cancel.

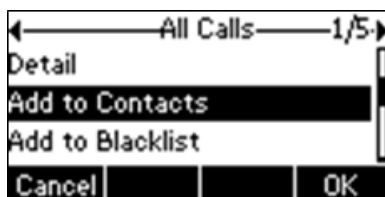


If the contact has existed in the directory, the LCD screen will prompt "Contact name existed!".

Adding Contacts from Call History

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

- Press the **History** soft key.
- Press ▲ or ▼ to select the desired entry.
- Press the **Option** soft key, and then select **Add to Contacts** from the prompt list.



- Enter the contact name.

- Press the **Save** soft key to save the change.



The entry is successfully saved to the local directory.

Adding Contacts from Remote Phone Book

To add a contact to the local directory from remote phone book:

- Press **Menu->Directory->Remote Phone Book**.

If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to enter remote phone book.

- Press  or  to select the desired contact.
- Press the **Option** soft key, and then select **Add to Contacts** from the prompt list.
- Press the **Save** soft key to save the contact in the local directory.

If the contact already exists in the local directory, the LCD screen will prompt “Overwrite the original contact?”. Press the **OK** soft key to overwrite the original contact in local directory or the **Cancel** soft key to cancel.

For more information on remote phone book operating, refer to [“Remote Phone Book”](#).

Editing Contacts



To edit a contact in the local directory:



- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Select the desired group and then press the **Enter** soft key.
- Press  or  to select the desired contact.
- Press the **Option** soft key, and then select **Detail** from the prompt list.

- Press  or  to scroll through the contact information and then edit.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.



Deleting Contacts

To delete a contact from the local directory:

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.

If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Select the desired group and then press the **Enter** soft key.
- Press  or  to select the desired contact.
- Press the **Option** soft key, and then select **Delete** from the prompt list.

The LCD screen prompts "Delete selected item?"

- Press the **OK** soft key to confirm the deletion or the **Cancel** soft key to cancel.

Placing Calls to Contacts



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

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Select the desired group and then press the **Enter** soft key.
- Press  or  to select the desired contact.

- Do one of the following:

- If only one number of the contact is stored in the local directory, press the **Send** soft key to dial out the number.
- If multiple numbers of the contact are stored in the local directory, press the **Send** soft key to display a list of numbers.

Press  or  to select the desired number.

Press the **Send** soft key to dial out the number.

Searching for Contacts



To search for a contact in the local directory:

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Press  or  to scroll to the **All Contacts** field.
- Press the **Search** soft key.
- Enter a few continuous characters of the contact name or number using the keypad.

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

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 may contain Local Directory, History and Remote Phone Book.

The search source list can be configured using a super search file. For more information on how to customize a super search template, refer ["Super Search Template"](#).

Procedure

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




Configuration File	<MAC>.cfg	Specify the access URL of the super search file. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Configure the search source list in dialing. Navigate to: http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

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

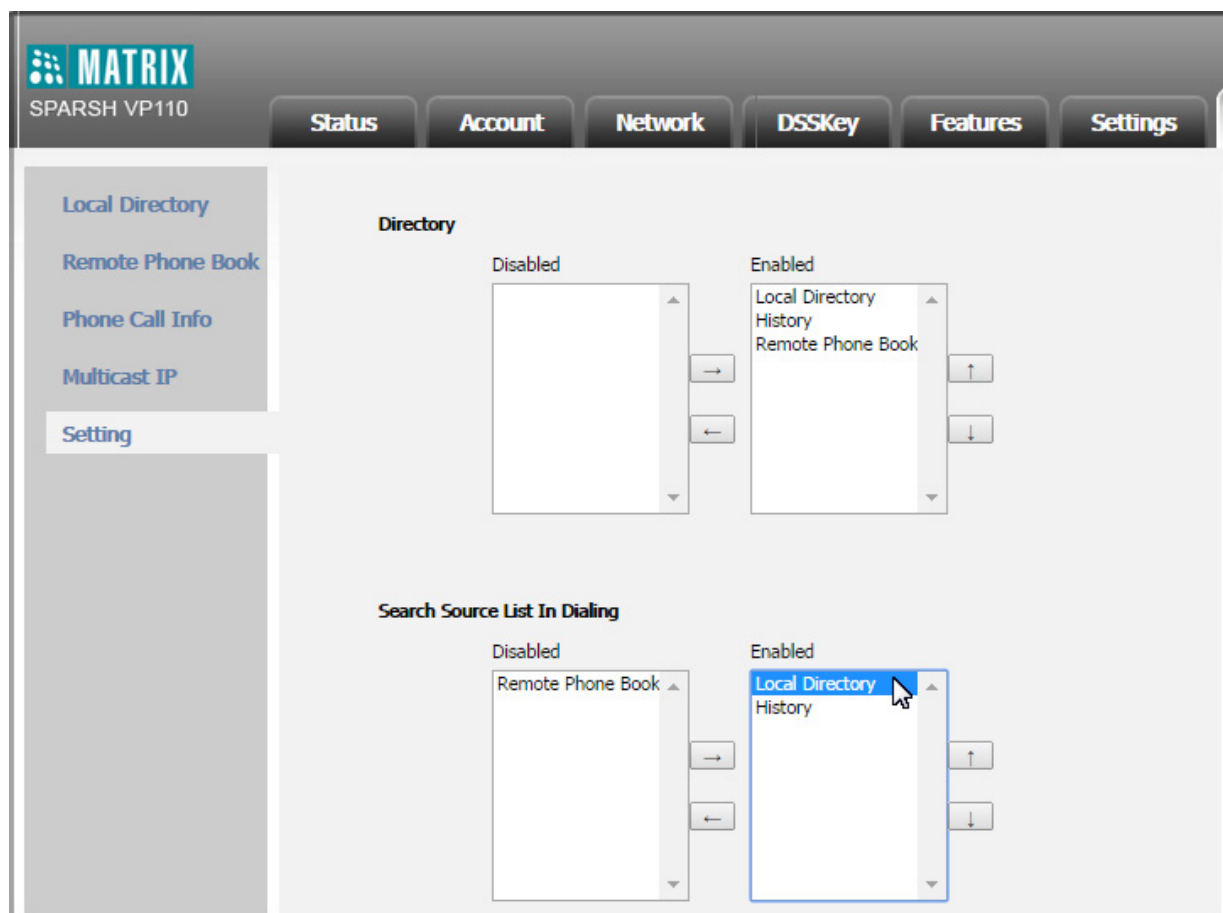
- Click on **Directory->Setting**.
- In the **Search Source List In Dialing** block, select the desired list from the **Disabled** column and then

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  .

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





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

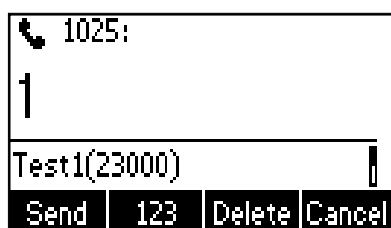


Search source list in dialing is configurable via web user interface only.

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

- Pickup the handset or press the speakerphone.
- Enter a few continuous characters of the contact name or number using the keypad.

The contacts in the enabled search source lists whose name or number matches the characters entered will appear on the LCD screen. You can press  or  to select the desired contact and then place a call to the contact.



Importing/Exporting Contact Lists

You can manage your phone's local directory via phone user interface or web user interface. But you can only import or export the contact list via web user interface.

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

- Click on **Directory->Local Directory**.
- Click **Browse** to locate a contact list file (file format must be *.xml) from your local system.

Index	Name	Office Number	Mobile Number	Other Number	All Contacts
1	Alice	26588856	989806336		All Contacts
2	Bob	96636665			All Contacts
3	JOHN DOE	6363446	9898905660		All Contacts
4	Ryan	789885656			All Contacts
5					
6					
7					
8					
9					
10					

- Click **Import XML** to import the contact list.

The web user interface prompts “The original contact will be covered, Continue?”.

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

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

- Click on **Directory->Local Directory**.
- Click **Browse** to locate a contact list file (file format must be *.csv) from your local system.
- (Optional) Select the **Show Title** check box.

It will prevent importing the title of the contact information which is located in the first line of the CSV file.

- Click **Import CSV** to import the contact list.

- (Optional) Click **On** in the **Delete Old Contacts** field.

It will delete all existing contacts while importing the contact list.

- (Optional) Select the contact information you want to import into the local directory from the **Index** list.
- Click **Import** to complete importing the contact list.

To export contact list via web user interface:

- Click on **Directory->Local Directory**.
- Click **Export XML** (or, **Export CSV**).

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SPARSH VP110

Navigation: Status, Account, Network, DSSKey, Features, Settings, **Directory**

Local Directory

Index	Name	Office Number	Mobile Number	Other Number	All Contacts
1	Alice	26588856	989806336		<input type="checkbox"/>
2	Bob	96636665			<input type="checkbox"/>
3	JOHN DOE	6363446	9898905660		<input type="checkbox"/>
4	Ryan	789885656			<input type="checkbox"/>
5					<input type="checkbox"/>
6					<input type="checkbox"/>
7					<input type="checkbox"/>
8					<input type="checkbox"/>
9					<input type="checkbox"/>
10					<input type="checkbox"/>

Page 1 | Prev | Next | Hang Up | Delete All | Delete | Move To | All Contacts

Group Setting

Group:
 Ring: Auto

Add Edit Delete Delete All

Import Local Directory File

Choose File No file chosen
 Import XML Export XML
 Choose File No file chosen
 Import CSV Export CSV ☐ Show Title

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



Importing/exporting contact lists are configurable via web user interface only.

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 manually:

- Press **Menu->Directory->Blacklist**.

- Press the **Add** soft key.
- Enter the name, office, mobile or other numbers in the corresponding fields.



- Press the **Add** soft key to save the change or the **Back** soft key to cancel.



To add a contact to the blacklist from local directory:

- Press the **Dir** soft key.

The IP phone enters the local directory directly as there is only Local Directory in the directory by default.



If Local Directory is removed from the directory, press **Menu->Directory->Local Directory** to enter the local directory.

- Press  or  to select the desired contact.
- Press the **Option** soft key and then select **Add to Blacklist** from the prompt list.

The LCD screen prompts "Move selected to blacklist?"



- Press the **OK** soft key to confirm the setting.

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 access your corporate directory through your phone. You can add local contacts, search for a contact, or simply dial a contact number from the corporate directory.

You can configure your new phone to access up to 5 corporate directories. The phone supports up to 2000 corporate directory entries.

Remote phone book can be customized. For more information how to customize a remote phone book, refer [“Remote XML Phone Book”](#).

Search 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	<p>Specify the access URL of the remote phonebook.</p> <p>Specify whether to query the entry name from the remote phone book for outgoing/incoming calls.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phonebook.</p> <p>Specify whether to refresh the local cache of the remote phone book at a time when accessing the remote phone book.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
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Local	Web User Interface	<p>Specify the access URL of the remote phonebook.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p> <p>Specify whether to query the entry name from the remote phone book for outgoing/incoming calls.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phonebook.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p>
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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.

MATRIX
SPARSH VP110

Local Directory

Remote Phone Book

Phone Call Info

MulticastIP

Setting

Status Account Network DSSKey Features Settings **Directory**

Index	Remote URL	Display Name
1	192.168.101.100	Main Directory
2		
3		
4		
5		

Search Remote Phonebook Name: Disabled

Search Flash Time(Seconds): 21600

Confirm Cancel

NOTE
Remote Phone Book
This feature allows you to download contact list from a remote server. Input the phonebook URL and rename the book.

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





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

To access remote phone book via phone user interface:

- Press **Menu->Directory->Remote Phone Book**.

If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to enter remote phone book.



- Press  or  to select the desired remote group, and then press the **Enter** soft key.

The phone connects to load the remote phone book, and then the contacts in the remote phone book appear on the LCD screen.

- Press the **Back** soft key to back to the previous interface.

To search for a contact in the remote phone book:

- Press **Menu->Directory->Remote Phone Book**.

If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to enter remote phone book.

- Select the desired remote group, and then press the **Enter** soft key to load the remote phone book.
- Press the **Search** soft key.
- Enter a few continuous characters of the contact name or number using the keypad.





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

To place a call from the remote phone book:

- Press **Menu->Directory->Remote Phone Book**.

If Remote Phone Book is added to the directory, press **Dir->Remote Phone Book** to enter remote phone book.

- Press  or  to select the desired remote group, and then press the **Enter** soft key to load the remote phone book.
- Select the desired contact in the remote phone book.
- Press the **Send** soft key.

In addition, you can enable the phone to present the caller identity stored in the remote phone book when receiving a call.

To enable the presentation of caller identity stored in remote phone book via web user interface:

- Click on **Directory->Remote Phone Book**.
- Select **Enabled** from the **Incoming/Outgoing Call Lookup** list.
- Enter the desired refresh period in the **Update Time Interval (Seconds)** field. The default value is 21600 seconds.
- Click **Confirm** to save the change.

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”](#).

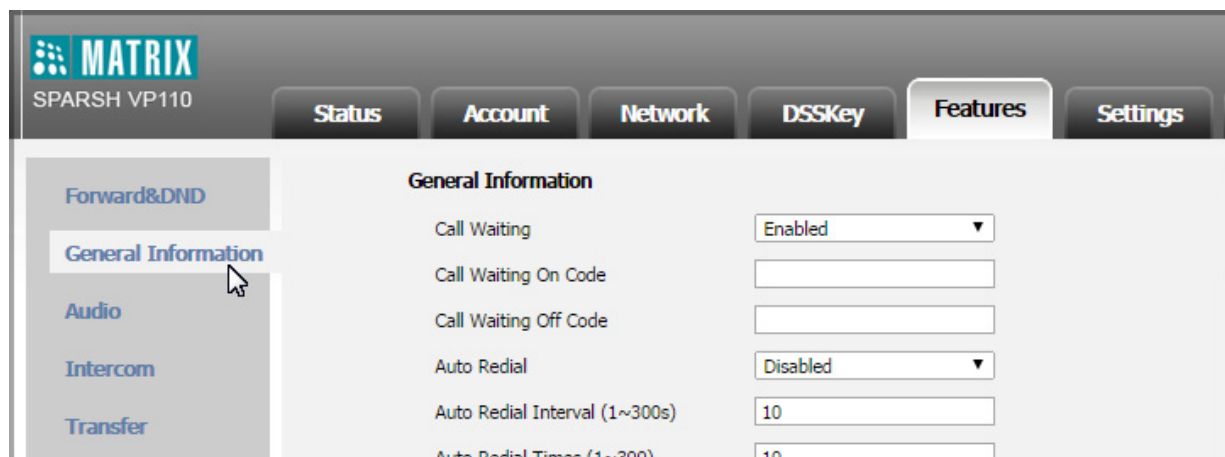
Procedure

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

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

To configure call log feature via web user interface:

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



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

PswDial	Disabled ▼
Save Call Log	Enabled ▼
Suppress DTMF Display	Disabled ▼
Suppress DTMF Display Delay	Enabled ▼
Play Local DTMF Tone	3 ▼
DTMF Repetition	Enabled ▼
Play Hold Tone	Enabled ▼

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

To enable history record via phone user interface:

- Press **Menu->Features->History Setting**.
- Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **History Record** field.

- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

To view Number instead of Name in the call history:

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

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

The screenshot shows a configuration menu with the following items:

- Call List Show Number**: A dropdown menu with 'Disabled' selected and 'Enabled' highlighted by a mouse cursor.
- Voice Mail Tone**: A text field.
- DHCP Hostname**: A text field containing 'Matrix SPARSH VP110'.
- Reboot in Talking**: A dropdown menu with 'Enabled' selected.
- Hide Feature Access Codes**: A dropdown menu with 'Disabled' selected.
- Display Method on Dialing**: A dropdown menu with 'User Name' selected.

At the bottom of the menu are two buttons: **Confirm** and **Cancel**.

- 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 view the call history:

- Press the **History** soft key.

The LCD screen displays All Calls list.

- Press or to switch between **All Calls**, **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
- Press or to select the desired entry.
- Press the **Option** soft key, and then select **Detail** from the prompt list.

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

To place a call from the call history list:

- Press the **History** soft key.
- Press or to switch between **All Calls**, **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
- Press or to scroll to the desired entry.
- Press the **Send** soft key.

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

- Press the **History** soft key.

- Press ◀ or ▶ to switch between **All Calls**, **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
- Press ▲ or ▼ to scroll to the desired entry.
- Press the **Option** soft key, and then select **Add to Contacts** (or, **Add to Blacklist**) from the prompt list.
- Enter the desired values in the corresponding fields, and then press the **Save** soft key.



For more information, refer "[Contact Management](#)".

To delete an entry from the call history list:

- Press the **History** soft key.
- Press ◀ or ▶ to switch between **All Calls**, **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
- Press ▲ or ▼ to scroll to the desired entry.
- Press the **Delete** soft key.

To delete all entries from the call history list:

- Press the **History** soft key.
- Press ◀ or ▶ to switch between **All Calls**, **Placed Calls**, **Received Calls**, **Missed Calls** and **Forwarded Calls**.
- Press the **Option** soft key, and then select **Delete All** from the prompt list.
- Press the **OK** soft key.

The LCD screen prompts "Delete all records?".



- Press the **OK** soft key to confirm the deletion or the **Cancel** soft key 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.

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?p=account-basic&q=load&acc=0

To configure missed call log via web user interface:

- Click on **Account-> Basic**.
- Select the desired value from the **Missed Call Log** list.

The screenshot displays the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account (selected), Network, DSSKey, Features, and Settings. On the left, a sidebar shows options for Register, Basic (selected), Codec, and Advanced. The main configuration area is titled 'Account' and contains several settings. The 'Missed Call Log' setting is currently set to 'Enabled' and its dropdown menu is open, showing 'Enabled', 'Disabled', and 'Enabled' as options. Other settings include 'Local Anonymous' (Off), 'Local Anonymous Rejection' (Off), 'Send Anonymous Code' (Off Code), 'On Code', 'Off Code', 'Send Anonymous Rejection Code' (Off Code), 'On Code', 'Off Code', and 'Auto Answer'. At the bottom of the configuration area are 'Confirm' and 'Cancel' buttons.

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

System Customization

Logo Customization

Logo customization allows unifying the IP phone appearance or displaying a custom image on the idle screen such as a company logo, instead of the default system logo.

The following table lists the logo file format, resolution and total files size for the phone model.

Phone Model	Logo File Format	Resolution	Total Files Size
SPARSH VP110	.dob	<=132*642 gray scale	<=200KB

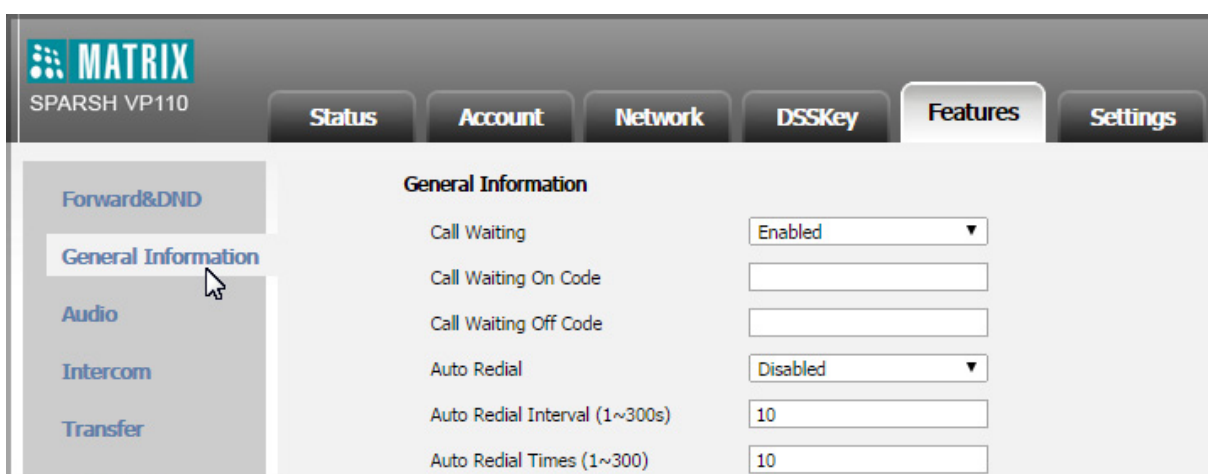
Procedure

The logo shown on the idle screen can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the logo shown on the idle screen. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Configure the logo shown on the idle screen. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load

To upload a custom logo via web user interface:

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



- Select **Custom Logo** from the **Use Logo** list.

- Click **Browse** to locate the logo file from your local system.
- Click **Upload** to upload the file.
- Click **Confirm** to save the change.



- *Delete option will appear after you upload a custom logo, you can click **Delete** to delete the custom logo.*

- *Logo customization is configurable via web user interface only.*

Programmable Keys

You can customize the soft keys, navigation keys and function keys on the keypad. The SPARSH VP110 IP phone supports 11 programmable keys.

To customize the programmable keys via web user interface:

- Click on **DSSKey->Programmable Key**.
- Customize specific features for these keys.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Directory	N/A			
SoftKey 3	DND	N/A			
SoftKey 4	Menu	N/A			
Up	History	Local History			
Down	N/A	N/A			
Left	N/A	N/A			
Right	N/A	N/A			
OK	Status	N/A			
MUTE	N/A	N/A			
TRAN	N/A	N/A			

- (Optional) Enter a string that will appear on the LCD screen in the **Label** field.

Label is configurable only when customizing four soft keys.

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

You can click **Reset to default** to reset custom settings to defaults.



Programmable keys are configurable via web user interface only.

Commonly used programmable key features are explained in the following sub-chapters in detail¹:

- Speed Dial
- Direct Pickup
- Group Pickup
- Prefix
- Local Directory
- Local Group
- XML Directory
- XML Group
- XML Browser
- SMS
- New SMS
- Zero Touch
- Keypad Lock
- Intercom
- Multicast Paging

Speed Dial

You can use this key feature to speed up dialing numbers often used or hard to remember.

Dependencies: Type (Speed Dial)

Value (the number you want to dial out)

Usage: Press the programmable key to dial out the number specified in the Value field.

Direct Pickup

You can use this key feature to answer someone else's incoming call on the phone.

Dependencies: Type (Direct Pickup)

¹. Programmable Keys are subject to change depending on various reasons. For more information, contact your vendor/re-seller.

Value (the direct pickup code followed by the specific phone number)

Usage: Press the programmable key on your phone when the specific phone number receives an incoming call.

The call is 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 (Group Pickup)

Value (the group pickup feature code)

Usage: Press the programmable key on your phone when a phone number in the group receives an incoming call.

The call is answered on your phone.

Prefix

You can use this key feature to add a specified prefix number before the dialed number.

Dependencies: Type (Prefix)

Value (the prefix number)

Usage: Press the programmable key when the phone is idle, then the phone will enter into the dialing screen and display the prefix number which you specified in the Value field. You can enter other digits and then dial out.

Local Directory

You can use this key feature to access the local directory quickly. For more information, refer [“Local Directory”](#).

Dependencies: Type (Local Directory)

Usage: Press the programmable key to access the local directory quickly.

Local Group

You can use this key feature to access the group in the local directory quickly. For more information, refer [“Local Directory”](#).

Dependencies: Type (Local Group)

Line (the group you want to access)

Usage: Press the programmable key to access the group specified in the Line field.

XML Directory

You can use this key feature to access the corporate directory quickly. For more information, refer [“Remote Phone Book”](#).

Dependencies: Type (XML Directory)

Usage: Press the programmable key to access the corporate directory quickly.

XML Group

You can use this key feature to access the remote group in the corporate directory quickly. You should configure remote phone book in advance. For more information, refer "[Remote Phone Book](#)".

Dependencies: Type (XML Group)

Line (the remote group you want to access if the remote phone book is configured)

Usage: Press the programmable key to access the remote group specified in the Line field.

XML Browser

You can use this key feature to access the XML browser quickly. The XML browser allows you to create custom services which meet your functional requirements on the server. You can customize practical applications, such as weather report, stock information, Google search, etc.

Dependencies: Type (XML Browser)

Value (the access URL for xml browser)

Usage: Press the programmable key to access the XML browser specified in the Value field.

SMS

You can use this key feature to quick access short message service. For more information, refer "[Short Message Service \(SMS\)](#)".

Dependencies: Type (SMS)

Usage: Press the programmable key when the phone is idle to access the short message service.

New SMS

You can use this key feature to quick access the new short message service. For more information, refer "[Short Message Service \(SMS\)](#)".

Dependencies: Type (New SMS)

Usage: Press the programmable key when the phone is idle to access the New Message screen. You can enter the text message and then send it.


Zero Touch

You can use this key feature to configure auto provision and network parameters quickly.


Dependencies: Type (Zero Touch)**Usage:**

- Press the programmable key to access the zero touch screen.
- Press the **OK** soft key in a few seconds.
- Configure the network parameters in the corresponding fields.
- Press the **Next** soft key.
- Configure the auto provision parameters in the corresponding fields.
- Press the **OK** soft key.
- The phone will reboot to update configurations.

Keypad Lock (Phone Lock)

You can use this key feature to immediately lock the keypad of your phone instead of long pressing . For more information, refer [“Keypad Lock \(Phone Lock\)”](#).

Dependencies: Type (Phone Lock)

Usage: Press the programmable key to immediately lock the keypad of your phone instead of long pressing .

Intercom

You can use this key to make outgoing intercom calls.

Dependencies: Type (Intercom)

Value (the remote extension number)

Usage: Press the programmable key to place an outgoing intercom call to the configured remote extension number.

Multicast Paging

You can use this key to make outgoing multicast paging calls.

Dependencies: Type (Multicast Paging)

Value (the multicast IP address and port number)

Usage: Press the programmable key to place an outgoing multicast paging call to the pre-configured multicast address; each IP phone pre-configured to listen to the multicast address can receive the RTP stream.

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 "123", "1234", "12345", "12abc", etc.
x	The "x" can be used as a placeholder for any character. Example: "12x" would match "121", "122", "123", "12a", 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 "9151234", "9161234", "9171234".
()	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 "923", "153", "673", 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: 1234567”. When trying to dial out the number “1234567”, you just need to enter “1” on the phone and then press the Send soft key.

You can also configure replace rules in batch using a replace rule template. For more information on how to customize a replace rule template, refer [“Replace Rule Template”](#).

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: <a href="http://<phoneIPAddress>/servlet?p=settings-dialplan&q=load">http://<phoneIPAddress>/servlet?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.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features **Settings**

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069
Voice Monitoring
SIP

Replace Rule Dial-now Area Code Block Out

Index	Prefix	Replace	
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>

Prefix Replace

- Click **Add** to add the replace rule.

When you enter the number “1” using the keypad and then press the Send soft key, the phone will dial out “1234” instead.

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.

Index	Prefix	Replace	
1	1	1234	<input checked="" type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>

Prefix Replace

- Edit the values in the **Prefix** and **Replace** fields.
- 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.

IP phones support up to 100 dial-now rules, which can be created either one by one or in batch using a dial-now rule template. For more information on how to customize a dial-now template, refer [“Dial-now Template”](#).

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: <code>http://<phoneIPAddress>/servlet?p=settings-dialnow&q=load</code> Configure the delay time for the dial-now rule. Navigate to: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>

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.

MATRIX SPARSH VP110

Status Account Network DSSKey Features Settings

Preference

Time & Date

Call Display

Upgrade

Auto Provision

Configuration

Dial Plan

Voice

Ring

Tones

Softkey Layout

TR069

Voice Monitoring

Replace Rule Dial-now Area Code Block Out

Index	Dial-now Rule
1	<input type="checkbox"/>
2	<input type="checkbox"/>
3	<input type="checkbox"/>
4	<input type="checkbox"/>
5	<input type="checkbox"/>
6	<input type="checkbox"/>
7	<input type="checkbox"/>
8	<input type="checkbox"/>
9	<input type="checkbox"/>
10	<input type="checkbox"/>

Rule 1234

Add Edit Del

- 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.

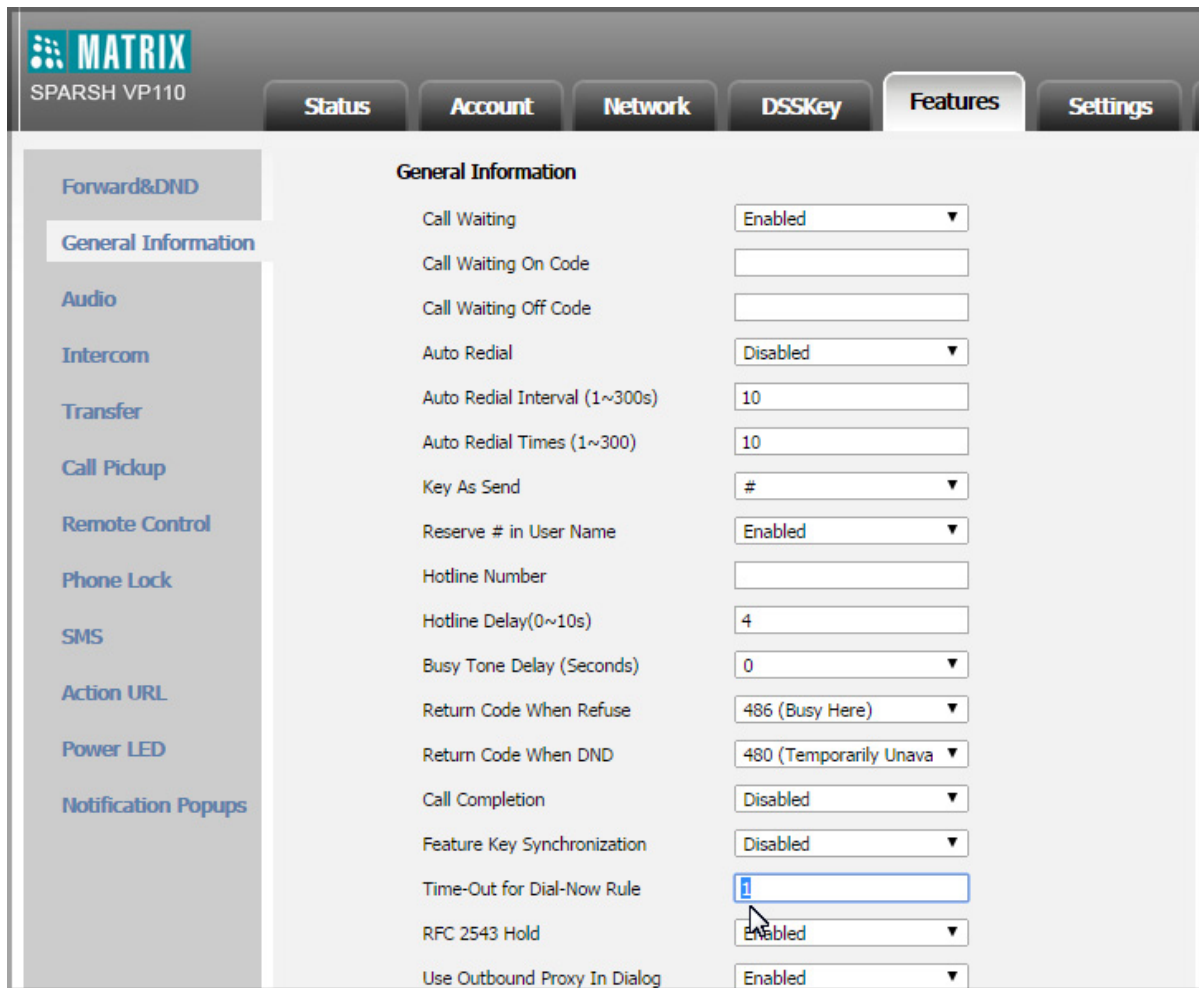


Dial-now rule is configurable via web user interface only.

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.



MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

SMS

Action URL

Power LED

Notification Popups

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	480 (Temporarily Unava
Call Completion	Disabled
Feature Key Synchronization	Disabled
Time-Out for Dial-Now Rule	1
RFC 2543 Hold	Enabled
Use Outbound Proxy In Dialog	Enabled

- 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?p=settings-areacode&q=load

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.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The Settings tab is active, and the left sidebar shows a tree view with options like Preference, Time & Date, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, and Ring. The main content area is titled 'Area Code' and contains three input fields: 'Code' (empty), 'Min Length (1-15)' (set to 1), and 'Max Length (1-15)' (set to 15). Below these fields are 'Confirm' and 'Cancel' buttons.

- 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 LCD screen will prompt "Forbidden Number". You can add 10 block out rules at most on your phone.

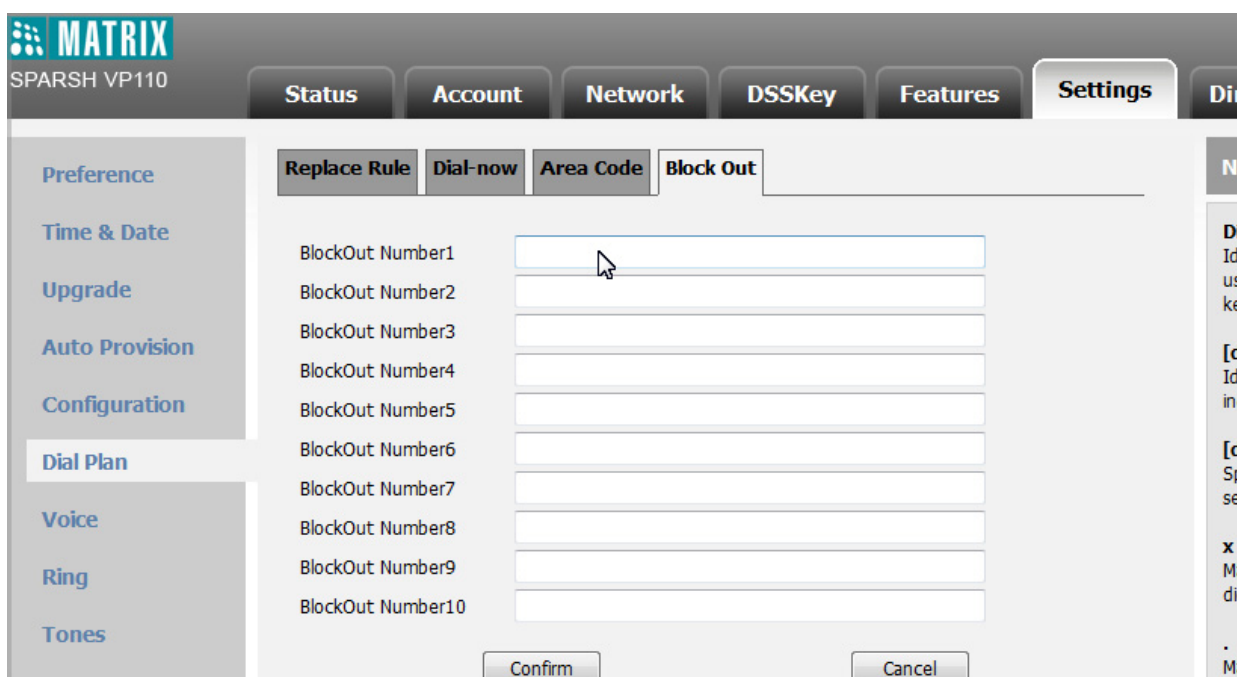
Procedure

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 desired line. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-blackout&q=load">http://<phoneIPAddress>/servlet?p=settings-blackout&q=load

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.



- Click **Confirm** to add the block out number.



Block out number is configurable via web user interface only.

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.



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.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', and 'Settings'. The 'Features' tab is active. On the left, a sidebar lists various features: Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock (highlighted), SMS, Action URL, Power LED, and Notification Popups. The main content area displays the 'Phone Lock' configuration. It includes a 'Phone Lock Enable' dropdown set to 'Disabled', a 'Phone Lock Type' dropdown set to 'All Keys', a 'Phone Unlock PIN(0~15 Digit)' field with asterisks, a 'Phone Lock Time Out(0~3600s)' field set to '0', and an 'Emergency' field containing '112,911,110'. At the bottom of the configuration area are 'Confirm' and 'Cancel' buttons.

- 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.

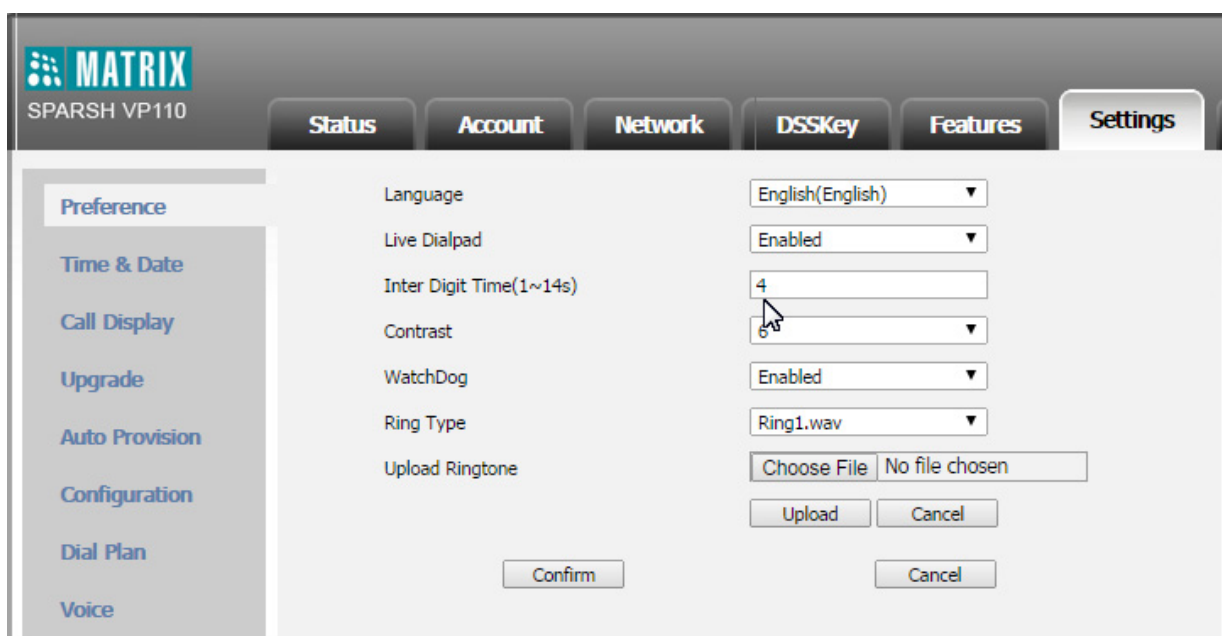
Procedure

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?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.



The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The Settings tab is active, and the left sidebar shows the Preference section selected. The main content area displays various settings: Language (English(English)), Live Dialpad (Enabled), Inter Digit Time(1~14s) (4), Contrast (6), WatchDog (Enabled), Ring Type (Ring1.wav), and Upload Ringtone (Choose File, No file chosen). At the bottom, there are Confirm and Cancel buttons.

- 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	<p>Configure the hotline number.</p> <p>Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure the hotline number.</p> <p>Specify the time (in seconds) the IP phone waits before automatically dial out the hotline number.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	Phone User Interface	<p>Configure the hotline number.</p> <p>Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number.</p>

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.

MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

SMS

Action URL

General Information

Call Waiting Enabled

Call Waiting On Code

Call Waiting Off Code

Auto Redial Disabled

Auto Redial Interval (1~300s) 10

Auto Redial Times (1~300) 10

Key As Send #

Reserve # in User Name Enabled

Hotline Number

Hotline Delay(0~10s)

Busy Tone Delay (Seconds) 0

Return Code When Refuse 486 (Busy Here)

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

To configure the hotline number via phone user interface:

- Press **Menu->Features->Hotline**.
- Enter the desired number in the **Hotline Number** field.

Hot Line

1. Hot Number:

1005

Back 123 Delete Save

- Enter the delay time (in seconds) in the **Hotline Time-out** field.

The valid values range from 0 to 10.

- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

Matrix SPARSH VP110 supports single SIP account which you can register with some ITSP's (Internet Telephony Service Provider) proxy server for making/receiving calls and access a wide range of features provided by the proxy server. Also 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.

This chapter provides detailed information regarding configuration of SIP account and related parameters for Proxy and Peer-to-Peer (P2P) calling using the SPARSH VP110 IP phones.

Configuration for Proxy Calling

Matrix SPARSH VP110 supports one SIP Account; thus allowing you to register one SIP account with the added advantage of server redundancy.



For more information about how server redundancy works, refer [“Server Redundancy”](#).

Procedure

SIP account and server redundancy can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the SIP account and server redundancy on the IP phone. For more information regarding specific configuration parameters refer following description, “To configure SIP account and server redundancy features using the configuration files”.
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Local	Web User Interface	Configure the server redundancy on the IP phone. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0
	Phone User Interface	Configure the SIP account and related parameters.

To configure SIP account and server redundancy features using the configuration files:

- Configure the following parameters in the <MAC>.cfg file as per the tables shown below:

Account Configuration and Registration

Parameter	Valid Values	Descriptions	Web Interface Path
account.1.enable	0 or 1	It enables or disables the account. 0 -Disabled 1 -Enabled The default value is 0.	Account->Register ->Line Active
account.1.register.enable	0 or 1	Enables or Disables the account registration. 0 -Disabled 1 -Enabled The default value is 1.	Account->Register->Register
account.1.label	String within 99 characters	It configures the label displayed on the LCD screen for the account. The default value is blank.	Account->Register ->Label
account.1.display_name	String within 99 characters	It configures the display name for account. The default value is blank.	Account->Register ->Display Name
account.1.auth_name	String within 99 characters	It configures the user name for register authentication for the account. The default value is blank.	Account->Register ->Register Name
account.1.user_name	String within 32 characters	It configures the register user name for the account. The default value is blank.	Account->Register ->User Name
account.1.password	String within 99 characters	It configures the password for register authentication for the account. The default value is blank.	Account->Register ->Password
account.1.sip_server. Y.address = (Y ranges from 1 to 2.)	IP address or domain name	It configures the IP address or domain name of the SIP server Y for the account. Example: account.1.sip_server.1.address = 10.2.1.128 The default value is blank.	Account->Register ->SIP Server Y->Server Host

account.1.sip_server. Y.port (Y ranges from 1 to 2)	Integer from 0 to 65535	It configures the port of SIP server Y for the account. The default value is 5060.	Account->Register - >SIP Server Y-> Port
account.1.sip_server. Y.expires (Y ranges from 1 to 2)	Integer from 30 to 2147483647	It configures the registration expiration time (in seconds) to SIP server Y for the account. The default value is 3600.	Account->Register - >SIP Server Y-> Server Expires
account.1.sip_server. Y.retry_counts (Y ranges from 1 to 2)	Integer from 0 to 20	It configures the times for the phone to retransmit the request when the SIP server Y is unavailable or there is no response from the SIP server Y for the account. The default value is 3.	Account->Register - >SIP Server Y - >Server Retry Counts
account.1.sip_server. Y.transport_type (Y ranges from 1 to 2)	Integer	It configures the transport type for the account. 0 -UDP 1 -TCP 2 -TLS 3 -DNS-NAPTR The default value is 0.	Account->Register - >Transport

DNS Resolution and Server Redundancy Configuration

Parameter	Valid Values	Descriptions	Web Interface Path
account.1.naptr_build	0 or 1	It configures the way of SRV query when there is no result from the NAPTR query for the account. 0 -SRV query using UDP only 1 -SRV query using UDP, TCP or TLS. The default value is 0.	NA

Fallback Mode

Parameter	Valid Values	Descriptions	Web Interface Path
account.1.fallback.red undancy_type	0 or 1	It configures the registration mode for the account in fallback mode. 0 -Concurrent registration 1 -Successive registration The default value is 0.	NA
account.1.fallback.tim eout	Integer from 10 to 2147483647	It configures the time interval (in seconds) for the phone to detect whether the working server is available by sending the registration request for the account. It is only applicable to successive registration mode. The default value is 120.	NA

Failover Mode

Parameter	Valid Values	Descriptions	Web Interface Path
account.1.sip_server. Y.failback_mode (Y ranges from 1 to 2)	0, 1, 2 or 3	It configures the mode for the phone to retry the primary server in failover mode for the account. 0 -newRequests: all requests are forwarded to the primary server first, regardless of the last used server. 1 -DNSTTL: the phone retries to use the primary server after the timeout of the DNSTTL configured for the SIP server. 2 -Registration: the phone retries to use the primary server when the SIP server's registration requires renewal. 3 -duration: the phone retries to use the primary server after the timeout defined by the parameter "account.1.failback_timeout". The default value is 0.	NA
account.1.sip_server. Y.failback_timeout (Y ranges from 1 to 2.)	Integer 0, from 60 to 65535	It configures the timeout (in seconds) for the phone to retry to use the primary server after failing over to the current working server for the account when the parameter "account.1.sip_server.Y.failback_mode" is set to 3 (duration). If you set the parameter between 1 and 59, the timeout will be 60 seconds. The default value is 3600.	NA
account.1.sip_server. Y.register_on_enable (Y ranges from 1 to 2.)	0 or 1	It enables or disables the phone to send registration requests to the secondary server for the account when encountering a failover. 0 -Disabled 1 -Enabled The default value is 0.	NA

- Save the configuration files to the root directory of the provisioning server.

The following shows an example of SIP account and server redundancy configurations for account 1 (since the IP phone supports single SIP account) in the configuration files:

```

###Account Configuration and Registration
account.1.enable = 1
account.1.register.enable = 1
account.1.label = 1234
account.1.display_name = 1234
account.1.auth_name = 1234
account.1.user_name = 1234
account.1.password = 1234
account.1.sip_server.1.address = matrix_ippbx.com

```

```
account.1.sip_server.1.port = 5060
account.1.sip_server.1.expires = 3600
account.1.sip_server.1.retry_counts = 3
account.1.sip_server.2.address = 192.168.1.15
account.1.sip_server.2.port = 5060
account.1.sip_server.2.expires = 3600
account.1.sip_server.2.retry_counts = 3
```

##DNS SRV

```
account.1.transport = 3
account.1.naptr_build = 0
```

##Fallback Mode

```
account.1.fallback.redundancy_type = 1
account.1.fallback.timeout = 120
```

##Failover Mode

```
account.1.sip_server.1.fallback_mode = 3
account.1.sip_server.1.fallback_timeout = 120
account.1.sip_server.1.register_on_enable = 0
account.1.sip_server.2.fallback_mode = 0
account.1.sip_server.2.register_on_enable = 0
```

To configure SIP Account and server redundancy feature via web user interface:

- Click on **Account->Register**.
- Select **Enabled** from the **Line Active** list.
- Select **Enabled** from the **Register** list.
- Enter the desired value in **Label**, **Display Name**, **Register Name**, **User Name**, **Password** and **SIP Server1/2** field respectively.
- Select **Enabled** from the **Enable Outbound Proxy Server** list and configure 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, if your ITSP uses an outbound server.

- Enter the desired interval in the **Proxy Fallback Interval** field.

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

To configure SIP Account and server redundancy feature via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234)->**Account** and press the **Enter** soft key.
- Press **◀** or **▶** or the **Switch** soft key to select **Enabled** from the **Active Line** field.
- Press **⌵** to configure the **Label**, **Display Name**, **Register Name**, **User Name**, **Password**, **SIP Server 1/2**, **Outbound Status**, **Outbound Proxy1/2**, **Proxy Fallback Interval**, **NAT Status** and press the **Save** soft key.

Configuration for Peer-to-Peer (P2P) Calling

Matrix SPARSH VP110 supports Peer-to-Peer (P2P) calling and Direct IP calling using the phone dial pad without registering to the SIP server.

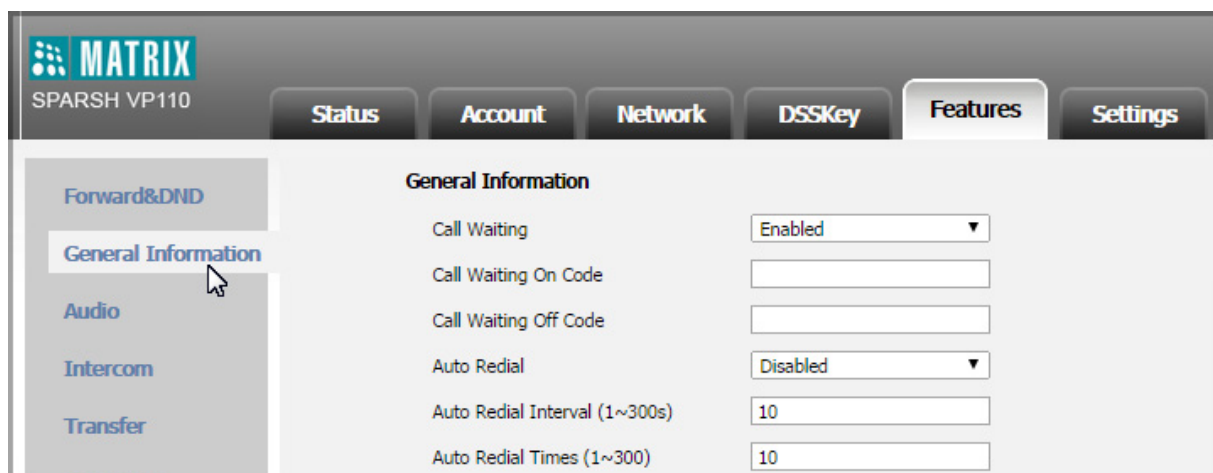
Procedure

Parameters for Peer-to-Peer (P2P) calling and Direct IP calling can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Configure the Allow IP Call and relevant SIP account parameters on the IP phone for Peer-to-Peer calling.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure the Allow IP Call parameters on the IP phone.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p> <p>Configure the SIP account parameters on the IP phone.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</p>

To configure Peer-to-Peer (P2P) calling feature via web user interface:

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



- Select **Enable** from the **Allow IP Call** list.

The screenshot shows a configuration window for the Matrix SPARSH VP110. The 'Allow IP Call' dropdown menu is open, showing 'Enabled' as the selected option. Other settings visible include 'Accept SIP Trust Server Only' (Disabled), 'IP Direct Auto Answer' (Enabled), 'Call List Show Number' (Disabled), 'Voice Mail Tone' (Enabled), 'DHCP Hostname' (Matrix SPARSH VP110), 'Reboot in Talking' (Enabled), 'Hide Feature Access Codes' (Disabled), and 'Display Method on Dialing' (User Name). 'Confirm' and 'Cancel' buttons are at the bottom.

- Click **Confirm** to save the change.
- Then click on **Account->Register**.
- Select **Enabled** from the **Line Active** list.
- Select **Disabled** from the **Register** list.
- Configure the **User Name**. You may configure other parameters if required.
- Configure parameters of the **SIP Server 1** in the corresponding fields

The screenshot shows the 'Account' tab in the Matrix SPARSH VP110 configuration interface. The 'Register' sub-tab is selected. The 'Line Active' dropdown is set to 'Enabled' and the 'Register' dropdown is set to 'Disabled'. Other fields include 'Label' (3301), 'Display Name' (3301), 'Register Name' (3301), 'User Name' (3301), and 'Password' (masked with dots). The 'Status' tab is also visible.

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

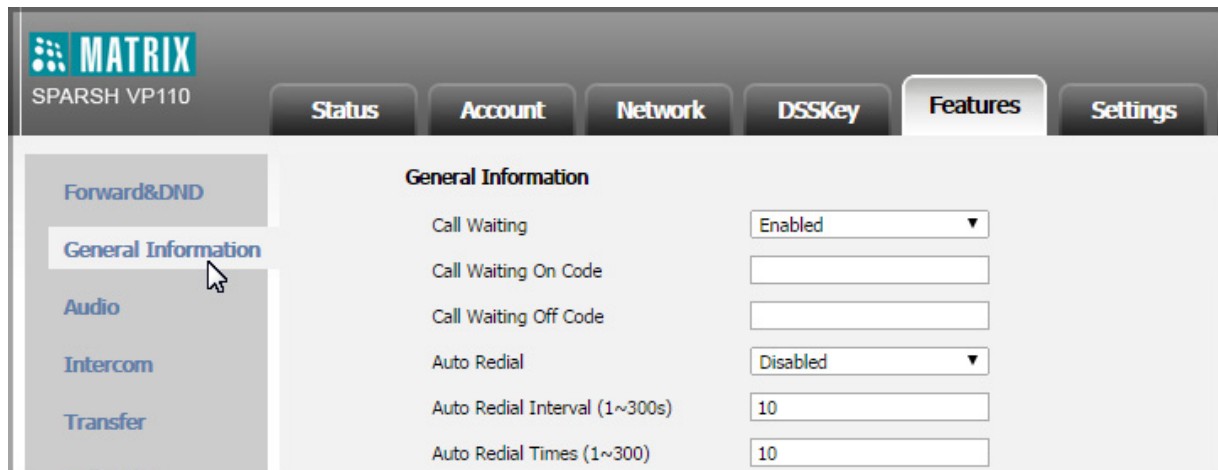
Making Peer-to-Peer (P2P) calls

- When you dial out any number (which is not an IP address) using the phone dial pad, the number will be dialed out as per the SIP Server 1 address configured. This way outgoing call without registration may be achieved.

- If you dial out an IP address using the phone dial pad, the IP address will be dialed out along with the destination port as 5060 (irrespective of the Port programmed for the SIP Server 1 parameter).

To configure auto answer for incoming Peer-to-Peer (P2P) calls via web user interface:

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



MATRIX SPARSH VP110

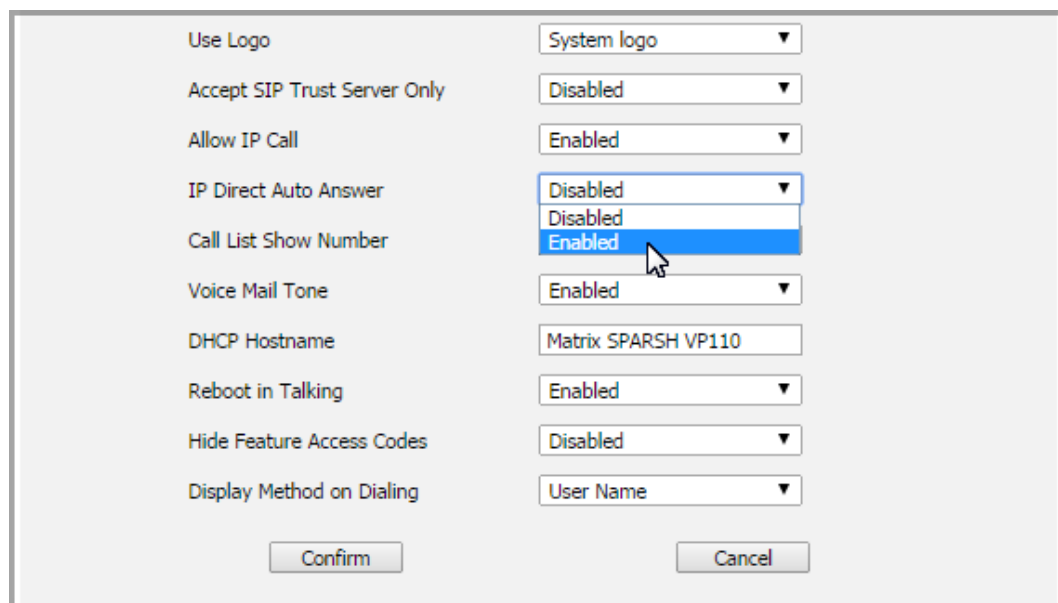
Status Account Network DSSKey Features Settings

Forward&DND General Information Audio Intercom Transfer

General Information

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10

- Select **Enable** from the **IP Direct Auto Answer** list.



Use Logo	System logo
Accept SIP Trust Server Only	Disabled
Allow IP Call	Enabled
IP Direct Auto Answer	Disabled Disabled Enabled
Call List Show Number	
Voice Mail Tone	Enabled
DHCP Hostname	Matrix SPARSH VP110
Reboot in Talking	Enabled
Hide Feature Access Codes	Disabled
Display Method on Dialing	User Name

Confirm Cancel

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

The SPARSH VP110 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
- Redialing Numbers
- Recent Call In Dialing
- Auto Answer
- Auto Redial
- Call Completion
- ReCall
- Call Mute
- Call Hold/Resume
- Do Not Disturb (DND)
- Call Forward
- Call Transfer
- Call Waiting
- Conference
- Call Pickup
- Anonymous Call
- Anonymous Call Rejection

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”](#).

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 active call is visible on the LCD screen. In the figure below, the call has lasted 26 seconds.




To place a call using the handset:

- Pick up the handset.
- Enter the desired number using the keypad.
- Press **OK** , , or the **Send** soft key.


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 [“Key as Send”](#).



- You can also dial using the SIP URI or IP address. To obtain the IP address of the phone, press the **OK**  key. The maximum SIP URI or IP address length is 32 characters. For example, SIP URI: 2210@sip.com, IP: 192.168.1.15.
- Your phone may not support direct IP dialing. Contact your network 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  to obtain a dial tone.




Enter the desired number using the keypad.

Press **OK** , , or the **Send** soft key.

- With the handset on-hook, enter the desired number using the keypad.

Press , **OK** , , or the **Send** soft key.

To place a call using the headset:



- With the optional headset connected, press  to activate the headset mode.
- Enter the desired number using the keypad.
- Press **OK** , , or the **Send** soft key.





To permanently enable your headset, refer [“Headset Prior”](#).

To place multiple calls:

You can have more than one call on your IP phone. To place a new call during an active call:

- Press the **Hold** soft key to place the original call on hold.
- Press the **NewCall** soft key.
- Enter the desired number using the keypad.
- Press **OK** , , or the **Send** soft key.

You can press  or  to switch between the calls, and then press the **Resume** soft key to retrieve the desired 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”](#).
- 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 LCD 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  .
- With the handset on-hook and headset mode deactivated, press the **Answer** soft key.

To answer a call using the headset:

Do one of the following:


- Press  .
- With the headset mode activated, press the **Answer** soft key.


Answering When in Another Call

If you have an active call, and an incoming call arrives on the phone, do one of the following:

- Press the **Answer** soft key.

The incoming call is answered and the original call is placed on hold.

- Press  to access the new call.

Press **OK**  or the **Answer** soft key.

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, press the **Cancel** soft key or hang up the handset.
- If you are using the headset, press the **Cancel** soft key.
- If you are using the speakerphone, press  or the **Cancel** soft key.



*To end a call placed on hold, you can press the **Cancel** soft key to end the call directly, or press the **Resume** soft key to resume the call before ending it.*





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.
- Press  or  to select the desired entry, and then press  or the **Send** soft key.

Recent Call In Dialing

You can view the placed calls list when the phone is in the pre-dialing screen. To do this, you should enable recent call in dialing in advance.

To enable recent call in dialing via web user interface:

- Click on **Directory->Setting**.
- Select **Enabled** from the **Recent Call In Dialing** list.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', and 'Directory'. The left sidebar has 'Local Directory', 'Remote Phone Book', 'Phone Call Info', 'Multicast IP', and 'Setting'. The main content area is titled 'Directory' and contains two sections: 'Directory' and 'Search Source List In Dialing'. Each section has 'Disabled' and 'Enabled' lists with arrows for moving items between them. At the bottom, the 'Recent Call In Dialing' dropdown menu is open, showing 'Disabled', 'Disabled', and 'Enabled' options. The 'Enabled' option is highlighted. Below the dropdown are 'Confirm' and 'Cancel' buttons. On the right, a 'NOTE' section provides information about the 'Directory', 'Search Source', and 'Recent Call In' features.



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

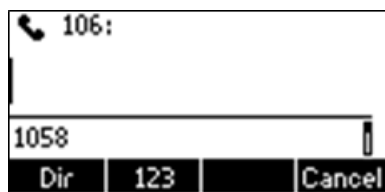


This parameter is configurable via web user interface only.

To view placed calls list when the phone is in the pre-dialing screen:

- Pickup the handset or press the speakerphone.

The LCD screen displays the placed calls list. You can press  or  to select the desired entry and then place a call to the contact.



Auto Answer

Auto answer allows IP phones to automatically answer an incoming call. IP phones will not automatically answer the incoming call during a call even if auto answer is enabled. Auto-Answer delay defines a period of delay time before the IP phone automatically answers incoming calls.

Procedure

Auto answer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure auto answer. For more information, refer "Appendix D - Configuration Parameters" .
	<MAC>.cfg	Specify a period of delay time for auto answer. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure auto answer. Navigate to: http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0 Specify a period of delay time for auto answer. Navigate to: http://<phoneIPAddress>servlet?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 value from the **Auto Answer** list.

The screenshot shows the MATRIX SPARSH VP110 web interface. The 'Account' tab is selected. On the left, there is a sidebar with 'Register', 'Basic', 'Codec', and 'Advanced' options. The 'Basic' section is active, showing various settings. The 'Auto Answer' setting is currently set to 'Disabled', and its dropdown menu is open, showing three options: 'Disabled', 'Disabled', and 'Enabled'. A mouse cursor is pointing at the 'Enabled' option. Other settings include 'Local Anonymous' (Off), 'Local Anonymous Rejection' (Off), 'Send Anonymous Code' (Off Code), 'On Code', 'Off Code', 'Send Anonymous Rejection Code' (Off Code), 'On Code', 'Off Code', 'Missed Call Log' (Enabled), and 'Confirm' and 'Cancel' buttons.

- 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 via phone user interface:

- Press **Menu->Features->Auto Answer**.
- Press the **Enter** soft key.
- Press **▲** or **▼** to select **Status**.
- Press **◀** or **▶**, or the **Switch** soft key to select **Enable** from the **Status** field.

The screenshot shows the phone user interface for the 'Auto Answer' screen. It displays '2. Status:' and 'Enabled'. At the bottom, there are three soft keys: 'Back', 'Switch', and 'Save'.

- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

The  icon appears on the LCD screen.



Auto answer is only applicable when there is no other call in progress on the phone.

Auto Redial

You can enable auto redial 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?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.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The Features tab is selected. On the left, a sidebar lists menu items: Forward&DND, General Information (selected), Audio, Intercom, Transfer, and Call Pickup. The main content area is titled 'General Information' and contains the following settings:

Setting	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Enabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#

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

To configure auto redial via phone user interface:

- Press **Menu->Features->Auto Redial**.
- Press ◀ or ▶, or the **Switch** soft key to select **Enable** from the **Auto Redial** field.



- Enter the desired time in the **Redial Interval** field.

The default time interval is 10s.

- Enter the desired times in the **Redial Times** field.

The default times are 10.

- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

To use auto redial:

When the called party is busy, the following prompt will appear on the LCD screen of the phone:



- Press the **OK** soft key to activate auto redial.

The following prompt will appear on the LCD screen of the phone:



- Wait for a period of time or press the **OK** soft key 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 users 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?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.

MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

SMS

Action URL

Power LED

Notification Popups

General Information

Call Waiting Enabled

Call Waiting On Code

Call Waiting Off Code

Auto Redial Disabled

Auto Redial Interval (1~300s) 10

Auto Redial Times (1~300) 10

Key As Send #

Reserve # in User Name Enabled

Hotline Number

Hotline Delay(0~10s) 4

Busy Tone Delay (Seconds) 0

Return Code When Refuse 486 (Busy Here)

Return Code When DND 480 (Temporarily Unava

Call Completion Disabled

Feature Key Synchronization Disabled

Time-Out for Dial-Now Rule 1

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

To configure call completion via phone user interface:

- Press **Menu->Features->Call Completion**.
- Press **◀** or **▶**, or the **Switch** soft key to select **Enable** from the **Call Completion** field.

Call Completion

1. Call Completion:

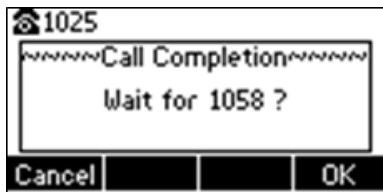
Enable

Back Switch Save

- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

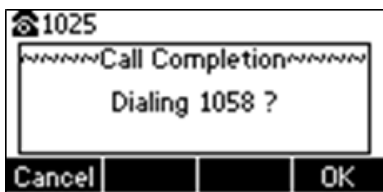
To use call completion:

When the called party is busy, the following prompt will appear on the LCD screen of the phone:



- Press the **OK** soft key, the phone returns to the idle screen and call completion is activated.

When the called party becomes idle, the following prompt appears on the LCD screen of the phone:



- Press the **OK** soft key to redial the number.



Call completion is not available on all servers. IP phones support 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.

ReCall

ReCall, also known as last call return, allows users to place a call back to the last caller. ReCall is implemented on IP phones using a ReCall key.

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: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0

To configure a ReCall key via web user interface:

- Click on **DSSKey->Programmable Key**.
- Select the desired programmable key.
- Select **ReCall** from the **Type** list.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	N/A	N/A			
SoftKey 3	Speed Dial	N/A			
SoftKey 4	Direct Pickup	N/A			
SoftKey 5	Group Pickup	N/A			
SoftKey 6	Intercom	N/A			
SoftKey 7	Prefix	N/A			
SoftKey 8	Local Group	N/A			
SoftKey 9	XML Group	N/A			
SoftKey 10	XML Browser	N/A			
SoftKey 11	History	Local History			
SoftKey 12	Menu	N/A			
SoftKey 13	Forward	N/A			
SoftKey 14	DND	N/A			
SoftKey 15	ReCall	N/A			
SoftKey 16	SMS	N/A			
SoftKey 17	New SMS	N/A			
SoftKey 18	XML Directory	N/A			
SoftKey 19	Status	N/A			
SoftKey 20	Multicast Paging	N/A			
SoftKey 21	Local Directory	N/A			
SoftKey 22	N/A	N/A			

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



A call return key is configurable via web user interface only.

Call Mute

You can mute the microphone of the active audio device during an active call, and then the other party cannot hear you.

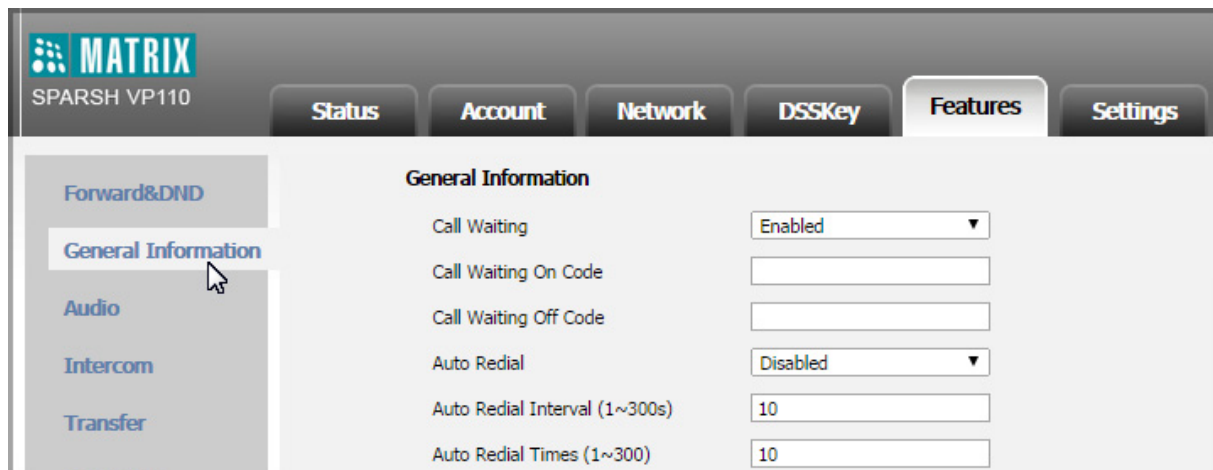
Procedure

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

Configuration File	<MAC>.cfg	Configure call mute. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Configure call mute. Navigate to: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>
	Phone User Interface	Configure call mute.

To configure allow mute via web user interface:

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




- Select the desired value from the **Allow Mute** list.

DTMF Repetition	3 ▼
Play Hold Tone	Enabled ▼
Play Hold Tone Delay	30
Allow Mute	Enabled ▼
Dual-Headset	Enabled ▼
Auto-Answer Delay(1~4s)	1
Enable auto answer tone	Enabled ▼
Headset Prior	Disabled ▼

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


To mute a call:

- Press  during an active call.

The LCD screen indicates that the call is on mute.



To un-mute a call:

- Press  again to un-mute the call.



*To mute an active call, you must set the **Allow Mute** parameter as **Enabled**. This parameter is accessible from the web user interface by clicking **Features -> General Information**.*

Call Hold and Resume

You can place an active call on hold. Only one active call can be in progress at any time. Other calls can be made and received while placing the original call on hold. When you place a call on hold, your IP PBX may play Music on Hold (MoH) to the other party while waiting.

For advanced users:

When a call is placed on hold, the IP phones 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 phones support two call hold methods, one is RFC 3264, which sets the “a” (media attribute) in the SDP to sendonly, recvonly or inactive (e.g., a=sendonly). The other is RFC 2543, which sets the “c” (connection addresses for the media streams) in the SDP to zero (e.g., c=0.0.0.0). Call hold tone allows IP phones to play a warning tone at regular intervals when there is a call on hold. The warning tone is played through the speakerphone.

IP phones also support Music on Hold (MoH) feature. MoH is the business practice of playing recorded music to fill the silence that would be heard by the party who has been placed on hold. To use this feature, specify a SIP URI pointing to an MoH server account. When a call is placed on hold, the IP phone will send an INVITE message to the specified MoH server account according to the SIP URI. The MoH server account automatically responds to the INVITE message and immediately plays audio from some source located anywhere (LAN, Internet) to the held party.

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: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?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.

MATRIX
SPARSH VP110

Status **Account** **Network** **DSSKey** **Features** **Settings**

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

SMS

Action URL

Power LED

Notification Popups

General Information

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	480 (Temporarily Unava
Call Completion	Disabled
Feature Key Synchronization	Disabled
Time-Out for Dial-Now Rule	1
RFC 2543 Hold	Enabled
Use Outbound Proxy In Dialog	Enabled
180 Ring Workaround	Enabled

- 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.

DTMF Repetition	3
Play Hold Tone	Enabled
Play Hold Tone Delay	30
Allow Mute	Enabled
Dual-Headset	Disabled
Auto-Answer Delay(1~4s)	1

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



To configure MoH, refer [“Music on Hold”](#).

To place a call on hold:

- Press the **Hold** soft key during a call.

The LCD screen indicates that the call is on hold.





By default, 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 the **Resume** soft key.

Multiple Calls on Hold:

If multiple calls are placed on hold:

- Press  or  to switch between the calls, and then press the **Resume** soft key to retrieve the desired call.

A numbered prompt appears on the LCD screen, for example “1/2”, indicating that this is the first call out of two calls.

Do Not Disturb

You can use DND to reject incoming calls automatically on the phone. Callers will receive a busy message. You can receive incoming calls from authorized numbers when DND is enabled.

A user can activate or deactivate DND using the DND key or DND soft key.

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 LCD screen displays the received return code.

Procedure

DND can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Assign a DND key.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p> <p>Configure DND in the phone mode.</p> <p>Specify the return code and the reason of the SIP response message when DND is enabled.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
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Local	Web User Interface	<p>Assign a DND key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</p> <p>Configure DND.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-forward&q=load">http://<phoneIPAddress>/servlet?p=features-forward&q=load</p> <p>Specify the return code and the reason of the SIP response message when DND is enabled.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	Phone User Interface	Configure DND.

To configure a DND key via web user interface:

- Click on **DSS Key ->Programmable Key**.
- In the desired DSS key field, select **DND** from the **Type** list.

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

To configure DND feature via web user interface:

- Click on **Features->Forward&DND**.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Features' tab is selected, and the left sidebar shows 'Forward&DND' as the active menu item. The main content area is titled 'Forward' and contains the following settings:

Setting	Value
Forward Emergency	Disabled
Forward Authorized Numbers	
Always Forward	On (selected), Off
Target	
On Code	
Off Code	
Busy Forward	On, Off (selected)
Target	

- Click the desired option in **DND Status**.

The screenshot shows a 'DND' configuration dialog box with a question mark icon. It contains the following settings:

Setting	Value
DND Emergency	Disabled
DND Authorized Numbers	
DND Status	On (selected), Off
DND On Code	
DND Off Code	

At the bottom of the dialog are 'Confirm' and 'Cancel' buttons.

- (Optional.) Enter the DND on code in the **DND On Code** field.
- (Optional.) Enter the DND off code in the **DND Off Code** field.
- 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.

MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND
General Information
Audio
Intercom
Transfer
Call Pickup
Remote Control
Phone Lock
SMS
Action URL
Power LED
Notification Popups

General Information

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	480 (Temporarily Unavailable) 404 (Not Found) 480 (Temporarily Unavailable) 486 (Busy Here) 603 (Decline)
Call Completion	
Feature Key Synchronization	
Time-Out for Dial-Now Rule	1

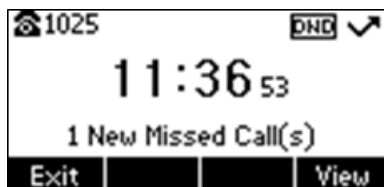
- Click **Confirm** to save the change.

To activate DND via phone user interface:

- Press the **DND** soft key when the phone is idle.

The icon on the idle screen indicates that DND is enabled.

Incoming calls will be rejected automatically and "n New Missed Call(s)" ("n" indicates the number of the missed calls) will appear on the LCD screen.





- The prompt message displays only if Missed Call Log for the line is enabled. **Missed Call Log** is configurable via web user interface at the path **Account->Basic**.
- Do Not Disturb is local to the phone, and may be overridden by the server settings. For more information, contact your ITSP.

To configure the DND authorized numbers via web user interface:

- Click on **Features->Forward&DND**.
- Select **Enabled** from the **DND Emergency** list.
- Enter the numbers in the **DND Authorized Numbers** field.

For multiple numbers, enter a comma between the numbers.

MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND

General Information
Audio
Intercom
Transfer
Call Pickup
Remote Control
Phone Lock
SMS
Action URL
Power LED
Notification Popups

Forward

Forward Emergency: Disabled

Forward Authorized Numbers:

Always Forward

On: ☐ Off: ☒

Target:

On Code:

Off Code:

Busy Forward

On: ☐ Off: ☒

Target:

On Code:

Off Code:

No Answer Forward

On: ☐ Off: ☒

After Ring Time(0~120s): 12

Target:

On Code:

Off Code:

DND

DND Emergency: Enabled

DND Authorized Numbers: 123

DND Status: On ☒ Off ☐

DND On Code:

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

When DND is enabled on the phone, the phone can still receive incoming calls from the numbers specified in the **DND Authorized Numbers** field.



DND authorized number is configurable via web user interface only.

Call Forward

You can configure your phone to forward incoming calls to another party by the static forwarding. You can also forward calls while your phone is ringing, refer to the dynamic forwarding.

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 period of time.

For advanced users:

Call forward allows users to redirect an incoming call to a third party. IP phones redirect 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.

IP phones support 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 phone of a call's history. For example, when a phone has been set to enable call forward, the Diversion/History-info header allows the receiving phone to indicate who the call was from, and from which phone number it was forwarded.

Forward International

Forward international allows users to forward an incoming call to an international telephone number. This feature is enabled by default.

Procedure

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

Configuration File	<MAC>.cfg	<p>Configure call forward in phone mode.</p> <p>Configure diversion/history-info feature.</p> <p>Configure forward international.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure call forward.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-forward&q=load">http://<phoneIPAddress>/servlet?p=features-forward&q=load</p> <p>Configure diversion/history-info feature.</p> <p>Configure forward international.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	Phone User Interface	Configure call forward.

To configure call forward via web user interface:

- Click on **Features->Forward& DND**.
- Select **Enabled** from the pull-down list of **Forward Emergency**.
- Configures the **Authorized Numbers** not to be forwarded even if call forward feature is enabled. For multiple numbers, enter a comma between every two numbers.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Features' tab is selected, and the 'Forward&DND' sub-tab is active. The 'Forward' section contains the following fields:

- Forward Emergency:** A dropdown menu set to 'Enabled'.
- Forward Authorized Numbers:** A text input field containing '1029, 1030'.
- Always Forward:** Radio buttons for 'On' and 'Off', with 'Off' selected.
- Target:** An empty text input field.
- On Code:** An empty text input field.
- Off Code:** An empty text input field.

- Click **Confirm** to accept the change.
- In the **Forward block**, click the desired option in **Always Forward/Busy Forward/No Answer Forward**.

MATRIX SPARSH VP110

Navigation tabs: Status, Account, Network, DSSKey, **Features**, Settings

Forward&DND

Forward

Forward Emergency: Enabled

Forward Authorized Numbers: 1029, 1030

Always Forward

On: ☒ On ☐ Off

Target:

On Code:

Off Code:

Busy Forward

On: ☐ On ☒ Off

Target:

On Code:

Off Code:

No Answer Forward

On: ☐ On ☒ Off

After Ring Time(0~120s): 12

Target:

On Code:

Off Code:

- Enter the destination number you want to forward in the **Target** field.
- (Optional.) Enter the on code and off code in the **On Code** and **Off Code** fields.
- Select the ring time to wait before forwarding from the **After Ring Time(0~120s)** list (only for the *No answer forward*).
- Click **Confirm** to save the change.

To configure Diversion/History-Info feature via web user interface:

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

MATRIX SPARSH VP110

Navigation tabs: Status, Account, Network, DSSKey, **Features**, Settings

Forward&DND

General Information

Call Waiting: Enabled

Call Waiting On Code:

Call Waiting Off Code:

Auto Redial: Disabled

Auto Redial Interval (1~300s): 10

Auto Redial Times (1~300): 10

- Select the desired value from the **Diversion/History-Info** list.

Fwd International	Enabled
Diversion/History-Info	Enabled
Allow Trans Exist Call	Disabled
Auto-Logout Time(1~1000min)	1000
Call Number Filter	-
Use Logo	System logo
Accept SIP Trust Server Only	Disabled

- 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.

DTMF Replace Tran	Disabled
Tran Send DTMF	
Send Pound Key	Disabled
Fwd International	Enabled
Diversion/History-Info	Disabled
Allow Trans Exist Call	Enabled
Auto-Logout Time(1~1000min)	1000
Call Number Filter	-







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



By default, the Fwd International prefix value is "00" and it is not configurable.


To enable call forward via phone user interface:

- Press **Menu->Features->Call Forward**.
- Press or to select the desired forwarding type, and then press the **Enter** soft key.
- Depending on your selection:
 - If you select **Always Forward**:
 - Press or , or the **Switch** soft key to select **Enable** from the **Always** field.
 - Enter the destination number you want to forward all incoming calls to in the **Forward to** field.

- (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.
- If you select **Busy Forward**:
 - Press  or , or the **Switch** soft key to select **Enable** from the **Busy** field.
 - Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward to** field.
 - (Optional.) Enter the busy forward on code or off code respectively in the **On Code** or **Off Code** field.
- If you select **No Answer Forward**:
 - Press  or , or the **Switch** soft key to select **Enable** from the **No Answer** field.
 - Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.
 - Press  or , or the **Switch** soft key to select the ring time to wait before forwarding from the **After Ring Time** field.

The default ring time is 12 seconds.

 - (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.






The  icon on the idle screen indicates call forward is enabled.



- You can also enter the SIP URL or IP address in the Forward to field. For more information on using the SIP URL or IP address, refer ["Placing Calls"](#).
- 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 between servers. For more information, contact your ITSP.

To disable call forward in phone mode:

Do one of the following:

- Press  when the phone is idle.
- Press **Menu->Features->Call Forward**.
- Press  or  to select the desired forwarding type, then press the **Enter** soft key.
- Press  or , or the **Switch** soft key to select **Disable** to disable the call forward.
- Press the **Save** soft key 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.



The screenshot shows a rectangular LCD screen with a black border. At the top, it says "Forward to:". Below that, the number "1058" is displayed in a large font. Underneath the number, there is a horizontal line. Below the line, the number "1058" is displayed in a smaller font. At the bottom of the screen, there is a black bar with four white buttons: "Send", "123", "Delete", and "Cancel".

- Press **OK** , , or the **Send** soft key.

The incoming call is forwarded. The LCD screen prompts a call forward message.

Call Transfer

Call transfer enables IP phones to transfer an existing call to another party. IP phones support call transfer using the REFER method specified in RFC 3515 and offer 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 LCD screen.

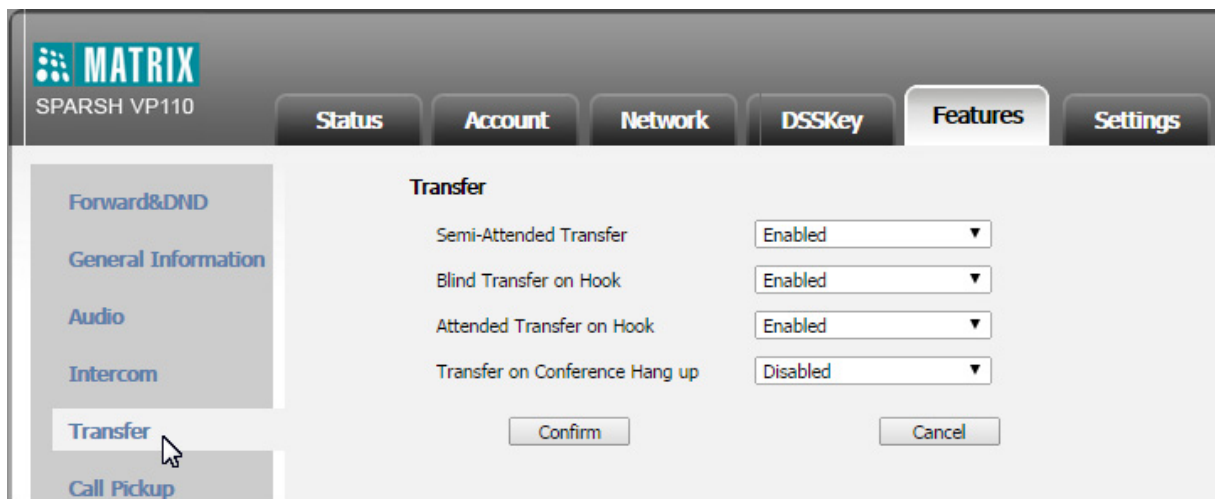
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=features-transfer&q=load">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 the **Semi-Attended Transfer**, **Blind Transfer On Hook** and **Semi-Attended Transfer On Hook** lists.
- Click **Confirm** to save the change.

To perform a blind transfer:




- Press  or the **Tran** soft key during a call.
- Enter the number you want to transfer the call to.
- Press  or the **Tran** soft key to complete call transfer.

Then the call is connected to the number to which you are transferring.







To perform a semi-attended transfer:

- Press  or the **Tran** soft key during a call.
- Do one of the following:
 - Enter the number you want to transfer the call to.
 - Press the **Dir** soft key, and then select **Local Directory**. Select the desired group, and search for the contact (Directory should be configured in advance. Refer [“Directory”](#) for more information.).
 - Press the **Dir** soft key, and then select **History**. Select the desired list and use  or  to select the entry (Directory should be configured in advance. Refer [“Directory”](#) for more information.).
 - Press the **Dir** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and remote phone book should be configured in advance. Refer [“Directory”](#) and

[“Remote Phone Book”](#) for more information.).

- Press **OK**  or  to dial out.
- Press  or the **Tran** soft key to complete call transfer when receiving the ring-back.

To perform an attended transfer:

- Press  or the **Tran** soft key during a call.
- Do one of the following:
 - Enter the number you want to transfer the call to.
 - Press the **Dir** soft key, and then select **Local Directory**. Select the desired group, and search for the contact (Directory should be configured in advance. Refer [“Directory”](#) for more information.).
 - Press the **Dir** soft key, and then select **History**. Select the desired list and use  or  to select the entry (Directory should be configured in advance. Refer [“Directory”](#) for more information.).
 - Press the **Dir** soft key, and then select **Remote Phone Book**. Select the desired group and search for the contact (Directory and remote phone book should be configured in advance. Refer [“Directory”](#) and [“Remote Phone Book”](#) for more information.).
- Press **OK**  or  to dial out.
- After the party answers the call, press  or the **Tran** soft key to complete call transfer.

If you are using a handset, call transfer can be completed by hanging up the handset.

You can cancel call transfer before the call is connected by pressing the **Cancel** soft key.

Call Waiting

You can enable or disable call waiting on the phone. If call waiting is enabled, you can receive another call when there is an active call on the phone. Otherwise, another incoming call is automatically rejected by the phone with a busy message when there is an active call on the phone. You can also enable or disable the phone to play a warning tone when receiving another call.

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?p=features-general&q=load Configure call waiting tone. Navigate to: http://<phoneIPAddress>/servlet?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.
- (Optional.) Enter the call waiting on code in the **Call Waiting On Code** field.

- (Optional.) Enter the call waiting off code in the **Call Waiting Off Code** field.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features' (selected), and 'Settings'. On the left, a sidebar lists 'Forward&DND', 'General Information' (selected), 'Audio', 'Intercom', 'Transfer', and 'Call Pickup'. The main content area is titled 'General Information' and contains the following settings:

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#

- 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.

The screenshot shows the MATRIX SPARSH VP110 web interface with the 'Features' tab selected. The left sidebar now has 'Audio' selected under the 'General Information' category. The main content area is titled 'Audio Settings' and contains the following settings:

Call Waiting Tone	Enabled
Key Tone	Enabled
Send Tone	Enabled
Redial Tone	
Ringer Device for Headset	Use Speaker

At the bottom of the 'Audio Settings' section are two buttons: 'Confirm' and 'Cancel'.

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

To configure call waiting via phone user interface:

- Press **Menu->Features->Call Waiting**.

The screenshot shows a phone's user interface. At the top, it says 'Call Waiting'. Below that, a menu item '1. Call Waiting:' is displayed with a value of 'Enable' and a right arrow. At the bottom, there are three buttons: 'Back', 'Switch', and 'Save'.

- Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **Call Waiting** field.
- Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **Play Tone** field.
- (Optional.)Enter the call waiting on code or off code respectively in the **On Code** or **Off Code** field.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.



Conference

You can create a conference with other parties using the phone's local conference. You can create a conference between an active call and a call on hold by pressing the **Conf** soft key.



Local Conference

The IP phone supports up to 3 parties (including yourself) in a conference call. This is the default method of conference called Local Conference.




To set up a local conference call:

- Place a call to the first party.
- When the first party answers the call, press the **NewCall** soft key to place a new call.
- The active call is placed on hold.
- Enter the number of the second party and press **OK** , , or the **Send** soft key.
- When the second party answers the call, press the **Conf** soft key again to join all parties in the conference.



- You can press  or , to see all parties in the conference.

You can do the following during a conference:

- Press the **Hold** soft key to place the conference on hold.
- Press the **Split** soft key to split the conference call into two individual calls on hold.
- Press the **Manage** soft key, and then press  or  to select the desired party:
 - Press the **FarMute** soft key to mute the party. The muted party can hear everyone, but no one can hear the muted party.
 - Press the **Remove** soft key to remove the party from the conference call.
 - Press the **New Call** soft key to place a new call.
 - Press the **Back** soft key to return to the previous screen.
- Press  to mute the conference call, all other participants can hear each other, but they cannot hear you.
- Press the **Cancel** soft key to drop the conference call.

Network Conference

You can use network conference on the IP phone to conduct a 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.

To use this feature, you must configure the network conference URI in advance.



- Network conference is not available on all servers. For more information, contact your ITSP.
- IP phones implement network conference using the REFER method specified in RFC 4579.

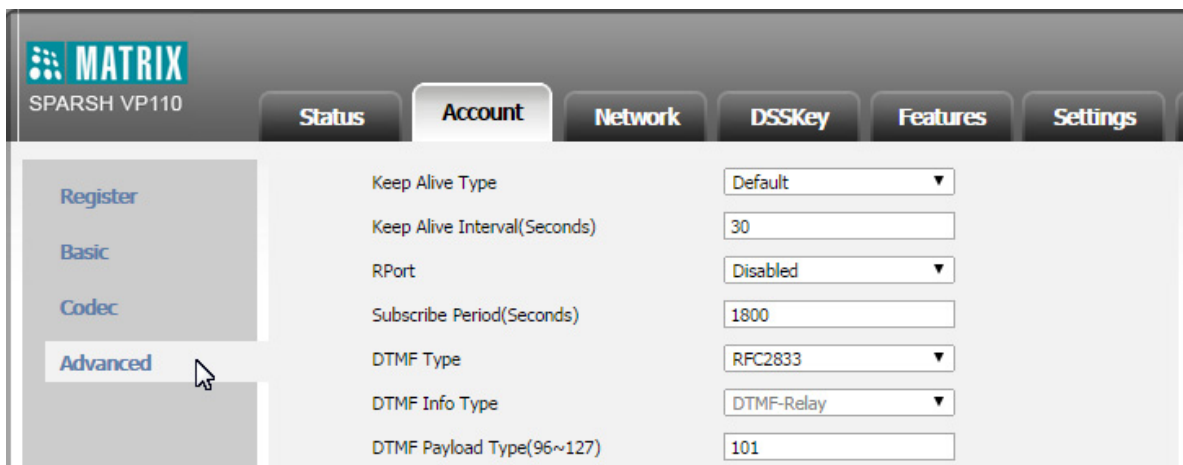
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

To configure network conference via web user interface:

- Click on **Account->Advanced**.







- Select **Network Conference** from the **Conference Type** list.

- Enter the conference URI (e.g., conference@example.com) in the **Conference URI** field.

SIP Registration Retry Timer(0~1800s)	<input type="text" value="30"/>
Conference Type	<input type="text" value="Network Conference"/>
Conference URI	<input type="text" value="conference@example.com"/>
Early Media	<input type="text" value="Disabled"/>
SIP Server Type	<input type="text" value="Default"/>
Music Server URI	<input type="text"/>

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

To set up a network conference call:

- Place a call to the first party.
- Press the **NewCall** soft key to place a new call.
- The active call is placed on hold.
- Enter the number of the second party and press **OK**  ,  , or the **Send** soft key.
- When the second party answers the call, press the **Conf** soft key to add the second party to the conference.
- Press the **NewCall** soft key to place a new call.
- The conference is placed on hold.
- Enter the number of the new party and then press **OK**  ,  , or the **Send** soft key.
- When the new party answers the call, press the **Conf** soft key to add the new party to the conference.
- 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

For a conference call, all parties drop the call when the conference initiator drops the conference call. For local conference, transfer on conference hang up allows the other two parties to remain connected when the conference initiator drops the conference call.

Procedure

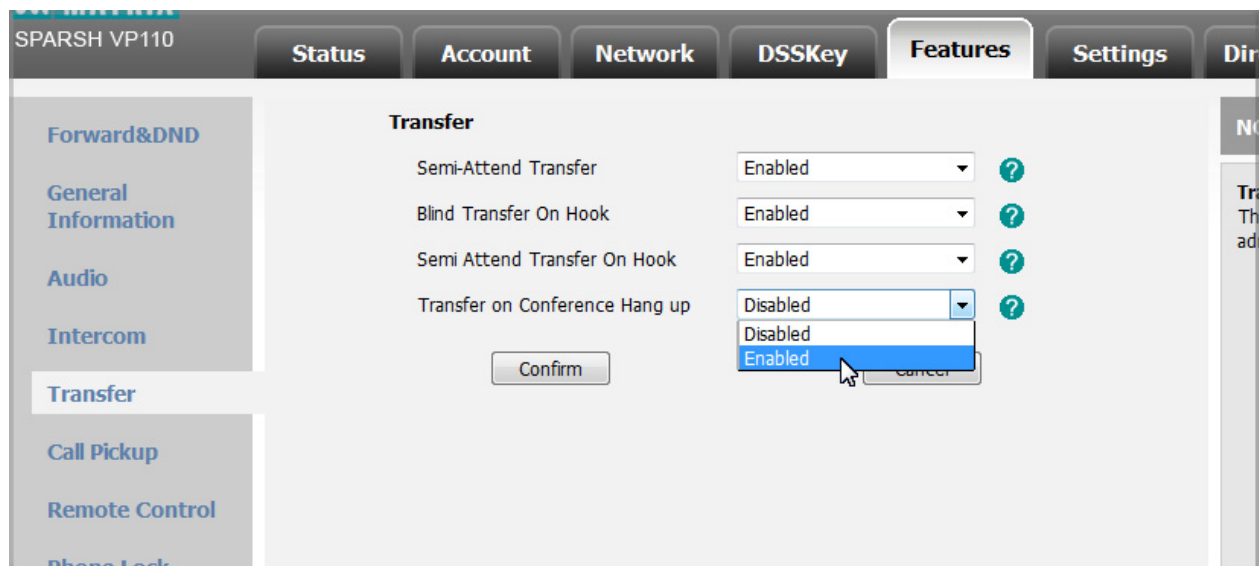
Transfer on conference hang up can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transfer on conference hang up. For more information, refer "Appendix D - Configuration Parameters" .
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Local	Web User Interface	<p>Configure the transfer on conference hang up.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-transfer&q=load">http://<phoneIPAddress>/servlet?p=features-transfer&q=load</p>
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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.

Call Pickup

You can use call pickup to answer someone else's incoming call on the phone. The IP phone supports directed call pickup and group call pickup. Directed call pickup is used for picking up a call that is ringing at a specific phone number. Group call pickup is used for picking up a call that is ringing at any phone number in the group. The pickup group should be predefined.

You can pickup an incoming call by using the **DPickup/GPickup** soft key. To use call pickup, you need to configure the call pickup code before-hand on a global basis via web user interface.



If there are many incoming calls at the same time, pressing the GPickup soft key on the phone will pick up the call that rings first.

Directed Call Pickup

Directed call pickup is used for picking up an incoming call on a specific extension. A user can pick up the incoming call using a directed pickup key or the **DPickup** soft key. This feature depends on support from a SIP server. For many SIP servers, directed call pickup requires a directed pickup code, which can be configured on a phone.



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 directed call pickup feature on a phone basis. For more information, refer "Appendix D - Configuration Parameters" .
	<MAC>.cfg	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: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Configure directed call pickup feature on a phone basis. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-callpickup&q=load">http://<phoneIPAddress>/servlet?p=features-callpickup&q=load
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To configure a directed call pickup key via web user interface:

- Click on **DSS Key->Programmable Key**.
- In the desired DSS key field, select **Direct Pickup** from the **Type** list.
- Enter the directed call pickup code followed by the specific extension in the **Value** field.

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Direct Pickup	N/A			
SoftKey 3	DND	N/A			
SoftKey 4	Menu	N/A			
Up	History	Local History			
Down	Direct Pickup	N/A			
Left	N/A	N/A			
Right	N/A	N/A			
OK	Status	N/A			

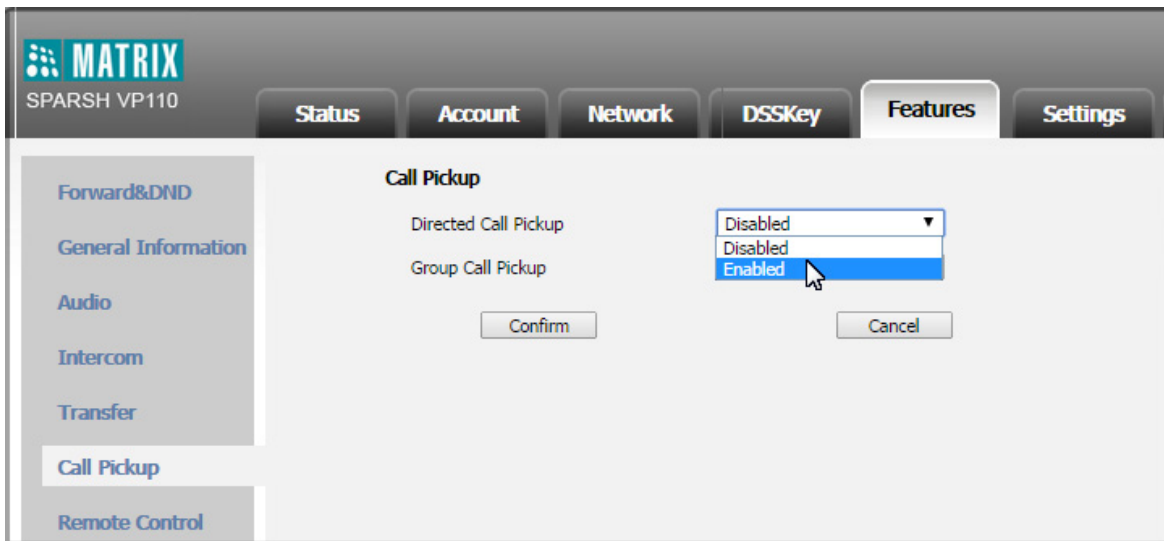
NOTE
Programmable Keys
Customizes the soft keys, navigation keys and function keys.

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

To enable the directed call pickup via web user interface:

- Click on **Features->Call Pickup**.

- Select **Enabled** from the **Directed Call Pickup** list.



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

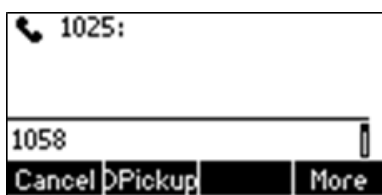
To configure the directed call pickup code on a global basis via web user interface:

- Click on **Account->Advanced**.
- Enter the directed call pickup code in the **Direct Call Pickup Code** field.
- Click **Confirm** to save the change.

To pick up a call directly:

- Pick up the handset.

The **DPickup** soft key appears on the LCD screen.



- Press the **DPickup** soft key on your phone when a phone receives an incoming call.
- Enter the phone number which receives an incoming call.
- Press the **DPickup** soft key again.

The call is answered on your phone.



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 incoming calls within a pre-defined group. If the group receives many incoming calls at once, the user will pick up the first incoming call, using a group pickup key or the GPickup soft key. 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.

Procedure

Group call pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure group call pickup feature on a phone basis. For more information, refer "Appendix D - Configuration Parameters" .
	<MAC>.cfg	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: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Configure group call pickup feature on a phone basis. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-callpickup&q=load">http://<phoneIPAddress>/servlet?p=features-callpickup&q=load

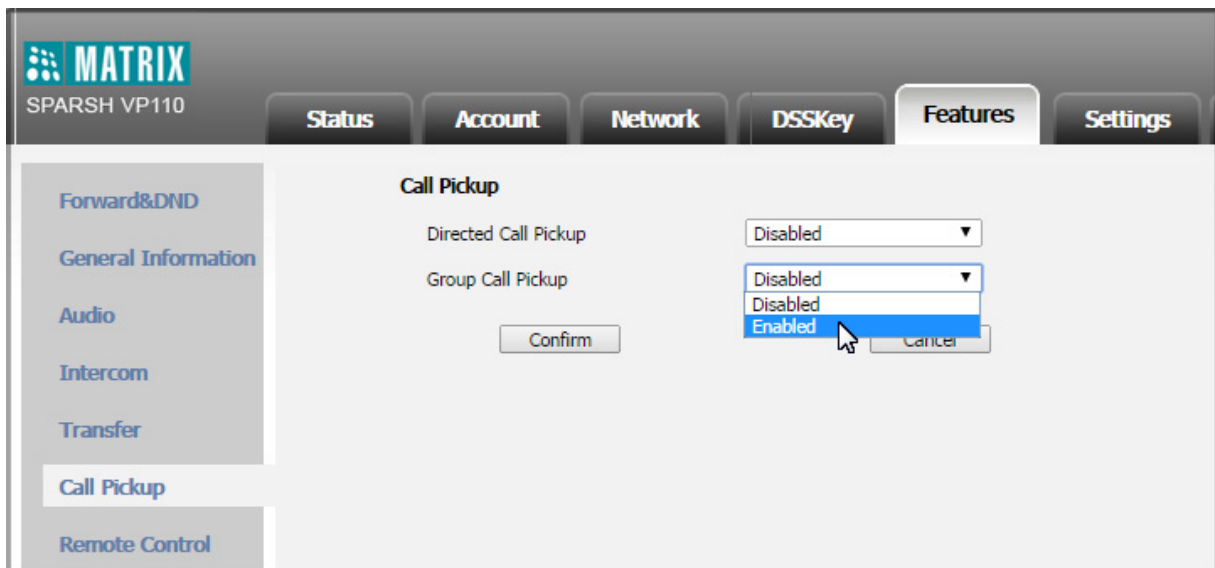
To configure a group call pickup key via web user interface:

- Click on **DSS Key-> Programmable Key**.
- In the desired DSS key field, select **Group Pickup** from the **Type** list.
- Enter the group call pickup code in the **Value** field.
- Click **Confirm** to save the change.

To enable the group call pickup via web user interface:

- Click on **Features->Call Pickup**.

- Select **Enabled** from the **Group Call Pickup** list.



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

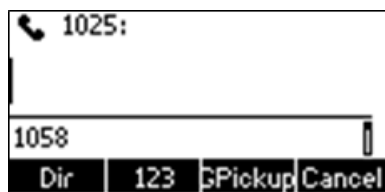
To configure the group call pickup code on a global basis via web user interface:

- Click on **Account->Advanced**.
- Enter the group call pickup code in the **Group Call Pickup Code** field.
- Click **Confirm** to save the change.

To pick up a call in the group:

- Pick up the handset.

The **GPickup** soft key appears on the LCD screen.



- Press the **GPickup** soft key on your phone when a phone in the group receives an incoming call.

The call is answered on your phone.



- *You can configure a programmable key as a group 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 group pickup key directly.*
- *The directed call pickup code and group call pickup code are predefined on the system server. Contact your ITSP for more information.*

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 want to call to consult some of the services, but don't want to be harassed. You can also configure the phone to send anonymous on/off code to the server to activate/deactivate anonymous call on the server side.



Anonymous call is not available on all servers. Contact your ITSP for the anonymous call on code and off code.

For advanced users:

Anonymous call allows the caller to conceal the identity information displayed on the callee's screen. The callee's phone LCD screen prompts an incoming call from anonymity.

Example of anonymous SIP header:

```
Via: SIP/2.0/UDP 10.2.8.183:5063;branch=z9hG4bK1535948896
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=128043702
To: <sip:1011@10.2.1.199>
Call-ID: 1773251036@10.2.8.183
CSeq: 1 INVITE
Contact: <sip:1012@10.2.8.183:5063>
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 VP110 2.72.0.1
Privacy: id
Supported: replaces
Allow-Events: talk,hold,conference,refer,check-sync
P-Preferred-Identity: <sip:1012@10.2.1.199>
Content-Length: 302
```

The anonymous call on code and anonymous call off code configured on IP phones are used to activate/deactivate the server-side anonymous call feature. They may vary on different servers. Send Anonymous Code feature allows IP phones to send anonymous on/off code to the server.

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?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 value from the **Local Anonymous** list.

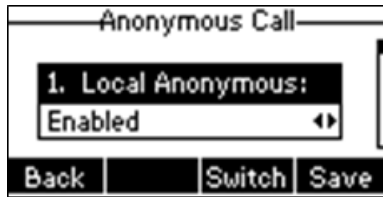
The screenshot shows the MATRIX SPARSH VP110 web interface. The 'Account' tab is selected, and the 'Basic' sub-tab is active. On the left sidebar, 'Register' is expanded, and 'Basic' is selected. The main configuration area shows several settings for anonymous calls. The 'Local Anonymous' dropdown is open, with 'On' selected. Other settings include 'Local Anonymous Rejection' (Off), 'Send Anonymous Code' (Off Code), 'On Code' (empty field), 'Off Code' (empty field), 'Send Anonymous Rejection Code' (Off Code), 'On Code' (empty field), 'Off Code' (empty field), 'Missed Call Log' (Enabled), and 'Auto Answer' (Disabled). There are 'Confirm' and 'Cancel' buttons at the bottom.

- Select the desired value from the **Send Anonymous Code** list.
- (Optional.) Enter the anonymous call on code in the **On Code** field.
- (Optional.) Enter the anonymous call off code in the **Off Code** field.
- Click **Confirm** to save the change.

To configure anonymous call via phone user interface:

- Press **Menu->Features->Anonymous Call**.

- Press ◀ or ▶ to select **Enabled** from the **Local Anonymous** field.



- (Optional.) Press ◀ or ▶ to select the desired value from the **Send Anony Code** field.
- (Optional.) Enter the anonymous call on code in the **On Code** field.
- (Optional.) Enter the anonymous call off code in the **Off Code** field.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

To place an anonymous call:

- Using the specific line on the phone to place a call to phone B.

The LCD screen of phone B prompts an incoming call from anonymity.



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 showing up.

The anonymous call rejection on code and anonymous call rejection off code configured on IP phones are used to activate/deactivate the server-side anonymous call rejection feature. They may vary on different servers.

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?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 value from the **Local Anonymous Rejection** list.

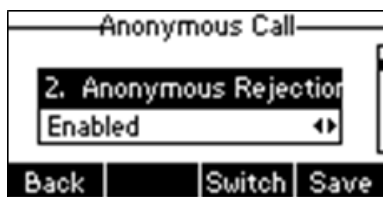
- Select the desired value from the **Send Anonymous Rejection code**.

The screenshot shows the MATRIX SPARSH VP110 web interface. The 'Account' tab is selected. In the 'Send Anonymous Code' dropdown, 'On' is highlighted. The interface includes a sidebar with 'Register', 'Basic', 'Codec', and 'Advanced' options. At the bottom, there are 'Confirm' and 'Cancel' buttons.

- (Optional.) Enter the anonymous call rejection on code in the **On Code** field.
- (Optional.) Enter the anonymous call rejection off code in the **Off Code** field.
- Click **Confirm** to save the change.

To configure anonymous call rejection via phone user interface:

- Press **Menu->Features->Anonymous Call**.
- Press \uparrow or \downarrow to scroll to the **Anonymous Rejection** field.
- Press \leftarrow or \rightarrow , or the **Switch** soft key to select **Enabled** from the **Anonymous Rejection** field.



- Press \uparrow or \downarrow to scroll to the **Send Rejection Code** field.
- Press \leftarrow or \rightarrow , or the **Switch** soft key to select desired value from the **Send Rejection Code** field.
- (Optional.) Enter the anonymous call rejection on code in the **Reject On Code** field.
- (Optional.) Enter the anonymous call rejection off code in the **Reject Off Code** field.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

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:

- Hot Desking
- Calling Line Identification Presentation
- Connected Line Identification Presentation
- DTMF
- Suppress DTMF Display
- Transfer via DTMF
- Intercom
- Multicast Paging
 - Sending RTP Stream
 - Receiving RTP Stream
- Music on Hold
- Messages
 - Short Message Service (SMS)
 - Voice Mail
 - Message Waiting Indicator (MWI)
- Web Server Type
- Softkey Layout
- Busy Tone Delay
- Return Code When Refuse
- Early Media
- 180 Ring Workaround
- Use Outbound Proxy in Dialog
- SIP Session Timer
- Session Timer
- Distinctive Ring Tones
- Tones
- Remote Phone Book

- Action URL
- Action URI
- Server Redundancy
 - SIP Server Domain Name Resolution
- LLDP
- VLAN
- Quality of Service
- Network Address Translation
- 802.1X Authentication
- IPv6 Support

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 not available on all servers. Contact your ITSP for more information.

Procedure

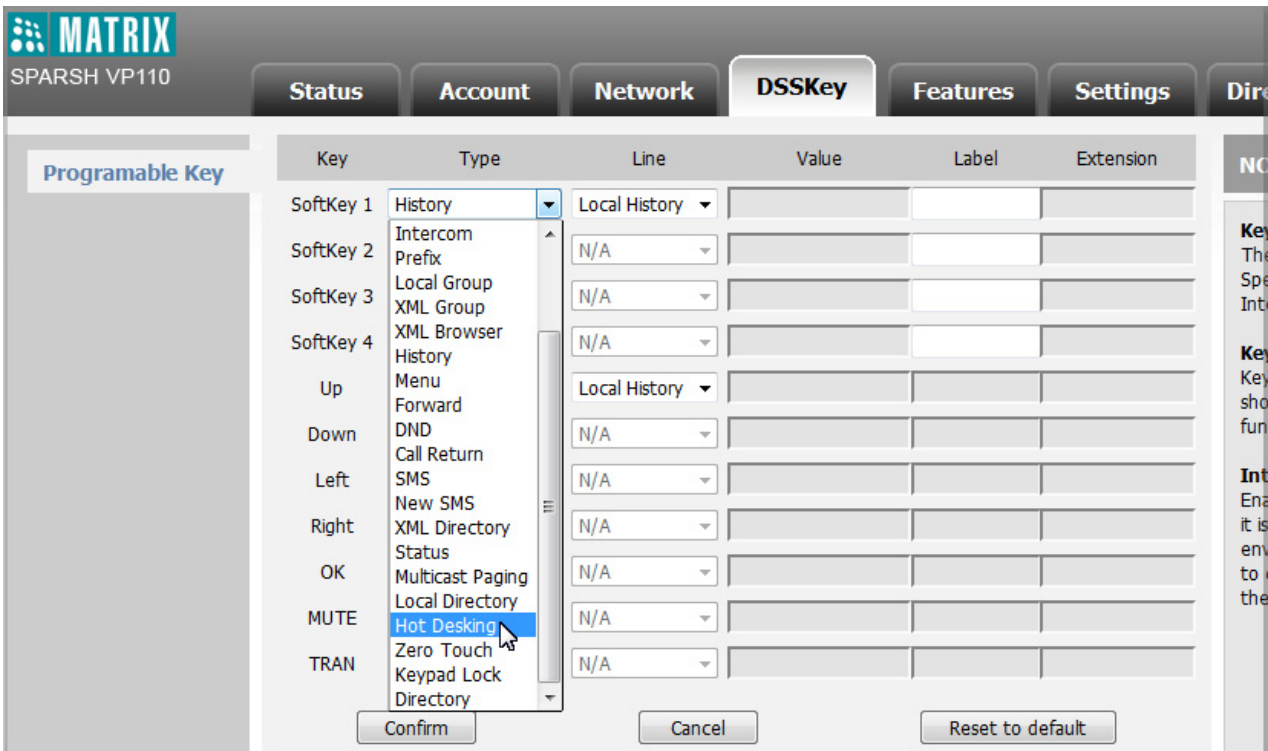
Hot desking key can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Assign a hot desking key. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Assign a hot desking key. Navigate to: <code>http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</code>

To configure a hot desking key via web user interface:

- Click on **DSSKey->Programmable Key**.
- Select the desired programmable key.


- Select **Hot Desking** from the **Type** list.



Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	Intercom	N/A			
SoftKey 3	Prefix	N/A			
SoftKey 4	Local Group	N/A			
	XML Group	N/A			
	XML Browser	N/A			
Up	History	Local History			
	Menu				
Down	Forward				
	DND	N/A			
Left	Call Return				
	SMS	N/A			
	New SMS				
Right	XML Directory	N/A			
	Status				
OK	Multicast Paging	N/A			
	Local Directory				
MUTE	Hot Desking	N/A			
	Zero Touch				
TRAN	Keypad Lock	N/A			
	Directory				

Buttons: Confirm, Cancel, Reset to default

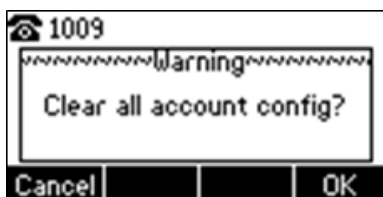
- Click **Confirm** to save the change.

 A hot desking key is configurable via web user interface only.

To use hot desking:

- Press the hot desking key when the phone is idle.

The LCD screen prompts “Clear all account config?”.



- Press the **OK** soft key, registration configurations (e.g., display name, register name, user name and password) of the account will be cleared immediately.

The login wizard will be shown as below:



- Enter the desired values in the **User Name** and **Password** fields respectively.
- Press the **Save** soft key to login or **Cancel** soft key to cancel.



You can use the Logon Wizard to configure the SIP account information of the IP phone from the phone user interface. Enabling the Logon Wizard (from the web interface, Features -> General Information) will prompt you to configure SIP account information using the logon wizard via phone user interface during the startup. To use the logon wizard, the SIP account configuration must be kept blank. This feature is similar to Hot Desking but pops-up automatically on the phone user interface by default when the phone reboots.

Calling Line Identification Presentation (CLIP)

Calling Line Identification Presentation (CLIP) allows IP phones to display the caller identity, derived from a SIP header contained in the INVITE message when receiving an incoming call. IP phones support deriving caller identity from three types of SIP header: From, P-Asserted-Identity and Remote-Party-ID. Identity presentation is based on the identity in the relevant SIP header.

If the caller has existed in the local directory, the local contact name assigned to the caller should be preferentially displayed and stored in the call log.

You may also refer [“Connected Line Identification Presentation \(COLP\)”](#).


Procedure

CLIP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the presentation of the caller identity. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure the presentation of the caller identity. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?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 value from the **Caller ID Source** list.



SPARSH VP110

Status

Account

Network

DSSKey

Features

Settings

Register

Basic

Codec

Advanced

Keep Alive Type

Keep Alive Interval(Seconds)

RPort

Subscribe Period(Seconds)

DTMF Type

DTMF Info Type

DTMF Payload Type(96~127)

Retransmission

Subscribe Register

Subscribe for MWI

MWI Subscription Period(Seconds)

Subscribe MWI To Voice Mail

Voice Mail

Voice Mail Display

Caller ID Source

Session Timer

Session Expires(90~7200s)

Session Refresher

Default

30

Disabled

1800

RFC2833

DTMF-Relay

101

Disabled

Disabled

Disabled

3600

Disabled

Enabled

FROM

FROM

PAI

PAI-FROM

RPID-PAI-FROM

PAI-RPID-FROM

RPID-FROM

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

Connected Line Identification Presentation (COLP)

Connected Line Identification Presentation (COLP) allows IP phones 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. Connected line identification presentation is also known as Called line identification presentation. 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. For more information, refer “Appendix D - Configuration Parameters” .
---------------------------	-----------	---

DTMF

DTMF (Dual Tone Multi-frequency), better 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 pressing the IP phone's keypad 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 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

Three methods of transmitting DTMF digits on SIP calls:

- **RFC 2833** -- DTMF digits are transmitted by RTP Events compliant to RFC2833.
- **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 support transmitting 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. For more information, refer "Appendix D - Configuration Parameters" .
	<MAC>.cfg	Configure the number of times for the IP phone to send the end RTP Event packet. 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: <code>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</code> Configure the number of times for the IP phone to send the end RTP Event packet. Navigate to: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>

To configure the method of transmitting DTMF digits via web user interface:

- Click on **Account-> Advanced**.
- 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.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features Settings

Register
Basic
Codec
Advanced

Keep Alive Type: Default
Keep Alive Interval(Seconds): 30
RPort: Disabled
Subscribe Period(Seconds): 1800
DTMF Type: RFC2833
DTMF Info Type: RFC2833
DTMF Payload Type(96~127): RFC2833
Retransmission: Disabled
Subscribe Register: Disabled
Subscribe for MWI: Disabled
MWI Subscription Period(Seconds): 3600
Subscribe MWI To Voice Mail: Disabled

- 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**.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features Settings

Forward&DND
General Information
Audio
Intercom
Transfer

General Information

Call Waiting: Enabled
Call Waiting On Code:
Call Waiting Off Code:
Auto Redial: Disabled
Auto Redial Interval (1~300s): 10
Auto Redial Times (1~300): 10

- Select the desired value (1-3) from the **DTMF Repetition** list.

PswLength	<input type="text"/>
PswDial	Disabled ▼
Save Call Log	Enabled ▼
Suppress DTMF Display	Disabled ▼
Suppress DTMF Display Delay	Disabled ▼
Play Local DTMF Tone	Enabled ▼
DTMF Repetition	3 ▼
Play Hold Tone	1 ▼
Play Hold Tone Delay	2 ▼
	3 ▼
	30 ▼
Allow Mute	Enabled ▼
Dual-Headset	Disabled ▼

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

Suppress DTMF Display

Suppress DTMF display allows IP phones to suppress the display of DTMF digits. DTMF digits are displayed as “*” on the LCD screen. Suppress DTMF display delay defines whether to display the DTMF digits for a short period of time before displaying as “*”.

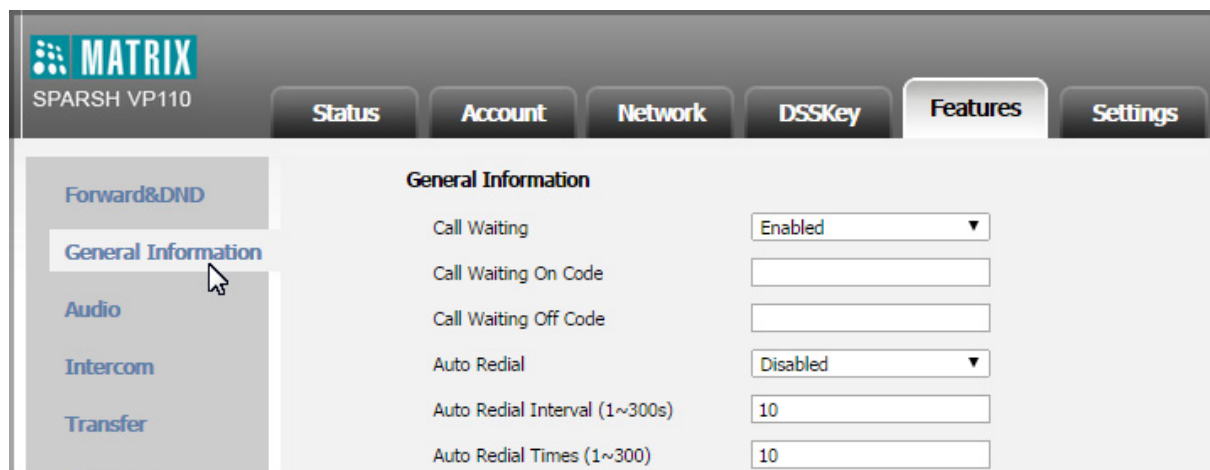
Procedure

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: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>

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.

Save Call Log	Enabled ▼
Suppress DTMF Display	Disabled ▼
Suppress DTMF Display Delay	Disabled ▼ Disabled Enabled
Play Local DTMF Tone	3 ▼
DTMF Repetition	Enabled ▼
Play Hold Tone	30
Play Hold Tone Delay	Enabled ▼
Allow Mute	Enabled ▼

- 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.

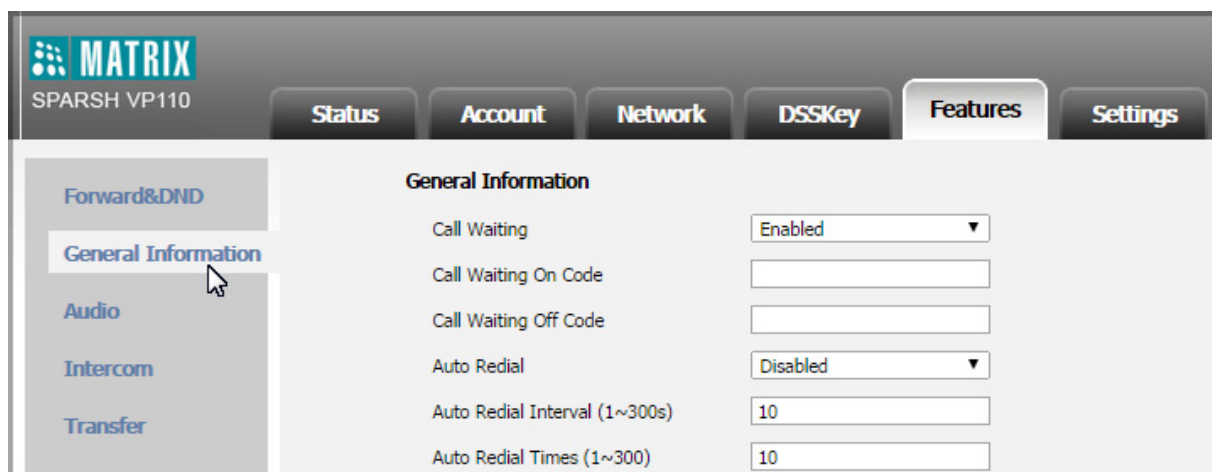
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?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.

Headset Prior	Disabled ▼
DTMF Replace Tran	Disabled ▼
Tran Send DTMF	<input type="text"/>
Send Pound Key	Disabled ▼
Fwd International	Enabled ▼
Diversion/History-Info	Enabled ▼
Allow Trans Exist Call	Enabled ▼
Auto-Logout Time(1~1000min)	1000

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

Play Local DTMF Tone

Play local DTMF tone allows IP phones to play a local DTMF tone during an active call. If this feature is enabled, you can hear the DTMF tone when pressing the IP phone's keypad during a call.

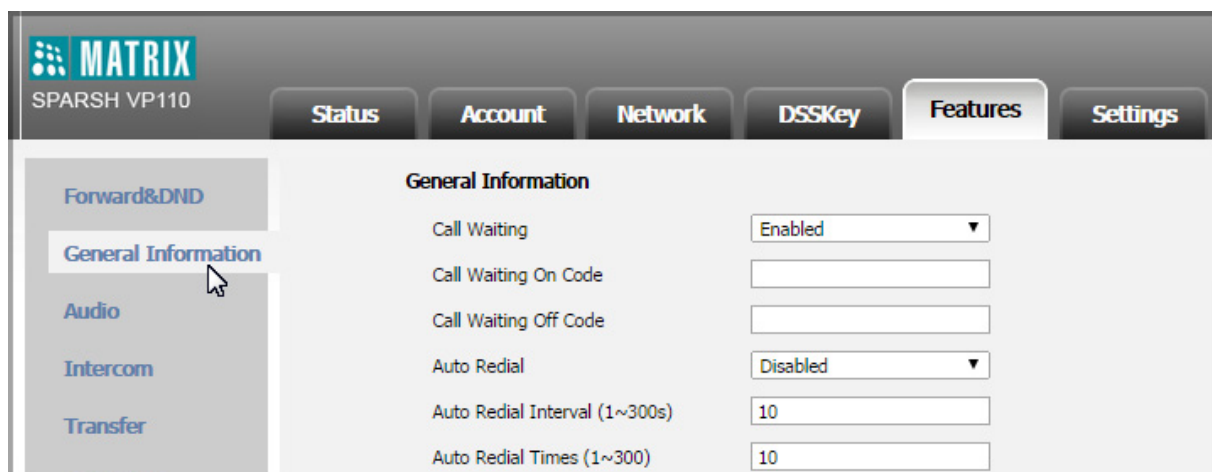
Procedure

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: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>

To configure play local DTMF tone via web user interface:

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



- Select **Enabled** from the **Play Local DTMF Tone**, if you want DTMF tone to be played when the keypad digits are pressed during a call.

Suppress DTMF Display	Disabled ▼
Suppress DTMF Display Delay	Disabled ▼
Play Local DTMF Tone	Enabled ▼
DTMF Repetition	3 ▼
Play Hold Tone	Enabled ▼

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

Intercom

Intercom allows establishing an audio conversation directly. The IP phone can answer intercom calls automatically. This feature depends on support from a SIP server.

Outgoing Intercom Calls

Intercom is a useful feature in office environments to quickly connect with an operator or secretary. Users can press an intercom key to automatically initiate an outgoing intercom call with a remote extension.

Procedure

Intercom key can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	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?p=dsskey&q=load&model=0

To configure an intercom key via web user interface:

- Click on **DSS Key-> Programmable Key**.
- In the desired DSS key field, select **Intercom** from the **Type** list.

- Enter the remote extension number in the **Value** field.

MATRIX
SPARSH VP110

Status **Account** **Network** **DSSKey** **Features** **Settings** **Dir**

Programable Key

Key	Type	Line	Value	Label	Extension
SoftKey 1	History	Local History			
SoftKey 2	N/A	N/A			
SoftKey 3	Speed Dial	N/A			
	Directed Pickup	N/A			
	Group Pickup	N/A			
SoftKey 4	Intercom	N/A			
	Prefix	N/A			
Up	Local Group	Local History			
	XML Group	N/A			
Down	XML Browser	N/A			
	History	N/A			
Left	Menu	N/A			
	Forward	N/A			
Right	DND	N/A			
	Call Return	N/A			
OK	SMS	N/A			
	New SMS	N/A			
MUTE	XML Directory	N/A			
	Status	N/A			
TRAN	Multicast Paging	N/A			
	Local Directory	N/A			

Confirm Cancel Reset to default

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

Incoming Intercom Calls

The IP phone supports automatically to answer 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.

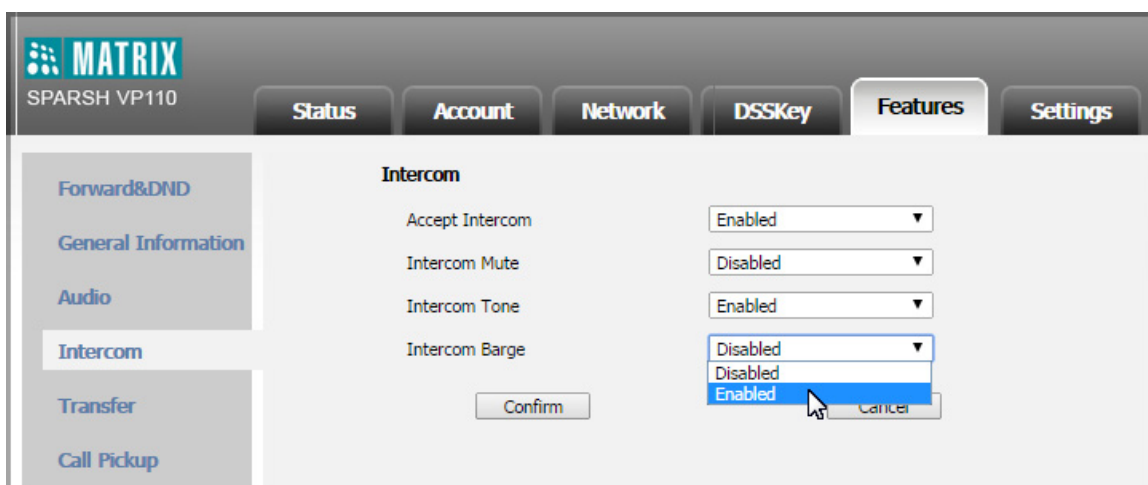
Procedure

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?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:

- Press **Menu->Features->Intercom**.

- Make the desired changes (Press  or  to switch between different intercom features).



- Press the **Save** soft key to save the change or the **Back** soft key to cancel.

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.



You can set a delay time before the phone automatically answers intercom calls.

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.

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.

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.

Multicast Paging

Multicast paging allows IP phones to send/receive Real-time Transport Protocol (RTP) streams to/from the pre-configured multicast address (es) without involving SIP signaling. Up to 10 listening multicast addresses can be specified on the IP phone.

Sending RTP Stream

Users can send an RTP stream without involving SIP signaling by pressing a configured multicast paging key. A multicast address (IP: Port) should be assigned to the multicast paging key, which is defined to transmit RTP stream to a group of designated IP phones. When the IP phone sends the RTP stream to a pre-configured multicast address, each IP phone pre-configured to listen to the multicast address 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	Assign a multicast paging key. For more information, refer " Appendix D - Configuration Parameters ". Specify a multicast codec for the IP phone to use for multicast RTP. For more information, refer " Appendix D - Configuration Parameters ".
Local	Web User Interface	Assign a multicast paging key. Navigate to: <code>http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</code>

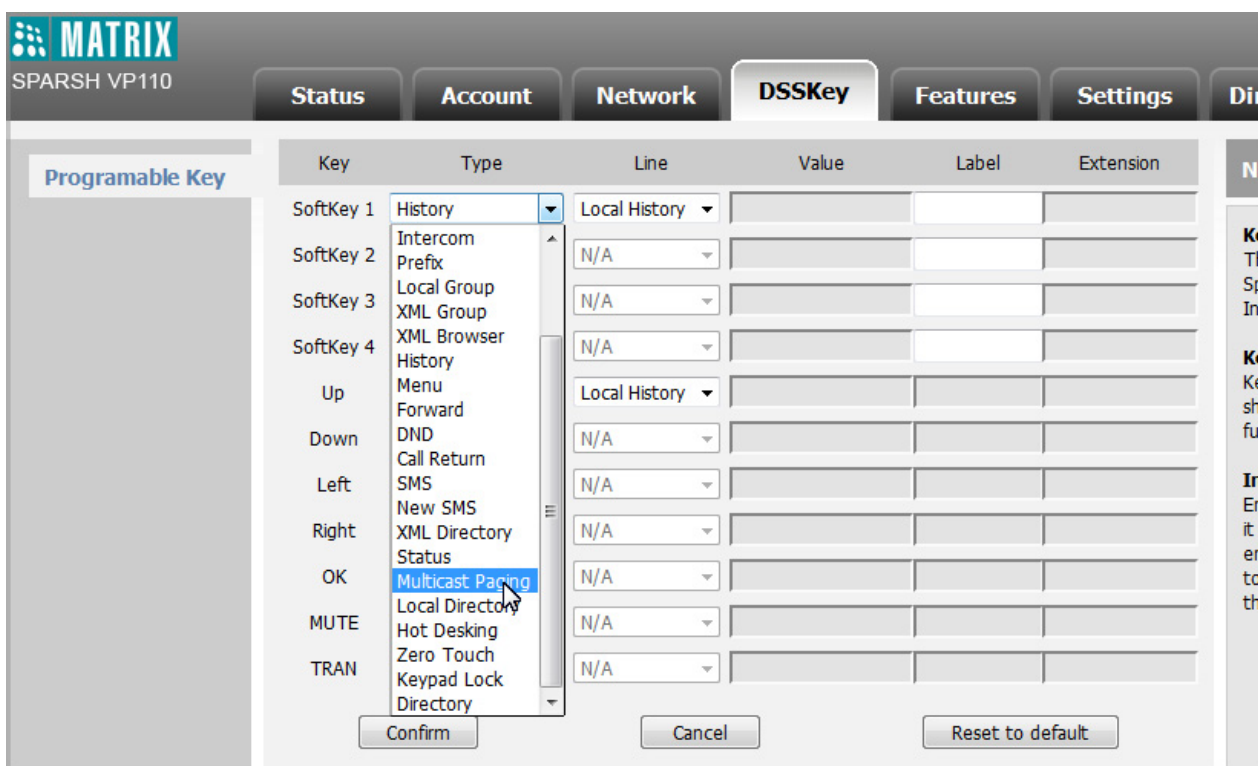


You can specify a multicast codec for the IP phone to use for multicast RTP using the Configuration File.

To configure a multicast paging key via web user interface:

- Click on **DSS Key-> Programmable 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.



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

Receiving RTP Stream

IP phones 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.

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 Disabled is selected from the Paging Barge list, the voice call in progress takes precedence over all incoming multicast paging calls. Valid values in the Paging Barge list:

- **1to10:** Define the priority of an active call, 1 with the highest priority, 10 with the lowest.
- **Disabled:** The voice call in progress shall take precedence over all incoming paging calls.

Paging Priority Active

Paging priority active parameter decides how the phone handles the incoming multicast paging calls when there is already a multicast paging call on the phone. If enabled, the phone will ignore incoming multicast paging calls with lower priorities, otherwise, the phone will answer incoming multicast paging calls automatically and place the previous multicast paging call on hold. If disabled, the 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	<p>Configure the listening multicast address.</p> <p>Configure Paging Barge and Paging Priority Active features.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure the listening multicast address.</p> <p>Configure Paging Barge and Paging Priority Active features.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load</p>

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.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

- Enter the label in the **Label** field.

Label will appear on the LCD screen when receiving the RTP multicast.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features Settings Directory

Local Directory
Remote Phone Book
Phone Call Info
Multicast IP
Setting

Multicast Listening

Paging Barge: 10
Paging Priority Active: Enabled

	IP Address	Listening Address	Label	Priority
1	IP Address			1
2	IP Address			2
3	IP Address			3
4	IP Address			4
5	IP Address			5
6	IP Address			6
7	IP Address			7
8	IP Address			8
9	IP Address			9
10	IP Address			10

NOTE
Multicast Paging
Multicast paging phones to send Transport Protocol to/from the pre-multicast address involving SIP signaling listening multicast be specified on

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



- The priorities of listening addresses are predefined: 1 with the highest priority, 10 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.

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 some source located anywhere (LAN, Internet) to the held party.



To learn about putting a call on hold and resuming the call, refer [“Call Hold and Resume”](#).

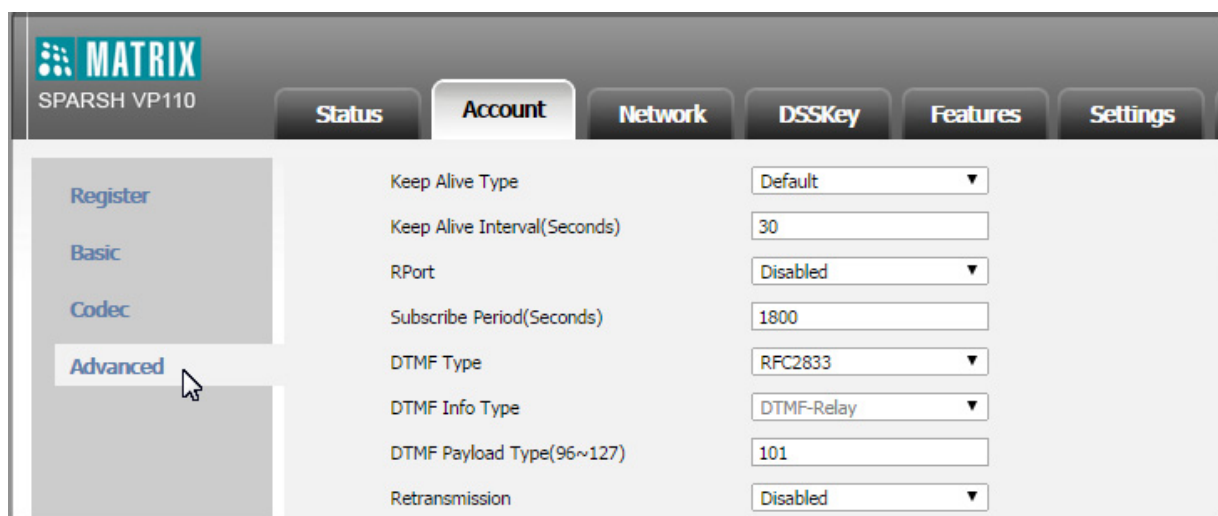
Procedure

Music on hold (MoH) can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure MoH. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure MoH. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

To configure MoH server via web user interface:

- Click on **Account->Advanced**.



- Enter the SIP URI (for example, sip:moh@sip.com) in the **Music Server URI** field.

SIP Registration Retry Timer(0~1800s)	<input type="text" value="30"/>
Conference Type	<input type="text" value="Local Conference"/>
Conference URI	<input type="text"/>
Early Media	<input type="text" value="Disabled"/>
SIP Server Type	<input type="text" value="Default"/>
Music Server URI	<input type="text" value="sip:moh@sip.com"/>
Directed Call Pickup Code	<input type="text"/>
Group Call Pickup Code	<input type="text"/>

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

When you have placed a call on hold, the held party can hear the music.



- *All involved parties cannot use encrypted RTP (SRTP).*
- *MoH server is configurable via web user interface only.*

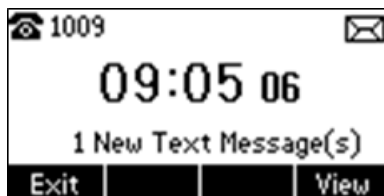
Messages

Messages service in the IP phone can be categorized as follows:

- Short Message Service (SMS)
- Voice Mail
- Message Waiting Indicator (MWI)

Short Message Service (SMS)

You can send and receive text messages using the IP phone. New text messages can be indicated both acoustically and visually. When receiving a new text message, the phone will play a warning tone. The LCD screen will prompt receiving new text messages with the number of waiting messages (e.g., 1 New Text Message(s)) and a flashing icon.



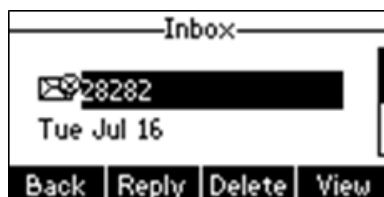
You can store text messages in your phone's Inbox, Sentbox, Outbox or Draftbox. Each of the boxes can store up to 100 text messages. If the number of the text messages in one box is more than 100, the phone will directly delete the oldest text message in the box.



SMS is not available on all servers.

To read a text message:

- Press **Menu->Messages->Text Message->Inbox**.



- Select the desired message and press the **View** soft key.



If the phone prompts receiving new text messages, you can also press the View soft key to read the new messages directly.

To send a text message via phone user interface:

- Press **Menu->Messages->Text Message->New Message**.
- Compose the new text message. You can press the **abc** soft key to change the input mode.



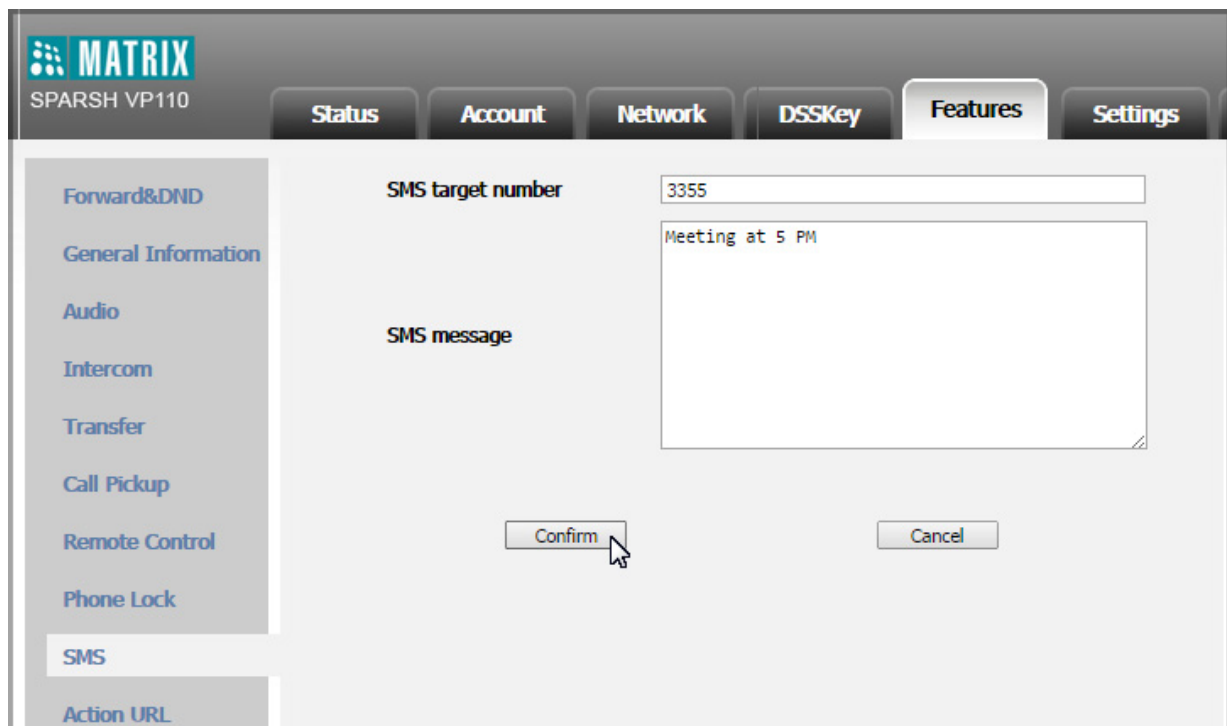
- Press the **Send** soft key after completing the content.
- Enter the number you want to send the message to in the **To** field.
- Press the **Send** soft key to send the message or the **Back** soft key to cancel.



To assign a SMS soft key and a New SMS soft key on the Idle screen, refer [“Programmable Keys”](#).

To send a text message via web user interface:

- Click on **Features->SMS**.
- Enter the number you want to send the message to in the **SMS target number** field.
- Compose the new text message in the **SMS message** field.



- Click **Confirm** to send the message.

To reply a text message:

- Press **Menu->Messages->Text Message->Inbox**.

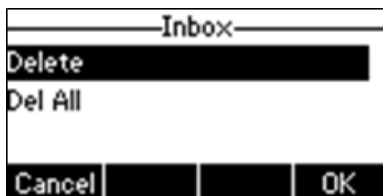
- Select the desired message and press the **Reply** soft key.



- Compose the new text message. You can press the **abc** soft key to change the input mode.
- Press the **Send** soft key after completing the content.
- You may change the destination number, if required in the **To** field.
- Press the **Send** soft key.

To delete a text message:

- Press **Menu->Messages->Text Message->Inbox (Sentbox, Outbox or Draftbox)**.
- Select the desired message and then press the **Delete** soft key.



- Select **Delete** to delete the desired message.

The LCD screen prompts "Delete Message?".



- Press the **OK** soft key to delete this message or the **Cancel** soft key to cancel.
- You can also delete all text messages by pressing the **Delete** soft key and then select **Del All**. For more information, refer to the above steps.



*You can also delete a specific message after retrieving by pressing the **Delete** soft key.*

Voice Mail

You can leave voice mails for someone else on the IP phone. You can also listen to the voice mails stored in a centralized location. When receiving a new voice mail, the phone will play a warning tone. The LCD screen will display a prompt message and flashing icon.



- Voice mail is not available on all servers.
- You can configure whether to play the warning tone or not when receiving a voice mail. To do that, on the web interface, access **Features -> General Information** and select **Enable** for the **Voice Mail Tone** parameter.

To leave a voice mail:

You can leave a voice mail for someone else when he/she 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.

To configure voice mail access codes via phone user interface:

- Press **Menu->Messages->Voice Mail->Set Voice Mail**.




- Press the **123** soft key to select the proper input mode and then enter the voice mail access code.
- Press the **Save** soft key to save the change or the **Back** soft key to cancel.



Voice mail access code must be predefined on the system server.

To listen to voice mails:

- When the phone user interface prompts receiving new voice mails, press  or the **Conn** soft key to dial out the voicemail access code.
- Follow the voice prompt to listen to voice mails.

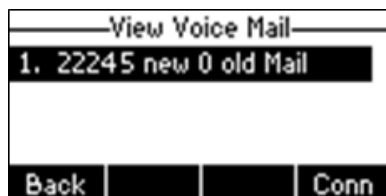


Before listening to voice mails, make sure the voice mail access code has been configured.

To view voice mails via phone user interface:

- Press **Menu->Messages->Voice Mail->View Voice Mail**.

The LCD screen displays the amount of new and old voice mails.



- Press the **Conn** soft key to listen to voice mails.

Message Waiting Indicator (MWI)

The IP phone supports MWI when receiving a new voice message. If someone leaves you a voicemail, you will receive a message waiting indicator. MWI will be indicated in two ways: a warning tone and a prompt message (including a voice mail icon) on the LCD screen. This will be cleared when you retrieve all voice mails or delete them. Message Waiting Indicator (MWI) informs users of the number of messages waiting in their mailbox without calling the mailbox.

The MWI service is unsolicited for some servers, so the IP phone only handles the MWI messages sent from the server. But for other servers, the MWI service is solicited, so the IP phone must enable subscription for MWI.

For advanced users:

IP phones support both solicited and unsolicited MWI. Unsolicited MWI is a server related feature.

The IP phone sends a SUBSCRIBE message to the server for message-summary updates. The server sends a message-summary NOTIFY within the subscription dialog each time the MWI status changes. For solicited MWI, you must enable MWI subscription feature on IP phones. IP phones support subscribing the MWI messages to the account or the voice mail number.

IP phones do not need to subscribe for message-summary updates. The server automatically sends a message-summary NOTIFY in a new dialog each time the MWI status changes.



MWI service is not available on all servers.

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 SUBSCRIBE expiration.
Subscribe MWI to Voice Mail	<p>Enable or disable a subscription to the voice mail number for MWI service.</p> <p>To use this feature, you should configure subscribe for MWI and the voice mail number in advance.</p>



Whether the phone sends SUBSCRIBE messages for MWI service to the account or to the voice mail number depends on the server.

Procedure

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

Configuration File	<MAC>.cfg	<p>Configure subscribe for MWI.</p> <p>Configure subscribe MWI to voice mail.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure subscribe for MWI.</p> <p>Configure subscribe MWI to voice mail.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0 </p>

To configure subscribe for MWI via web user interface:

- Click on **Account->Advanced**.
- Select **Enabled** from the **Subscribe for MWI** list.

- Enter the period time in the **MWI Subscription Period(Seconds)** field.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Account' tab is active. On the left, a sidebar menu shows 'Register', 'Basic', 'Codec', and 'Advanced' (highlighted). The main content area displays various settings for the 'Account' section. The 'Subscribe for MWI' dropdown menu is open, showing three options: 'Disabled', 'Disabled', and 'Enabled'. A mouse cursor is pointing at the 'Enabled' option.

Setting	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Disabled
MWI Subscription Period(Seconds)	Enabled
Subscribe MWI To Voice Mail	Disabled
Voice Mail	

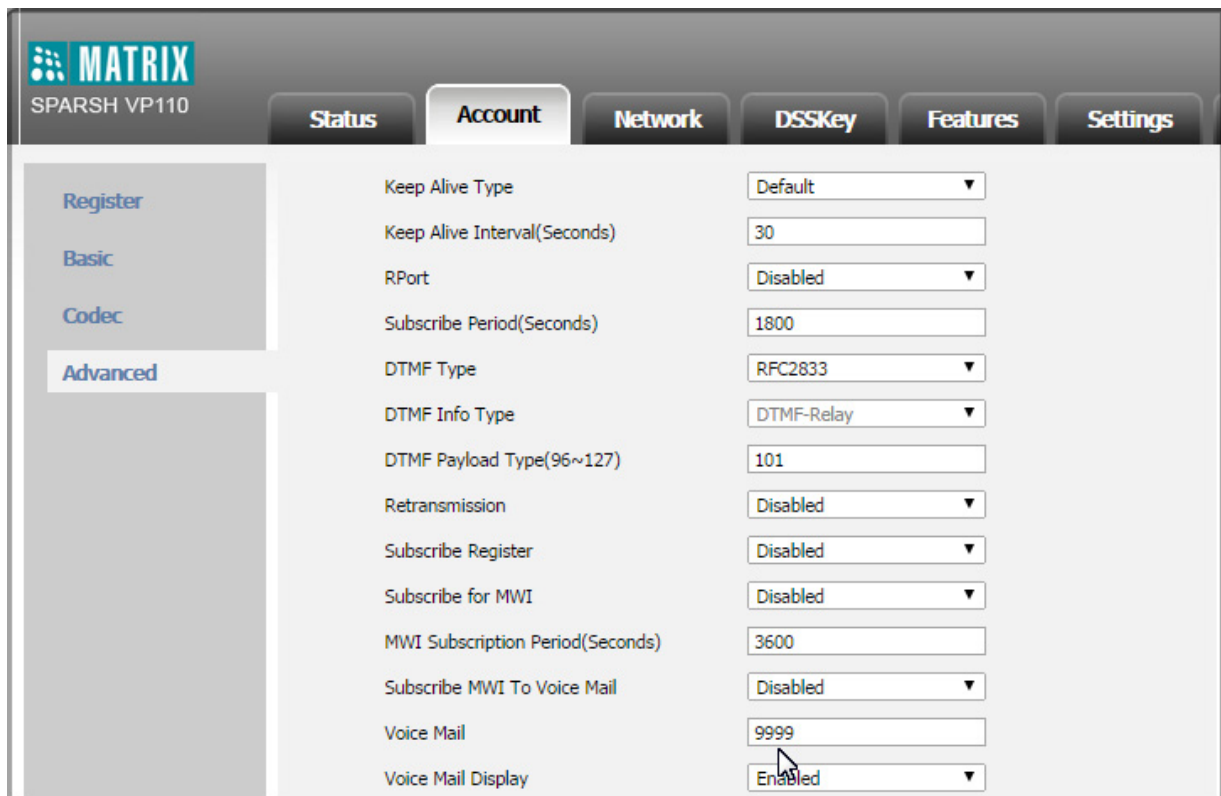
- Click **Confirm** to save the change.

The IP phone will subscribe to the account number for MWI service by default.

To enable **Subscribe MWI to Voice Mail** via web user interface:

- Click on **Account->Advanced**.
- Select **Enabled** from the **Subscribe MWI To Voice Mail** list.

- Enter the desired voice mail number in the **Voice Mail** field.



The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account (selected), Network, DSSKey, Features, and Settings. On the left, a sidebar contains links for Register, Basic, Codec, and Advanced (selected). The main content area displays various configuration parameters for the Account tab. The 'Voice Mail' field is set to 9999, and the 'Voice Mail Display' dropdown is set to Enabled. A mouse cursor is pointing at the 'Voice Mail Display' dropdown.

Parameter	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Disabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Disabled
Voice Mail	9999
Voice Mail Display	Enabled

- 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.



- *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).*



MWI subscription is configurable via web user interface only.

Web Server Type

Web server type determines access protocol of the IP phone's web user interface. IP phones support both HTTP and HTTPS protocols for accessing the 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.

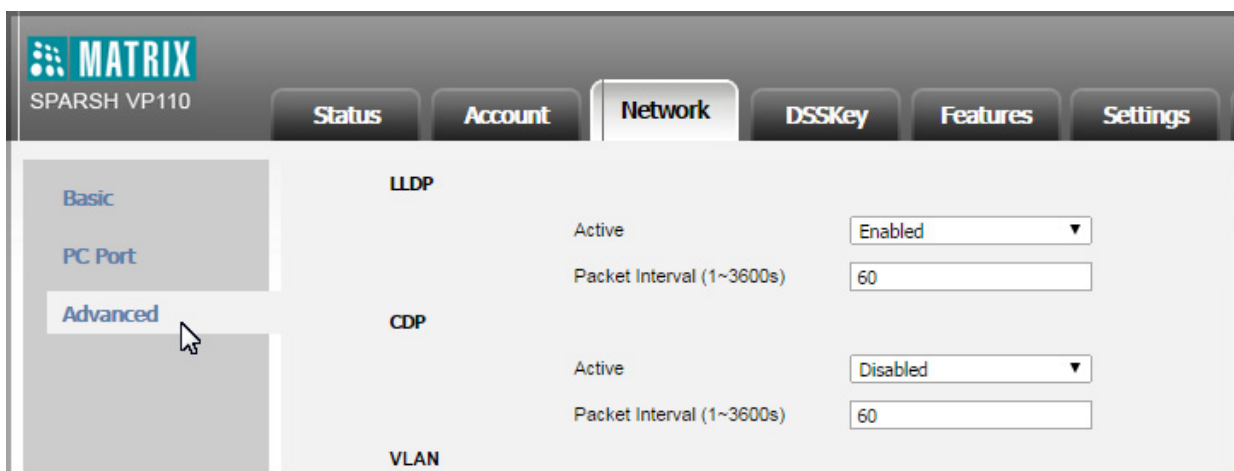
Procedure

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: <code>http://<phoneIPAddress>/servlet?p=network-adv&q=load</code>
	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.

Web Server	
HTTP	Enabled ▼
HTTP Port (1~65535)	80
HTTPS	Enabled ▼
HTTPS Port (1~65535)	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:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Network->Webserver Type**.
- Press ◀ or ▶, or the **Switch** soft key to select the desired value from the **HTTP Status** field.
- Enter the HTTP port number in the **HTTP Port** field.
- Press ◀ or ▶, or the **Switch** soft key to select the desired value from the **HTTPS Status** field.
- Enter the HTTPS port number in the **HTTPS Port** field.
- Press the **Save** soft key to save the change.

The IP phone reboots automatically to make settings effective after a period of time.

Softkey Layout

Softkey layout is used to customize the soft keys at the bottom of the LCD screen to best meet users' requirements. In addition to specifying which soft keys to display, you can determine their display order. It can be configured based on call states.

You can configure the soft key layout using the softkey layout templates for different call states. For more information on how to configure a softkey layout template, refer ["Softkey Layout Template"](#).

The following table lists soft keys available for IP phones in different call states.

Call State		Default Soft Keys	Optional Soft Keys
CallFailed		NewCall Empty Empty Empty	Empty Switch Cancel
CallIn		Answer Forward Silence Reject	Empty Switch
Connecting	Connecting	Empty Empty Empty Cancel	Empty Switch
	SemiAttendTrans	Transfer Empty Empty Cancel	Empty Switch
Dialing		Send IME Delete Cancel	Empty History Switch Dir GPickup DPickup
RingBack	RingBack	Empty Empty Empty Cancel	Empty Switch CC
	SemiAttendTransBack	Transfer Empty Empty Cancel	Empty Switch CC

Talking	Talk	Transfer Hold Conference Cancel	Empty Mute SWAP NewCall Switch Answer Reject
	Hold	Transfer Resume NewCall Cancel	Empty Switch Answer Reject
	Held	Empty Empty Empty Cancel	Empty Switch Answer Reject NewCall
	PreTrans (Transfer To)	Transfer IME Delete Cancel	Empty Directory Switch Send
	Conferenced	Empty Hold Split Cancel	Empty Switch Answer Reject Mute



CC soft key is not applicable to the IP phone and it has no functional importance. You can remove it from the Auto Provisioning configuration also.

Procedure

Softkey layout can be configured using the configuration files or locally.




Configuration File	<MAC>.cfg	Specify the access URL of the softkey layout template. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure the softkey layout. Navigate to: http://<phoneIPAddress>/servlet?p=settings-softkey&q=load

To configure softkey layout via web user interface:

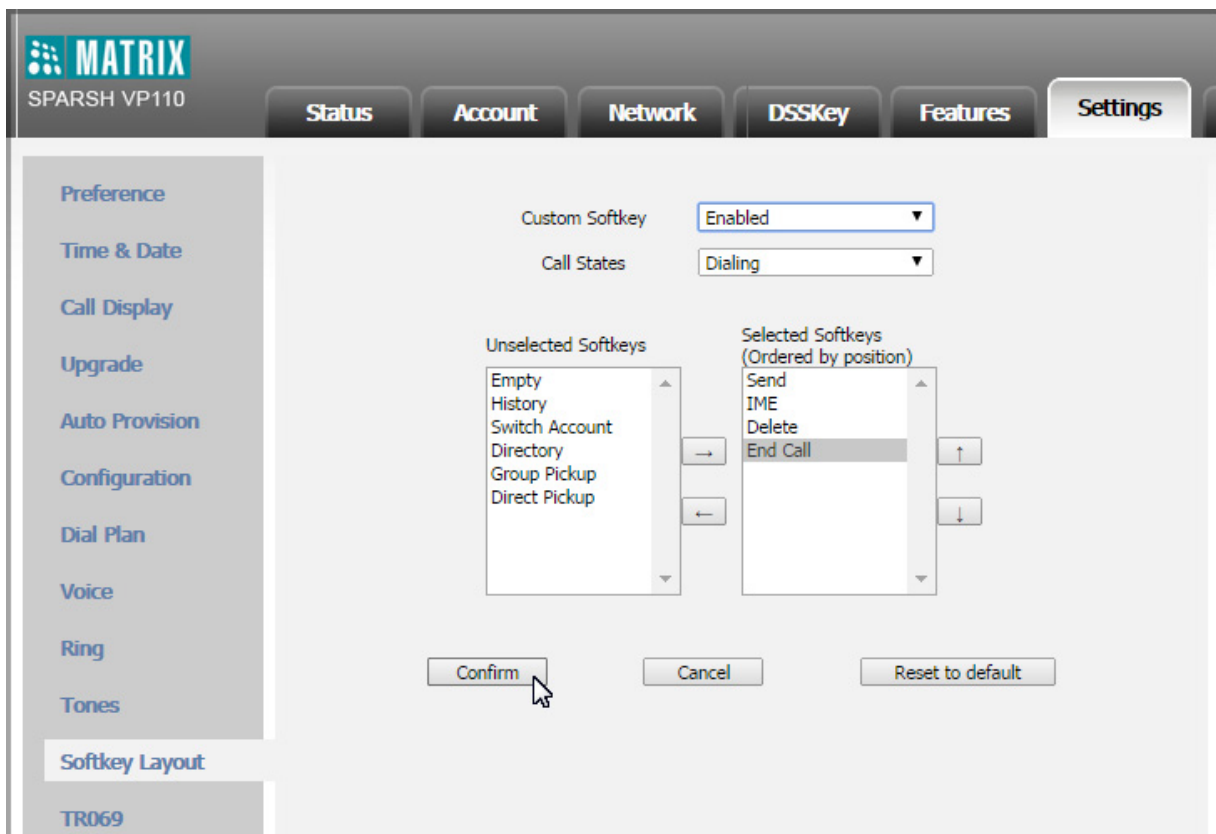
- Click on **Settings->Softkey Layout**.
- Select the desired value from the **Custom Softkey** list.

- 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.

- 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 LCD screen displays the soft keys in the adjusted order.



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

Busy Tone Delay

Busy tone is audible to the other party, indicating that the call connection has been broken when one party releases a call. Busy tone delay can define a period of time during which the busy tone is audible.

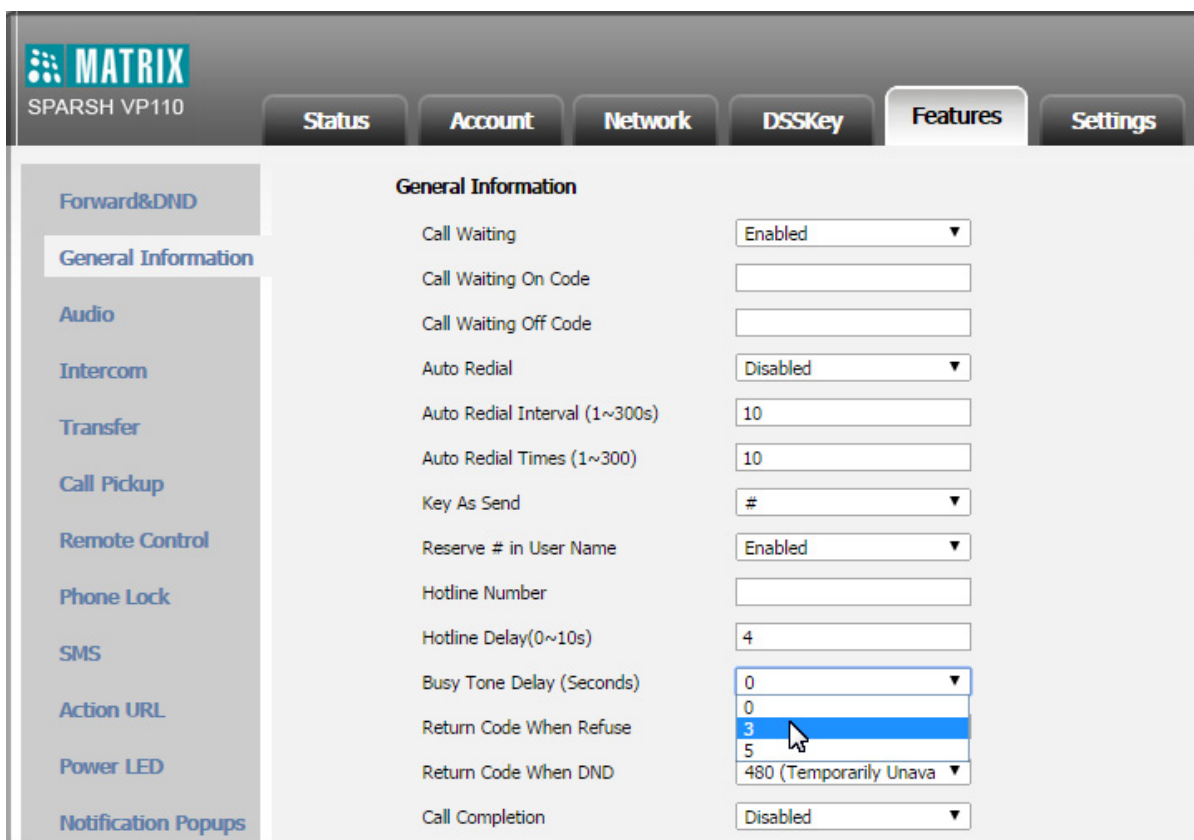
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?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.



The screenshot displays the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Features' tab is active, and the 'General Information' sub-tab is selected. On the left sidebar, various feature categories are listed, including Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control, Phone Lock, SMS, Action URL, Power LED, and Notification Popups. The main content area shows the 'General Information' configuration page with various settings. The 'Busy Tone Delay (Seconds)' dropdown menu is open, showing options 0, 3, 5, and 480 (Temporarily Unavailable). The value 3 is currently selected.

- 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 phone LCD 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	<p>Specify the return code and the reason of the SIP response message when refusing a call.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Specify the return code and the reason of the SIP response message when refusing a call.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load</p>

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.

MATRIX
SPARSH VP110

Status **Account** **Network** **DSSKey** **Features** **Settings**

Forward&DND

General Information

Audio

Intercom

Transfer

Call Pickup

Remote Control

Phone Lock

SMS

Action URL

Power LED

Notification Popups

General Information

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	404 (Not Found)
Call Completion	480 (Temporarily Unavailable)
	486 (Busy Here)
	603 (Decline)

- 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.

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 IP phones to resume and play the local ring-back tone upon a subsequent 180 message received.

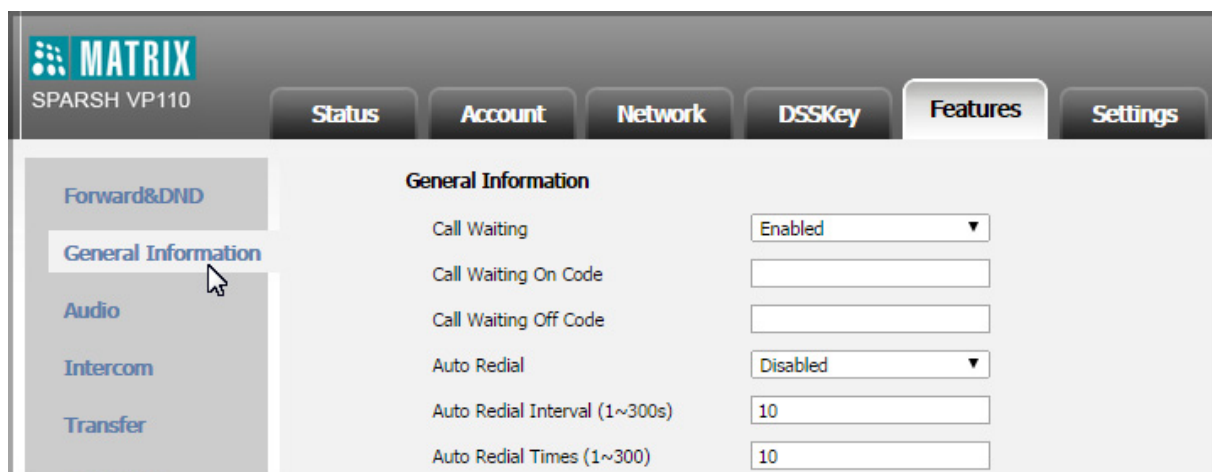
Procedure

180 ring workaround can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure 180 ring work around. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure 180 ring work around. Navigate to: <code>http://<phoneIPAddress>/servlet?p=features-general&q=load</code>

To configure 180 ring workaround via web user interface:

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



- Select the desired value from the **180 Ring Workaround** list.

Time-Out for Dial-Now Rule	<input type="text" value="1"/>
RFC 2543 Hold	<input type="text" value="Enabled"/>
Use Outbound Proxy In Dialog	<input type="text" value="Enabled"/>
180 Ring Workaround	<input type="text" value="Disabled"/>
Logon Wizard	<input type="text" value="Disabled"/>
PswPrefix	<input type="text"/>
PswLength	<input type="text"/>
PswDial	<input type="text" value="Disabled"/>

- 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 forcefully.



To use this feature, make sure the outbound server has been correctly configured on the IP phone.

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: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load

To specify whether to use outbound proxy server in a dialog via web user interface:

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

- Select the desired value from the **Use Outbound Proxy In Dialog** list.

MATRIX
SPARSH VP110

Status Account Network DSSKey **Features** Settings

Forward&DND
General Information
Audio
Intercom
Transfer
Call Pickup
Remote Control
Phone Lock
SMS
Action URL
Power LED
Notification Popups

General Information

Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	480 (Temporarily Unava
Call Completion	Disabled
Feature Key Synchronization	Disabled
Time-Out for Dial-Now Rule	1
RFC 2543 Hold	Enabled
Use Outbound Proxy In Dialog	Enabled Disabled Enabled
180 Ring Workaround	Enabled
Logon Wizard	Disabled

- 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. Timer T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server. Timer T2 represents the maximum retransmit interval for non-INVITE requests and INVITE responses. Timer T4 represents the maximum duration a message will remain in the network. These session timers are configurable on IP phones.

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?p=settings-sip&q=load

To configure session timer via web user interface:

- Click on **Settings-> SIP**.
- Enter the desired value in the **SIP Session Timer T1 (0.5~10s)** field.

The default value is 0.5s.

- Enter the desired value in the **SIP Session Timer T2 (2~40s)** field.

The default value is 4s.

- Enter the desired value in the **SIP Session Timer T4 (2.5~60s)** field.

The default value is 5s.

The screenshot displays the 'MATRIX SPARSH VP110' web interface. At the top, there are tabs for 'Status', 'Account', 'Network', 'DSSKey', 'Features', and 'Settings'. The 'Settings' tab is active. On the left, a sidebar menu lists various settings categories: 'Preference', 'Time & Date', 'Call Display', 'Upgrade', 'Auto Provision', 'Configuration', 'Dial Plan', 'Voice', 'Ring', 'Tones', 'Softkey Layout', 'TR069', 'Voice Monitoring', and 'SIP'. The 'SIP' category is selected. The main content area is titled 'SIP Config' and contains the following configuration items:

Parameter	Value
SIP Session Timer T1 (0.5~10s)	0.5
SIP Session Timer T2 (2~40s)	4
SIP Session Timer T4 (2.5~60s)	5
Local SIP Port	5060
TLS SIP Port	5061

At the bottom of the configuration area, there are two buttons: 'Confirm' and 'Cancel'.

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

Session Timer

Session timer allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active. Session timer is specified in RFC 4028. IP phones support two refresher modes: UAC and UAS. The UAC mode means refreshing the session from the client, while the UAS mode means refreshing the session from the server. The session expiration and session refresher are negotiated via the Session-Expires header in the INVITE message. The negotiated refresher will send a re-INVITE/UPDATE request at or before the negotiated session expiration.

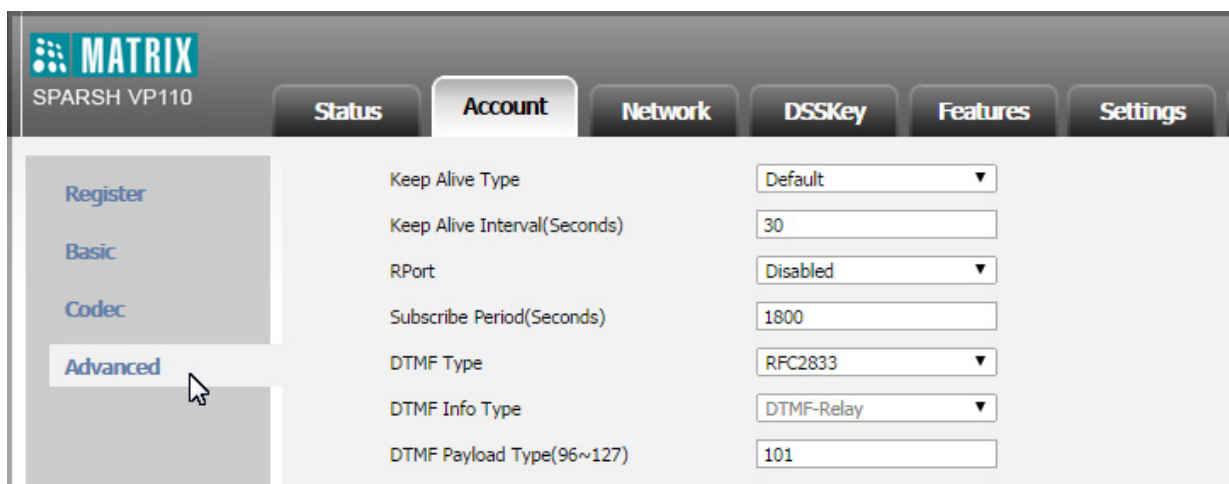
Procedure

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: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

To configure session timer via web user interface:

- Click on **Account-> Advanced**.



- 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.

Voice Mail	<input type="text"/>
Voice Mail Display	Enabled ▼
Caller ID Source	FROM ▼
Session Timer	Disabled ▼
Session Expires(90~7200s)	1800
Session Refresher	UAC ▼
Send user=phone	UAC
	UAS
RTP Encryption(SRTP)	Disabled ▼

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

Distinctive Ring Tones

Distinctive ring tones allows certain incoming calls to trigger IP phones 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 and keyword parameter and maps them to the appropriate ring tone.



If the caller has existed in the local directory, the ring tone assigned to the caller should be preferentially played.

Alert-Info headers in the following formats can be configured:

- Alert-Info: 127.0.0.1/Bellcore-drN
- Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)
- Alert-Info: <URL>
- Alert-Info: info=info text;x-line-id=0

Refer following details about each header format:

- If the Alter-Info header contains the keyword "Bellcore-drN", and the parameter "features.alert_info_tone" is set to 1, the IP phone will play the Bellcore-drN (N=1, 2, 3, 4 or 5) ring tone. Otherwise, the IP phone will play the pre-configured local ring tone in about ten seconds.

Example:

Alert-Info: http://127.0.0.1/Bellcore-dr1

The following table identifies the different Bellcore ring tone patterns and cadences.

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



"Bellcore-dr5" is a ring splash tone that reminds the user that the DND or Always Call Forward feature is enabled on the server side.

- If the Alert-Info header contains the keyword "ringtone-N" or "MyMelodyN", the IP phone will play the corresponding local ring tone (RingN.wav). Otherwise, the IP phone will play the pre-configured local ring tone in about ten seconds.

Example:

Alert-Info: ringtone-2

Alert-Info: MyMelody2

The following table identifies different ring tones.

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

- If the Alert-Info header contains a remote URL, and the parameter “account.X.alert_info_url_enable” is set to 1 (the item called “Distinctive Ring Tones” on the web user interface is Enabled), the IP phone will try to download the WAV ring tone file from the URL and then play the remote ring tone. Otherwise, the IP phone will play the pre-configured local ring tone in about ten seconds.

Example:

Alert-Info: http://192.168.0.12:8080/Custom.wav

- If the Alert-Info header contains an info text, the IP phone will map the text with the internal ringer text pre-configured, and then play the ring tone associated with the internal ringer text. Otherwise, the IP phone will play the pre-configured local ring tone in about ten seconds.

Example:

Alert-Info: info=family;x-line-id=0

Procedure

Distinctive ring tones can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure distinctive ring tones. For more information, refer “Appendix D - Configuration Parameters” .
	<MAC>.cfg	Configure the internal ringer text and internal ringer file. For more information, refer “Appendix D - Configuration Parameters” .

Local	Web User Interface	<p>Configure distinctive ring tones.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p> <p>Configure the internal ringer text and internal ringer file.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-ring&q=load">http://<phoneIPAddress>/servlet?p=settings-ring&q=load</p>
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To configure distinctive ring tones via web user interface:

- Click on **Account-> Advanced**.

- 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**.

The screenshot shows the MATRIX SPARSH VP110 web interface. At the top, there is a header with the MATRIX logo and the model name SPARSH VP110. Below the header is a navigation bar with tabs: Status, Account, Network, DSSKey, Features, and Settings. The Settings tab is selected. On the left side of the Settings page, there is a sidebar with a list of configuration categories: Preference, Time & Date, Call Display, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, Ring, Tones, and Softkey Layout. The 'Ring' category is selected, and a mouse cursor is pointing at it. The main content area displays a table with 7 rows, each representing a configuration item. Each row has three columns: an index number, a label, and a value field. The labels are 'Internal Ringer Text' and 'Internal Ringer File'. The value fields for 'Internal Ringer Text' are text input boxes, and the value fields for 'Internal Ringer File' are dropdown menus showing 'Ring1.wav'.

Index	Label	Value
1	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
2	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
3	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
4	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
5	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
6	Internal Ringer Text	<input type="text"/>
	Internal Ringer File	<input type="text" value="Ring1.wav"/>
7	Internal Ringer Text	<input type="text"/>

- Enter the keywords in the **Internal Ringer Text** fields.
- Select the desired ring tones for each text from the **Internal Ringer File** lists.
- Click **Confirm** to save the change.

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 IP phones for the following conditions.

Condition	Description
Dial	When in the pre-dialing interface

Ring Back	Ring-back tone
Busy	When the callee is busy
Congestion	When the network is congested
Call Waiting	Call waiting tone
Dial Recall	When receiving a call back
Info	When receiving a special message
Stutter	When receiving a voice mail
Message	When receiving a text message
Auto Answer	When automatically answering a call

Procedure

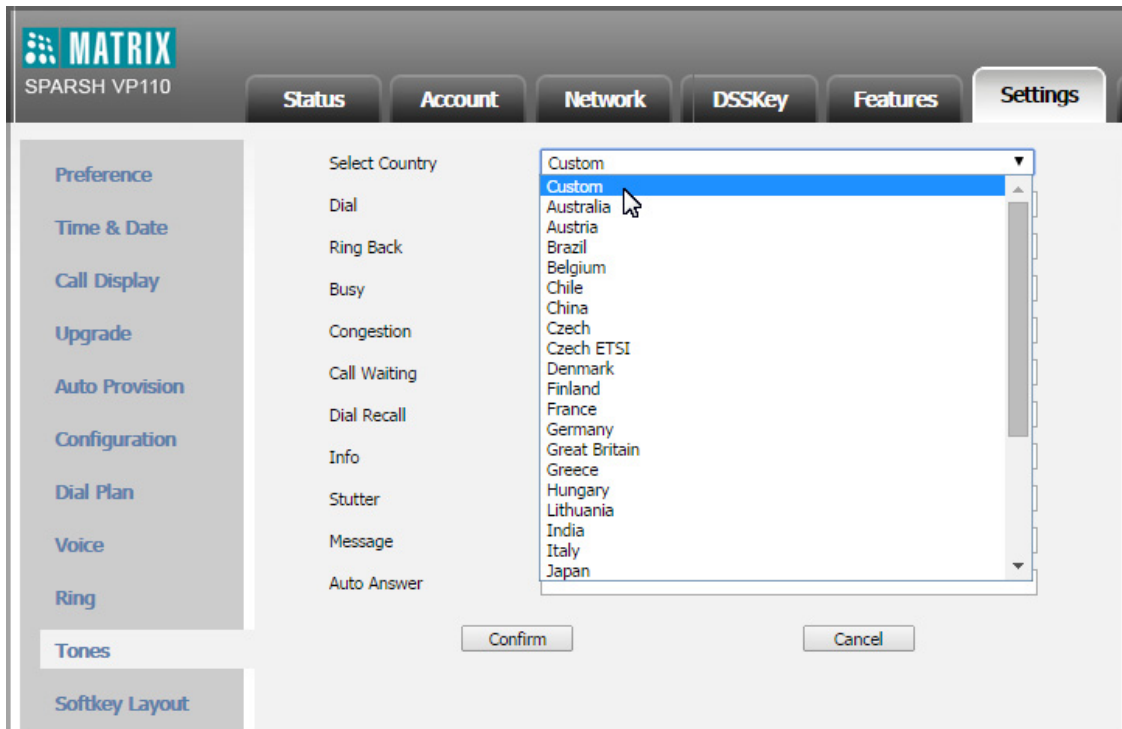
Tones can be configured using the configuration files or locally.

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: <code>http://<phoneIPAddress>/servlet?p=settings-tones&q=load</code>

To configure tones via web user interface:

- Click on **Settings->Tones**.
- Select the desired type 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.

Action URL

Action URL allows IP phones 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., log on). The valid URL format is: `http(s)://IP address of the server/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.
Autop 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)://IP address of server/help.xml?variable name=$variable`. Variable name can be customized by users, while the variable value is pre-defined. For example, a URL “`http://192.168.1.10/help.xml?mac=$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: <code>http://<phoneIPAddress>/ servlet?p=features-actionurl&q=load</code>

To configure action URL via web user interface:

- Click on **Features->Action URL**.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The Features tab is active, and the left sidebar shows the 'Action URL' option selected. The main content area displays a list of features with their current status and input fields for configuration:

Feature	Status	Input Field
Forward&DND	Setup Completed	<input type="text"/>
General Information	Registered	<input type="text"/>
	Unregistered	<input type="text"/>
	Register Failed	<input type="text"/>
Audio	Off Hook	<input type="text"/>
Intercom	On Hook	<input type="text"/>
Transfer	Incoming Call	<input type="text"/>
Call Pickup	Outgoing call	<input type="text"/>
Remote Control	Established	<input type="text"/>
Phone Lock	Terminated	<input type="text"/>
SMS	Open DND	<input type="text"/>
	Close DND	<input type="text"/>
Action URL	Open Always Forward	<input type="text"/>
	Close Always Forward	<input type="text"/>
Power LED	Open Busy Forward	<input type="text"/>
Notification Popups	Close Busy Forward	<input type="text"/>

- Enter the action URLs in the corresponding fields.
- Click **Confirm** to save the change.

Action URI

Opposite to action URL, action URI allows IP phones to interact with web server application by receiving and handling an HTTP or HTTPS GET request. When 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)://phone IPaddress/servlet?key=variable value.

SIP Notify Message

In addition, Yealink 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 especially useful for users always working in the small office/home office where a secure firewall may prevent the HTTP or HTTPS GET request from the external network.

Example of a SIP Notify with the variable value (OK):


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: Yealink SIP-T23G
Event: ACTION-URI
Content-Type: message/sipfrag
Content-Length: 6

Message Body

key=OK

The following table lists pre-defined variable values:

Variable Value	Phone Action
OK	Press the OK  key.
ENTER	Press the Enter soft key.
SPEAKER	Press the Speakerphone key.
F_TRANSFER	Transfers a 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.
X	Cancel actions or reject incoming calls.
Cancel/CANCEL	Cancel actions or reject incoming calls or end a call.
0-9/*/POUND	Press the keypad (0-9, * or #).
F_CONFERENCE	Press the Conference soft key.
F1-F4	Press the soft keys.
MSG	Press the MESSAGE key.
HEADSET	Press the HEADSET key.
RD	Press the RD key.
UP/DOWN/LEFT/RIGHT	Press the navigation keys.
Reboot	Reboot the IP 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.
OFFHOOK	Pick up the handset.
ONHOOK	Hang up the handset.
ANSWER/ASW/Asw	Answer a call.
Reset	Reset a phone.
ATrans=xxx	Perform a semi-attended/attended transfer to xxx.
BTrans=xxx	Perform a blind transfer to xxx.
CALLEND	End a call.
phonecfg=get[&accounts=x][&dnd=x][&fw=x]	<p>Get firmware version, registration, DND or forward configuration information.</p> <p>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.</p> <p>Note: The valid URI is: http(s)://phone IP address/ servlet?phonecfg=get[&accounts=x][&dnd=x][&fw=x]</p>



- The variable value is not applicable to all events. For example, the variable value “MUTE” is only applicable when the IP phone is 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.

For security reasons, IP phones do not receive and handle HTTP/HTTPS GET requests by default. You need to specify the trusted IP address for action URI. When the IP phone receives a GET request from the trusted IP address for the first time, the LCD 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.

Procedure

Specify the trusted IP address for action URI using the configuration files or locally.

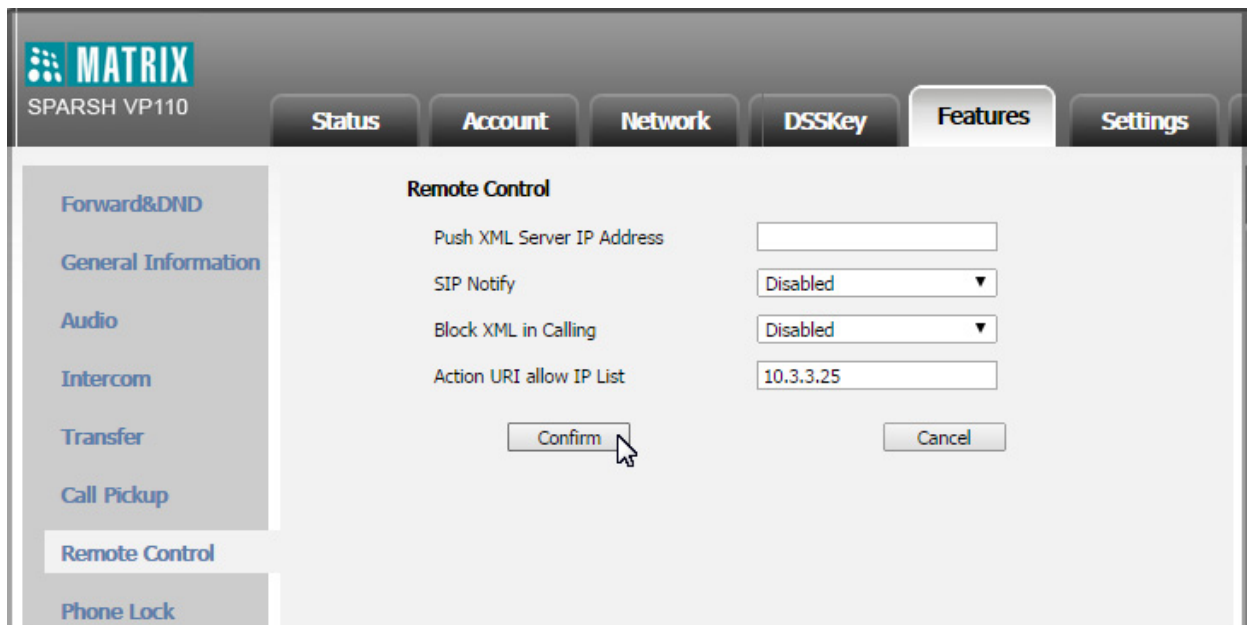
Configuration File	<MAC>.cfg	<p>Specify the trusted IP address(es) for sending the action URI to the IP phone.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Specify the trusted IP address(es) for sending the action URI to the IP phone.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=features-remotecontrl&q=load</p>

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.



The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The left sidebar lists various settings categories: Forward&DND, General Information, Audio, Intercom, Transfer, Call Pickup, Remote Control (highlighted), and Phone Lock. The main content area is titled 'Remote Control' and contains the following settings:

Setting	Value
Push XML Server IP Address	<input type="text"/>
SIP Notify	Disabled
Block XML in Calling	Disabled
Action URI allow IP List	10.3.3.25

At the bottom of the settings area, there are two buttons: 'Confirm' and 'Cancel'. A mouse cursor is pointing at the 'Confirm' button.

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

Server Redundancy

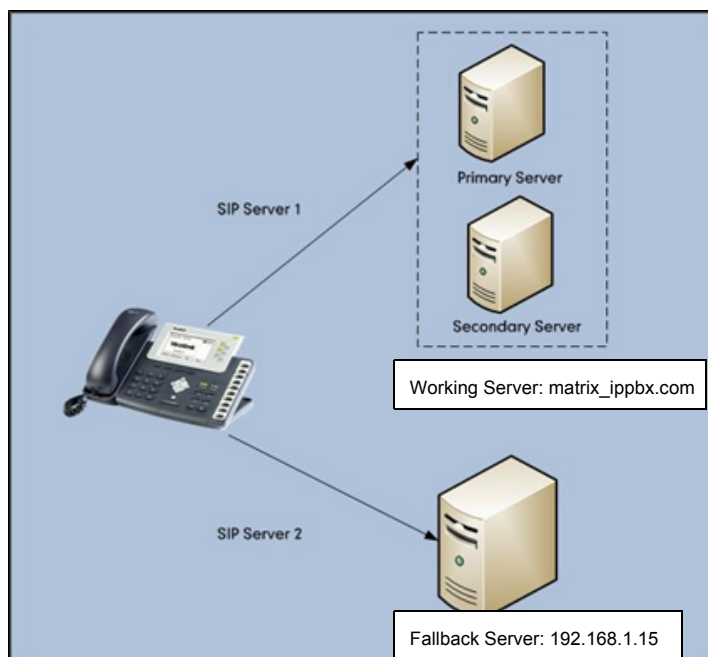
Server redundancy is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

Two types of redundancy are possible. In some cases, a combination of the two may be deployed:

- **Failover:** In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using the DNS mechanism from the primary to the secondary server.
- **Fallback:** In this mode, a second less featured call server with SIP capability takes over call control to provide basic calling capability, but without some advanced features offered by the working server (for example, MWI). IP phones support configuration of two SIP servers per SIP registration for fallback purpose.

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 as below. In the example, server redundancy for fallback and failover purposes is deployed. Two separate SIP servers (a working server and a fallback server) are configured for account (line) registration.



- **Working Server:** Server 1 is configured with the domain name of the working server. For example, 'matrix_ippbx.com'. DNS mechanism is used such that the working server is resolved to multiple SIP servers for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server has the highest priority server in a cluster of servers resolved by

the DNS server. The secondary server backs up a primary server when the primary server fails and offers the same functionality as the primary server.

- **Fallback Server:** Server 2 is configured with the IP address of the fallback server. For example, 192.168.1.15. A fallback server offers less functionality than the working server.

Phone Registration

Registration methods of the fallback mode include:

- **Concurrent registration:** The IP phone registers to two SIP servers (working server and fallback server) at the same time. In a failure situation, a fallback server can take over the basic calling capability, but without some of the advanced features offered by the working server (default registration method).
- **Successive registration:** The IP phone only registers to one server at a time. The IP phone first registers to the working server. In a failure situation, the IP phone registers to the fallback server.

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	<p>Configure the server redundancy on the IP phone.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure the server redundancy on the IP phone.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</p>

To configure server redundancy for fallback purpose via web user interface:

- Click on **Account->Register**.

- Configure registration parameters of the account in the corresponding fields.
- Configure parameters of **SIP Server 1** and **SIP 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**.
- Configure registration parameters of the account in the corresponding fields.
- Configure parameters of the **SIP Server 1** or **SIP Server 2** in the corresponding fields.
- Select **DNS-NAPTR** from the **Transport** list.

You must set the port of SIP server to 0 for NAPTR, SRV and A queries.

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



If the outbound proxy server is required and the transport is set to DNS-NAPTR, you must set the port of outbound proxy server to 0 for NAPTR, SRV and A queries.

SIP 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. The IP phone performs NAPTR query for the NAPTR pointer and transport protocol (UDP, TCP and TLS), the SRV query on the record returned from the NAPTR for the target domain name and the port number, and the 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_ippbx.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_ippbx.com
IN NAPTR	100	50	"s"	"SIP+D2U"	""	_sip._udp.matrix_ippbx.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_ippbx.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_ippbx.com
IN SRV	0	2	5060	server2.matrix_ippbx.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 will be 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 an A query. So in this case, the IP phone uses "server1.matrix_ippbx.com" and "server2.matrix_ippbx.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_ippbx.com IN A 192.168.1.13
Server2.matrix_ippbx.com IN A 192.168.1.14
```

The IP phone picks the IP address "192.168.1.14" first.

Outgoing Call When the Working Server Connection Fails

When a user initiates a call, the phone will go through the following steps to connect the call:

- Sends the INVITE request to the primary server.

- If the primary server does not respond correctly to the INVITE, then tries to make the call using the secondary server.
- If the secondary server is also unavailable, the IP phone will try the 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 and this is the last server, then the signaling fails after the complete UDP time-out defined in RFC 3261. If it is not the last server in the list, the maximum number of retries depends on the configured retry count.

Procedure

Server redundancy can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Configure the transport type on the IP phone.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure the transport type on the IP phone.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</p>

LLDP

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol, which allows IP phones 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. LLDP transmits information as packets called LLDP Data Units (LLDPDUs). An LLDPDU consists of a set of Type-Length-Value (TLV) elements, each of which contains a particular type of information about the device or the port transmitting it.

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 IP phones to determine the capabilities that the connected switch supports and has enabled.
- Network Policy -- provides voice VLAN configuration to notify IP phones 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 IP phones are powered, power priority, and how much power IP phones need.
- Inventory Management -- provides a means to effectively manage IP phones 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	<p>The supported and enabled capabilities of the IP phone.</p> <p>The supported capabilities are Bridge and Telephone. The enabled capabilities are Bridge and Telephone by default.</p>
	Port Description	<p>Description of port that sends data unit.</p> <p>The default value is "WAN PORT".</p>
IEEE Std 802.3 Organizationally Specific TLV	MAC/PHY Configuration/Status	<p>Duplex and bit rate settings of the IP phone.</p> <p>The Auto Negotiation is supported and enabled by default.</p> <p>The advertised capabilities of PMD.</p> <p>Auto-Negotiation is: 100BASE-TX (full duplex mode), 100BASE-TX (half duplex mode), 10BASE-T (full duplex mode), or 10BASE-T (half duplex mode).</p>

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

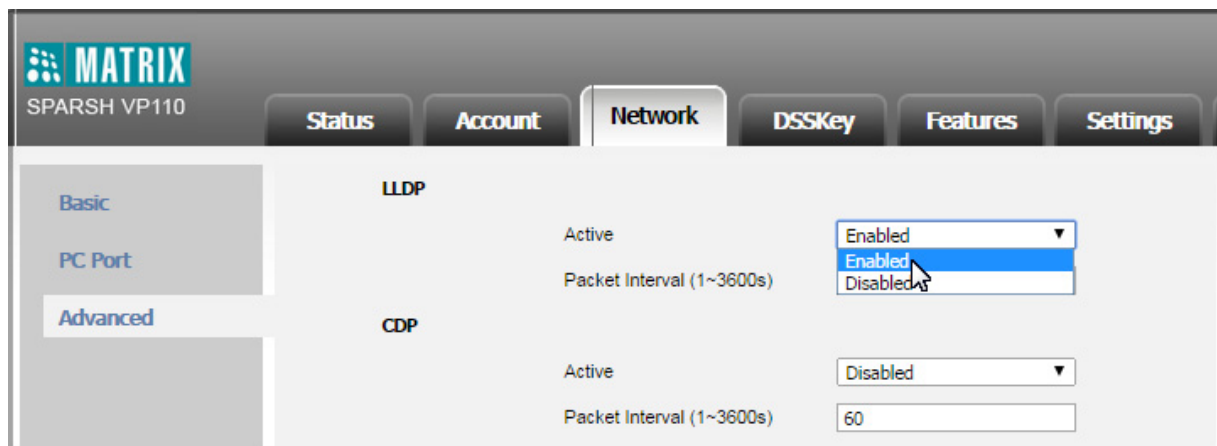
LLDP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure LLDP. For more information, refer “Appendix D - Configuration Parameters” .
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Local	Web User Interface	Configure LLDP. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network-adv&q=load">http://<phoneIPAddress>/servlet?p=network-adv&q=load
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To configure LLDP 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.

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 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 IP phones 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 or DHCP. The assignment takes effect in this order: assignment via LLDP, manual configuration, then assignment via DHCP.

VLAN Discovery via DHCP

IP phones support VLAN discovery via DHCP. When the VLAN Discovery method is set to DHCP, the IP phone will examine 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 VLAN for the Internet port and PC port manually. Configure DHCP VLAN discovery feature. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure VLAN for the Internet port and PC port. Configure DHCP VLAN discovery feature. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network-adv&q=load">http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure VLAN for the Internet port and PC port.

To configure VLAN for Internet port via web user interface:

- Click on **Network->Advanced**.
- In the **VLAN** block, select the desired value from the **WAN Port Active** list.
- Enter the VLAN ID in the **VID (1-4094)** field.
- Select the desired value (0-7) from the **Priority** list.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The left sidebar has a tree view with Basic, PC Port, and Advanced. The main content area is titled 'Network' and contains several configuration blocks: LLDP, CDP, VLAN, and DHCP VLAN. The VLAN block is expanded, showing settings for WAN Port, PC Port, and DHCP VLAN. The WAN Port Active dropdown is open, showing options Disabled, Disabled, and Enabled, with a mouse cursor pointing at Enabled.

Section	Parameter	Value
LLDP	Active	Enabled
	Packet Interval (1~3600s)	60
CDP	Active	Disabled
	Packet Interval (1~3600s)	60
VLAN	WAN Port Active	Enabled
	VID (1-4094)	
	Priority	0
PC Port	Active	Disabled
	VID (1-4094)	1
	Priority	0
DHCP VLAN	Active	Enabled
	Option (1-255)	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 IP phone.

To configure VLAN for PC port via web user interface:

- Click on **Network->Advanced**.
- In the **VLAN** block, select the desired value from the **PC Port 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 DHCP VLAN discovery via web user interface:

- Click on **Network->Advanced**.
- In the **VLAN** block, select the desired value from the **DHCP VLAN Active** list.
- Enter the desired option in the **Option** field.



The default option is 132.

- 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 VLAN for Internet port (or PC port) via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Network->VLAN->WAN Port (or PC Port)**.
- Press  or , or the **Switch** soft key to select the desired value from the **VLAN Status** field.
- Enter the VLAN ID (1-4094) in the **VID** field.
- Enter the priority value (0-7) in the **Priority** field.
- Press the **Save** soft key to save the change
- The IP phone reboots automatically to make settings effective after a period of time.

Quality of Service (QoS)

Quality of Service (QoS) is the ability to provide different priorities for different packets in the network, allowing the transport of traffic with special requirements. QoS guarantees are important for applications that require fixed bit rate and are delay sensitive when the network capacity is insufficient. There are four major QoS factors to be considered when configuring a modern QoS implementation: 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 guarantees 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. 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. Each router on the network can provide QoS simply based on the DiffServ class. 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 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 nodes that implement IP precedence-based classification and forwarding.
- **Expedited Forwarding PHB** -- the key ingredient in DiffServ model for providing a low-loss, low-latency, low-jitter and assured bandwidth service.
- **Assured Forwarding PHB** -- defines a method by which BAs (Bandwidth Allocations) can be given different forwarding assurances.
- **Default PHB** -- 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 not be delayed or dropped due to interference from 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. IP phones support 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.

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 phones should be configured with a high transmission priority.

DSCPs for voice and SIP packets can be specified respectively.

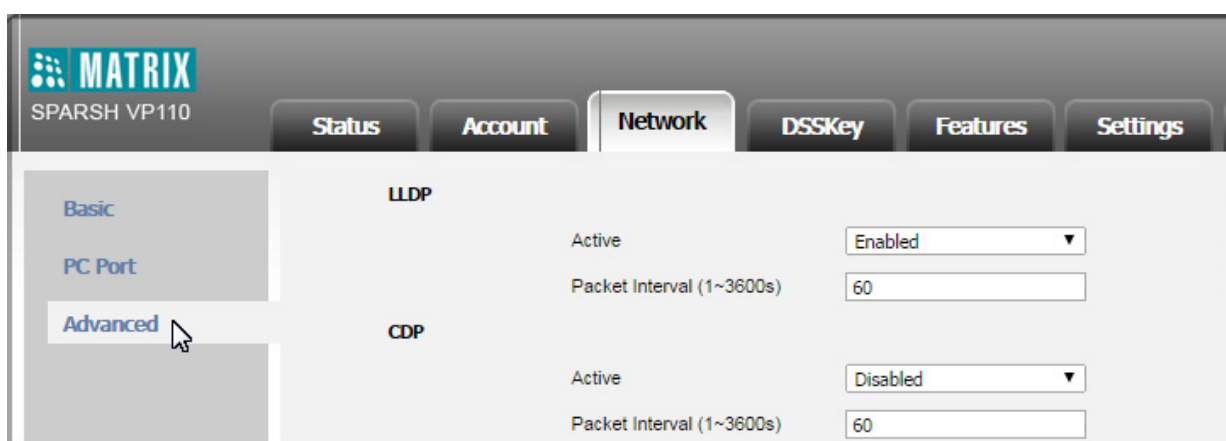
Procedure

QoS can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the DSCPs for voice packets and SIP packets. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure the DSCPs for voice packets and SIP packets. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network-adv&q=load">http://<phoneIPAddress>/servlet?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 **Voice QoS(0~63)** field.

- Enter the desired value in the **SIP QoS(0~63)** field.

Voice QoS	
Voice QoS (0~63)	<input type="text" value="46"/>
SIP QoS (0~63)	<input type="text" value="26"/>
Local RTP Port	
Max RTP Port (1~65535)	<input type="text" value="12780"/>
Min RTP Port (1~65535)	<input type="text" value="11780"/>

- 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.

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 since each outgoing or incoming request must first go through a translation process. But in the VoIP environment, NAT breaks end-to-end connectivity.

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 IP phones.

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 applications to operate behind a NAT to discover the presence of the network address translator, 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.

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 IP phone. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure NAT traversal and STUN server on the IP phone. Navigate to: <code>http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</code>
	Phone User Interface	Configure auto answer.

To configure NAT traversal via web user interface:

- Click on **Account->Register**.

- Select **STUN** from the **NAT** list.

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

To configure 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 **Port** field.

- Click **Confirm** to accept 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 STUN server via phone user interface:

- Press **Menu->Settings->Advanced Settings** (default password:1234) **->Network->NAT->NAT Status**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from 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 **Port** field.
- Press the **Save** soft key to accept the change.

The IP phone reboots automatically to make settings effective after a period of time.

To configure NAT traversal for a specific account via phone user interface:

- Press **Menu->Settings->Advanced Settings** (default password: 1234) ->**Accounts**.
- Press ◀ or ▶ , or the **Switch** soft key to select the desired value from the **NAT Status** field.
- Press the **Save** soft key to accept the change.

802.1X Authentication

IEEE802.1X authentication is an IEEE standard for Port-based Network Access Control (PNAC), part of the 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: a supplicant, an authenticator and an authentication server. The supplicant is the IP phone that wishes to attach to the 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 are valid, the IP phone is allowed to access resources located on the protected side of the network.

IP phones support protocols EAP-MD5, EAP-TLS, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 for 802.1X authentication.

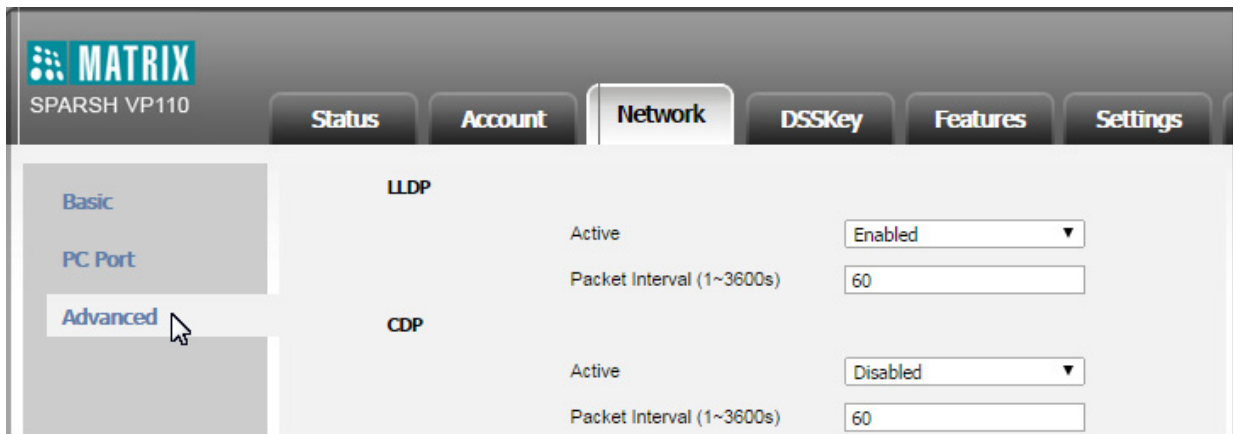
Procedure

802.1X authentication can be configured using the configuration files or locally.

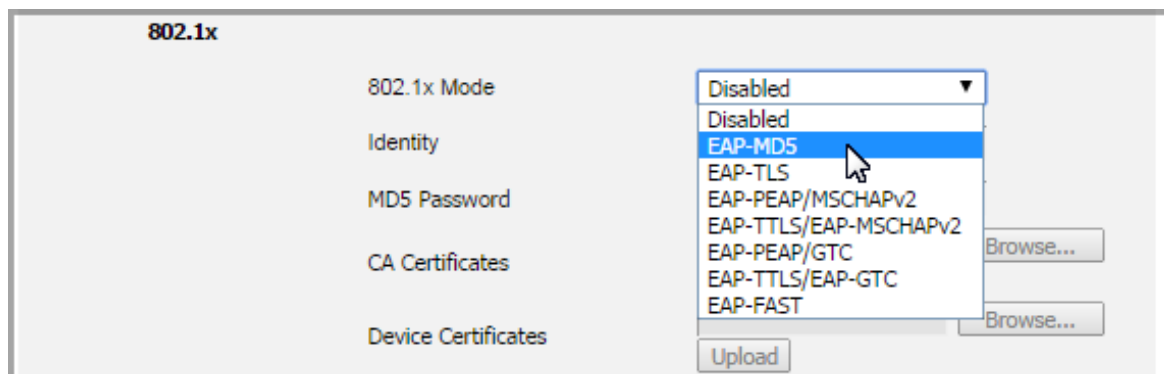
Configuration File	<MAC>.cfg	Configure the 802.1X authentication. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure the 802.1X authentication. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network-adv&q=load">http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure the 802.1X authentication.

To configure the 802.1X authentication 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - In the **Device Certificates** field, click **Browse** to select the desired client (*.pem or *.cer) certificate from your local system.
 - Click **Upload** to upload the certificates.

- 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - Click **Upload** to upload the certificate.

- 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - Click **Upload** to upload the certificate.

- If you select **EAP-TTLS/EAP-MSCHAPv2**:
 - 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - Click **Upload** to upload the certificate.
 - Click **Upload** to upload the certificate.

- 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - Click **Upload** to upload the certificate.



- 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 the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - Click **Upload** to upload the certificate.

- 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:

- Press **Menu->Settings->Advanced Settings** (password: 1234) **->Network->802.1x Settings**.
- Press  or  , or the **Switch** soft key to select the desired value from 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 **PEAP-MSCHAPv2**:
 - 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-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.
- Click **Save** to save the change.

The IP phone reboots automatically to make the settings effective after a period of time.

IPv6 Support

IPv6 is the next generation network layer protocol, designed as a replacement for the current IPv4 protocol. IPv6 is developed by the Internet Engineering Task Force (IETF) to deal with the long-anticipated problem of IPv4 address exhaustion. IPv6 uses a 128-bit address, consisting of eight groups of four hexadecimal digits separated by colons. VoIP network based on IPv6 can ensure QoS, a set of service requirements to deliver performance guarantee while transporting traffic over the network.

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):** 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 connected. 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.

Procedure

IPv6 can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the IPv6 address assignment method. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure the IPv6 address assignment method. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network&q=load">http://<phoneIPAddress>/servlet?p=network&q=load

To configure IPv6 address assignment method via web user interface:

- Click on **Network->Basic**.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account', 'Network' (selected), 'DSSKey', 'Features', and 'Settings'. On the left, a sidebar shows 'Basic' (selected), 'PC Port', and 'Advanced'. The main content area is titled 'Internet Port' and 'IPv4 Config'. Under 'Internet Port', there is a 'Mode(IPv4/IPv6)' dropdown menu set to 'IPv4'. Under 'IPv4 Config', there are two radio buttons: 'DHCP' and 'Static IP Address' (selected). Below these, there are input fields for 'IP Address' (containing '192.168.101.145') and 'Subnet Mask' (containing '255.255.255.0').

- Select the desired address mode (IPv6 or IPv4&IPv6) from the **Mode (IPv4/IPv6)** list.
- In the **IPv6 Config** block, do one of the following:
 - If you click **Static IP Address**, configure the IPv6 address and other configuration parameters in the corresponding fields.

The screenshot shows the 'IPv6 Config' dialog box. It has two radio buttons: 'DHCP' and 'Static IP Address' (selected). Below the radio buttons, there are input fields for 'IP Address', 'IPv6 Prefix(0~128)' (containing '64'), and 'Gateway'. There are also radio buttons for 'IPv6 Static DNS' set to 'On'. Below these are input fields for 'Primary DNS' and 'Secondary DNS'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

- (Optional.) If you click **DHCP**, you can configure the static DNS address in the corresponding fields.

IPv6 Config

☒ DHCP
☐ Static IP Address

IP Address

IPv6 Prefix(0~128)

Gateway

IPv6 Static DNS ☒ On ☐ Off

Primary DNS

Secondary DNS

- 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 IPv6 address assignment method via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: **1234**) **->Network->WAN Port**.
- Press **◀** or **▶** to select **IPv4&IPv6** or **IPv6** from the **IP Mode** field.
- Press **▲** or **▼** to highlight **IPv6** and press the **Enter** soft key.
- Press **▲** or **▼** to select the desired IPv6 address assignment method.
- If you select the **Static IPv6 Client**, configure the IPv6 address and other network parameters in the corresponding fields.
- Press the **Save** soft key to save the change.

The IP phone reboots automatically to make settings effective after a period of time.

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

- 1. Press **Menu->Settings->Advanced Settings** (password: **1234**) **->Network->WAN Port.->IPv6->DHCP IPv6 Client**.
- Press **◀** or **▶** or the **Switch** soft key to select **Enabled** from the **Static DNS** field.
- Enter the desired value in the **IPv6 Pri.DNS** and **IPv6 Sec.DNS** field respectively.

- Press the **Save** soft key to accept the change.

The IP phone reboots automatically to make settings effective after a period of time.

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”](#).

Headset Mode Activation/De-activation

To activate the headset mode:

- Press  on the phone.

The headset icon on the idle screen indicates that the headset mode is activated. Press the **Answer** soft key to answer a call. The call will connect to your headset automatically. Enter the desired number and press the **Send** soft key, then the phone will place a call using the headset automatically. For more information on using the headset to place a call, refer [“Placing Calls”](#).

To deactivate the headset mode:

- Press  again on the phone.

The headset icon disappears from the idle screen indicates the headset mode is deactivated.

Headset Prior

Headset prior allows users 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.

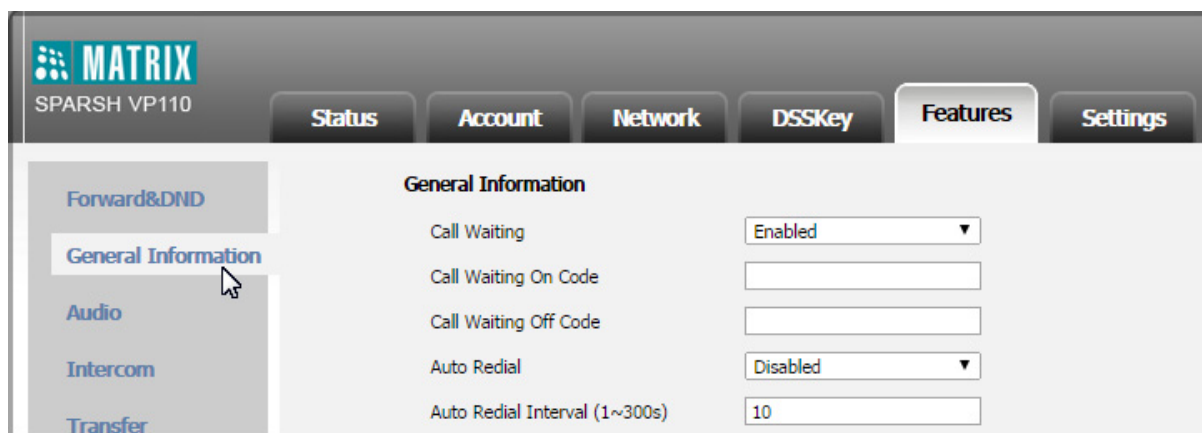
Procedure

Headset prior can be configured using the configuration files or locally.

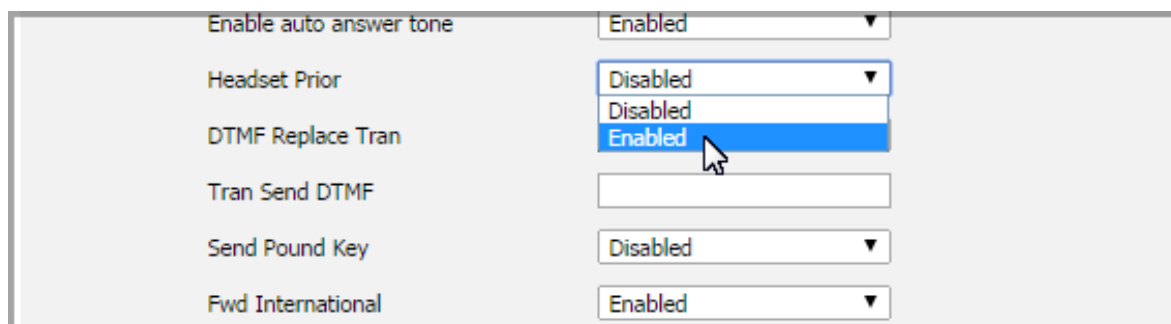
Configuration File	<MAC>.cfg	Configure headset prior. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure headset prior. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load

To enable headset prior via web user interface:

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




- Select **Enabled** from the **Headset Prior** list.



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

To use headset prior feature, 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.*
- *Headset prior is configurable via web user interface only.*

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 object 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 default codecs used on IP phones are summarized in the following table:

Codec	Algorithm	Bit Rate	Sample Rate	Packetization Time
PCMA	G.711 a-law	64 Kbps	8 Ksps	20ms
PCMU	G.711 μ -law	64 Kbps	8 Ksps	20ms
G729	G.729	8 Kbps	8 Ksps	20ms
G722	G.722	64 Kbps	16 Ksps	20ms

In addition to the codecs introduced above, IP phones also support codecs: G726_16, G726_24, G726-32, G726-40. Codecs and priorities of these codecs are configurable. 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 are listed as follows:

Codec	Configuration Methods	Priority	RTPmap
G722	Configuration Files Web User Interface	1	9
PCMU	Configuration Files Web User Interface	2	0
PCMA	Configuration Files Web User Interface	3	8
G729	Configuration Files Web User Interface	4	18
G726_16	Configuration Files Web User Interface	0	103
G723_24	Configuration Files Web User Interface	0	104

G726-32	Configuration Files Web User Interface	0	102
G726-40	Configuration Files Web User Interface	0	105
iLBC	Configuration Files Web User Interface	0	106

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.

Procedure




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

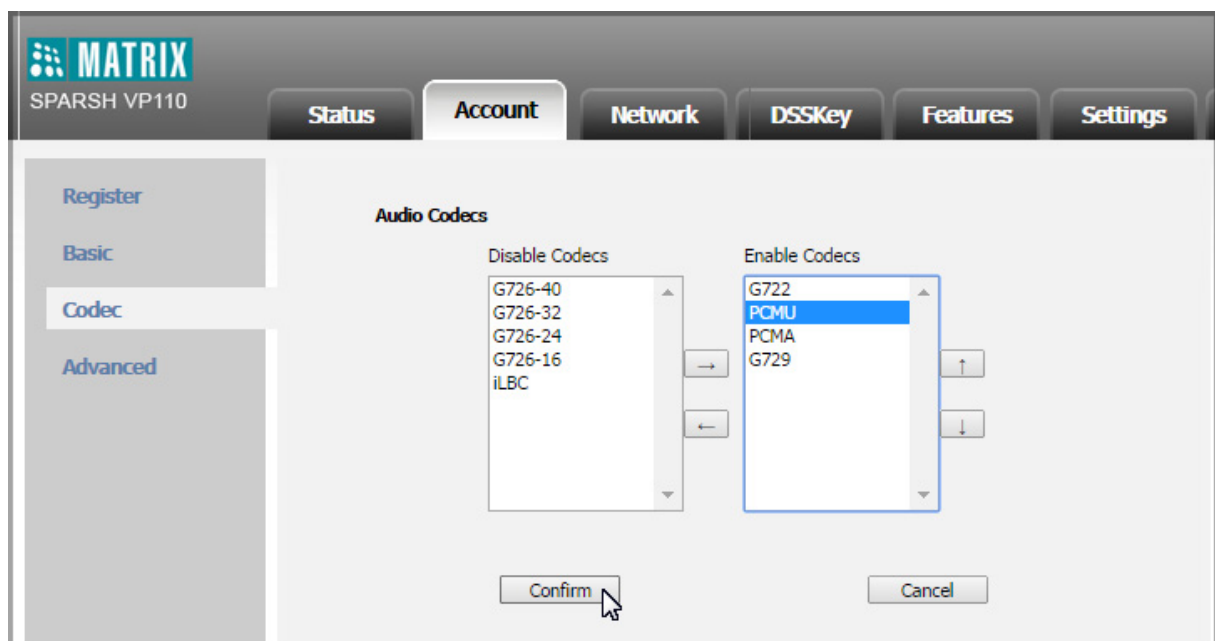
Configuration File	<MAC>.cfg	<p>Configure the codecs to use.</p> <p>Configure the priority and rtpmap for the enabled codec.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p> <p>Configure the ptime.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure the codecs to use and adjust the priority of the enabled codecs.</p> <p>Configure the ptime.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-codec&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-codec&q=load&acc=0</p>

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 codec from the **Disable Codecs** column and then click  .

The selected codec appears in the **Enable Codecs** column.

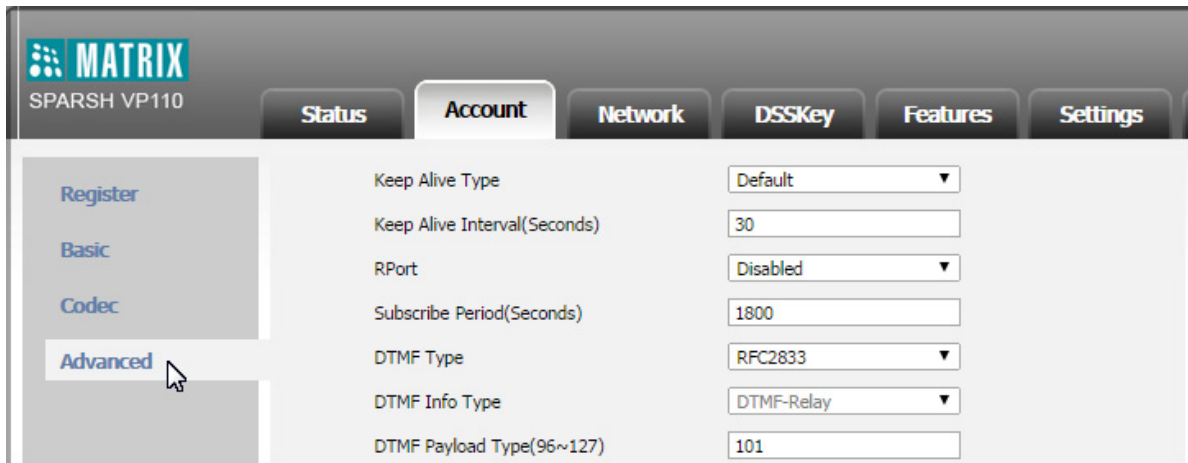
- Repeat the above step to add more codecs to the **Enable Codecs** column.
- To remove the codec from the **Enable Codecs** column, select the desired codec and then click  .
- To adjust the priority of codecs, select the desired codec and then click  or  .



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

To configure the ptime via web user interface:

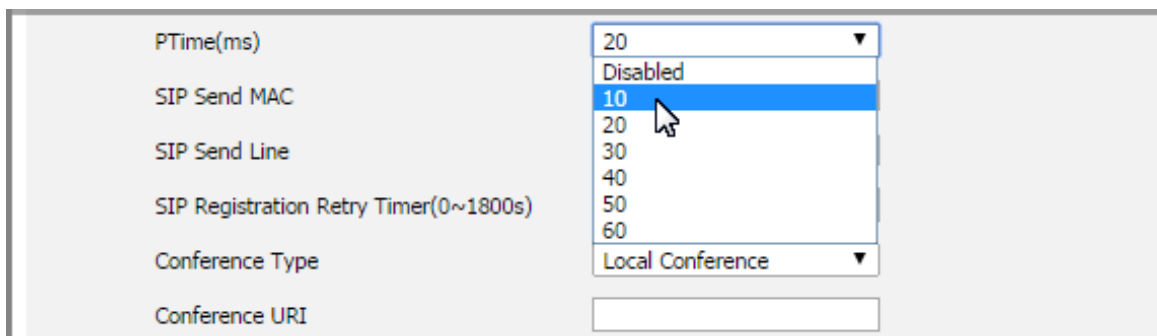
- Click on **Account** -> **Advanced**.



The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The 'Account' tab is selected, and the 'Advanced' sub-tab is active in the left sidebar. The main content area displays several configuration parameters:

Parameter	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101

- Select the desired value from the **PTime (ms)** list.



This screenshot shows a close-up of the 'PTime(ms)' dropdown menu. The menu is open, displaying a list of options: 20, Disabled, 10, 20, 30, 40, 50, and 60. The option '10' is currently selected and highlighted in blue. Below the list, the 'Local Conference' dropdown is also visible, set to 'Local Conference'.

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

Acoustic Clarity Technology

Acoustic Clarity Technology involves the following:

- Acoustic Echo Cancellation
- Voice Activity Detection
- Comfort Noise Generation
- Jitter Buffer

Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) is used to remove 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 a network. IP phones employ advanced AEC for hands-free operation. Echo cancellation is achieved using the echo canceller.



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 AEC. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure AEC. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure AEC via web user interface:

- Click on **Settings->Voice**.
- Select the desired value from the **ECHO** list.

The screenshot shows the Matrix SPARSH VP110 web interface. The 'Settings' tab is selected, and the 'Voice' sub-tab is active. On the left, a sidebar lists various settings categories: Preference, Time & Date, Call Display, Upgrade, Auto Provision, Configuration, Dial Plan, Voice (selected), and Ring. The main content area is titled 'Echo Cancellation' and contains three dropdown menus: 'ECHO' (set to 'Enabled'), 'VAD' (set to 'Enabled'), and 'CNG' (set to 'Enabled'). Below these is the 'JITTER BUFFER' section, which includes a 'Type' radio button group with 'Adaptive' selected and 'Fixed' unselected. There are also three input fields: 'Min Delay' (60), 'Max Delay' (240), and 'Normal' (120). At the bottom of the main content area are 'Confirm' and 'Cancel' buttons.

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

Voice Activity Detection

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 can facilitate 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 VAD. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?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 the line has released. The purpose of VAD and CNG is to maintain an acceptable perceived QoS while simultaneously keeping transmission costs and bandwidth usage as low as possible.

Procedure

CNG can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure CNG. For more information, refer "Appendix D - Configuration Parameters" .
Local	Web User Interface	Configure CNG. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?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 is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. 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. IP phones support two types of jitter buffers: fixed and adaptive. A fixed jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. A 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 be also configured on IP phones.

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. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure Jitter Buffer via web user interface:

- Click on **Settings->Voice**.
- Click the desired option in **Type**.
- Enter the minimum delay time for adaptive jitter buffer in the **Min Delay** field.

Valid values range from 0 to 300.

- Enter the maximum delay time for adaptive jitter buffer in the **Max Delay** field.

Valid values range from 0 to 300.

- Enter the fixed delay time for fixed jitter buffer in the **Normal** field.

Valid values range from 0 to 300.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features **Settings**

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring

Echo Cancellation

ECHO Enabled ▼

VAD Disabled ▼

CNG Enabled ▼

JITTER BUFFER

Type ☒ Adaptive ☐ Fixed

Min Delay 60

Max Delay 240

Normal 120

Confirm Cancel

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

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

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
- DHE-RSA-AES128-SHA
- DHE-DSS-AES128-SHA
- AES128-SHA
- IDEA-CBC-SHA
- DHE-DSS-RC4-SHA
- RC4-SHA
- RC4-MD5
- EXP1024-DHE-DSS-DES-CBC-SHA
- EXP1024-DES-CBC-SHA
- EDH-RSA-DES-CBC-SHA
- EDH-DSS-DES-CBC-SHA
- DES-CBC-SHA
- 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-RC4-MD5

The following figure illustrates the TLS messages exchanged between the IP phone and TLS server to establish an encrypted communication channel:

No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.3.86	192.168.0.230	SSLv3	Client Hello
2	0.021345	192.168.0.230	192.168.3.86	SSLv3	Server Hello, Certificate, Server Key Exchange, Server Hello Done
3	0.954947	192.168.3.86	192.168.0.230	SSLv3	Client Key Exchange, Change Cipher Spec, Encrypted Handshake Message
4	0.970099	192.168.0.230	192.168.3.86	SSLv3	Change Cipher Spec, Encrypted Handshake Message
5	1.012195	192.168.3.86	192.168.0.230	SSLv3	Application Data, Application Data
6	1.013562	192.168.0.230	192.168.3.86	SSLv3	Application Data
7	1.013667	192.168.0.230	192.168.3.86	SSLv3	Application Data

Frame 13: 652 bytes on wire (5216 bits), 652 bytes captured (5216 bits) on interface 0
 # Ethernet II, Src: Vmware_72:c9:2e (00:0c:29:72:c9:2e), Dst: xiaomeng_11:12:b7 (00:15:65:11:12:b7)
 # Internet Protocol, Src: 192.168.0.230 (192.168.0.230), Dst: 192.168.3.86 (192.168.3.86)
 # Transmission Control Protocol, Src Port: https (443), Dst Port: nssserver (2244), Seq: 1482, Ack: 437, Len: 586
 # Secure Socket Layer

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 phones 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 LCD 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	<p>Configure TLS.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
	<MAC>.cfg	<p>Configure trusted certificates feature.</p> <p>Configure server certificates feature.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p> <p>Upload the trusted certificates.</p> <p>Upload the server certificates.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure TLS.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</p> <p>Configure trusted certificates feature.</p> <p>Upload the trusted certificates.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=trusted-cert&q=load">http://<phoneIPAddress>/servlet?p=trusted-cert&q=load</p> <p>Configure server certificates feature.</p> <p>Upload the server certificates.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=server-cert&q=load">http://<phoneIPAddress>/servlet?p=server-cert&q=load</p>

To configure TLS via web user interface:

- Click on **Account->Register**.

- Select **TLS** from the **Transport** list.

The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The Account tab is active. On the left, there is a sidebar with links for Register, Basic, Codec, and Advanced. The main content area displays configuration fields for a SIP account. The 'Transport' dropdown menu is open, showing the following options: UDP, TCP, TLS (highlighted with a mouse cursor), and DNS-NAPTR. Other visible fields include Register Status (Registered), Line Active (Enabled), Register (Enabled), Label (3301), Display Name (3301), Register Name (3301), User Name (3301), Password (masked with dots), SIP Server 1 (Server Host: 192.168.101.146, Port: 5060), and SIP Server 2.

- 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.

MATRIX
SPARSH VP110

Status **Account** **Network** **DSSKey** **Features** **Settings**

Trusted Certificates

Index ID	Issued To	Issued By	Expiration	Delete
1				<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>
10				<input type="checkbox"/>

Only Accept Trusted Certificates

Common Name Validation

CA Certificates

Import Trusted Certificates

Load trusted certificates file No file chosen

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

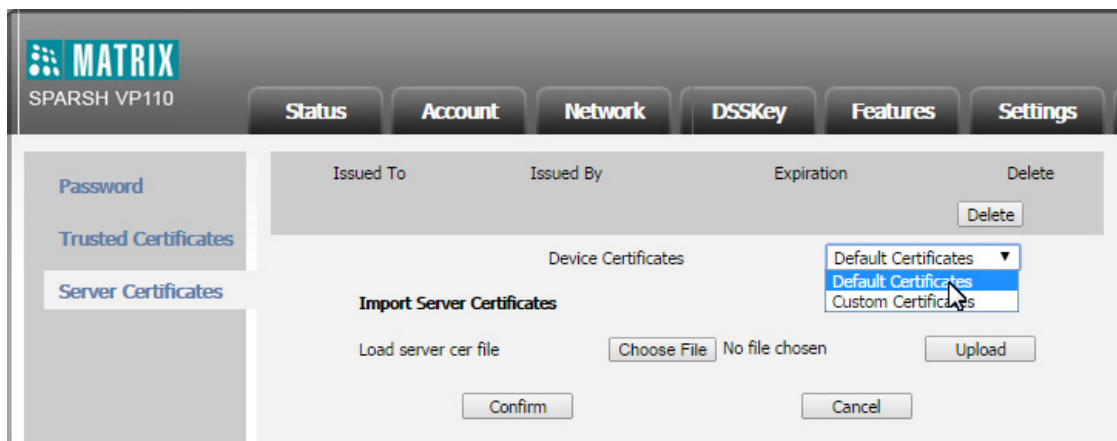
To upload a trusted certificate via web user interface:

- 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 11780 RTP/SAVP 0 8 18 9 101
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NzFINTUwZDk2OGVIOTc3YzNkYTkwZWVkbMTM1YWFj
a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:NzkyM2FjNzQ2ZDgxYjg0MzQwMGVmMGUxMzdmNWFm
a=crypto:3 F8_128_HMAC_SHA1_80 inline:NDliMWIzZGE1ZTAwZjA5ZGFhNjQ5YmEANTMzYzA0
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:9 G722/8000
a=fmtp:9 0-15
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

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 11780 RTP/SAVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NGY4OGViMDYzZjQzYTNiOTNkOWRiYzRlMjM0Yzcz

a=sendrecv

a=ptime:20

a=fmtp:101 0-15

```

When SRTP is enabled on both IP phones, RTP streams will be encrypted, and a lock icon appears on the LCD 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”](#).

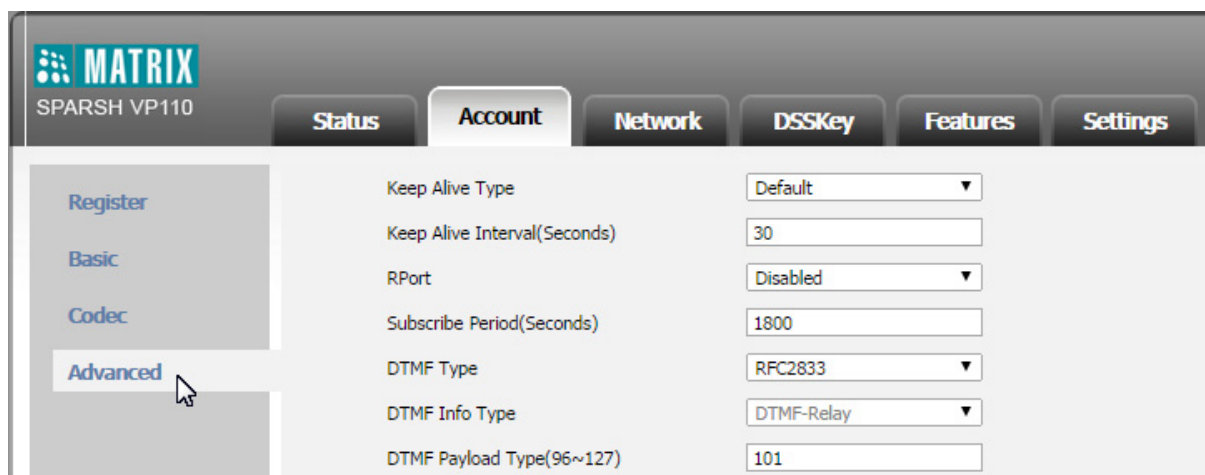
Procedure

SRTP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Configure SRTP feature.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Local	Web User Interface	<p>Configure SRTP feature.</p> <p>Navigate to:</p> <p><code>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</code></p>

To configure SRTP feature via web user interface:

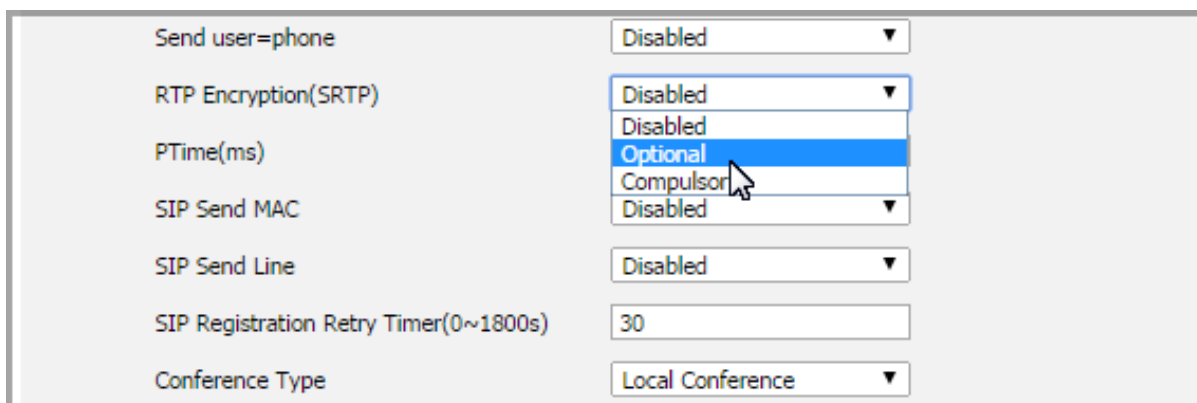
- Click on **Account** -> **Advanced**.



The screenshot shows the Matrix SPARSH VP110 web interface. The top navigation bar includes 'Status', 'Account' (selected), 'Network', 'DSSKey', 'Features', and 'Settings'. On the left, a sidebar contains 'Register', 'Basic', 'Codec', and 'Advanced' (highlighted with a mouse cursor). The main content area displays the following settings:

Parameter	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
RPort	Disabled
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101

- Select the desired value from the **RTP Encryption(SRTP)** list.



This screenshot shows a close-up of the 'RTP Encryption(SRTP)' dropdown menu. The menu is open, showing the following options: 'Disabled', 'Disabled', 'Optional' (highlighted with a blue background and a mouse cursor), 'Compulsory', and 'Disabled'. The other settings visible in the background are:

Parameter	Value
Send user=phone	Disabled
PTime(ms)	
SIP Send MAC	Disabled
SIP Send Line	Disabled
SIP Registration Retry Timer(0~1800s)	30
Conference Type	Local Conference

- 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 VP110 (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 the decryption method. Configure AES keys. For more information, refer “Appendix D - Configuration Parameters” .
Local	Web User Interface	Configure AES keys. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-autop&q=load">http://<phoneIPAddress>/servlet?p=settings-autop&q=load

To configure AES keys via web user interface:

- Click on **Settings->Auto Provision**.
- Enter the values in the **MAC-Oriented AES Key** fields.

AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z.



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.

The screenshot shows the 'MATRIX SPARSH VP110' web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', and 'Settings'. The left sidebar lists various settings categories: 'Preference', 'Time & Date', 'Call Display', 'Upgrade', 'Auto Provision' (selected), 'Configuration', 'Dial Plan', 'Voice', 'Ring', and 'Tones'. The main content area is titled 'Auto Provision' and contains the following settings:

Setting	Value
PNP Active	<input checked="" type="radio"/> On <input type="radio"/> Off
DHCP Active	<input checked="" type="radio"/> On <input type="radio"/> Off
Custom Option(128~254)	<input type="text"/>
DHCP Option Value	MATRIX SPARSH
Server URL	<input type="text"/>
User Name	<input type="text"/>
Password	<input type="password"/>
Attempt Expired Time(s)	5
Common AES Key	<input type="password"/>
MAC-Oriented AES Key	<input type="password"/>
Zero Active	Disabled
Wait Time(1~100s)	5

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

This chapter describes the following features in details:

- Upgrading Firmware
- SNMP
- 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 VP110	53.80.196.x.rom	53.80.196.x.rom



If your phone firmware is 31.72.196.x.rom, then it is not upgradable.

Do not unplug the network and power cables when the IP phone is upgrading firmware.

Upgrade via Web User Interface

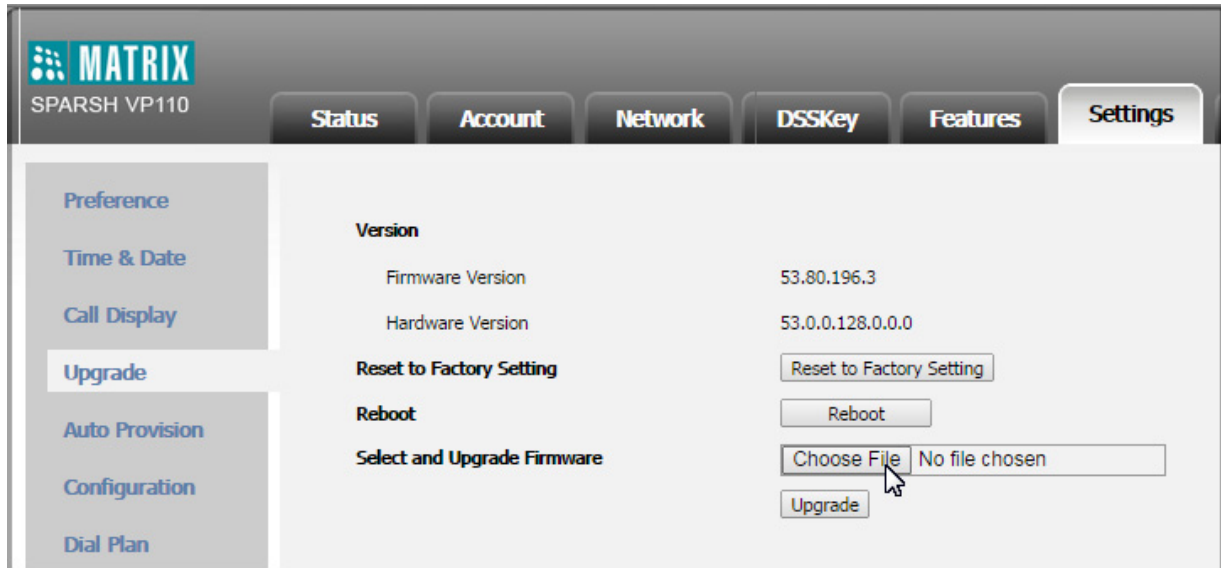
To manually upgrade firmware via web user interface, you need to store firmware to your local system in advance.

To upgrade firmware manually via web user interface:

- Click on **Settings->Upgrade**.

- Click **Browse**.
- Select firmware from the local system.
- Click **Upgrade**.

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	<p>Configure the way for the IP phone to check for configuration files.</p> <p>Specify the access URL of firmware.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure the way for the IP phone to check for configuration files.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=settings-autop&q=load</p>

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).

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The left sidebar lists various configuration options: Preference, Time & Date, Call Display, Upgrade, Auto Provision (highlighted with a mouse cursor), Configuration, Dial Plan, Voice, Ring, Tones, Softkey Layout, TR069, and Voice Monitoring. The main content area displays the 'Auto Provision' settings, which include:

- PNP Active:** Radio buttons for On (selected) and Off.
- DHCP Active:** Radio buttons for On (selected) and Off.
- Custom Option(128~254):** An empty text input field.
- DHCP Option Value:** A text input field containing 'MATRIX SPARSH'.
- Server URL:** An empty text input field.
- User Name:** An empty text input field.
- Password:** A text input field with masked characters (dots).
- Attempt Expired Time(s):** A text input field containing '5'.
- Common AES Key:** A text input field with masked characters (dots).
- MAC-Oriented AES Key:** A text input field with masked characters (dots).
- Zero Active:** A dropdown menu set to 'Disabled'.
- Wait Time(1~100s):** A text input field containing '5'.
- Power On:** Radio buttons for On (selected) and Off.
- Repeatedly:** Radio buttons for On and Off (selected).
- Interval(Minutes):** A text input field containing '1440'.
- Weekly:** Radio buttons for On and Off (selected).

- 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.

SNMP

SNMP (Simple Network Management Protocol) is an Internet-standard protocol for managing devices on IP networks. It is used mostly in network management systems to monitor network-attached devices for conditions that warrant administrative attention. SNMP exposes management data in the form of variables on the managed systems, which describe the system configuration. These variables can then be queried (and sometimes set) by managing applications. The variables accessible via SNMP are organized in hierarchies, which are described by Management Information Bases (MIBs).

IP phones only support SNMPv1 and SNMPv2. They act as SNMP clients, receiving requests from the SNMP server. The SNMP server may send requests from any available source port to the configured port on the client, while the client responds to the source port on the SNMP server. IP phones only support the GET request from the SNMP server.

The following table lists the basic object identifiers (OIDs) supported by IP phones.

OID	Description
1.3.6.1.2.1.37459.2.1.1.0	The textual identification of the contact person for the IP phone, together with the contact information. For example, Sysadmin (root@localhost)
1.3.6.1.2.1.37459.2.1.2.0	An administratively-assigned name for the IP phone. If the name is unknown, the value is a zero-length string. For example, IPPHONE
1.3.6.1.2.1.37459.2.1.3.0	The physical location of the IP phone. For example, Server Room
1.3.6.1.2.1.37459.2.1.4.0	The time (in milliseconds) since the network management portion of the system was last re-initialized.
1.3.6.1.2.1.37459.2.1.5.0	The firmware version of the IP phone.
1.3.6.1.2.1.37459.2.1.6.0	The hardware version of the IP phone.
1.3.6.1.2.1.37459.2.1.7.0	The IP phone's model.
1.3.6.1.2.1.37459.2.1.8.0	The MAC address of the IP phone.
1.3.6.1.2.1.37459.2.1.9.0	The IP address of the IP phone.
1.3.6.1.2.1.37459.2.1.10.0	The target version to which the current version is automatically updated. Format: MacVersion[*]ComVersion[*] For example, MacVersion[0.0.0.1]ComVersion[0.0.0.1]

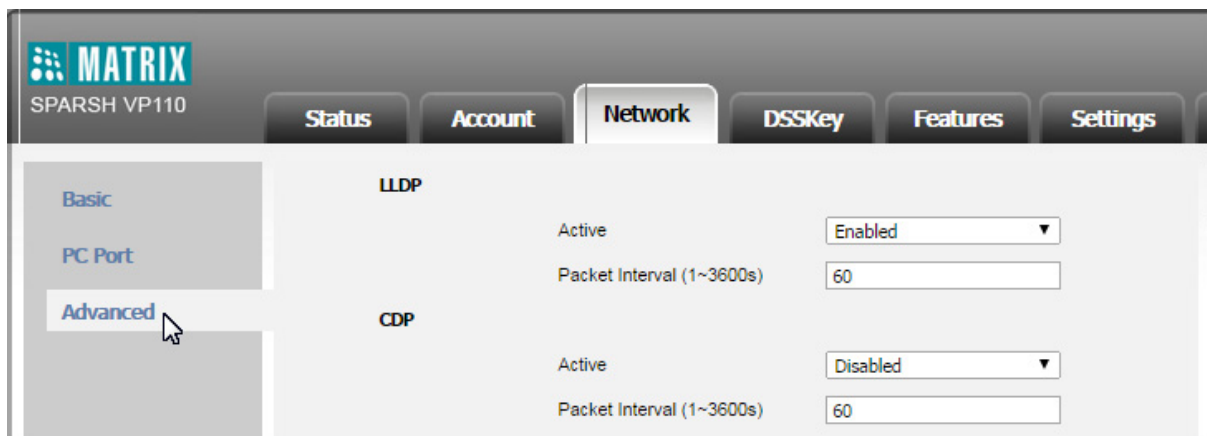
Procedure

SNMP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Enable or disable the SNMP feature.</p> <p>Configure the SNMP port.</p> <p>Configure IP address(es) or domain name of the trusted SNMP server.</p> <p>Multiple IP addresses should be separated by space.</p> <p>For more information, refer "Appendix D - Configuration Parameters".</p>
Local	Web User Interface	<p>Configure SNMP feature.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=network-adv&q=load</p>

To configure SNMP feature via web user interface:

- Click on **Network ->Advanced**.



- In the **SNMP** block, select **Enabled** from the **Active** list.
- Enter the SNMP port in the **Port (1~65535)** field.
- Enter the IP address or domain name of the SNMP server in the **Trusted Address** field.

Multiple IP addresses should be separated by space.

SNMP	
Active	Enabled ▼
Port (1~65535)	161
Trusted Address	

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

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

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

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: <ul style="list-style-type: none">• 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: <ul style="list-style-type: none">• 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: <a href="http://<phoneIPAddress>/servlet?p=settings-tr069&q=load">http://<phoneIPAddress>/servlet?p=settings-tr069&q=load

To configure TR-069 via web user interface:

- Click on **Settings->TR069**.

The screenshot shows the Matrix SPARSH VP110 web interface. At the top, there's a header with the Matrix logo and 'SPARSH VP110'. Below the header is a navigation bar with tabs: Status, Account, Network, DSSKey, Features, and Settings. The Settings tab is selected. On the left side, there's a sidebar menu with various settings categories: Preference, Time & Date, Call Display, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, Ring, Tones, Softkey Layout, TR069, and Voice Monitoring. The TR069 option is highlighted with a mouse cursor. The main content area displays the TR069 configuration page. It has a title 'TR069' and several configuration fields: 'Enable TR069' (a dropdown menu set to 'Disabled'), 'ACS Username' (a text input field), 'ACS Password' (a password input field with dots), 'ACS URL' (a text input field), 'Enable Periodic Inform' (a dropdown menu set to 'Enabled'), 'Periodic Inform Interval (seconds)' (a text input field with '60'), 'Connection Request Username' (a text input field), and 'Connection Request Password' (a password input field with dots). At the bottom of the configuration area, there are 'Confirm' and 'Cancel' buttons.

- 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.

IP phones 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
- Analyzing Configuration File



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:

- **syslog.mode:** Specify the system log to be exported to a server or local system.
- **syslog.server:** Specify the IP address or domain name of the syslog server to which the log will be exported.
- **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

For more information on the system log setting parameters, refer [“Appendix D - Configuration Parameters”](#).

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.

The screenshot shows the MATRIX SPARSH VP110 web interface. The top navigation bar includes tabs for Status, Account, Network, DSSKey, Features, and Settings. The left sidebar lists various configuration categories: Preference, Time & Date, Call Display, Upgrade, Auto Provision, Configuration (highlighted), Dial Plan, Voice, Ring, Tones, Softkey Layout, TR069, Voice Monitoring, and SIP. The main content area is the 'Configuration' section, specifically the 'System Log Level' setting. The 'System Log Level' dropdown menu is open, displaying a list of values: 0, 1, 2, 3, 4 (highlighted), 5, and 6. A mouse cursor is pointing at the value 4. Other settings visible include 'Export or Import Configuration' with 'Choose File' and 'No file chosen' buttons, 'Export' and 'Import' buttons; 'Export CFG Configuration File' with a 'Local Configuratic' dropdown and 'Export' button; 'Import CFG Configuration File' with 'Choose File', 'No file chosen', 'Local Configuratic' dropdown, and 'Import' button; 'Pcap Feature' with 'Start', 'Stop', and 'Export' buttons; and 'Export System Log' with radio buttons for 'Local' (selected), 'Server', and 'Ftp/Tftp Server', and an 'Export' button. A 'Confirm' button is at the bottom of the configuration area.

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

A dialog box pops up to prompt “Do you want to restart your machine?”. The configuration will take effect after a reboot.

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

After a reboot, the system log level is set as 6, the informational level.



Informational level may make some sensitive information accessible (e.g., password-dial number), we recommend that you reset the system log level to 3 after having the syslog file provided.

To configure the phone to export the system log to a syslog server via web user interface:

- Click on **Settings->Configuration**.
- Click **Server** 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 “Do you want to restart your machine?”. 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.

To export a log file to the local system via web user interface:

- Click on **Settings->Configuration**.
- Click **Local** in **Export System Log**.
- Reproduce the issue.

- Click **Export** to open file download window, and then save the file to your local system.

MATRIX
SPARSH VP110

Status Account Network DSSKey Features **Settings**

Preference
Time & Date
Call Display
Upgrade
Auto Provision
Configuration
Dial Plan
Voice
Ring
Tones
Softkey Layout
TR069

Export or Import Configuration No file chosen

Export CFG Configuration File Local Configuratic

Import CFG Configuration File No file chosen
Local Configuratic

Pcap Feature

Export System Log ☒ Local ☐ Server ☐ Ftp/Tftp Server

System Log Level 3

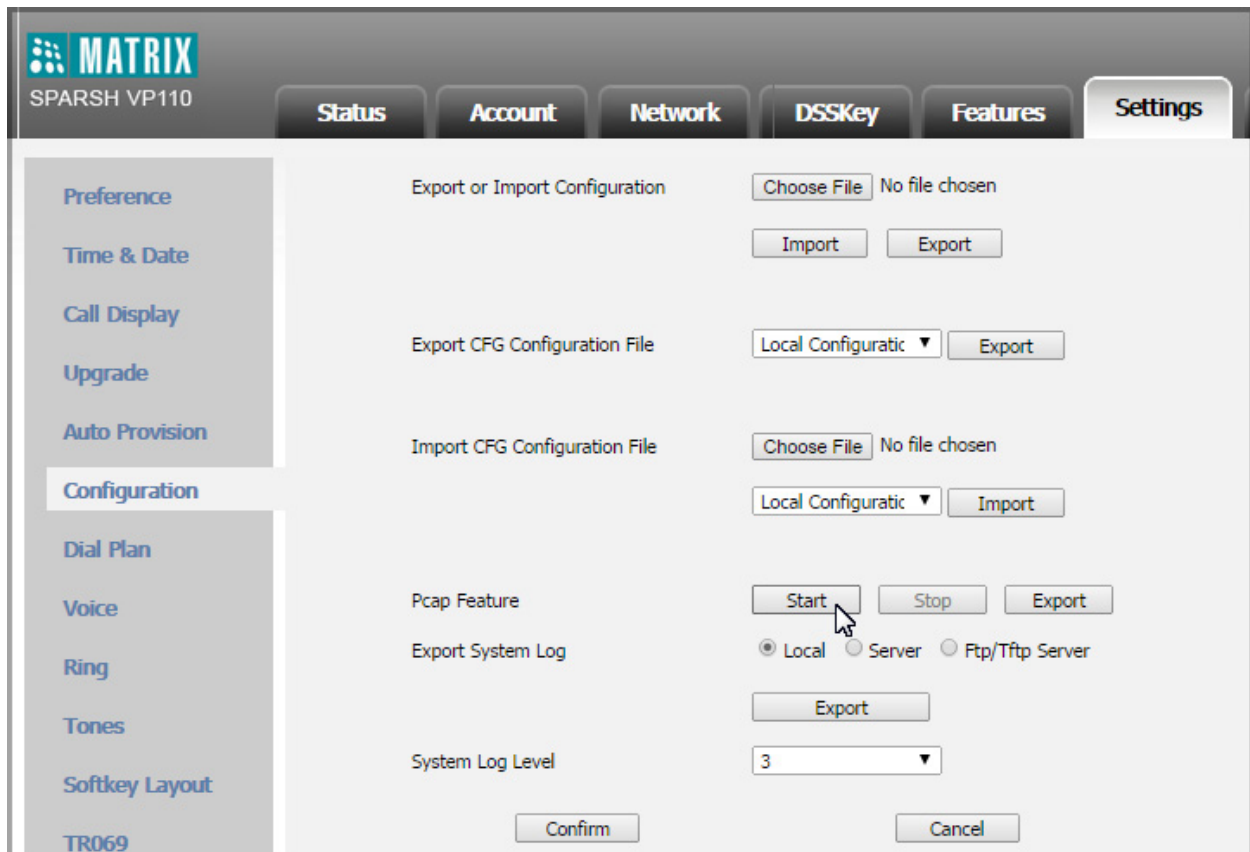
Capturing Packets (Packet Capture)

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.

To capture packets via web user interface:

- Click on **Settings->Configuration**.
- On **Pcap Feature**, 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.



To capture packets using the Ethernet software:

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.

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.

You can use the “watch_dog.enable” parameter to configure watchdog feature in the configuration files. For more information, refer [“Appendix D - Configuration Parameters”](#).

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.

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.

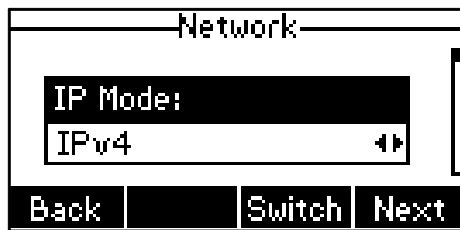
- Configure the wait time in the **Wait Time (1~100s)** field.

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

When Zero Touch is enabled, there will be a configuration wizard during startup as shown below

- Press the **OK** soft key.

The network parameters are configurable via phone user interface:



- Press the **Next** soft key after finishing network setting.

Configure the provisioning server address, authentication user name (optional) and password (optional) in the **Auto Provision** screen.


An example of screen-shot is shown as below:



Getting Information from Status Indicators

Status indicators may consist of the Power LED and the on-screen icon(s).

The following shows two examples of obtaining the phone information from status indicators:

- If a LINK failure of the IP phone is detected, a prompting message “Network Unavailable” and the  icon will appear on the LCD screen.
- If a voice mail is received, the Power LED illuminates.

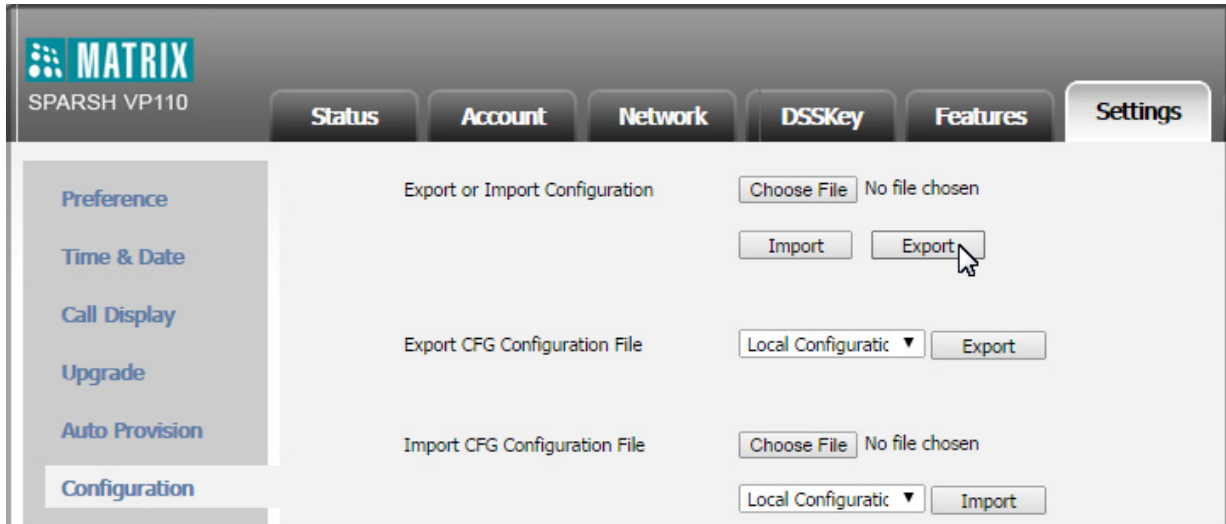
For more information on the icons, refer [“Icon Instructions”](#).

Analyzing Configuration File

Wrong configurations may have an impact on your phone use. You can export configuration file to check the current configuration of the IP phone and troubleshoot if necessary.

To export configuration file via web user interface:

- Click on **Settings->Configuration**.
- In the **Export or Import Configuration** block, click **Export** to open the file download window, and then save the file to your local system.



When configuring particular features, you may need to upload resource files (e.g., local contact directory, remote phone book) to IP phones. The resources files can be local contact directory, remote phone book and so on. Ask Matrix field application engineer for resource file templates. To enable the desired phone to use the resource file(s), the resource file access URL should be programmed in the <MAC>.cfg file.

This chapter provides the detailed information on how to customize the following resource files and specify the access URL:

- Replace Rule Template
- Dial-now Template
- Softkey Layout Template
- Directory Template
- Super Search Template
- Local Contact File
- Remote XML Phone Book
- Specifying the Access URL of Resource Files

Replace Rule Template

The replace rule template helps with the creation of multiple replace rules. After setup, place the replace rule template to the provisioning server and specify the access URL in the configuration files.

When editing a replace rule template, learn the following:

- <DialRule> indicates the start of a template and </DialRule> indicates the end of a template.
- Create replace rules between <DialRule> and </DialRule>.
- At most 100 replace rules can be added to the IP phone.
- The expression syntax in the replace rule template is the same as that introduced in the section [“Dial Plan”](#).

Procedure

Use the following procedures to customize a replace rule template.

To customize a replace rule template:

- Open the template file using an ASCII editor.
- Add the following string to the template, each starting on a separate line:

```
<Data Prefix="" Replace="" LineID=""/>
```

Where:

Prefix="" specifies the numbers to be replaced.

Replace="" specifies the alternate string instead of what the user enters.

LineID="" specifies the desired line(s) for this rule. Leave it blank or enter 0, to apply this replace rule to all lines.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of a replace rule template:

```
<DialRule>
  <Data Prefix="1" Replace="05928665234" LineID=""/>
  <Data Prefix="2(xx)" Replace="002$1" LineID="0"/>
  <Data Prefix="5([6-9])(.)" Replace="3$2" LineID="0"/>
  <Data Prefix="0(.)" Replace="9$1" LineID="0"/>
  <Data Prefix="1009" Replace="05921009" LineID="0"/>
</DialRule>
```

Dial-now Template

The dial-now template helps with the creation of multiple dial-now rules. After setup, place the dial-now template to the provisioning server and specify the access URL in the configuration files.

When editing a dial-now template, learn the following:

- <DialNow> indicates the start of a template and </DialNow> indicates the end of a template.
- Create dial-now rules between <DialNow> and </DialNow>.
- At most 100 rules can be added to the IP phone.
- The expression syntax in the dial-now rule template is the same as that introduced in the section [“Dial Plan”](#).

Procedure

Use the following procedures to customize a dial-now template.

To customize a dial-now template:

- Open the template file using an ASCII editor.
- Add the following string to the template, each starting on a separate line:

```
<Data DialNowRule="" LineID=""/>
```

Where:

DialNowRule="" specifies the dial-now rule.

LineID="" specifies the desired line(s) for this rule. When you leave it blank or enter 0, this dial-now rule will apply to all lines.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of a dial-now template:

```
<DialNow>
  <Data DialNowRule="1234" LineID="0"/>
  <Data DialNowRule="52[0-6]" LineID="0"/>
  <Data DialNowRule="xxxxxx" LineID=""/>
</DialNow>
```

Softkey Layout Template

The softkey layout template allows you to customize soft key layout for different call states. The call states include CallFailed, CallIn, Connecting, Dialing, RingBack and Talking. After setup, place the templates to the provisioning server and specify the access URL in the configuration files.

When editing a softkey layout template, learn the following:

- <Call States> indicates the start of a template and </Call States> indicates the end of a template. For example, <CallFailed></CallFailed>.
- <Disable> indicates the start of the disabled soft key list and </Disable> indicates the end of the soft key list. The disabled soft keys are not displayed on the LCD screen.
- Create disabled soft keys between <Disable> and </Disable>.
- <Enable> indicates the start of the enabled soft key list and </Enable> indicates the end of the soft key list. The enabled soft keys are displayed on the LCD screen.
- Create enabled soft keys between <Enable> and </Enable>.
- <Default> indicates the start of the default soft key list and </Default> indicates the end of the default soft key list. The default soft keys are displayed on the LCD screen by default.

Procedure

Use the following procedures to customize a softkey layout template.

To customize a softkey layout template:

- Open the template file using an ASCII editor.
- For each soft key that you want to enable, add the following string between <Enable> and </Enable> in the file. Each starts on a separate line:

```
<Key Type=""/>
```

Where:

Key Type="" specifies the enabled soft key (This value cannot be blank).

For each disabled soft key and each default soft key that you want to add, add the same string introduced above.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of the CallFailed template:

```
<CallFailed>
  <Disable>
    <Key Type="Empty"/>
    <Key Type="Switch"/>
    <Key Type="Cancel"/>
  </Disable>
  <Enable>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Enable>
  <Default>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Default>
</CallFailed>
```

Directory Template

Directory provides easy access to frequently used lists. Users can access lists by pressing the Directory soft key when the IP phone is idle. The lists may contain Local Directory, History and Remote Phone Book. You can add the desired list(s) to Directory using the supplied directory template (favorite_setting.xml). After setup, place the directory template to the provisioning server and specify the access URL in the configuration files.

When editing a directory template, learn the following:

- Do not rename the directory template.
- <root_favorite_set> indicates the start of a template and </root_favorite_set> indicates the end of a template.
- The default display names of the directory lists are Local Directory, History and Remote Phone Book.
- When specifying the display priority of the directory list, the valid values are 1, 2 and 3. 1 is the highest priority, 3 is the lowest.
- When enabling or disabling the desired directory list, the valid values are 0 and 1. 0 stands for Disabled, 1 stands for Enabled.

Procedure

Use the following procedures to customize a directory template.

Customizing a directory template:

- Open the template file using an ASCII editor.
- For each directory list that you want to configure, edit the corresponding string in the file. For example, configure the local directory list, edit the following strings:

```
<item id_name="localdirectory" display_name="Local Directory" priority="1" enable="1" />
```

Where:

id_name="" specifies the existing directory list("localdirectory" for the local directory list). Do not edit this field.

display_name="" specifies the display name of the directory list. We recommend you do not edit this field.

priority="" specifies the display priority of the directory list.

enable="" enables or disables the directory list.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of a directory template:

```
<root_favorite_set>
  <item id_name="localdirectory" display_name="Local Directory" priority="1" enable="1" />
  <item id_name="history" display_name="History" priority="2" enable="0" />
```

```
<item id_name="remotedirectory" display_name="Remote Phone Book" priority="3" enable="0" />
</root_favorite_set>
```

Super Search Template

Search source list in dialing allows the IP phone to search for entries from the desired lists based on the entered string when in the pre-dialing screen, and then the user can select the desired entry to dial out quickly. The lists may contain Local Directory, History and Remote Phone Book. You can configure the search source list in dialing using the supplied super search template (super_search.xml). After setup, place the super search template to the provisioning server and specify the access URL in the configuration files.

When editing a super search template, learn the following:

- Do not rename the super search template.
- `<root_super_search>` indicates the start of a template and `</root_super_search>` indicates the end of a template.
- The default display names of the directory lists are Local Directory, History and Remote Phone Book.
- When specifying the priority of search results, the valid values are 1, 2, 3. 1 is the highest priority, 3 is the lowest.
- When enabling or disabling the desired directory list, the valid values are 0 and 1. 0 stands for Disabled, 1 stands for Enabled.

Procedure

Use the following procedures to customize a super search template.

Customizing a super search template:

- Open the template file using an ASCII editor.
- For each directory list that you want to configure, edit the corresponding string in the file. For example, configure the local directory list, edit the following strings:

```
<item id_name="local_directory_search" display_name="Local Directory" priority="1" enable="1" />
```

Where:

`id_name=""` specifies the directory list ("local_directory_search" for the local directory list). Do not edit this field.

`display_name=""` specifies the display name of the directory list. We do not recommend editing this field.

`priority=""` specifies the priority of search results.

`enable=""` enables or disables the directory list.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of a super search template:

```
<root_super_search>
  <item id_name="local_directory_search" display_name="Local Directory" priority="1" enable="1" />
  <item id_name="callog_search" display_name="History" priority="2" enable="1" />
  <item id_name="remote_directory_search" display_name="Remote Phone Book" priority="3" enable="0" /
>
</root_super_search>
```

Local Contact File

You can add contacts one by one on the IP phone directly. You can also add multiple contacts at a time and/or share contacts between IP phones using the local contact template file. After setup, place the template file to the provisioning server and specify the access URL of the template file in the configuration files.

When editing a local contact template, learn the following:

- <root_contact> indicates the start of a contact list and </root_contact> indicates the end of a contact list.
- <root_group> indicates the start of a group list and </root_group> indicates the end of a group list.
- When specifying a ring tone for a contact or a group, the format of the value must be Auto (the first registered line), Resource:RingN.wav (system ringtone, integer N ranges from 1 to 5) or Custom:Name.wav (custom ringtone).
- When specifying a desired line for a contact, the valid values are 0 and line ID, 0 stands for the first available account. Multiple line IDs are separated by commas.
- At most 5 groups can be added to the IP phone.
- At most 1000 local contacts can be added to the IP phone.

Procedure

Use the following procedures to customize a local contact template file.

To customize a local contact file:

- Open the template file using an ASCII editor.
- For each group that you want to add, add the following string to the file. Each starts on a separate line:

```
<group display_name="" ring="" />
```

Where:

display_name="" specifies the name of the group.

ring="" specifies the desired ring tone for this group.

- For each contact that you want to add, add the following string to the file. Each starts on a separate line:

```
<contact display_name="" office_number="" mobile_number="" other_number="" line="" ring=""
group_id_name=""/>
```

Where:

display_name="" specifies the name of the contact (This value cannot be blank or duplicated).

office_number="" specifies the office number of the contact.

mobile_number="" specifies the mobile number of the contact.

other_number="" specifies the other number of the contact.

line="" specifies the line you want to add this contact to. Keep it '0' or blank for IP phone.

ring="" specifies the ring tone for this contact.

group_id_name="" specifies the existing group you want to add the contact to.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of a local contact file:

```
<root_group>
  <group display_name="Friend" ring=""/>
  <group display_name="Family" ring="Resource:Ring1.wav"/>
</root_group>
<root_contact>
  <contact display_name="John" office_number="1001" mobile_number="12345678910" other_number=""
line="0" ring="Auto" group_id_name="All Contacts"/>
  <contact display_name="Alice" office_number="1002" mobile_number="" other_number="" line="0"
ring="Resource:Ring2.wav" group_id_name="Friend"/>
</root_contact>
```



*IP phones support both *.xml and *.csv formats.*

Remote XML Phone Book

IP phones can access 5 remote phone books. You can customize the remote XML phone book for IP phones as required. Before specifying the access URL of the remote phone book in the configuration files, you need to create a remote XML phone book and then place it to the provisioning server.

When creating an XML phone book, learn the following:

- <xxxIPPhoneDirectory> indicates the start of a phone book and </xxxIPPhoneDirectory> indicates the end of a phone book.
- <DirectoryEntry> indicates the start of a contact and </DirectoryEntry> indicates the end of a contact.

Procedure

Use the following procedures to customize an XML phone book.

Customizing an XML phone book:

- Open the template file using an ASCII editor.
- For each contact that you want to add, add the following strings to the phone book. Each starts on a separate line:

```
<Name>Mary</Name>  
<Telephone>1001</Telephone>
```

Where:

Specify the contact name between <Name> and </Name>.

Specify the contact number between <Telephone> and </Telephone>.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following shows an example of an XML phone book:

```
<xxxIPPhoneDirectory clearlight="true">  
<Title>Phonelist</Title>  
<Prompt>Prompt</Prompt>  
  <DirectoryEntry>  
    <Name>Support</Name>  
    <Telephone>8881234567</Telephone>  
  </DirectoryEntry>  
  <DirectoryEntry>  
    <Name>Sales</Name>  
    <Telephone>8887654321</Telephone>  
    <Telephone>8887654322</Telephone>  
  </DirectoryEntry>  
  <DirectoryEntry>  
    <Name>Billing</Name>  
    <Telephone>8881231231</Telephone>  
  </DirectoryEntry>  
  <SoftKeyItem>  
    <Name>#</Name>  
  </SoftKeyItem>  
  <SoftKeyItem>  
    <Name>*</Name>  
  </SoftKeyItem>
```

Specifying the Access URL of Resource Files

Access URL of the resource file can be configured using the configuration files:

Configuration File	<MAC>.cfg	<p>Configure the access URL of the replace rule template.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Configuration File	<MAC>.cfg	<p>Configure the access URL of the dial-now rule template.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Configuration File	<MAC>.cfg	<p>Configure the access URL of the softkey layout template.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Configuration File	<MAC>.cfg	<p>Configure the access URL of the local contact file.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Configuration File	<MAC>.cfg	<p>Configure the access URL of the remote XML phonebook.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>

Configuration File	<MAC>.cfg	<p>Configure the access URL of the directory template.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>
Configuration File	<MAC>.cfg	<p>Configure the access URL of the super search template.</p> <p>For more information, refer “Appendix D - Configuration Parameters”.</p>

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 VP110 reseller for further support.

Why is the LCD screen blank?

- Ensure that the phone is properly plugged into a functional AC outlet.
- Ensure that the phone is plugged into a socket controlled by a switch that is on.
- If the phone is plugged into a power strip, try plugging it directly into a wall outlet instead.
- If your IP phone is powered from PoE, ensure you use a PoE-compliant switch or hub.

Why does the phone display “Network Unavailable”?

- Ensure that the Ethernet cable is plugged into the Internet port on the phone and the Ethernet cable is not loose.
- Ensure that the switch or hub in your network is operational.
- Contact your network administrator for more information.


Why does the phone display “No Service”?

The LCD screen displays “No Service” when no SIP account is registered successfully.

Why doesn’t the phone display time and date correctly?

Check if you have configured the phone to obtain the time and date from the SNTP server automatically. If the phone fails to connect the SNTP server, configure the time and date manually.

How do I find the basic information of the phone?

Press the OK  Key when the phone is idle to check the basic information of the phone, such as the IP address and firmware version. For more information, refer the [“Viewing the Phone Status”](#).

How to obtain the MAC address of a phone when the phone is not powered on?

Three ways you can obtain the MAC address of a phone:

- You can ask your supplier for shipping information sheet which includes MAC addresses according to the corresponding PO (Purchase Order).

- You can find the MAC address in the label of carton box.
- You can find the MAC address from the phone's bar code on the back of the phone.

Why can't I get a dial tone?

- Check for any loose connections and that the phone has been installed properly. For the installation instructions, refer ["Phone Installation"](#).
- Switch between the Handset, Headset (if present) and Hands-Free Speakerphone to check whether the dial tone is present for one of the audio modes.
- If the dial tone exists on another audio mode, connect a different handset or headset to isolate the problem.

Why doesn't the phone ring?

Check that the ringer volume on the phone. To adjust the ringer volume setting, press the Volume key when the phone is idle. For more information, refer ["Volume"](#).

Why can't I receive calls?

- Check the SIP registration.
- Check that DND (Do Not Disturb) feature is disabled on the phone. Refer ["Do Not Disturb"](#).
- Check that call forward is disabled on the phone. Refer ["Call Forward"](#).
- Check whether the caller number is stored in the blacklist. Refer ["Blacklist"](#).

Why is my handset not working?

Check that the handset cord is fully connected to both the handset jack on the phone and handset. Refer ["Phone Installation"](#).

Why is my headset not working?

- Check that the headset cord is fully connected to the headset jack on the phone. Refer ["Phone Installation"](#).
- Check that the headset mode is activated. Refer ["Headset Mode Activation/De-activation"](#).
- Check that the headset volume is adjusted to an appropriate level. Refer ["Volume"](#).

What is the difference between user name, register name and display name?

Both user name and register name are defined by the server. A user name is used to identify the account while a register name matched with a password is used for authentication if the server requires. Display name is the caller ID that will be displayed on the callee's LCD screen. Server configurations may override local configurations.

Why does the phone play a tone when there is a call on hold? How to disable it?

When there is a call on hold, the phone will play a hold tone every 30 seconds. Play hold tone is enabled by default. Play hold tone and play hold tone delay are configurable via web user interface only.

To configure play hold tone and play 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 seconds) in the **Play Hold Tone Delay** field.
- Click **Confirm** to save the change.

Why can't I send a SMS to any other phone?

SMS depends on support from a SIP server. Contact your ITSP for more information.

How to change the user password?

To change the user password via web user interface:

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

- *User password is configurable via web user interface only.*

How to make a call using SRTP?

You can enable SRTP to encrypt the audio stream(s) of phone calls. The parties participating in the call should enable SRTP. To enable SRTP via web user interface, refer ["SRTP"](#).

How to reboot the phone?

To reboot the phone via web user interface:

- Click on **Settings->Upgrade**.
- Click **Reboot** to reboot the phone.



Any reboot of the phone may take a few minutes.

How to export PCAP trace?

We may need you to provide a PCAP trace to help analyze your problem. To export a PCAP trace via web user interface, refer ["Capturing Packets \(Packet Capture\)"](#).

How to export system log?

We may need you to provide debug log information to help analyze problem. By default, you can export the debug system log information to local PC. To export the system log to local PC via web user interface, refer ["Viewing Log Files"](#).

How to export/import phone configurations?

We may need you to provide the phone configurations to help analyze your problem. In some instance, you may need to import configurations to the phone.

To export the phone configurations via web user interface:

- Click on **Settings->Configuration**.
- Click **Export** to open file download window, and then save the file to your local system.

To import the phone configurations via web user interface:

- Click on **Settings->Configuration**.
- Click **Browse** to locate a configuration file from your local system.
- Click **Import** to import the configuration file.



*The file format of configuration file must be *.bin.*

How to upgrade firmware?

To upgrade firmware via web user interface:

- Click on **Settings->Upgrade**.
- Click **Browse** to locate the firmware from your local system.
- Click **Upgrade** to upgrade the firmware.

The web user interface prompts “Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!”.

- Click **OK** to confirm upgrading.


How to reset the phone to factory default settings using phone user interface?

Reset the phone when other troubleshooting suggestions do not correct the problem. You need to note that all customized settings will be overwritten after resetting. We recommend asking your vendor for advice before resetting the phone.

To reset the phone via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: 1234).
- Press  or  to scroll to **Reset to Factory**, and then press the **Enter** soft key.

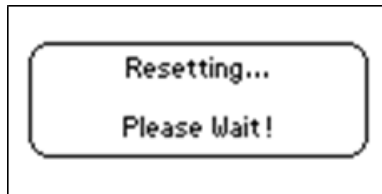
Or,

- Press the **OK**  key for more than 7 seconds.

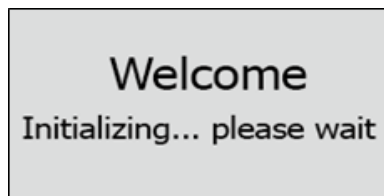
The LCD screen prompts “Reset to Factory?”.

- Press the **OK** soft key.

The LCD screen prompts “Resetting...Please Wait!”.



The LCD screen prompts “Welcome Initializing...please wait”.



The phone will be reset to factory successfully after startup.



Resetting of your phone may take a few minutes. Do not power off until the phone starts up successfully.

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.

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., time-out handling, 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 call and redial.

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.

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.

How to reboot the IP phone remotely?

IP phones support remote reboot by a SIP NOTIFY message with “Event: check-sync” header. When receiving a NOTIFY message with the parameter “reboot=true”, the IP phone reboots 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
```

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 pre-configured 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.

Why doesn't the IP phone update the configuration?

Do one of the following:

- Ensure that the configuration is set correctly.
- Reboot the IP 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.

What do “on code” and “off code” mean?

They are 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.

How to solve the IP conflict problem?

Do one of the following:

- Reset another available IP address for the IP phone.
- Check network configuration via phone user interface at the path - **Menu->Settings->Advanced Settings->Network->WAN Port->IPv4**. If Static IP Client is selected, select **DHCP** IP Client instead.

Why the web user interface logs out automatically after inactivity of a few minutes?

It happens as the **Auto-Logout Time(1~1000min)** value by default is set to 5 minutes.

Do the following:

- Click **Features -> General Information**.
- Configure the timer value of **Auto-Logout Time(1~1000min)** to any desired value of your choice, preferably a higher value if desired. However, configuring a much higher value for this timer may cause undesired security breaches and allow access of the web user interface to non-authorized individuals if you do not log out.

How does 'Reboot In Talking' parameter function?

This parameter is accessible from the web interface, **Features -> General Information**. If this parameter is enabled and SIP NOTIFY message with reboot event is received while the IP phone is in active call, it will reboot automatically.

If this parameter is disabled and SIP NOTIFY message with reboot event is received while the IP phone is in active call, then the IP phone will reboot only after the call ends (that is, after it becomes idle).

If the IP phone is idle and SIP NOTIFY message with reboot event is received, then the IP phone will reboot immediately irrespective of this parameter whether enabled or disabled.

Appendix

Physical Features of IP Phones

This section lists the available physical features of SPARSH VP110 IP phones.

- 132 x 64 Graphic LCD
- Single VoIP account
- 29 keys including 4 soft keys
- 1 x RJ9 (4P4C) Handset port
- 1 x RJ9 (4P4C) Headset port
- 2 x RJ45 10/100Mbps Ethernet ports
- 1 LED: 1 x Power
- Power Adapter (Optional): AC 100~240V input and DC 5V/600mA output
- Power over Ethernet (IEEE 802.3af)

Technical Specifications

Audio Features

- Full-duplex Hands-free Speakerphone with AEC
- Codecs: G.711(A/μ), G.722, G.723, G.729, G.726, iLBC
- DTMF: In-band, Out-of-band (RFC 2833) and SIP INFO
- VAD, CNG, AEC, PLC, AJB, AGC

Phone Book

- Local Phone book up to 1000 entries Black List
- XML Remote Phone book
- Intelligent Search Method
- Phone book Search/Import/Export
- Call History: Dialed/Received/Missed/Forwarded

Phone Features

- Single VoIP Account
- Call Hold, Mute, DND
- One-touch Speed Dial, Hotline
- Redial, Call Return, Auto Answer
- Call Forward, Call Waiting, Call Transfer
- Local 3-way Conference
- Direct IP Call without SIP Proxy
- Ringtone Selection/Import/Delete
- Keypad Lock, Emergency Call
- Set Date & Time Manually or Automatically
- Dial Plan, XML Browser, Action URL/URI
- Instant Messaging (Web UI and Phone)

Call Management

- Anonymous Call (CLIR)
- Anonymous Call Rejection
- Message Waiting Indicator (MWI)
- Voicemail, Call Pickup
- Intercom, Music on Hold
- Call Completion, Hot-desking
- Dial out Number from Web UI

Display

- 132x64-pixel Graphical LCD
- LED for Indicating Incoming calls, Voice/Text Messages, Mute, Call Hold/Held, On call
- Intuitive User Interface with Icons and Soft keys
- Multiple Language Options
- Caller ID with Name, Number

Networking and Security

- SIP v1 (RFC2543), v2 (RFC3261)
- IPv6
- NAT Transverse: STUN Mode
- Proxy Mode and Peer-to-Peer SIP Link Mode
- IP Assignment: Static/DHCP/PPPoE
- HTTP/HTTPS Web Server
- Time and Date Synchronization using SNTP

- UDP/TCP/DNS-SRV (RFC 3263)
- QoS: 802.1p/Q Tagging (VLAN), Layer 3 ToS, DSCP
- SRTP for Voice
- Transport Layer Security (TLS)
- HTTPS Certificate Manager
- AES Encryption for Configuration File
- Digest Authentication
- IEEE802.1X
- SNMP v1/v2

Management

- Configuration: Browser/Phone/Auto-Provision
- Auto Provision via FTP/TFTP/HTTP/HTTPS for Mass Deploy
- Server Redundancy
- Factory Reset
- Soft Reboot
- Packet Tracing Export
- System Log

Physical Features

- 2 x 10/100 Mbps LAN & PC Ports
- 29 keys including 4 Soft Keys
- 1 x RJ9 Handset Port
- 1 x RJ9 Headset Port
- Dimension (W x D x H): 185 x 188 x 143 mm

Power Supply

- Power Adapter (Optional): 5VDC/600mA
- Power over Ethernet (IEEE 802.3af)
- Power Consumption: 5W (Typical)
- Connector: DC Power Jack

Mechanical

- Packaging: 10 Qty/CTN
- Net Weight: 9.8 Kg
- Gross Weight: 10.8 Kg
- Gift Box: 215 x 200 x 121 mm
- Installation: Wall Mount, Table-top

- Color: Gray

Environmental

- Operating Temperature: -10° C to 50°C (14° F to 122° F)
- Operating Humidity: 10 - 95% (Non-Condensing)

Certifications

- CE, FCC-15 (Class-B), RCM, RoHS

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)

Time Zone	Time Zone Name
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,Melbourne,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 802.1 group 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 an 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 VP110 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



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- network.internet_port.type	Configuration File <MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0 -DHCP 1 -PPPoE 2 -Static IP Address
Example	network.internet_port.type= 0

Parameter- network.static_dns_enable	Configuration File <MAC>.cfg
Description	Enables or disables the phone to use manually configured static IPv4 DNS when the parameter "network.internet_port.type" is set to 0 (DHCP). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	network.static_dns_enable= 0

Static Network Settings

Parameter- network.internet_port.type	Configuration File <MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-PPPoE 2-Static IP Address
Example	network.internet_port.type = 2

Parameter- 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.
Format	Integer
Default Value	0
Range	Valid values are: 0-IPv4 1-IPv6 2-IPv4&IPv6
Example	network.ip_address_mode = 0

Parameter- 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.
Format	IPv4Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.ip = 192.168.1.20

Parameter- 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.
Format	Subnet Mask
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.mask = 255.255.255.0

Parameter- 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.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.gateway = 192.168.1.254

Parameter- network.primary_dns	Configuration File <MAC>.cfg
Description	Configures the primary DNS server 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.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.primary_dns = 202.101.103.55

Parameter- network.secondary_dns	Configuration File <MAC>.cfg
Description	Configures the secondary DNS server 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.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.secondary_dns = 202.101.103.54

PPPoE

Parameter- network.internet_port.type	Configuration File <MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-PPPoE 2-Static IP Address
Example	network.internet_port.type= 1

Parameter- 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.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	network.pppoe.user = test.matrix

Parameter- 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.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	network.pppoe.password = matrix

Internet and PC Ports Transmission Methods

Internet Port Transmission Method

Parameter- network.internet_port.speed_duplex	Configuration File <MAC>.cfg
Description	Configures the transmission method of Internet port. Note: 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.
Format	Integer
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
Example	network.internet_port.speed_duplex = 0

PC Port Transmission Method

Parameter- network.pc_port.speed_duplex	Configuration File <MAC>.cfg
Description	Configures the transmission method of PC port. Note: 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.
Format	Integer
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
Example	network.pc_port.speed_duplex = 0

PC Port Mode

Parameter- 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.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Auto Negotiation
Example	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
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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".
Format	Boolean
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".
Format	Boolean
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	<p>Enables or disables the power indicator LED to flash when a call is mute.</p> <p>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".</p>
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	<p>Enables or disables the power indicator LED to flash when a call is placed on hold or is held.</p> <p>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".</p>
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

Contrast

Parameter- phone_setting.contrast	Configuration File <MAC>.cfg
Description	Configures the contrast of the LCD screen. It configures the LCD's contrast of the IP phone only. Note: We recommend that you set the contrast of the LCD screen to 6 as a more comfortable level.
Format	Integer
Default Value	6
Range	1 to 10
Example	phone_setting.contrast = 6

Web Server Type

Parameter- wui.http_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to access its web user interface using HTTP protocol. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled

Example	wui.http_enable = 1
----------------	---------------------

Parameter- network.port.http	Configuration File <MAC>.cfg
Description	Configures the HTTP port used to access the web user interface of the IP phone. The default HTTP port is 80. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	80
Range	1 to 65535
Example	network.port.http = 80

Parameter- wui.https_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to access its web user interface using HTTPS protocol. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	wui.https_enable = 1

Parameter- network.port.https	Configuration File <MAC>.cfg
Description	Configures the HTTPS port used to access the web user interface of the IP phone. The default HTTPS port is 443. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer

Default Value	443
Range	1 to 65535
Example	network.port.https = 443

User Password

Parameter- security.user_password	Configuration File <MAC>.cfg
Description	Configures the password of the user for web server access. The IP phone uses “1111” as the default user password. Note: IP phones support ASCII characters 32-126(0x20-0x7E) in passwords.
Format	username:new password
Default Value	1111
Range	String within 32 characters
Example	security.user_password = user:1111

Administrator Password

Parameter- security.user_password	Configuration File <MAC>.cfg
Description	Configures the password of the administrator for web server access. The IP phone uses “1234” as the default administrator password. Note: IP phones support ASCII characters 32-126(0x20-0x7E) in passwords.
Format	administrator username:new password
Default Value	1234
Range	String within 32 characters
Example	security.user_password = admin:1234

Phone Lock

Parameter- phone_setting.phone_lock.enable	Configuration File <MAC>.cfg
Description	It enables or disables the phone lock feature. 0-Disabled 1-Enabled

Format	Integer
Default Value	0
Range	0 or 1
Example	phone_setting.phone_lock.enable = 1

Parameter- phone_setting.phone_lock.lock_key_type	Configuration File <MAC>.cfg
Description	<p>Configures the type of phone lock.</p> <p>Menu Key: The Menu soft key and MESSAGE key are locked.</p> <p>Function Keys: MESSAGE, RD, MUTE, TRAN, OK, navigation keys, soft keys are locked.</p> <p>All Keys: All keys are locked except the volume key. You are only allowed to dial emergency numbers, answer incoming calls by lifting the handset, pressing the Speakerphone key, the HEADSET key or the OK key, place an active call on hold by pressing the Hold soft key, resume the held call by pressing the Resume soft key, and end the call by hanging up the handset, pressing the Speakerphone key.</p> <p>It works only if the parameter "phone_setting.phone_lock.enable" is set to 1(Enabled).</p>
Format	Integer
Default Value	0
Range	Valid values are: 0 -All Keys 1 -Function Keys 2 -Menu Key
Example	phone_setting.phone_lock.lock_key_type =1

Parameter- phone_setting.phone_lock.unlock_pin	Configuration File <MAC>.cfg
Description	Configures a new unlock PIN. Once the IP phone is locked, you can use the default password "123" to unlock it.
Format	numeric characters
Default Value	123
Range	characters within 15 digits

Example	phone_setting.phone_lock.unlock_pin = 123
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Parameter- phone_setting.phone_lock.lock_time_out	Configuration File <MAC>.cfg
Description	Configures the IP phone to automatically lock the keypad after a delay time (in seconds). If it is set to 0 (0s), the keypad will not be locked automatically. In this case, you need to long press the pound key to lock the keypad. Note: This parameter works only if the IP phone lock type is preset.
Format	Integer
Default Value	0
Range	0 to 3600
Example	phone_setting.phone_lock.lock_time_out = 8

Time and Date

Parameter- local_time.manual_time_enable	Configuration File <MAC>.cfg
Description	Configures the phone to obtain time from NTP server or manual settings.
Format	Integer
Default Value	1
Range	Valid values are: 0 -Manual 1 -NTP
Example	local_time.manual_time_enable = 1

NTP Server

Parameter- local_time.manual_ntp_srv_prior	Configuration File <MAC>.cfg
Description	Enables or disables the phone to use manually configured NTP server preferentially.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled (use the NTP server obtained by DHCP preferentially) 1 -Enabled

Example	local_time.manual_ntp_srv_prior =0
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Parameter- local_time.ntp_server1	Configuration File <MAC>.cfg
Description	Configures the IP address or the domain name of the primary NTP server.
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	String within 99 characters
Example	local_time.ntp_server1 = cn.pool.ntp.org

Parameter- local_time.ntp_server2.	Configuration File <MAC>.cfg
Description	Configures the IP address or the domain name of the secondary NTP server. If the primary NTP server is not configured or cannot be accessed, the IP phone will request the time and date from the secondary NTP server.
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	String within 99 characters
Example	local_time.ntp_server2 = cn.pool.ntp.org

Parameter- local_time.interval	Configuration File <MAC>.cfg
Description	Configures the IP phone to update time and date from the NTP server at regular intervals (in seconds).
Format	Integer
Default Value	1000
Range	15 to 86400
Example	local_time.interval = 1000

Time Zone

Parameter- local_time.time_zone	Configuration File <MAC>.cfg
Description	Configures the time zone. For more available time zones, refer “Appendix A - Time Zone” .
Format	String
Default Value	+5:30
Range	-11 to +14
Example	local_time.time_zone = +8

Parameter- local_time.time_zone_name	Configuration File <MAC>.cfg
Description	Configures the desired time zone name. For more available time zones, refer “Appendix A - Time Zone” .
Format	String
Default Value	India (Calcutta)
Range	String within 32 characters
Example	local_time.time_zone_name = India (Calcutta)

DST

Parameter- local_time.summer_time	Configuration File <MAC>.cfg
Description	Enables or disables Daylight Saving Time (DST) feature.
Format	Integer
Default Value	2
Range	Valid values are: 0-Disabled 1-Enabled 2-Automatic
Example	local_time.summer_time = 2

Parameter- local_time.dst_time_type	Configuration File <MAC>.cfg
Description	Configures the DST type. Note: It works only if the parameter “local_time.summer_time” is set to 1 (Enabled).
Format	Integer
Default Value	0
Range	Valid values are: 0-By Date 1-By Week
Example	local_time.dst_time_type = 0

Parameter- local_time.start_time	Configuration File <MAC>.cfg
Description	Configures the time to start DST. If “local_time.dst_time_type” is set to 0 (By Date), use the mapping: MM: 1=Jan, 2=Feb,..., 12=Dec DD:1=the first day in a month,..., 31= the last day in a month HH:0=1am, 1=2am,..., 23=12pm If “local_time.dst_time_type” is set to 1 (By Week), use the mapping: Month: 1=Jan, 2=Feb,..., 12=Dec Week of Month: 1=the first week in a month,..., 5=the last week in a month Day of Week: 1=Mon, 2=Tues,..., 7=Sun Hour of Day: 0=1am, 1=2am,..., 23=12pm Note: It works only if the parameter “local_time.summer_time” is set to 1 (Enabled).
Format	The value formats are: <ul style="list-style-type: none"> MM/DD/HH (For By Date) Month/Day of week last in Month/Day of Week/Hour of Day (for By Week)
Default Value	1/1/0

Range	1 to 12/1 to 31/0 to 23 (for By Date) 1 to 12/1 to 5/1 to 7/0 to 23 (for By Week)
Example	local_time.start_time = 1/1/0

Parameter- local_time.end_time	Configuration File <MAC>.cfg
Description	<p>Configures the time to end DST.</p> <p>If “local_time.dst_time_type” is set to 0 (By Date), use the mapping: MM: 1=Jan, 2=Feb,..., 12=Dec DD:1=the first day in a month,..., 31= the last day in a month HH:0=1am, 1=2am,..., 23=12pm</p> <p>If “local_time.dst_time_type” is set to 1 (By Week), use the mapping: Month: 1=Jan, 2=Feb,..., 12=Dec Week of Month: 1=the first week in a month,..., 5=the last week in a month Day of Week: 1=Mon, 2=Tues,..., 7=Sun Hour of Day: 0=1am, 1=2am,..., 23=12pm</p> <p>Note: It works only if the parameter “local_time.summer_time” is set to 1 (Enabled).</p>
Format	<p>The value formats are:</p> <ul style="list-style-type: none"> MM/DD/HH (For By Date) Month/Day of week last in Month/Day of Week/Hour of Day (for By Week)
Default Value	12/31/23
Range	1 to 12/1 to 31/0 to 23 (For By Date) 1 to 12/1 to 5/1 to 7/0 to 23 (For By Week)
Example	local_time.end_time = 12/31/23

Parameter- local_time.dhcp_time	Configuration File <MAC>.cfg
Description	<p>Enables or disables the phone to update time with the offset time obtained from the DHCP server.</p> <p>Note: It is only available to offset from GMT 0.</p>

Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	local_time.dhcp_time = 0

Parameter- local_time.offset_time	Configuration File <MAC>.cfg
Description	Configures the offset time (in minutes).
Format	Integer
Default Value	Blank
Range	-300 to +300
Example	local_time.offset_time = 120

Time Format

Parameter- local_time.time_format	Configuration File <MAC>.cfg
Description	Configures the time format. If it is set to 0 (12 Hour), the time display will use 12 hour format. If it is set to 1 (24 Hour), the time display will use 24 hour format.
Format	Integer
Default Value	1
Range	Valid values are: 0-12 Hour 1-24 Hour
Example	local_time.time_format = 1

Date Format

Parameter- local_time.date_format	Configuration File <MAC>.cfg
Description	Configures the date format. IP phones support various date formats. You can change the desired format according to your requirement.
Format	Integer
Default Value	0

Range	<p>Valid values are:</p> <p>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</p> <p>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.</p>
Example	local_time.date_format = 0

Language

Parameter- gui_lang.url	Configuration File <MAC>.cfg
Description	<p>Configures the access URL of the language pack.</p> <p>Note: The language packs you load are dependent on available language packs from the provisioning server. You can download the language pack to the phone user interface only.</p>
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the language pack “lang+English.txt” from the provisioning server 192.168.10.25.</p> <p>gui_lang.url = http://192.168.10.25/lang+English.txt</p> <p>Note: This is not supported in this version. So keep the url blank for now.</p>

Parameter- lang.gui	Configuration File <MAC>.cfg
Description	Configures the language used on the phone user interface.
Format	String
Default Value	English

Range	Valid values are: English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish, Turkish, Russian or the custom language name
Example	lang.gui = English

Parameter- lang.wui	Configuration File <MAC>.cfg
Description	Configures the language used on the web user interface. Note: The default language used on the web user interface depends on the language preferences of your browser. If the language of your browser is not supported by the IP phone, the web user interface will use English by default.
Format	String
Default Value	Blank
Range	Valid values are: English, Chinese_S, Chinese_T, French, German, Italian, Polish, Portuguese, Spanish, Turkish, Russian or the custom language name
Example	lang.wui = English

Logo Customization

Parameter- phone_setting.lcd_logo.mode	Configuration File <MAC>.cfg
Description	Configures the logo mode of the LCD screen. If it is set to 0 (Disabled), the IP phone is not allowed to display a logo. If it is set to 1 (System logo), the LCD screen will display the system logo. If it is set to 2 (Custom logo), the LCD screen will display the custom logo (you need to upload a custom logo file to the phone).
Format	Integer
Default Value	1
Range	Valid values are: 0-Disabled 1-System logo 2-Custom logo

Example	phone_setting.lcd_logo.mode = 1
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Parameter- lcd_logo.url	Configuration File <MAC>.cfg
Description	Configures the access URL of custom logo file.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	The following example uses HTTP to download the custom logo file (logo.dob) from the provisioning server 192.168.10.25. lcd_logo.url = http://192.168.10.25/logo.dob

Key as Send

Parameter- features.key_as_send	Configuration File <MAC>.cfg
Description	Configures the "#" or "*" key as the send key. If it is set to 0 (Disabled), neither "#" nor "*" can be used as a 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.
Format	Integer
Default Value	1
Range	Valid values are: 0 -Disabled 1 -# key 2 -* 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 tone when a user presses a key. If it is set to 1 (Enabled), the IP phone will play a tone when a user presses a key.
Format	Boolean
Default Value	1
Range	0 -Disabled 1 -Enabled

Example	features.key_tone= 1
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Parameter- features.send_key_tone	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to play a tone when a user presses a send key. If it is set to 1 (Enabled), the IP phone will play a tone when a user presses a send key. Note: It works only if the parameter “features.key_tone” is set to 1 (Enabled).
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.send_key_tone = 1

Dial Plan

Replace Rule

Parameter- dialplan.replace.prefix.X	Configuration File <MAC>.cfg
Description	Configures the string you want to replace. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.prefix.1 = 123

Parameter- dialplan.replace.replace.X	Configuration File <MAC>.cfg
Description	Configures the alternate string instead of what the user enters. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.replace.1 = 1

Dial-now

Parameter- dialplan.dialnow.rule.X	Configuration File <MAC>.cfg
Description	Configures 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. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 511 characters
Example	dialplan.dialnow.rule.1 = 123

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 specified delay time.
Format	Integer
Default Value	0
Range	1 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 add before the entered numbers.
Format	String
Default Value	Blank
Range	String within 16 characters
Example	dialplan.area_code.code = 010

Parameter- dialplan.area_code.min_len	Configuration File <MAC>.cfg
Description	Configures the minimum length of the entered numbers.
Format	Integer
Default Value	1
Range	1 to 15
Example	dialplan.area_code.min_len = 1

Parameter- dialplan.area_code.max_len	Configuration File <MAC>.cfg
Description	Configures the maximum length of the entered numbers. Note: The value must be larger than the minimum length.
Format	Integer
Default Value	15
Range	1 to 15
Example	dialplan.area_code.max_len = 15

Block Out

Parameter- dialplan.block_out.number.X	Configuration File <MAC>.cfg
Description	Configures the block out numbers. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.block_out.number.1 = 1234

Hotline

Parameter-features.hotline_number	Configuration File<MAC>.cfg
Description	Configures the hotline number. It configures a number that the IP phone automatically dials out when lifting the handset, pressing the speakerphone key. Leaving it blank disables hotline feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	features.hotline_number = 3601

Parameter-features.hotline_delay	Configuration File<MAC>.cfg
Description	Configures the waiting time (in seconds) the IP phone automatically dials out the hotline number. If it is set to 0 (0s), the IP phone will immediately dial out the pre-configured hotline number when you lift the handset or press the speakerphone key. If it is set to a value greater than 0, the IP phone will wait the specified seconds before dialing out the predefined hotline number when you lift the handset or press the speakerphone key.
Format	Integer
Default Value	4
Range	0 to 10
Example	features.hotline_delay = 4

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

Missed Call Log

Parameter- account.X.missed_callog	Configuration File <MAC>.cfg
Description	<p>Enables or disables missed call log feature for account X.</p> <p>If it is set to 0 (Disabled), there is no indicator displaying on the LCD screen, the IP phone does not log the missed call in the Missed Calls list.</p> <p>If it is set to 1 (Enabled), a prompt message "<number> New Missed Call(s)" along with an indicator icon is displayed on the IP phone idle screen when the IP phone misses calls.</p> <p>X is 1.</p>
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.missed_callog = 1

Live Dialpad

Parameter- phone_setting.predial_autodial	Configuration File <MAC>.cfg
Description	<p>Enables or disables live dialpad feature.</p> <p>If it is set to 1 (Enabled), the IP phone will automatically dial out the entered phone number without having to press any key.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	phone_setting.predial_autodial = 1

Parameter- phone_setting.inter_digit_time	Configuration File <MAC>.cfg
Description	Configures the time (in seconds) for the phone to automatically dial out the entered digits without pressing any other key. Note: It works only if the parameter “phone_setting.predial_autodial” is set to 1 (Enabled).
Format	Integer
Default Value	4
Range	1 to 14
Example	phone_setting.inter_digit_time = 4

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 message while during a call. If it is set to 1 (Enabled), the LCD screen will present a new incoming call while during a call.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	call_waiting.enable = 1

Parameter- call_waiting.tone	Configuration File <MAC>.cfg
Description	Enables or disables the playing of a 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 parameter “call_waiting.enable” is set to 1 (Enabled).

Format	Boolean
Default Value	1
Range	Valid values are: 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.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.on_code = *72

Parameter- call_waiting.off_code	Configuration File <MAC>.cfg
Description	Configures the call waiting off code to deactivate the server-side call waiting feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.off_code = *73

Auto Redial

Parameter- auto_redial.enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to automatically redial the called number when it is busy. If it is set to 1 (Enabled), the IP phone will dial the previous dialed out number automatically when the dialed number is busy.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled

Example	auto_redial.enable = 1
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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.
Format	Integer
Default Value	10
Range	1 to 300
Example	auto_redial.interval = 10

Parameter- auto_redial.times	Configuration File <MAC>.cfg
Description	Configures the redial times for the IP phone. The IP phone tries to redial the dialed number as many times as configured till the callee answers the call.
Format	Integer
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 is 1. Note: The IP phone cannot automatically answer the incoming call during a call even if auto answer is enabled.
Format	Boolean
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 phone automatically answers an incoming call.
Format	Integer
Default Value	1
Range	1 to 4
Example	features.auto_answer_delay = 1

Call Completion

Parameter- features.call_completion_enable	Configuration File <MAC>.cfg
Description	<p>Enables or disables call completion feature.</p> <p>If a user places a call and the callee is temporarily not available to answer the call, call completion feature allows notifying the user when the callee becomes available to receive a call.</p> <p>If it is set to 1 (Enabled), the caller is notified when the callee becomes available to receive a call.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.call_completion_enable = 1

Anonymous Call

Parameter- account.X.anonymous_call	Configuration File <MAC>.cfg
Description	<p>Enables or disables anonymous call feature for account X.</p> <p>If it is set to 1 (Enabled), the IP phone will block its identity from showing up to the callee when placing a call. The callee's phone LCD screen presents anonymous instead of the caller's identity.</p> <p>X is 1.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.anonymous_call = 1

Parameter- account.X.send_anonymous_code	Configuration File <MAC>.cfg
Description	<p>Configures the phone to send anonymous on/off code to activate/deactivate the server-side anonymous call feature for account X.</p> <p>If it is set to 0 (Off Code), the IP phone will send anonymous off code to deactivate the server-side anonymous call feature.</p> <p>If it is set to 1 (On Code), the IP phone will send anonymous on code to activate the server-side anonymous call feature.</p> <p>X is 1.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Off Code 1 -On Code
Example	account.1.send_anonymous_code = 0

Parameter- account.X.anonymous_call_oncode	Configuration File <MAC>.cfg
Description	Configures the anonymous call on code to activate the server-side anonymous call feature for account X. X is 1. Note: It works only if the parameter "account.X.send_anonymous_code" is set to 1 (Enabled).
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_oncode = *72

Parameter- account.X.anonymous_call_offcode	Configuration File <MAC>.cfg
Description	Configures the anonymous call off code to deactivate the server-side anonymous call feature for account X. X is 1. Note: It works only if the parameter "account.X.send_anonymous_code" is set to 1 (Enabled).
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_offcode = *73

Anonymous Call Rejection

Parameter- account.X.reject_anonymous_call	Configuration File <MAC>.cfg
Description	Enables or disables anonymous call rejection feature for account X. If it is set to 1 (Enabled), the IP phone will automatically reject incoming calls from users enabled anonymous call feature. The anonymous user's phone LCD screen presents "Anonymity Disallowed". X is 1.
Format	Boolean
Default Value	0

Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.reject_anonymous_call = 1

Parameter- account.X.anonymous_reject_oncode	Configuration File <MAC>.cfg
Description	Configures the anonymous call rejection on code to activate the server-side anonymous call rejection feature for account X. X is 1.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_oncode = *74

Parameter- account.X.anonymous_reject_offcode	Configuration File <MAC>.cfg
Description	Configures the anonymous call rejection off code to deactivate the server-side anonymous call rejection feature for account X. X is 1.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_offcode = *75

Do Not Disturb

Return Message When DND

Parameter- features.dnd_refuse_code	Configuration File <MAC>.cfg
Description	Configures a return code and reason of SIP response messages when the phone rejects an incoming call by DND. A specific reason is displayed on the caller's phone LCD screen. If it is set to 486 (Busy here), the caller's phone LCD screen will display the reason "Busy here" when the callee enables DND feature.
Format	Integer
Default Value	480

Range	Valid values are: 404 -No Found 480 -Temporarily not available 486 -Busy here
Example	features.dnd_refuse_code = 480

DND Mode

Parameter- features.dnd_mode	Configuration File <MAC>.cfg
Description	Configures the DND mode for the IP phone. It must be set to 0 (Phone) so that DND feature is effective for the IP phone.
Format	Integer
Default Value	0
Range	0-Phone
Example	features.dnd_mode = 0

Parameter- features.dnd.enable	Configuration File <MAC>.cfg
Description	Enables or disables DND feature. If it is set to 1 (Enabled), the IP phone will reject incoming calls on all accounts.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.dnd.enable = 1

Parameter- features.dnd.on_code	Configuration File <MAC>.cfg
Description	Configures the DND on code to activate the server-side DND feature.
Format	String
Default Value	Blank
Range	String within 32 characters

Example	features.dnd.on_code = *71
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Parameter- features.dnd.off_code	Configuration File <MAC>.cfg
Description	Configures the DND off code to deactivate the server-side DND feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	features.dnd.off_code = *72

Busy Tone Delay

Parameter- features.busy_tone_delay	Configuration File <MAC>.cfg
Description	<p>Configures a period of time (in seconds) for which the busy tone is audible on the IP phone.</p> <p>When one party releases the call, a busy tone is audible to the other party indicating that the call connection breaks.</p> <p>If it is set to 3 (3s), a busy tone is audible for 3 seconds on the IP phone.</p>
Format	Integer
Default Value	0
Range	Valid values are: 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	<p>Configures a return code and reason of SIP response messages when the phone rejects an incoming call. A specific reason is displayed on the caller's phone LCD screen.</p> <p>If it is set to 486 (Busy here), the caller's phone LCD screen will display the message "Busy here" when the callee rejects the incoming call.</p>

Format	Integer
Default Value	486
Range	Valid values are: 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	<p>Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message.</p> <p>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.</p>
Format	Boolean
Default Value	1
Range	Valid values are: 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	<p>Enables or disables the IP phone to send the SIP requests to the outbound proxy server.</p> <p>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.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p>
Format	Boolean
Default Value	1
Range	Valid values are: 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) for account X. T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server. X is 1.
Format	Float
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 session timer T2 (in seconds) for account X. T2 represents the maximum retransmit interval for non-INVITE requests and INVITE responses. X is 1.
Format	Float
Default Value	4
Range	2 to 40
Example	sip.timer_t2 = 4

Parameter- sip.timer_t4	Configuration File <MAC>.cfg
Description	Configures the session timer of T4 (in seconds) for account X. T4 represents the maximum duration a message will remain in the network. X is 1.
Format	Float
Default Value	5
Range	2.5 to 60
Example	sip.timer_t4 = 5

Session Timer

Parameter- account.X.session_timer.enable	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 re-INVITE requests to refresh the session during a call. X is 1.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.session_timer.enable = 1

Parameter- account.X.session_timer.expires	Configuration File <MAC>.cfg
Description	Configures the IP phone to refresh the session during a call at regular intervals (in seconds) for account X. If it is set to 1800 (1800s), the IP phone will refresh the session during a call before 1800 seconds. X is 1.
Format	Integer
Default Value	1800
Range	30 to 7200
Example	account.1.session_timer.expires = 1800

Parameter- account.X.session_timer.refresher	Configuration File <MAC>.cfg
Description	Configures the session timer refresher for account X. If it is set to 0 (UAC), refreshing the session is performed by the IP phone. If it is set to 1 (UAS), refreshing the session is performed by a SIP server. X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 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 tone when there is a hold call on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 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 plays a hold tone. If it is set to 30 (30s), the IP phone will play a hold tone every 30 seconds when there is a hold call on the IP phone. Note: It works only if the parameter "features.play_hold_tone.enable" is set to 1 (Enabled).
Format	Integer
Default Value	30
Range	3 to 3600
Example	features.play_hold_tone.delay = 30

Parameter- sip.rfc2543_hold	Configuration File <MAC>.cfg
Description	Configures whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. 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.
Format	Boolean
Default Value	0

Range	Valid values are: 0 -Disabled 1 -Enabled
Example	sip.rfc2543_hold = 0

Parameter- account.X.music_server_uri	Configuration File <MAC>.cfg
Description	Configures the Music on Hold server address. Examples for valid values: <10.1.3.165>, 10.1.3.165, sip:moh@sip.com etc. X is 1. Note: The DNS query in this parameter only supports A query.
Format	String
Default Value	Blank
Range	String within 256 characters
Example	account.1.music_server_uri =<10.1.3.165>

Call Forward

Call Forward Mode

Parameter- features.fwd_mode	Configuration File <MAC>.cfg
Description	Configures the call forward mode for the IP phone. It must be set to 0 (Phone) so that call forward feature is effective for the IP phone.
Format	Integer
Default Value	0
Range	0 -Phone
Example	features.fwd_mode = 0

Always Forward

Parameter- forward.always.enable	Configuration File <MAC>.cfg
Description	Enables or disables always forward feature. If it is set to 1 (Enabled), incoming calls are forwarded to the destination number immediately.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled

Example	forward.always.enable = 1
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Parameter- forward.always.target	Configuration File <MAC>.cfg
Description	Configures the destination number of the always forward.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.always.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.
Format	String
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.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.always.off_code = *73

Busy Forward

Parameter- forward.busy.enable	Configuration File <MAC>.cfg
Description	Enables or disables busy forward feature. If it is set to 1 (Enabled), incoming calls are forwarded to the destination number when the callee is busy.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	forward.busy.enable = 1

Parameter- forward.busy.target	Configuration File <MAC>.cfg
Description	Configures the destination number of the busy forward.
Format	String
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.
Format	String
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.
Format	String

Default Value	Blank
Range	String within 32 characters
Example	forward.busy.off_code = *75

No Answer Forward

Parameter- forward.no_answer.enable	Configuration File <MAC>.cfg
Description	Enables or disables no answer forward feature. If it is set to 1 (Enabled), incoming calls are forwarded to the destination number after a period of ring time.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
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.
Format	String
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.
Format	Integer
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.
Format	String
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.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.off_code = *77

Diversion

Parameter- features.fwd_diversion_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to present the diversion information when the call is forwarded to your IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.fwd_diversion_enable = 1

Fwd International

Parameter- forward.international.enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to forward an incoming call to an international phone number.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled

Example	forward.international.enable = 1
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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.
Format	Boolean
Default Value	1
Range	Valid values are: 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 transfer or the attended transfer through on-hook.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	transfer.on_hook_trans_enable = 1

Parameter- transfer.semi_attend_tran_enable	Configuration File <MAC>.cfg
Description	Configures whether to display the missed call prompt on the destination party's phone when performing a semi-attended transfer.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Enabled 1 -Disabled

Example	transfer.semi_attend_tran_enable = 1
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Network Conference

Parameter- account.X.conf_type	Configuration File <MAC>.cfg
Description	Configures the 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. X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 0 -Local Conference 2 -Network Conference
Example	account.1.conf_type = 0

Parameter- account.X.conf_uri	Configuration File <MAC>.cfg
Description	Configures the conference URI for account X. X is 1. Note: It works only if the parameter "account.X.conf_type" is set to 2 (Network Conference).
Format	SIP URI
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 Transfer on Conference Hang Up feature. If it is set to 1 (Enabled), the other two parties remain connected when the conference initiator drops the conference call. Note: It is only applicable to the local conference.

Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	transfer.tran_others_after_conf_enable=1

Directed Call Pickup

Phone Basis

Parameter- features.pickup.direct_pickup_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to display the DPickup soft key when the IP phone is off-hook.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.pickup.direct_pickup_enable = 1

Parameter- account.X.direct_pickup_code	Configuration File <MAC>.cfg
Description	Configures the directed call pickup code on a phone basis.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.X.direct_pickup_code = *68

Group Call Pickup

Phone Basis

Parameter- features.pickup.group_pickup_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to display the GPickup soft key when the IP phone is off-hook.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled

Example	features.pickup.group_pickup_enable = 1
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Parameter- account.X.group_pickup_code	Configuration File <MAC>.cfg
Description	Configures the group call pickup code on a phone basis.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.X.group_pickup_code = *69

Calling Line Identification Presentation

Parameter- account.X.cid_source	Configuration File <MAC>.cfg
Description	<p>Configures the presentation of the caller identity for account X.</p> <p>0-FROM (Derives the name and number of the caller from the "From" header).</p> <p>1-PAI (Derives the name and number of the caller from the "PAI" header).</p> <p>If the server does not send the "PAI" header, displays "anonymity" on the callee's phone).</p> <p>2-PAI-FROM (Derives the name and number of the caller from the "PAI" header preferentially. If the server does not send the "PAI" header, derives from the "From" header).</p> <p>3-RPID-PAI-FROM</p> <p>4-PAI-RPID-FROM</p> <p>5-RPID-FROM</p> <p>X is 1.</p>
Format	Integer
Default Value	0
Range	0 to 5
Example	account.1.cid_source = 0

Connected Line Identification Presentation

Parameter- account.X.cp_source	Configuration File <MAC>.cfg
Description	<p>Configures the presentation of the callee's identity for account X.</p> <p>0-PAI-RPID (Derives the name and number of the callee from the "PAI" header preferentially. If the server does not send the "PAI" header, derives from the "RPID" header).</p> <p>1-Dialed Digits (Preferentially displays the dialed digits on the caller's phone).</p> <p>2-RFC4916 (Derives the name and number of the callee from "From" header in the Update message).</p> <p>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.</p> <p>X is 1.</p>
Format	Integer
Default Value	0
Range	0 to 2
Example	account.1.cp_source = 0

DTMF

Parameter- account.X.dtmf.type	Configuration File <MAC>.cfg
Description	<p>Configures the DTMF type for account X.</p> <p>If it is set to 0 (INBAND), DTMF digits are transmitted in the voice band.</p> <p>If it is set to 1 (RFC 2833), DTMF digits are transmitted by RTP Events compliant to RFC 2833.</p> <p>If it is set to 2 (SIP INFO), DTMF digits are transmitted by the SIP INFO messages.</p> <p>If it is set to 3 (AUTO or SIP INFO), the IP phone negotiates with the other end to use INBAND or RFC 2833, if there is no negotiation, using SIP INFO by default.</p> <p>X is 1.</p>
Format	Integer
Default Value	1
Range	<p>Valid values are:</p> <p>0-INBAND</p> <p>1-RFC 2833</p> <p>2-SIP INFO</p> <p>3-AUTO or SIP INFO</p>
Example	account.1.dtmf.type = 1

Parameter- account.X.dtmf.dtmf_payload	Configuration File <MAC>.cfg
Description	<p>Configures the RFC 2833 payload type.</p> <p>X is 1.</p>
Format	Integer
Default Value	101
Range	96 to 127
Example	account.1.dtmf.dtmf_payload = 101

Parameter- account.X.dtmf.info_type	Configuration File <MAC>.cfg
Description	Configures the DTMF info type when the DTMF type is configured as "SIP INFO", "AUTO or SIP INFO". X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 0-Disabled 1-DTMF-Relay 2-DTMF 3-Telephone-Event
Example	account.1.dtmf.info_type = 0

Parameter- features.dtmf.repetition	Configuration File <MAC>.cfg
Description	Configures the number of times for the IP phone to send the end RTP EVENT packet.
Format	Integer
Default Value	3
Range	1 to 3
Example	features.dtmf.repetition = 3

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. If it is set to 1 (Enabled), the DTMF digits are displayed as asterisks.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.dtmf.hide = 1

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. Note: It works only if the parameter “features.dtmf.hide” is set to 1 (Enabled).
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.dtmf.hide_delay = 1

Transfer via DTMF

Parameter- features.dtmf.replace_tran	Configuration File <MAC>.cfg
Description	Enables or disables transfer via DTMF feature. If it is set to 0 (Disabled), the IP phone will perform the transfer as normal when pressing the transfer key during a call. If it is set to 1 (Enabled), the IP phone will transmit the specified DTMF digits to the server for completing call transfer when pressing the transfer key during a call.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.dtmf.replace_tran = 1

Parameter- features.dtmf.transfer	Configuration File <MAC>.cfg
Description	Configures the DTMF digits to be transmitted to complete the transfer. Note: It works only if the parameter “features.dtmf.replace_tran” is set to 1 (Enabled).
Format	String

Default Value	Blank
Range	Valid values are: 0-9, *, # and A-D. 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	It enables or disables the IP phone to play a local DTMF tone. 0-Disabled 1-Enabled
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.play_local_dtmf_tone_enable = 1

Incoming Intercom calls

Parameter- features.dtmf.replace_tran	Configuration File <MAC>.cfg
Description	Enables or disables transfer via DTMF feature. If it is set to 0 (Disabled), the IP phone will perform the transfer as normal when pressing the transfer key during a call. If it is set to 1 (Enabled), the IP phone will transmit the specified DTMF digits to the server for completing call transfer when pressing the transfer key during a call.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.dtmf.replace_tran = 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 0 (Disabled), the microphone is unmuted for incoming calls. If it is set to 1 (Enabled), the microphone is muted for intercom calls.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.intercom.mute = 1

Parameter- features.intercom.tone	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to play a warning tone when receiving an intercom call. If it is set to 0 (Disabled), the IP phone will automatically answer the intercom call without a warning tone. If it is set to 1 (Enabled), the IP phone will play a warning tone to alert you before answering the intercom call.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.intercom.tone = 1

Parameter-features.intercom.barge	Configuration File<MAC>.cfg
Description	<p>Enables or disables the IP phone to automatically answer an incoming intercom call while there is already an active call on the IP phone.</p> <p>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.</p> <p>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.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.intercom.barge = 1

Distinctive Ring Tones

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.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.alert_info_tone = 1

Parameter-account.X.alert_info_url_enable	Configuration File<MAC>.cfg
Description	<p>Enables or disables the IP phone to download the ring tone from the URL contained in the Alert-Info header for account X.</p> <p>X is 1.</p>
Format	Boolean
Default Value	1

Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.alert_info_url_enable= 1

Parameter- distinctive_ring_tones.alert_info.X.text	Configuration File <MAC>.cfg
Description	Configures the texts to map the keywords contained in the SIP header. X ranges from 1 to 10.
Format	String
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	Configuration File <MAC>.cfg
Description	Configures the desired ring tones for each text. The value ranges from 1 to 5, the digit stands for the appropriate ring tone. X ranges from 1 to 10.
Format	Integer
Default Value	1
Range	Valid values are: 1 -Ring1.wav 2 -Ring2.wav 3 -Ring3.wav 4 -Ring4.wav 5 -Ring5.wav
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.
Format	String
Default Value	Custom

<p>Range</p>	<p>Valid values are:</p> <ul style="list-style-type: none"> • Custom • 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
<p>Example</p>	<p>voice.tone.country = Custom</p>

Parameter- voice.tone.dial voice.tone.ring voice.tone.busy voice.tone.congestion voice.tone.callwaiting voice.tone.dialrecall voice.tone.info voice.tone.stutter voice.tone.message voice.tone.autoanswer	Configuration File <MAC>.cfg
Description	<p>Configures the tone for each condition. tonelist = element[,element] [,element]...</p> <p>Where element = [!]Freq1[+Freq2][+Freq3][+Freq4] / Duration</p> <p>Freq: the frequency of the tone (ranges from 200 to 7000 Hz). If it is set to 0 (0 Hz), it means the tone is not played. A tone is comprised of at most four different frequencies.</p> <p>Duration: the time duration (in milliseconds, ranges from 0 to 30000ms) of the ring tone.</p> <p>You can configure at most eight different tones for one condition, and separate tones by commas (e.g., 250/200, !0/1000, 200+300/500, 600+700+800+1000/2000). The exclamation point (!) can be added optionally, which means these tones are only played once.</p> <p>Note: It works only if the parameter "voice.tone.country" is set to Custom.</p>
Format	Refer to the introduction above
Default Value	Blank
Range	Not Applicable
Example	voice.tone.dial = 800+200/1000, 0/100, 500/1200, 500+600+950+1500/5000

Remote Phone Book

Parameter- remote_phonebook.data.X.url	Configuration File <MAC>.cfg
Description	<p>Configures the access URL of the remote XML phonebook.</p> <p>X ranges from 1 to 5.</p>
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	remote_phonebook.data.1.url = http:// 192.168.1.20/phonebook.xml
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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- remote_phonebook.display_name	Configuration File <MAC>.cfg
Description	Configures the display name of the remote phone book. If you leave it blank, Remote Phone Book is displayed on the LCD screen at the path Menu->Directory.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.display_name = Remote Phone Book

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	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to subscribe the message waiting indicator to the account 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. X is 1.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	account.1.subscribe_mwi=0

Parameter- account.X.subscribe_mwi_expires	Configuration File <MAC>.cfg
Description	<p>Configures MWI subscribe expiry time (in seconds) for account X.</p> <p>The IP phone is able to successfully refresh the SUBSCRIBE for message-summary events before expiration of the SUBSCRIBE dialog.</p> <p>X is 1.</p> <p>Note: It works only if the parameter “account.X.subscribe_mwi” is set to 1 (Enabled).</p>
Format	Integer
Default Value	3600
Value	0 to 84600
Example	account.1.subscribe_mwi_expires = 3600

Parameter- voice_mail.number.X	Configuration File <MAC>.cfg
Description	<p>Configures the voice mail number for account X.</p> <p>X is 1.</p>
Format	String
Default Value	Blank
Value	String within 99 characters
Example	voice_mail.number.1 = 1234

Parameter- account.X.subscribe_mwi_to_vm	Configuration File <MAC>.cfg
Description	<p>Enables or disables the IP phone to subscribe the message waiting indicator to the voice mail number for account X.</p> <p>X is 1.</p> <p>Note: It works only if the parameters “account.X.subscribe_mwi” is set to 1 (Enabled) and “voice_mail.number.X” is configured.</p>
Format	Boolean
Default Value	0
Value	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>

Example	account.1.subscribe_mwi_to_vm = 0
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Sending RTP Stream

Parameter- multicast.codec	Configuration File <MAC>.cfg
Description	Configures a multicast codec for the IP phone to use to send an RTP stream.
Format	string
Default Value	G722
Range	Valid values are: <ul style="list-style-type: none"> • PCMU • PCMA • G729 • G722 • G723_53 • G723_63 • iLBC • G726-32
Example	multicast.codec = G722

Receiving RTP Stream

Parameter- multicast.receive_priority.enable	Configuration File <MAC>.cfg
Description	<p>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.</p> <p>If it is set to 1 (Enabled), the IP phone will answer the incoming multicast paging call with a higher priority and ignore that with a lower priority.</p>
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	multicast.receive_priority.enable =1

Parameter- multicast.receive_priority.priority	Configuration File <MAC>.cfg
Description	Configures the priority of multicast paging calls. 1 is the highest priority, 10 is the lowest priority. If it is set to 0, all incoming multicast paging calls will be automatically ignored.
Format	Integer
Default Value	10
Range	0 to 10
Example	multicast.receive_priority.priority = 10

Parameter- multicast.listen_address.X.label	Configuration File <MAC>.cfg
Description	Configures the label to be displayed on the LCD screen when receiving the RTP multicast. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	multicast.listen_address.1.label = Paging1

Parameter- multicast.listen_address.X.ip_address	Configuration File <MAC>.cfg
Description	Configures the multicast address and port number that the IP phone listens to. X ranges from 1 to 10. Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Format	String
Default Value	Blank
Range	Not Applicable
Example	multicast.listen_address.1.ip_address = 224.5.6.20:10008

Action URL

Parameter-	Configuration File
action_url.setup_completed	<MAC>.cfg
action_url.registered	
action_url.unregisterd	
action_url.register_failed	
action_url.off_hook	
action_url.on_hook	
action_url.incoming_call	
action_url.outgoing_call	
action_url.call_established	
action_url.dnd_on	
action_url.dnd_off	
action_url.always_fwd_on	
action_url.always_fwd_off	
action_url.busy_fwd_on	
action_url.busy_fwd_off	
action_url.no_answer_fwd_on	
action_url.no_answer_fwd_off	
action_url.transfer_call	
action_url.blind_transfer_call	
action_url.attended_transfer_call	
action_url.hold	
action_url.unhold	
action_url.mute	
action_url.unmute	
action_url.missed_call	
action_url.call_terminated	
action_url.busy_to_idle	
action_url.idle_to_busy	
action_url.ip_change	
action_url.forward_incoming_call	
action_url.reject_incoming_call	
action_url.answer_new_incoming_call	
action_url.transfer_finished	
action_url.transfer_failed	

Description	<p>Configures the URL for the predefined event.</p> <p>The value format is: http(s)://IP address of server/help.xml? variable name=variable value.</p> <p>Valid variable values are:</p> <ul style="list-style-type: none"> • \$mac • \$ip • \$model • \$firmware • \$active_url • \$active_user • \$active_host • \$local • \$remote • \$display_local • \$display_remote • \$call_id
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<pre>action_url.mute = http://192.168.0.20/ help.xml?model=\$model</pre>

Action URI

Parameter- features.action_uri_limit_ip	Configuration File <MAC>.cfg
Description	<p>Configures the address(es) from which Action URI will be accepted.</p> <p>For discontinuous IP addresses, multiple IP addresses are separated by commas.</p> <p>For continuous IP addresses, the format likes *.*.* and the "*" stands for the values 0~255.</p> <p>For example: 10.10.*.* stands for the IP addresses that range from 10.10.0.0 to 10.10.255.255.</p> <p>If left blank, the IP phone cannot receive or handle any HTTP GET request.</p> <p>If it is set to "any", the IP phone will accept and handle HTTP GET requests from any IP address.</p>
Format	IP Address or any
Default Value	Blank
Range	String within 511 characters
Example	<pre>features.action_uri_limit_ip = any</pre>

Server Redundancy

Parameter- account.X.sip_server.Y.address	Configuration File <MAC>.cfg
Description	Configures the IP address or domain name of the SIP server Y for account X. X is 1. Y ranges from 1 to 2.
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 256 characters
Example	account.1.sip_server.1.address = matrix.ip_pbx.com

Parameter- account.X.sip_server.Y.port	Configuration File <MAC>.cfg
Description	Configures the port of the SIP server Y for account X. X is 1. Y ranges from 1 to 2.
Format	Integer
Default Value	5060
Range	0 to 65535
Example	account.1.sip_server.1.port = 5060

Parameter- account.X.sip_server.Y.expires	Configuration File <MAC>.cfg
Description	Configures the registration expires (in seconds) of the SIP server Y for account X. X is 1. Y ranges from 1 to 2.
Format	Integer
Default Value	3600
Range	30 to 2147483647
Example	account.1.sip_server.1.expires = 3600

Parameter- account.X.sip_server.Y.retry_counts	Configuration File <MAC>.cfg
Description	Configures the retry times for the IP phone to resend requests when the SIP server Y does not respond correctly for account X. X is 1. Y ranges from 1 to 2.
Format	Integer
Default Value	3
Range	0 to 20
Example	account.1.sip_server.1.retry_counts = 3

Fallback Mode

Parameter- account.X.fallback.redundancy_type	Configuration File <MAC>.cfg
Description	Configures the registration mode for the IP phone in fallback mode. X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 0 -Concurrent registration 1 -Successive registration
Example	account.1.fallback.redundancy_type = 0

Parameter- account.X.fallback.timeout	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 after the fallback server takes over call control. It is only applicable to successive registration mode. X is 1.
Format	Integer
Default Value	120
Range	10 to 2147483647
Example	account.1.fallback.timeout = 120

Failover Mode

Parameter- account.X.sip_server.Y.failback_mode	Configuration File <MAC>.cfg
Description	Configures the mode for the IP phone to retry the primary server in failover mode. X is 1. Y ranges from 1 to 2.
Format	Integer
Default Value	0
Range	Valid values are: 0 -newRequests: all requests are sent to the primary server first, regardless of the last used server. 1 -DNSTTL: the IP phone retries to send requests to the primary server after the time-out equal to the DNSTTL configured for the server that the IP phone is registered to. 2 -registration: the IP phone retries to send REGISTER requests to the primary server when registration renewal. 3 -duration: the IP phone retries to send requests to the primary server after the time-out defined by the account.X.sip_server.Y.failback_timeout parameter.
Example	account.1.sip_server.1.failback_mode = 0

Parameter- account.X.sip_server.Y.failback_timeout	Configuration File <MAC>.cfg
Description	Configures the time-out (in seconds) for the IP phone to retry to send requests to the primary server after failing over to the current working server when the parameter "account.X.sip_server.Y.failback_mode" is set to 3 (duration). 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. X is 1. Y ranges from 1 to 2.
Format	Integer
Default Value	3600
Range	0, 60 to 65535
Example	account.1.sip_server.1.failback_timeout = 3600

Parameter- account.X.sip_server.Y.register_on_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to register to the secondary server before sending requests to the secondary server in the failover mode. X is 1. Y ranges from 1 to 2.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.sip_server.1.register_on_enable = 0

SIP Server Domain Name Resolution

Parameter- account.X.transport	Configuration File <MAC>.cfg
Description	Configures the transport type for account X. If the parameter is set to 3 (DNS-NAPTR) and no server port is given, the IP phone performs the DNS NAPTR and SRV queries for the service type and port. Note: If the outbound proxy server is required and the transport is set to DNS-NAPTR, you must set the port of outbound proxy server to 0 for NAPTR, SRV and A queries. X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 0 -UDP 1 -TCP 2 -TLS 3 -DNS-NAPTR
Example	account.1.transport = 3

Parameter- account.X.naptr_build	Configuration File <MAC>.cfg
Description	Configures UDP SRV query or TCP/TLS SRV query for the IP phone to be performed when no result is returned from NAPTR query. X is 1.
Format	Integer
Default Value	0
Range	Valid values are: 0 -UDP 1 -TCP or TLS.
Example	account.1.naptr_build = 0

Allow IP call for Peer-to-Peer (P2P) calling

Parameter- features.direct_ip_call_enable	Configuration File <MAC>.cfg
Description	It enables or disables the phone to make an IP call directly.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	features.direct_ip_call_enable= 1

LLDP

Parameter- network.lldp.enable	Configuration File <MAC>.cfg
Description	Enables or disables LLDP feature on the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled

Example	network.lldp.enable = 1
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Parameter- network.lldp.packet_interval	Configuration File <MAC>.cfg
Description	Configures the amount of time (in seconds) between the transmissions of LLDP packet. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It works only if the parameter “network.lldp.enable” is set to 1 (Enabled).
Format	Integer
Default Value	60
Range	1 to 3600
Example	network.lldp.packet_interval = 60

VLAN

Internet Port

Parameter- network.vlan.internet_port_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to insert VLAN tag on packet from the Internet port. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	network.vlan.internet_port_enable = 1

Parameter- network.vlan.internet_port_vid	Configuration File <MAC>.cfg
Description	Configures the VLAN ID that is associated with the particular VLAN. Note: If you change this parameter, the IP phone will reboot to make the change take effect.

Format	Integer
Default Value	1
Range	1 to 4094
Example	network.vlan.internet_port_vid = 1

Parameter- network.vlan.internet_port_priority	Configuration File <MAC>.cfg
Description	Configures the priority value used for passing VLAN packets. 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.
Format	Integer
Default Value	0
Range	0 to 7
Example	network.vlan.internet_port_priority = 0

PC Port

Parameter- network.vlan.pc_port_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to insert VLAN tag on packet from the PC port. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	network.vlan.pc_port_enable = 1

Parameter- network.vlan.pc_port_vid	Configuration File <MAC>.cfg
Description	Configures the VLAN ID that is associated with the particular VLAN. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	1
Range	1 to 4094
Example	network.vlan.pc_port_vid = 1

Parameter- network.vlan.pc_port_priority	Configuration File <MAC>.cfg
Description	Configures the priority value used for passing VLAN packets. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	0 to 7
Example	network.vlan.pc_port_priority = 0

DHCP VLAN Discovery

Parameter- 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.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	network.vlan.dhcp_enable = 1

Parameter- network.vlan.dhcp_option	Configuration File <MAC>.cfg
Description	Configures the DHCP option used to request the VLAN ID. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	132
Range	128 to 254
Example	network.vlan.dhcp_option = 132

QoS

Parameter- network.qos.rtpptos	Configuration File <MAC>.cfg
Description	Configures the DSCP 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.
Format	Integer
Default Value	46
Range	0 to 63
Example	network.qos.rtpptos = 46

Parameter- network.qos.signalptos	Configuration File <MAC>.cfg
Description	Configures the DSCP 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.
Format	Integer
Default Value	26
Range	0 to 63
Example	network.qos.signalptos = 26

Network Address Translation

Parameter- account.X.nat.nat_traversal	Configuration File <MAC>.cfg
Description	Enables or disables the NAT traversal for account X. X is 1.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	account.1.nat.nat_traversal = 0

Parameter- sip.nat_stun.enable	Configuration File <MAC>.cfg
Description	It enables or disables the STUN (Simple Traversal of UDP over NATs) feature on the IP phone. 0-Disabled 1-Enabled
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 99 characters
Example	sip.nat_stun.server = 218.107.220.201

Parameter- sip.nat_stun.server	Configuration File <MAC>.cfg
Description	Configures the IP address or the domain name of the STUN server.
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 99 characters
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 server.
Format	Integer
Default Value	3478
Range	1024 to 65000
Example	sip.nat_stun.port = 3478

802.1X

Parameter- network.802_1x.mode	Configuration File <MAC>.cfg
Description	Configures the types of the 802.1X authentication to use on the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-Disabled 1-EAP-MD5 2-EAP-TLS 3-PEAP-MSCHAPv2 4-EAP-TTLS/EAP-MSCHAPv2
Example	network.802_1x.mode = 1

Parameter- network.802_1x.identity	Configuration File <MAC>.cfg
Description	Configures the identity used for authenticating the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	network.802_1x.identity = admin

Parameter- network.802_1x.md5_password	Configuration File <MAC>.cfg
Description	Configures the password used for authenticating the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to EAP-MD5, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 protocols.
Format	String

Default Value	Blank
Range	String within 32 characters
Example	network.802_1x.md5_password = admin123

Parameter- network.802_1x.root_cert_url	Configuration File <MAC>.cfg
Description	Configures the access URL of the CA certificate used for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to EAP-TLS, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 protocols. The format of the certificate must be *.pem, *.crt, *.cer or *.der.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	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 used for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to the EAP-TLS protocol. The format of the certificate must be *.pem or *.cer.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	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
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IPv6

Parameter- 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.
Format	Integer
Default Value	0
Range	Valid values are: 0-IPv4 1-IPv6 2-IPv4&IPv6
Example	network.ip_address_mode = 1

Parameter- network.ipv6_internet_port.type	Configuration File <MAC>.cfg
Description	Configures the IPv6 address assignment method. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-Static IP Address
Example	network.ipv6_internet_port.type = 0

Parameter- network.ipv6_static_dns_enable	Configuration File <MAC>.cfg
Description	Enables or disables the phone to use manually configured static IPv6 DNS when the parameter "network.ipv6_internet_port.type" is set to 0 (DHCP). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled

Example	network.ipv6_static_dns_enable= 0
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Parameter- network.ipv6_internet_port.ip	Configuration File <MAC>.cfg
Description	Configures the IPv6 address when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4 & IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_internet_port.ip = 2026:1234:1:1:215:65ff:fe1f:caa

Parameter- network.ipv6_prefix	Configuration File <MAC>.cfg
Description	Configures the prefix of the IPv6 address when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	64
Range	0 to 128
Example	network.ipv6_prefix = 64

Parameter- network.ipv6_internet_port.gateway	Configuration File <MAC>.cfg
Description	Configures the gateway when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_internet_port.gateway = 3036:1:1:c3c7:c11c:5447:23a6:255

Parameter- network.ipv6_primary_dns	Configuration File <MAC>.cfg
Description	Configures the primary DNS server when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_primary_dns = 3036:1:1:c3c7:c11c:5447:23a6:256

Parameter- network.ipv6_secondary_dns	Configuration File <MAC>.cfg
Description	Configures the secondary DNS server when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_secondary_dns = 2026:1234:1:1:c3c7:c11c:5447:23a6

Audio Feature Parameters

Headset Prior

Parameter-features.headset_prior	Configuration File<MAC>.cfg
Description	Enables or disables headset prior feature. If it is set to 1 (enabled), a user needs to press the HEADSET key to activate the headset mode. The headset mode will not be deactivated until the user presses the HEADSET key again.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.headset_prior = 1

Audio Codecs

Parameter-account.X.codec.Y.enable	Configuration File<MAC>.cfg
Description	Enables or disables the IP phone to use the specific codec for account X. X is 1. Y ranges from 1 to 9.
Format	Boolean
Default Value	When Y=1, the default value is 1; When Y=2, the default value is 1; When Y=3, the default value is 1; When Y=4, the default value is 1; When Y=5, the default value is 0; When Y=6, the default value is 0; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0.
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.codec.1.enable = 1

Parameter- account.X.codec.Y.payload_type	Configuration File <MAC>.cfg
Description	Configures the codec for account X to use. X is 1. Y ranges from 1 to 8.
Format	String
Default Value	When Y=1, the default value is PCMU; When Y=2, the default value is PCMA; When Y=3, the default value is G729; When Y=4, the default value is G722; When Y=5, the default value is iLBC; When Y=6, the default value is G726-16; When Y=7, the default value is G726-24; When Y=8, the default value is G726-32; When Y=9, the default value is G726-40.
Range	Valid values are: PCMU PCMA G729 G722 G726-16 G726-24 G726-32 G726-40 iLBC
Example	account.1.codec.1.payload_type = PCMU

Parameter- account.X.codec.Y.priority	Configuration File <MAC>.cfg
Description	Configures the priority for the codec. X is 1. Y ranges from 1 to 9.
Format	Integer
Default Value	When Y=1, the default value is 2; When Y=2, the default value is 3; When Y=3, the default value is 4; When Y=4, the default value is 1; When Y=5, the default value is 0; When Y=6, the default value is 0; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0.
Range	0 to 11

Example	account.1.codec.1.priority = 1
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Parameter- account.X.codec.Y.rtpmap	Configuration File <MAC>.cfg
Description	Configures the rtpmap. X is 1. Y ranges from 1 to 9.
Format	Integer
Default Value	When Y=1, the default value is 0; When Y=2, the default value is 8; When Y=3, the default value is 18; When Y=4, the default value is 9; When Y=5, the default value is 106; When Y=6, the default value is 103; When Y=7, the default value is 104; When Y=8, the default value is 102; When Y=9, the default value is 105.
Range	0 to 127
Example	account.1.codec.1.rtpmap = 0

Ptime

Parameter- account.X.ptime	Configuration File <MAC>.cfg
Description	Configures the ptime (in milliseconds) for the codec. X is 1.
Format	Integer
Default Value	20
Range	Valid values are: 0 (Disabled) 10, 20, 30, 40, 50, 60
Example	account.1.ptime = 20

Acoustic Echo Cancellation

Parameter- voice.echo_cancellation	Configuration File <MAC>.cfg
Description	Enables or disables AEC feature on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled

Example	voice.echo_cancellation = 1
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Voice Activity Detection

Parameter- voice.vad	Configuration File <MAC>.cfg
Description	Enables or disables VAD feature on the IP phone.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	voice.vad = 1

Comfort Noise Generation

Parameter- voice.cng	Configuration File <MAC>.cfg
Description	Enables or disables CNG feature on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	voice.cng = 1

Jitter Buffer

Parameter- voice.jib.adaptive	Configuration File <MAC>.cfg
Description	Configures the type of jitter buffer.
Format	Integer
Default Value	1
Range	Valid values are: 0-Fixed 1-Adaptive
Example	voice.jib.adaptive = 1

Parameter- voice.jib.min	Configuration File <MAC>.cfg
Description	Configures the minimum delay time for jitter buffer. Note: It works only if the parameter “voice.jib.adaptive” is set to 1 (Adaptive).
Format	Integer
Default Value	60
Range	0 to 400
Example	voice.jib.min = 60

Parameter- voice.jib.max	Configuration File <MAC>.cfg
Description	Configures the maximum delay time for jitter buffer. Note: It works only if the parameter “voice.jib.adaptive” is set to 1 (Adaptive).
Format	Integer
Default Value	240
Range	0 to 400
Example	voice.jib.max = 300

Parameter- voice.jib.normal	Configuration File <MAC>.cfg
Description	Configures the fixed delay time for jitter buffer. Note: It works only if the parameter “voice.jib.adaptive” is set to 0 (Fixed).
Format	Integer
Default Value	120
Range	0 to 400
Example	voice.jib.normal = 120

Security Feature Parameters

TLS

Parameter- account.X.transport	Configuration File <MAC>.cfg
Description	Configures the transport type for account X. If it is set to 2 (TLS), the SIP message of this account will be encrypted after the successful TLS negotiation. X is 1.
Format	Integer
Default Value	0 (UDP)
Range	Valid values are: 0 -UDP 1 -TCP 2 -TLS 3 -DNS-NAPTR
Example	account.1.transport = 2

Parameter- security.trust_certificates	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to authenticate the connecting server based on the trusted certificates list. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	security.trust_certificates = 1

Parameter- security.ca_cert	Configuration File <MAC>.cfg
Description	Configures the type of certificates the IP phone used to authenticate the connecting server. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	2
Range	Valid values are: 0-Default certificates 1-Custom certificates 2-All certificates
Example	security.ca_cert = 2

Parameter- 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 connecting server. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	security.cn_validation = 0

Parameter- security.dev_cert	Configuration File <MAC>.cfg
Description	Configures the type of certificates the IP phone sends for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0

Range	Valid values are: 0 -Default certificates 1 -Custom certificates
Example	security.dev_cert = 0

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

SRTP

Parameter- account.X.srtp_encryption	Configuration File <MAC>.cfg
Description	Configures whether to use 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. X is 1.
Format	Integer
Default Value	0
Value	Valid values are: 0 -Disabled 1 -Optional 2 -Compulsory
Example	account.1.srtp_encryption = 0

Configuring Decryption Method

Parameter- auto_provision.aes_key_in_file	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to decrypt configuration files using the encrypted AES keys.
Format	Boolean
Default Value	0
Value	Valid values are: 0 -Disabled 1 -Enabled
Example	auto_provision.aes_key_in_file = 0

Parameter- auto_provision.aes_key_16.mac	Configuration File <MAC>.cfg
Description	Configures the plaintext AES key which is used to decrypt the <MAC>.cfg file. Note: It works only if the parameter "auto_provision.aes_key_in_file" is set to 0 (Disabled).
Format	String
Default Value	Blank

Range	16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z
Example	auto_provision.aes_key_16.mac = 0123456789abmins

Parameter- auto_provision.update_file_mode	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to update encrypted configuration settings only during auto provisioning.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	auto_provision.update_file_mode =0

Resource Files

Access URL of Replace Rule Template

Parameter- dialplan_replace_rule.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the replace rule template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	dialplan_replace_rule.url = http:// 192.168.10.25/dialplan.xml

Access URL of Dial-now Template

Parameter- dialplan_dialnow.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the dial-now template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	dialplan_dialnow.url = http://192.168.10.25/ dialnow.xml

Access URL of Softkey Layout Template

Parameter- custom_softkey_call_failed.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the Call Failed state.
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	<p>The following example uses HTTP to download the CallFailed state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_call_failed.url = http://10.2.8.16:8080/XMLfiles/CallFailed.xml</p>
----------------	--

Parameter- custom_softkey_call_in.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the CallIn state.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the CallIn state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_call_in.url = http://10.2.8.16:8080/XMLfiles/CallIn.xml</p>

Parameter- custom_softkey_connecting.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the Connecting state.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the Connecting state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_connecting.url = http://10.2.8.16:8080/XMLfiles/Connecting.xml</p>

Parameter- custom_softkey_dialing.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the Dialing state.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the Dialing state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_dialing.url = http://10.2.8.16:8080/XMLfiles/Dialing.xml</p>

Parameter- custom_softkey_ring_back.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the RingBack state.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the RingBack state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_ring_back.url = http://10.2.8.16:8080/XMLfiles/RingBack.xml</p>

Parameter- custom_softkey_talking.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the custom file for the softkey presented on the LCD screen when in the Talking state.
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	<p>The following example uses HTTP to download the Talking state file from the “XMLfiles” directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_talking.url = http://10.2.8.16:8080/XMLfiles/Talking.xml</p>
----------------	--

Access URL of Directory Template

Parameter- directory_setting.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the directory template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	directory_setting.url = http://192.168.1.20/favorite_setting.xml

Access URL of Super Search Template

Parameter- super_search.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the super search template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	super_search.url = http://192.168.1.20/super_search.xml

Access URL of Local Contact File

Parameter- local_contact.data.url	Configuration File <MAC>.cfg
Description	Configures the access URL of the local contact file.
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	local_contact.data.url = http://192.168.10.25/ contact.xml
----------------	---

Access URL of Remote XML 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

Troubleshooting

Log Settings

Parameter- syslog.mode	Configuration File <MAC>.cfg
Description	Configures the syslog mode. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	0-Local 1-Server
Example	syslog.mode = 1

Parameter- syslog.server	Configuration File <MAC>.cfg
Description	Configures the IP address or domain name of the syslog server where to export the log files. Note: It works only if the parameter “syslog.mode” is set to 1 (Server). If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 99 characters
Example	syslog.server = 192.168.1.50

Parameter- syslog.log_level	Configuration File <MAC>.cfg
Description	Configures the severity level of the logs to be reported to a log file. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	3
Range	0 to 6

Example	syslog.log_level = 3
----------------	----------------------

WatchDog

Parameter- watch_dog.enable	Configuration File <MAC>.cfg
Description	Enables or disables WatchDog feature.
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	watch_dog.enable = 1

Configuring DSS Key

This section provides the DSS key parameters you can configure on IP phones. DSS key consists of the programmable keys. There are 11 programmable keys available for SPARSH VP110.



The programmable key takes effect only if the IP phone is idle.

DSS key can be assigned with various key features. The parameters of the DSS key are detailed in the following:

Parameter- programablekey.X.type	Configuration File <MAC>.cfg
Description	<p>Configures key feature for the DSS key.</p> <p>For the programmable key, x ranges from 1 to 14 (For IP phones, x=1-9, 13, 14).</p> <p>For programmable keys:</p> <p>Valid types are:</p> <ul style="list-style-type: none">• N/A• Forward• DND• Recall• SMS• Directed Pickup• Speed Dial• XML Group• Group Pickup• Intercom• Multicast Paging• XML Browser• History• Menu• New SMS• Status• Hot Desking• Prefix• Zero Touch• Local Directory• Local Group• XML Directory• Keypad Lock• Directory
Format	Integer

Default Value	<p>For the programmable key,</p> <p>when x=1, the default value is 28.</p> <p>when x=2, the default value is 61.</p> <p>when x=3, the default value is 5.</p> <p>when x=4, the default value is 30.</p> <p>when x=5, the default value is 28.</p> <p>when x=6, the default value is 0.</p> <p>when x=7, the default value is 0.</p> <p>when x=8, the default value is 0.</p> <p>when x=9, the default value is 33.</p> <p>when x=14, the default value is 2.</p>
Range	<p>Valid values are:</p> <p>0-N/A</p> <p>2-Forward</p> <p>5-DND</p> <p>7-Recall</p> <p>8-SMS</p> <p>9-Directed Pickup</p> <p>13-Speed Dial</p> <p>14-Intercom</p> <p>22-XML Group</p> <p>23-Group Pickup</p> <p>24-Multicast Paging</p> <p>27-XML Browser</p> <p>28-History</p> <p>30-Menu</p> <p>32-New SMS</p> <p>33-Status</p> <p>34-Hot Desking</p> <p>40-Prefix</p> <p>41-Zero Touch</p> <p>43-Local Directory</p> <p>45-Local Group</p> <p>47-XML Directory</p> <p>50-Keypad Lock</p> <p>61-Directory</p>
Example	<p>programmablekey.7.type = 32</p>

Parameter- programablekey.X.line Note: <i>This is applicable to Local Group only; so it is applicable when programablekey.X.type = 45 only.</i>	Configuration File <MAC>.cfg
Description	Configures the desired line to apply the key feature. For the programmable key, x ranges from 1 to 14 (For IP phones, x=1-9, 13, 14). When assigning the following features, you do not need to configure this parameter: <ul style="list-style-type: none"> Local Group
Format	Integer
Default Value	1
Range	Valid values are: 1 – All Contacts 2 – Company 3 – Family 4 - Friends
Example	programablekey.X.line = 2

Parameter- programablekey.X.value	Configuration File <MAC>.cfg
Description	Configures the value for some key features. For the programmable key, x ranges from 1 to 14 (For IP phones, x = 1-9, 13, 14).
Format	String
Default Value	Blank
Range	String within 99 characters
Example	When you assign the Speed Dial to the memory key, this parameter is used to specify the number you want to dial out. programablekey.7.value= 565

Intercom Key

Parameter- programablekey.X.type	Configuration File <MAC>.cfg
Description	Configures a DSS key as an intercom key. The digit 14 stands for the key type Intercom .
Format	Integer
Value	14

Example	programmablekey.1.type = 14
----------------	-----------------------------

Parameter- programmablekey.X.value	Configuration File <MAC>.cfg
Description	Configures the intercom number. For the programmable key, x ranges from 1 to 14.
Format	String
Range	String within 99 characters
Example	programmablekey.1.value = 1008

Multicast Paging Key

Parameter- programmablekey.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 .
Format	Integer
Value	24
Example	programmablekey.1.type = 24

Parameter- programmablekey.X.value	Configuration File <MAC>.cfg
Description	Configures the multicast IP address and port number. Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Format	IP Address
Range	224.0.0.0 to 239.255.255.255
Example	programmablekey.1.value = 224.5.5.6:10008

Appendix E - Sample Configuration File

This section provides the sample configuration file necessary to configure the IP phone. Any line beginning with a pound sign (#) is considered to be a comment, unless the # is contained within double quotes.

For the Boolean fields, the meaning are as follows:

0 = Disabled

1 = Enabled.

This file contains sample configurations for the <MAC>.cfg file. The parameters included here are examples only. Not all possible parameters are shown in the sample configuration file. You can configure or comment the values as required.

You can copy the contents of the following sample configuration file and paste on a Word Processor (for example, *Notepad* which is available by default with Windows OS or third party tools like *Notepad++*) and edit the contents as per your requirements.



You must copy the contents of the Sample Configuration file which is available in the Matrix's public FTP server instead of the following file. The following Sample Configuration File is provided just for your reference and copying contents from it may result in erroneous formatting due to limitations of PDF files and thus the configuration file may not work properly. To obtain a copy of the latest Sample Configuration File available in the Matrix's public FTP server, please contact Matrix Technical Support.

Sample Configuration File

```
#!version:1.0.0.1 - Sunny
```

```
##File header "#!version:1.0.0.1" can not be edited or deleted, and must be placed in the first line.##
```

```
##If you want to reset the configuration of a parameter, set the value of the parameter to !NULL! or %NULL%. For example, local_time.ntp_server1 = %NULL%.
```

```
After the auto provisioning process is completed, the NTP server 1 will be reset to ?cn.pool.ntp.org?.##
```

```
##It is recommended that the Parameter with %NULL% or !NULL! value should not be changed while Auto Provisioning.It may affect the functionality.
```

```
#####PROVISIONER.NET#####
```

```
#Action URL,DSS Keys Configuration & Soft Key Layout,TR069 are not included in the sample configuration file.
```

```
# This Configuration file was generated from the Provisioner.net Library by {$provisioner_processor_info}
```

```
# Please don't copy sample configuration file from user manual as the updated copy of the same file is available at Matrix FTP.Please contact Matrix Technical
```

```

Support for details.
# Generated on: 01/10/2014
#
#
#####
#####
# Matrix SPARSH VP110 Auto Config Sample File

#####
#####
##              Auto Provisioning
##
#####
#####
#Configure the auto provision mode;
#0-Disabled , 1-Power on (default), 4-Repeatedly, 5-Weekly, 6-Power on +
Repeatedly, 7-Power on + Weekly;
auto_provision.mode = 1
#The above parameter can be separately configured by following parameters also.
#auto_provision.power_on =1
#auto_provision.repeat.enable =1
#auto_provision.weekly.enable =1

#Enable or disable the Plug and Play feature; 0-Disabled, 1-Enabled
(default);##This parameter needs to be 0 when the Phone is used with matrix
PBX.##
auto_provision.pnp_enable =0

#Configure the interval (in minutes) for the phone to check new configuration
files. It ranges from 1 to 43200, the default value is 1440.
#It is only applicable to "Repeatedly" and "Power on + Repeatedly" modes.
auto_provision.repeat.minutes = 1800

#Configure the start time of the day for the phone to check new configuration
files. The default value is 00:00.
#It is only applicable to "Weekly" and "Power on + Weekly" modes.

```


#If the desired start time of the day is seven forty-five a.m., the value format is 07:45.

```
auto_provision.schedule.time_from = 01:00
```

#Configure the end time of the day for the phone to check new configuration files. The default time is 00:00.

#It is only applicable to "Weekly" and "Power on + Weekly" modes.

#If the desired end time of the day is seven forty-five p.m., the value format is 19:45.

```
auto_provision.schedule.time_to = 23:00
```

#Configure the day of week for the phone to check new configuration files. The default value is 0123456.

#0-Sunday,1-Monday,2-Tuesday,3-Wednesday,4-Thursday,5-Friday,6-Saturday;

#It is only applicable to "Weekly" and "Power on + Weekly" modes.

#If the desired week is Monday, Tuesday and Wednesday, the value format is 012.

```
auto_provision.schedule.dayofweek = 0
```

#Configure the URL of the auto provisioning server.(Where TFTP server with the MAC.cfg file is configured & alive)

```
auto_provision.server.url = 192.168.153.27
```

#Configure the username and password for downloading.(As per the credentials of the TFTP server for the MAC.cfg file)

```
auto_provision.server.username = Matrix
```

```
auto_provision.server.password = 1234
```

#Enable or disable DHCP option mode; 0-Disabled, 1-Enabled (default);

#Default DHCP option=66(For TFTP server)

```
auto_provision.dhcp_option.enable = 1
```

#Configure the value (manufacturer of the device) of DHCP option 60.

```
auto_provision.dhcp_option.option60_value = Matrix SPARSH VP110
```

#Configure the custom DHCP option number. It ranges from 128 to 254.

```

auto_provision.dhcp_option.list_user_options = !NULL!

#Enables or Disable to use the AES key for the Auto Provision.0(default) or 1.
auto_provision.aes_key_in_file = 0

#Configure AES key (16 characters) for decrypting the MAC-Oriented CFG file.
auto_provision.aes_key_16.mac =

#Configure the mode of downloading configuration files for the phone. The value
is 0(default) or 1.
auto_provision.update_file_mode = 0

#enables or disables zero touch for the phone to perform provisioning during
startup.0(default) or 1.
zero_touch.enable=0

#configures the duration time (in seconds) of the phone displaying the zero-sp-
touch configuration screen when powered on. The default value is 5.Integer from 1
to 100
zero_touch.wait_time =10

#####
#####
##          Configure the access URL of firmware
##
#####
#####
#Before using this parameter, you should store the desired firmware (.70.x.x.rom)
to the provisioning server.
firmware.url = http://192.168.153.27:8081/Matrix SPARSH VP110/31.72.196.1.rom

```

```
#####
#####
##                               Network Settings                               ##
#####
#####

###   Basic Network Parameter#####
#Network Settings##Configures the Internet port type.
#If you change any of the parameter, the IP phone will reboot to make the change
take effect.
#Configure the WAN port type; 0-DHCP(Default), 1-PPPoE, 2-Static IP Address.
network.internet_port.type = 2
#"network.static_dns_enable" Enables or disables the phone to use manually
configured static IPv4 DNS when the parameter "network.internet_port.type" #is
set to 0 (DHCP).
network.static_dns_enable = 1
#Static Network Settings#
#network.ip_address_mode Configures the IP address mode.IP phones support using
the IPv4 address only,the IPv6 address only or both IPv4 and IPv6 addresses.
#0-IPv4(Default),1-IPv6,2-IPv4&IPv6
network.ip_address_mode=0
#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.
network.internet_port.ip = 192.168.153.57
#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.
network.internet_port.mask = 255.255.255.0
#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.
network.internet_port.gateway = 192.168.153.1
#Configures the primary DNS server when the Internet port type is configured as
Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6.
network.primary_dns= 8.8.8.8
#Configures the secondary DNS server when the Internet port type is configured as
Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6.
network.secondary_dns = 4.2.2.2
```

```
##### PPPoE for IPv4#####
#Configures the PPPoE user name when the Internet port type is configured as
PPPoE and the IP address mode is configured as IPv4 or IPv4&IPv6.
network.pppoe.user=test
#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.
network.pppoe.password=matrix123

##### IPv6 Network Parameters#####
#The Parameter Lines are intentionally kept commented.If you select the Network
mode as IPv6 or Both IPv4 & IPv6 you can uncomment it.)
#Configures the IPv6 address assignment method.0-DHCP,1-Static IP Address
#network.ipv6_internet_port.type = 0
#Enables or disables the phone to use manually configured static IPv6 DNS when
the parameter "network.ipv6_internet_port.type" is set to 0 (DHCP).0-
Disabled(Default),1-Enabled.
#network.ipv6_static_dns_enable= 0
#Configures the IPv6 address when the IPv6 address assignment method is
configured as Static IP Address and the IP address mode is configured as IPv6 or
IPv4&IPv6.Default value blank.
#network.ipv6_internet_port.ip =2026:1234:1:1:215:65ff:felf:caa
#Configures the prefix of the IPv6 address when the IPv6 address assignment
method is configured as Static IP Address and the IP address mode is configured
as IPv6 or IPv4&IPv6.Default 64.Range 0 to 128.
#network.ipv6_prefix = 64
#Configures the gateway when the IPv6 address assignment method is configured as
Static IP Address and the IP address mode is configured as IPv6 or
IPv4&IPv6.Default value blank.
#network.ipv6_internet_port.gateway =3036:1:1:c3c7:c11c:5447:23a6:255
#Configures the primary DNS server when the IPv6 address assignment method is
configured as Static IP Address and the IP address mode is configured as IPv6 or
IPv4&IPv6.Default 64.Range 0 to 128
#network.ipv6_primary_dns =3036:1:1:c3c7: c11c:5447:23a6:256
#Configures the Secondary DNS server when the IPv6 address assignment method is
configured as Static IP Address and the IP address mode is configured as IPv6 or
IPv4&IPv6.Default 64.Range 0 to 128
```

```
#network.ipv6_Secondary_dns =2026:1234:1:1:c3c7:c11c:5447:23a6
```

```
#### Port_Link #####
```

```
#Configures the transmission method of Internet port.0-Auto
```

```
negotiate(Default),1-Full duplex, 10Mbps,2-Full duplex, 100Mbps,3-Half duplex,  
10Mbps,4-Half duplex, 100Mbps
```

```
#We recommend that you do not change this parameter.
```

```
network.internet_port.speed_duplex = 0
```

```
#Configures the transmission method of PC port.0-Auto negotiate(Default),1-Full  
duplex, 10Mbps,2-Full duplex, 100Mbps,3-Half duplex, 10Mbps,4-Half duplex,  
100Mbps
```

```
#We recommend that you do not change this parameter.
```

```
network.pc_port.speed_duplex = 0
```

```
#Enables or disables the PC port.0-Disabled,1-Auto Negotiation(Default)
```

```
network.PC_port.enable = 1
```

```
#### LLDP ####
```

```
#Enables or disables LLDP feature on the IP phone.0-Disabled,1-Enabled(Default).
```

```
network.lldp.enable = 1
```

```
#Configures the amount of time (in seconds) between the transmissions of LLDP  
packet.Range from 1 to 3600.Default 60.
```

```
network.lldp.packet_interval = 80
```

```
#### VLAN #####
```

```
#Enables or disables the IP phone to insert VLAN tag on packet from the Internet  
port.
```

```
network.vlan.internet_port_enable = 0
```

```
#Configures the VLAN ID that is associated with the particular VLAN.Range 1 to  
4094.
```

```
network.vlan.internet_port_vid = 15
```

```
#Configures the priority value used for passing VLAN packets.7 is the highest  
priority, 0 is the lowest priority.
```

```
network.vlan.internet_port_priority=5
```

```

#Enables or disables the IP phone to insert VLAN tag on packet from the PC port.
network.vlan.pc_port_enable = 0
#Configures the VLAN ID that is associated with the particular VLAN.Range 1 to
4094.
network.vlan.pc_port_vid = 12
#Configures the priority value used for passing VLAN packets.7 is the highest
priority, 0 is the lowest priority.
network.vlan.pc_port_priority=2

#Enables or disables DHCP VLAN discovery feature on the IP phone.0-Disabled,1-
Enabled(Default).
network.vlan.dhcp_enable = 1
#Configures the DHCP option used to request the VLAN ID.Range 128 to 254.132
(Default).
network.vlan.dhcp_option = 132

###   Voice_QoS #####
#Configure the voice QOS. It ranges from 0 to 63, the default value is 46.
#Require reboot;
network.qos.rtptos = 40
#Configure the SIP QOS. It ranges from 0 to 63, the default value is 26.
#Require reboot;
network.qos.signalptos = 26

###   Local_RTP_Port#####
#Configure the maximum local RTP port. It ranges from 0 to 65535, the default
value is 11800.
#Require reboot;
network.port.max_rtpport = 11800
#Configure the minimum local RTP port. It ranges from 0 to 65535, the default
value is 11780.
#Require reboot;
network.port.min_rtpport = 11780

###   SNMP client####
#Configures the SNMP client.0-Disabled(Default),1-Enabled.

```

```

network.snmp.enable =1
#Configures the SNMP Port.Range from 1 to 65535.Default 161.
network.snmp.port =161
#Configures the SNMP server IP address.
network.snmp.trust_ip =192.168.153.222

###   WebServer####
#enables or disables the HTTP protocol for web server access.0-Disabled,1-
Enabled(Default).
wui.https_enable =1
#configures the HTTP port for webserver access.The default value is 80.Range 1 to
65535.
network.port.http =80
#enables or disables the HTTPS protocol for web server access.0-Disabled,1-
Enabled(Default).
wui.https_enable =1
#configures the HTTPS port for webserver access.The default value is 443.Range 1
to 65535.
network.port.https =443

##### 802.1x #####
#Configures the types of the 802.1X authentication to use on the IP phone.0-
Disabled (Default),1-EAP-MD5,2-EAP-TLS,3-PEAP-MSCHAPv2,4-EAP-TTLS/EAP-MSCHAPv2
network.802_1x.mode = 0
#Configures the identity used for authenticating the IP phone.String within 32
characters.
network.802_1x.identity = admin
#Configures the password used for authenticating the IP phone.The default value
is blank.String within 32 characters.
network.802_1x.md5_password =admin123
#configures the access URL of the CA certificate when the 802.1x mode is
configured as EAP-TLS,PEAP-MSCHAPV2 or EAP-TTLS/EAP-MSCHAPV2.The default value
is blank.URL within 511 characters.
network.802_1x.root_cert_url =http://192.168.1.10/ca.pem
#configures the access URL of the device certificate when the 802.1x mode is
configured as EAP-TLS.The default value is blank.URL within 511 characters.

```

```
network.802_1x.client_cert_url =http://192.168.1.10/ client.pem
```

```
### span to pc(Port Mirroring)#####
```

```
#enables or disables the phone to span data packets received in the WAN port to  
the PC port. If it is enabled, all packets from WAN port can be received by PC  
port.0-Disabled(Default).1-Enabled.
```

```
network.span_to_pc_port =1
```

```
#### Reg_Surge_Prev#####
```

```
#configures the maximum duration (in seconds) for account register after  
startup.The default value is 0.Integer from 0 to 60.
```

```
sip.reg_surge_prevention =20
```

```
#####  
#####
```

```
## SIP Account Settings ##
```

```
#####  
#####
```

```
### Basic Account Settings####
```

```
#Enables or disables the account.The.0-Disabled(Default),1-Enabled
```

```
account.1.enable =1
```

```
#Enable the Account registration to the Programmed server.0-Disabled,1-  
Enabled(Default).
```

```
account.1.register.enable=0
```

```
#configures the label displayed on the LCD screen for account.The default value  
is blank.String within 99 characters.
```

```
account.1.label =Ranger
```

```
#configures the display name for account.The default value is blank.String within  
99 characters.
```

```
account.1.display_name =Awsome
```

```
#configures the user name for register authentication for account.The default  
value is blank.String within 99 characters.
```

```
account.1.auth_name =3301
```

```
#configures the register user name for account.The default value is blank.String  
within 32 characters.
```

```
account.1.user_name =3301
```



```

#configures the password for register authentication for account.The default
value is blank.String within 99 characters.
account.1.password =12345
#Enables the Outbound Proxy for the Account.0-Disabled(Default),1-Enabled
account.1.outbound_proxy_enable =1
#Configures the Outbound server address.
account.1.outbound_host =voipmaster.matrix.com
#Configures the Outbound server address.Range 0 to 65535.Default 5060.
account.1.outbound_port =5060
#configures the local SIP port for account.The default value is 5060.Range from
1024 to 65535
account.1.sip_listen_port =8060
#configures the transport type for account.0-UDP,1-TCP,2-TLS,3-DNS-NAPTR, The
default value is 0.
account.1.sip_server.Y.transport_type =0
#It configures the transport type for the account. 0-UDP, 1-TCP, 2-TLS, 3-DNS-
NAPTR. Default is UDP.
account.1.naptr_build = 0
#enables or disables the NAT traversal for account.0-Disabled(Default),1-
Enabled.
account.X.nat.nat_traversal = 1
#Enables or disables the NAT traversal for account X.Default 0.
sip.nat_stun.enable = 1
#It enables or disables the STUN (Simple Traversal of UDP over NATs) feature on
the IP phone.Default 0.
sip.nat_stun.server = stun.voipbuster.com
#Configures the IP address or the domain name of the STUN server.Default is
blank.String within 99 characters
sip.nat_stun.port = 3478
#configures the port of the STUN server for account.The default value is
3478.Range 1024 to 65000
### Failback(Server Redundancy)####
#Configures the registration mode for the IP phone in fallback mode.0-Concurrent
registration(Default),1-Successive registration.
account.1.fallback.redundancy_type =0
#Configures the time interval (in seconds) for the IP phone to detect whether the

```

working server is available by sending the registration request after the fallback server takes over call control.

#It is only applicable to successive registration mode.Range 10 to 2147483647
Default:120

account.1.fallback.timeout =120

#Configures the IP address or domain name of the SIP server 1 for account.Default Blank.String within 256 characters

account.1.sip_server.1.address =matrix.pbx.com

#Configures the port of the SIP server 1 for account.0 to 65535.Default 5060.

account.1.sip_server.1.port =5060

#Configures the registration expires (in seconds) of the SIP server 1 for account.Default 3600.Range 30 to 2147483647.

account.1.sip_server.1.expires =2500

#Configures the retry times for the IP phone to resend requests when the SIP server 1 does not respond correctly for account.Range 0 to 20.Default 3.

account.1.sip_server.1.retry_counts =4

#configures the mode for the phone to retry the primary server in failover mode for account.

#0-newRequests: all requests are forwarded to the primary server first,regardless of the last used server.1-DNSTTL: the phone retries to use the primary server after the timeout of the DNSTTL configured for the SIP server.

#2-Registration: the phone retries to use the primary server when the SIP server's registration requires renewal.

#3-duration: the phone retries to use the primary server after the timeout defined by the parameter "account.1.fallback_timeout".

#The default value is 0.

account.1.sip_server.1.failback_mode =0

#configures the timeout (in seconds) for the phone to retry to use the primary server after failing over to the current working server for account when the parameter "account.1.sip_server.1.failback_mode " is set to 3 (duration).

#If you set the parameter between 1 and 59, the timeout will be 60 seconds.The default value is 3600.Range 0 & 60 to 65535

account.1.sip_server.1.failback_timeout =60

#enables or disables the phone to send registration requests to the secondary server for account when encountering a failover.0-Disabled(Default),1-Enabled.

account.1.sip_server.1.register_on_enable =0

```

#The following configuration for server 2 is same as above description.
account.1.sip_server.2.address =matrix.pbx2.com
account.1.sip_server.2.port =5090
account.1.sip_server.2.expires =3000
account.1.sip_server.2.retry_counts =5
account.1.sip_server.2.failback_mode =0
account.1.sip_server.2.failback_timeout =70
account.1.sip_server.2.register_on_enable =0

###   Features on the Account Page#####
#enables or disables anonymous call feature for account.0-Disabled(Default),1-
Enabled
account.1.anonymous_call = 0
#configures the phone to send anonymous on/off code to activate/deactivate the
server-side anonymous call feature for account.0-Off Code(Default),1-On Code.
account.1.send_anonymous_code = 0
#configures the code for activating the server-side anonymous call feature for
account  when the parameter "account.1.send_anonymous_code" is set to 1 (On
Code) .
#The default value is blank.String within 32 characters.
account.1.anonymous_call_oncode =182
##configures the code for deactivating the server-side anonymous call feature for
account  when the parameter "account.1.send_anonymous_code" is set to 0 (Off
Code) .
#The default value is blank.String within 32 characters.
account.1.anonymous_call_offcode =185
#enables or disables anonymous call rejection feature for account.0-
Disabled(Default),1-Enabled.
account.1.reject_anonymous_call = 1
#configures the code for activating the server-side anonymous call rejection
feature for account.The default value is blank.String within 32 characters
account.1.anonymous_reject_oncode = *74
##configures the code for deactivating the server-side anonymous call rejection
feature for account.The default value is blank.String within 32 characters
account.1.anonymous_reject_offcode = *75
#enables or disables the phone to record the missed call of account.0-Disabled,0-

```

```

Disabled(Default) .
account.1.missed_calllog=1
#enables or disables auto answer feature for account.0-Disabled(Default),1-
Enabled.
account.1.auto_answer =0

#### CODEC Settings #####
#Different values of "Y" according to the CODEC.1-PCMU;2-PCMA;3-G723_53;4-
G723_63;5-G729;6-G722;7-iLBC;8-G726-32
#Enables or disables the IP phone to use the specific codec for account.0-
Disabled,1-Enabled. (Y=1,2,5,6 are enabled by Default) .
account.1.codec.1.enable = 1
account.1.codec.2.enable = 1
account.1.codec.3.enable = 1
account.1.codec.4.enable = 1
account.1.codec.5.enable = 1
account.1.codec.6.enable = 1
account.1.codec.7.enable = 1
account.1.codec.8.enable = 0

#Configures the priority for the codec.Range 0 to 7.0-Highest and 7-Lowest local
priority.
account.1.codec.5.priority = 0
account.1.codec.2.priority = 1
account.1.codec.1.priority = 2
account.1.codec.4.priority = 3
account.1.codec.3.priority = 4
account.1.codec.7.priority = 5
account.1.codec.8.priority = 5
account.1.codec.6.priority = 7

#####
#####
##                               SIP Account Advanced Settings                               ##
#####
#####

```

```

#configures the type of keep-alive packets sent by the phone to the NAT device to
keep the communication port open so that NAT can continue to function for account
#0-Disabled,1-Default: the phone sends UDP packets to the server,2-Option: the
phone sends SIP OPTION packets to the server,3-Notify: the phone sends SIP NOTIFY
packets to the server.
account.1.nat.udp_update_enable =0
#configures the keep-alive interval (in seconds) for account.The default value is
30.Range is 15 to 2147483647
account.1.nat.udp_update_time =60
#enables or disables NAT Rport feature for account.0-Disabled(Default),1-
Enabled.
account.1.nat.rport =1
#configures the session timer T1 (in seconds) for account.The default value is
0.5.Float from 0.5-10s.
sip.timer_t1 =0.5
#configures the session timer T2 (in seconds) for account.The default value is
4.Float from 2-40s.
sip.timer_t2 =4
#configures the session timer T4 (in seconds) for account.The default value is
5.Float from 2.5-60s.
sip.timer_t4=5
#configures the DTMF type for account.0-INBAND,1-RFC2833(Default),2-SIP INFO,3-
AUTO or SIP INFO.
account.1.dtmf.type =1
#configures the RFC2833 payload for account.The default value is 101.From 96 to
127.
account.1.dtmf.dtmf_payload =101
#configures the DTMF info type when the DTMF type is configured as "SIPINFO",
"AUTO or SIP INFO" for account.
#0-Disabled(Default),1-DTMF-Relay,2-DTMF,3-Telephone-Event.
account.1.dtmf.info_type =0
#Enables the PRACK for the account.enables or disables 100 reliable
retransmission feature for account.0-Disabled(Default),1-Enabled.
account.1.100rel_enable =0
#enables or disables the phone to subscribe the message waiting indicator for
account.0-Disabled(Default),1-Enabled.

```

```

account.1.subscribe_mwi =1
#configures the interval (in seconds) of MWI subscription for account.The default
value is 3600.Range 0 to 84600.
account.1.subscribe_mwi_expires =3600
#enables or disables the phone to subscribe to the voice mail number for the
message waiting indicator for account.
#This method is not supported by Matrix PBX.Here the SUBSCRIBE for VM sent to
Voice mail retrieval number.0-Disabled(Default),1-Enabled.
account.1.subscribe_mwi_to_vm =0
#configures the voice mail number for account.(Voice Mail Retrieval Number).The
default value is blank.String within 99 characters.
voice_mail.number.1=3931
#configures the source caller identity for presentation when receiving an
incoming call for account.
#0-FROM(Default),1-PAI,2-PAI-FROM,3-PRID-PAI-FROM,4-PAI-RPID-FROM,5-RPID-FROM
account.1.cid_source =1
#enables or disables the session timer for account.0-Disabled(Default),1-
Enabled.
account.1.session_timer.enable =0
#configures the interval (in seconds) for refreshing the SIP session for
account.The default value is 1800.Range The default value is 1800.
account.1.session_timer.expires =1800
#configures the refresher of the session timer for account.0-UAC(Default),1-UAS.
account.1.session_timer.refresher =0
#enables or disables the "user=phone" carried in the INVITE message for
account.0-Disabled(Default),1-Enabled.
account.1.enable_user_equal_phone =1
#configures whether to use voice encryption service for account.0-
Disabled(Default),1-Optional,2-Compulsory.
account.1.srtp_encryption =0
#configures the RTP packet time for account.The default value is 0.Valid values 0
(Disabled),10,20,30,40,50 or 60.
account.1.ptime =0
#enables or disables the phone to carry the MAC address of Phone in the REGISTER
message for account.0-Disabled(Default),1-Enabled.
account.1.register_mac =0

```

```

#enables or disables the phone to carry the line number in the REGISTER message
for account.However SPARSH VP110 has only one Line.0-Disabled(Default),1-
Enabled.

account.1.register_line =0

#configures the conference type for account.0-Local Conference(Default),2-
Network Conference.

account.1.conf_type =0

#configures the network conference URI for account.SIP URI within 511
characters.The default value is blank.

account.1.conf_uri =conference@example.com

#configures the URI of the Music On Hold server for account.The default value is
blank.SIP URI within 256 characters.

account.1.music_server_uri =<10.1.3.165>

#enables or disables the distinctive ring tones by the Alert-Info SIP header for
account.0-Disabled(Default),1-Enabled.

account.1.alert_info_url_enable =0

#enables or disables the phone to un-register account after a reboot prior to
register.0-Disabled(Default),1-Enabled.

account.1.unregister_on_reboot =0

#configures the period (in seconds) of the BLF subscription for account.The
default value is 1800.Integer from 30 to 2147483647.

account.1.blf.subscribe_period =30


#####
#####
##                               Call Forward                               ##
#####
#####

#Enable or disable the always forward feature; 0-Disabled (default), 1-Enabled;
forward.always.enable = 0

#Configure the target number that the phone will forward the call to.
forward.always.target = 3312

#Configure the always forward on code and off code.

```

```

forward.always.on_code = *17
forward.always.off_code = *18

#Enable or disable the busy forward feature; 0-Disabled (default), 1-Enabled;
forward.busy.enable = 0
forward.busy.target = 3313
forward.busy.on_code = *11
forward.busy.off_code = *12

#Enable or disable the no answer forward feature; 0-Disabled (default), 1-
Enabled;
#Configure the waiting ring times before forwarding. Incoming calls will be
forwarded when not answered after N*6 seconds.N ranges from 0 to 20, the default
value is 2.##
forward.no_answer.enable = 1
forward.no_answer.target = 3318
forward.no_answer.timeout = 4
forward.no_answer.on_code = *15
forward.no_answer.off_code = *16

#Enable or disable the phone to forward the call to the international number (the
prefix is 00); 0-Disabled , 1-Enabled(default);
forward.international.enable = 1

#####
#####
##                                DND                                ##
#####
#####
#enables or disables the phone to receive incoming calls from authorized numbers
when DND feature is enabled.0-Disabled(default), 1-Enabled.
features.dnd.emergency_enable =0
#configures the numbers the phone will receive incoming calls from when DND
feature is enabled.Multiple numbers are separated by commas.The default value is
blank.String within 511 characters.
features.dnd.emergency_authorized_number =511,512

```



```

#It enables or disables DND feature.0-Disabled(default), 1-Enabled.
features.dnd.enable=0
#configures the DND on code when.the DND mode is configured as Phone.The default
value is Blank.String within 32 characters.
features.dnd.on_code =188
#configures the DND off code when.the DND mode is configured as Phone.The default
value is Blank.String within 32 characters.
features.dnd.off_code =189

#####
#####
##                               Audio                               ##
#####
#####
#enables or disables the phone to play the call waiting tone.0-Disabled , 1-
Enabled(default)
call_waiting.tone =1
#enables or disables the phone to play key tone when pressing any key.0-Disabled,
1-Enabled(default)
features.key_tone =1
#enables or disables the phone to play key tone when pressing the send key.0-
Disabled , 1-Enabled(default)
features.send_key_tone =1
#configures the phone to continue to play the dial tone after inputting the
preset numbers in the dialing screen.Integer within 6 digits.The default value is
blank.
#As per below example The phone will continue to play the dial tone after
inputting "125" in the dialing screen.
#If it is left blank, the phone will not play the dial tone after inputting
numbers in the dialing screen.
features.redial_tone = 125
#configures the ringer device for the phone in the headset mode.0-Use
Speaker(Default),1-Use Headset,2-Use Headset & Speaker.
features.ringer_device.is_use_headset =0

```

```
#####
#####
##                               Intercom                               ##
#####
#####
#enables or disables the phone to automatically answer an incoming intercom
call.0-Disabled , 1-Enabled(default).
features.intercom.allow =1
#enables or disables the phone to mute the speaker when answering an intercom
call.0-Disabled(default), 1-Enabled.
features.intercom.mute =0
#enables or disables the phone to play a warning tone when answering an intercom
call.0-Disabled , 1-Enabled(default).
features.intercom.tone =1
#enables or disables the phone to barge in an intercom call.0-Disabled(default),
1-Enabled.
features.intercom.barge =0

#####
#####
##                               Transfer                               ##
#####
#####
#enables or disables the transferee party's phone to prompt a missed call on the
LCD screen before displaying the caller ID when performing a semi-attended
transfer.0-Disabled , 1-Enabled(default).
transfer.semi_attend_tran_enable =1
#enables or disables the phone to complete the blind transfer through on-hook.0-
Disabled , 1-Enabled(default).
transfer.blind_tran_on_hook_enable =1
#enables or disables the phone to complete the attended transfer through on-
hook.0-Disabled , 1-Enabled(default).
transfer.on_hook_trans_enable =1
#enables or disables the phone to transfer call to the two parties after a local
conference call hangs up.0-Disabled(default), 1-Enabled.
transfer.tran_others_after_conf_enable =0
```

```
#####
#####
##                               Call Pickup                               ##
#####
#####
#enables or disables the phone to display the DPickup soft key when the phone is
in the pre-dialing screen.0-Disabled(default), 1-Enabled.
features.pickup.direct_pickup_enable =0
#configures the directed call pickup code.The default value is blank.String
within 32 characters.
account.1.direct_pickup_code =12
#enables or disables the phone to display the GPickup soft key when the phone is
in the pre-dialing screen.0-Disabled(default), 1-Enabled.
features.pickup.group_pickup_enable =0
#configures the group call pickup code.The default value is blank.String within
32 characters.
account.1.group_pickup_code =*4

#####
#####
##                               Remote Control                               ##
#####
#####
#configures the IP address of the push XML server.The default value is blank.IP
address.
push_xml.server =192.168.153.16
#enables or disables the phone to use the push XML via SIP NOTIFY message.0-
Disabled(default), 1-Enabled.
push_xml.sip_notify =0
#enables or disables the phone to block displaying the push XML screen when in
calling status.0-Disabled(default), 1-Enabled.
push_xml.block_in_calling =0
#configures the IP address of server from which the phone receives the action URI
requests.
#Multiple IP addresses are separated by commas.
#If it is set to any, the phone will receive action URI requests from any
```

server.If it is left blank, the phone will not receive action URI requests.
#The default value is blank.Valid is IP address or any.
features.action_uri_limit_ip=any

```
#####  
#####  
##                               Phone Lock                               ##  
#####  
#####  
#enable or disable the feature Phone Lock.  
phone_setting.phone_lock.enable = 1  
#configures the Phone lock type.0-All Key,1-Function Keys,2-Menu Keys.  
phone_setting.phone_lock.lock_key_type = 0  
#configures the password for unlocking the keypad.The default value is  
123.characters within 15 digits.  
phone_setting.phone_lock.unlock_pin =123  
#configures the interval (in seconds) to automatically lock the keypad.  
#The default value is 0 (the keypad is locked only by long pressing the pound key  
or pressing the keypad lockkey).Integer from 0 to 3600.  
phone_setting.phone_lock.lock_time_out =0  
#configures emergency numbers.Multiple emergency numbers are separated by  
commas.The default value is 112,911,110.String within 99 characters  
phone_setting.emergency.number =100,101,102,108
```

```
#####  
#####  
##                               Power LED                               ##  
#####  
#####  
#Different Power LED configuration.Values 0 or 1.Ring & Voice/Text Mail Power  
Light Flash are enabled by default.  
phone_setting.common_power_led_enable =0  
phone_setting.ring_power_led_flash_enable =1  
phone_setting.mail_power_led_flash_enable =1  
phone_setting.mute_power_led_flash_enable =0  
phone_setting.hold_and_held_power_led_flash_enable=0
```

phone_setting.talk_and_dial_power_led_enable =0

#####

Access URL of Resource Files

#####

#Various XML files URL can be configured this way.

dialplan_dialnow.url =http://192.168.153.27:8081/dialnow.xml

dialplan_replace_rule.url =http://192.168.153.27:8081/replacerule.xml

local_contact.data.url =http://192.168.153.27:8081/localcontact.xml

remote_phonebook.data.1.url =http://192.168.153.27:8081/remotephonebook.xml

directory_setting.url =http://192.168.153.27:8081/favorite_setting.xml

super_search.url =http://192.168.153.27:8081/super_search.xml

#####

Features -General Information

#####

#enables or disables call waiting

call_waiting.enable =0-Disabled , 1-Enabled(default).

#configures the call waiting on code.The default value is blank.String within 32 characters.

call_waiting.on_code =333

#configures the call waiting off code.The default value is blank.String within 32 characters.

call_waiting.off_code =334

#enables or disables the phone to automatically redial the called number when the called party is temporarily unavailable.

#0-Disabled(default), 1-Enabled.

auto_redial.enable = 0

#configures the interval (in seconds) for the phone to wait before redial.The default value is 10.Integer from 1 to 300.

auto_redial.interval = 10

#configures the auto redial times when the called party is temporarily

unavailable. The default value is 10. Integer from 1 to 300.

```
auto_redial.times = 10
```

#configures the "#" or "*" key as a send key. 0-Disabled, 1-# key(Default), 2-* key.

```
features.key_as_send =1
```

#enables or disables the phone to reserve the pound sign when dialing out. 0-Disabled (convert the pound sign into "%23"), 1-Enabled(Default).

#Web UI Path:Features-->General Information-->Reserve # in User Name

```
sip.use_23_as_pound =1
```

#configures the hotline number. The default value is blank. String within 32 characters.

```
features.hotline_number =1212
```

#configures the delay time (in seconds) for the phone to dial out the hotline number automatically. The default value of delay time is 4. Integer from 0 to 10.

```
features.hotline_delay =4
```

#configures the duration time (in seconds) for the busy tone. The default value is 0.0, 3 or 5

```
features.busy_tone_delay =0
```

#configures a return code and reason of SIP response messages when rejecting an incoming call. 404-No Found, 480-Temporarily not available, 486-Busy here(Default).

```
features.normal_refuse_code =486
```

#configures a return code and reason of SIP response messages when rejecting an incoming call by DND. 404-No Found, 480-Temporarily not available (Default), 486-Busy here.

```
features.dnd_refuse_code =480
```

#enables or disables call completion feature. 0-Disabled(default), 1-Enabled.

```
features.call_completion_enable =0
```

#configures the delay time (in seconds) for the dial-now rule. The default value is 0. Integer from 1 to 14.

```
phone_setting.dialnow_delay =0
```

#enables or disables the phone to support RFC 2543 hold (c=0.0.0.0). 0-Disabled(default), 1-Enabled.

```
sip.rfc2543_hold =0
```

#enables or disables the phone to keep sending SIP requests to the outbound server in a dialog. 0-Disabled, 1-Enabled(default). It takes effect after a reboot.

```
sip.use_out_bound_in_dialog =1
```

```

#enables or disables the phone to deal with the 180 SIP message received after
the 183 SIP message.0-Disabled, 1-Enabled(default).
phone_setting.is_deal180 =1
#enables or disables the phone to provide the logon wizard during startup.0-
Disabled(default), 1-Enabled.
phone_setting.logon_wizard=0
#configures the prefix of the password-dial number.
#Web UI Path--->Features-->General Information-->PswPrefix.
#For example, set the prefix to 12 and the length to 3, when you want to dial the
number 123456, the entered number is displayed as 12***6 on the LCD screen.
#The default value is blank.String within 32 characters.
features.password_dial.prefix =12
#configures the number of digits to be hidden.The hidden digits are displayed as
asterisks on the LCD screen. The default value is blank.Integer from 0 to 99.
#Web UI Path--->Features-->General Information-->PswLength.
features.password_dial.length =6
#enables or disables password dial feature for the phone.0-Disabled(default), 1-
Enabled.
#Web UI Path--->Features-->General Information-->PswDial.
features.password_dial.enable =0
#enables or disables the phone to save the call history.0-Disabled, 1-
Enabled(default).
features.save_call_history =1
#enables or disables the phone to suppress the display of DTMF digits.0-
Disabled(default), 1-Enabled.
features.dtmf.hide =0
#enables or disables the phone to display the DTMF digits for a short period
before displaying as asterisks when the parameter "features.dtmf.hide" is set to
1.0-Disabled(default), 1-Enabled.
features.dtmf.hide_delay =0
#enables or disables the phone to play a local DTMF tone.0-Disabled, 1-
Enabled(default).
features.play_local_dtmf_tone_enable=1
#configures the repetition times for sending the DTMF packets.The default value
is 3.1, 2 or 3
features.dtmf.repetition =3

```

```

#enables or disables the phone to play a warning tone when there is a call on
hold.0-Disabled, 1-Enabled(default).
features.play_hold_tone.enable =1
#configures the interval (in seconds) for playing a hold warning tone.The default
value is 30.Integer from 3 to 3600.
features.play_hold_tone.delay =30
#enables or disables the phone to mute an active call.0-Disabled, 1-
Enabled(default).
features.allow_mute =1
#configures the delay time (in seconds) before the phone automatically answers an
incomingcall.The default value is 1.Integer from Integer from 1 to 4
features.auto_answer_delay=1
#enables or disables headset prior.0-Disabled(default), 1-Enabled.
features.headset_prior =0
#enables or disables the phone to send DTMF sequences for transfer function when
pressing the transfer soft key or the TRAN key.0-Disabled(default), 1-Enabled.
features.dtmf.replace_tran =0
#configures DTMF sequences for transfer key function to be sent. It consists of
0-9, A-D, * and #. The default value is blank.String within 32 characters.
features.dtmf.transfer =123
#configures whether to send one pound key by pressing the pound key twice when
the pound key is configured as a send key.0-Send one pound key(Default),1-Do not
send any pound key
features.send_pound_key =0
#enables or disables the phone to forward incoming calls to international
numbers.0-Disabled, 1-Enabled(default).
forward.international.enable=1
#enables or disables forward diversion feature.0-Disabled, 1-Enabled(default).
features.fwd_diversion_enable =1
#enables or disables the phone to enter Transfer to menu during multiple calls
when pressing the transfer soft key or TRAN key.0-Disabled, 1-Enabled(default).
transfer.multi_call_trans_enable =1
#configures the web access time-out (in minutes).The default value is 5.It takes
effect after a reboot.Integer from 1 to 1000.
features.relog_offtime =600
#configures the characters the phone filters when dialing.

```



```

#If the dialed number contains configured characters, the phone will
automatically filter these characters when dialing.The default value is "-,".
#String within 99 characters.
#Web UI Path--->Features-->General Information-->Call Number Filter.
features.call_num_filter =-,
#configures the logo mode of the LCD screen.0-Disabled,1-System logo(Default),2-
Custom logo
phone_setting.lcd_logo.mode =1
#configures the access URL of logo file.The default value is blank.URL within 511
characters.
lcd_logo.url =http://192.168.153.27:8081/matrix.dob
#enables or disables the phone to make an IP call directly.0-Disabled, 1-
Enabled(default).
features.direct_ip_call_enabe =1
#enables or disables the phone to play the warning tone when receiving a voice
mail.0-Disabled, 1-Enabled(default).
features.voice_mail_tone_enable =1
#configures the client host name for DHCP option 12.The default value is Matrix
SPARSH VP110.
#Web UI Path--->Features-->General Information-->DHCP Hostname.
network.dhcp_host_name =Matrix SPARSH VP110

#####
#####
##                                LANGUAGE Settings                                ##
#####
#####
#configures the language of the web user interface.Options are
English,Chinese_S,German,Italian,Polish,Turkish.
lang.wui =English
#configures the language of the phone user interface.Options are
English,French,German,Spanish,Portuguese,Italian,Polish,Turkish,Chinese_S
lang.gui =English

```

```

#####
#####
##                               Live Dialpad                               ##
#####
#####
#enables or disables the phone to automatically dial out the entered digits in
the pre-dialing screen.0-Disabled, 1-Enabled(default).
phone_setting.predial_autodial =1
#configures the time (in seconds) for the phone to automatically dial out the
entered digits without pressing a send key.The default value is 5.Integer from 1
to 14.
phone_setting.inter_digit_time =5

#####
#####
##                               LCD parameter                               ##
#####
#####
#configures the contrast of the LCD screen.Integer from 1 to 10.The default value
is 6.
phone_setting.contrast =6

#####
#####
##                               Watchdog                               ##
#####
#####
#enables or disables Watch Dog feature. If it is enabled, the phone will reboot
automatically when the system is broken down.0-Disabled, 1-Enabled(default).
watch_dog.enable =1

#####
#####
##                               Phone Ring                               ##
#####
#####

```

```

#configures the ring tone for the phone.The default value is
Ring1.wav.Ring1.wav,Ring2.wav,Ring3.wav,Ring4.wav,Ring5.wav
phone_setting.ring_type =Ring1.wav
#configures the access URL of the custom ring tone file.The default value is
blank.URL within 511 characters.
ringtone.url =http://192.168.153.27:8081/Ringtone9.wav
#deletes all custom ring tone files.The valid value is: http://localhost/all.The
default value is blank.
ringtone.delete =!NULL!

#####
#####
##                               Time Settings                               ##
#####
#####

#Configure the time zone and time zone name. The time zone ranges from -11 to
+12, the default value is +8.
#The default time zone name is India(Calcutta).
#Refer to User Guide for more available time zones and time zone names.
local_time.time_zone = +5:30
#local_time.time_zone_name = India(Calcutta)
local_time.time_zone_name = India(Calcutta)

#Configure the domain name or the IP address of the NTP server. The default value
is cn.pool.ntp.org.
local_time.ntp_server1 = !NULL!
local_time.ntp_server2 = 192.168.153.222

#Configure the update interval (in seconds) when using the NTP server. The
default value is 1000.15 to 86400s
local_time.interval = 1000

#Configure the daylight saving time feature; 0-Disabled, 1-Enabled, 2-Automatic
(default);
local_time.summer_time = 0

```

```

#Configure the DST type when the DST feature is enabled; 0-By Date (default), 1-
By Week;
local_time.dst_time_type = 0

#Configure the start time of DST. The default value is 1/1/0.
#If the DST type is configured as By Date, the value format is Month/Day/Hour.
For example, the value 5/20/10 means the start time is at 10:00 on May 20.
#If the DST type is configured as By Week, the value format is Month/Day of Week/
Day of Week Last in Month/Hour of Day.
#For example, the value 1/4/2/5 means the start time is at 5 o'clock on Tuesday
of the 4th week in January.
local_time.start_time = !NULL!

#Configure the end time of DST. The default value is 12/31/23. The value format
is the same to the start time.
local_time.end_time = !NULL!

#Configure the offset time (in seconds). It ranges from -300 to 300, the default
value is 60.
local_time.offset_time = !NULL!

#Configure the time format; 0-12 Hour, 1-24 Hour (default);
local_time.time_format = 1

#Configure the date format; 0-WWW MMM DD (default), 1-DD-MMM-YY, 2-YYYY-MM-DD, 3-
DD/MM/YYYY, 4-MM/DD/YY, 5-DD MMM YYYY, 6-WWW DD MMM;
local_time.date_format = 0

#enables or disables the phone to update time with the offset time obtained from
the DHCP server.; 0-Disabled (default), 1-Enabled;
local_time.dhcp_time = 0

#Enable or disable the manual time setting. 0-Disabled (default), 1-Enabled;
local_time.manual_time_enable = 0

```

```
#####
#####
##                               Syslog                               ##
#####
#####

#configures the phone to export log files to a syslog server or the local
system.0-Local (Default),1-Server.It takes effect after a reboot.
syslog.mode=1
#configures the IP address or domain name of the syslog server when exporting log
to the syslog server.
#It takes effect only if the parameter "syslog.mode" is configured as Server.The
default value is blank.It takes effect after a reboot.
syslog.server=192.168.153.27
#configures the detail level of syslog information to be exported.0 means nothing
and 6 means all.The default value is 3.It takes effect after a reboot.
syslog.log_level =6

#####
#####
##                               Dial Plan                               ##
#####
#####

#configures the entered number to be replaced.String within 32 characters.X
ranges from 1 to 100.String within 32 characters.
dialplan.replace.prefix.X =
#configures the alternate number to replace the entered number. The default value
is blank.String within 32 characters.
dialplan.replace.replace.X=
#configures the dial now rule.The default value is blank.(X ranges from 1 to
100).String within 511 characters.
dialplan.dialnow.rule.1 =5xx
#configures the delay time (in seconds) for the dial-now rule.)The default value
is 0.Integer from 1 to 14.
phone_setting.dialnow_delay =0
#configures the area code.The default value is blank.String within 16 characters.
```

```

dialplan.area_code.code =+91
#configures the minimum length of the number prefixed with the area code.The
default value is 1.Integer from 1 to 15.
dialplan.area_code.min_len=5
#configures the Maximum length of the number prefixed with the area code.The
default value is 1.Integer from 1 to 15.The value must be larger than the minimum
length.
dialplan.area_code.mac_len=8
#configures the block out string.The default value is blank.String within 32
characters.
dialplan.block_out.number.1 = 5532
dialplan.block_out.number.2 = 5858
dialplan.block_out.number.3 =
dialplan.block_out.number.4 =
dialplan.block_out.number.5 =
dialplan.block_out.number.6 =
dialplan.block_out.number.7 =
dialplan.block_out.number.8 =
dialplan.block_out.number.9 =
dialplan.block_out.number.10 =

#####
#####
##                               Voice Tone                               ##
#####
#####
voice.vad = 0
voice.cng = 1
voice.echo_cancellation = 1
voice.jib.adaptive = 1
voice.jib.min = 0
voice.jib.max = 300
voice.jib.normal = 120

```

```
#####
#####
##                                Rings(Distinctive Rings) ##
#####
#####
#distinctive_ring_tones.alert_info.1.text =XYZ(configures the internal ringer
text for distinctive ring tone).String within 32 characters.
#distinctive_ring_tones.alert_info.1.ringer =1(configures the desired ring tones
for each text.)Integer from 1 to 5.
distinctive_ring_tones.alert_info.1.text = Friends
distinctive_ring_tones.alert_info.1.ringer = 1
distinctive_ring_tones.alert_info.2.text =
distinctive_ring_tones.alert_info.2.ringer =
distinctive_ring_tones.alert_info.3.text =
distinctive_ring_tones.alert_info.3.ringer =
distinctive_ring_tones.alert_info.4.text =
distinctive_ring_tones.alert_info.4.ringer =
distinctive_ring_tones.alert_info.5.text =
distinctive_ring_tones.alert_info.5.ringer =
distinctive_ring_tones.alert_info.6.text =
distinctive_ring_tones.alert_info.6.ringer =
distinctive_ring_tones.alert_info.7.text =
distinctive_ring_tones.alert_info.7.ringer =
distinctive_ring_tones.alert_info.8.text =
distinctive_ring_tones.alert_info.8.ringer =
distinctive_ring_tones.alert_info.9.text =
distinctive_ring_tones.alert_info.9.ringer =
distinctive_ring_tones.alert_info.10.text =
distinctive_ring_tones.alert_info.10.ringer =

#####
#####
##                                Tones                                ##
#####
#####
#Define the voice tone, the valid values can be Custom (default) or voice tone of
```

different countries. For example, United States, France, Germany and so on.

```
voice.tone.country = Custom
```

#Customize the tone when the "voice.tone.country" is configured as Custom.

#The value format: Frequency/Duration.

#Frequency ranges from 200 to 7000. When 0 is used to define the frequency, it means a pause between tones.

#A tone can be composited at most four different frequencies (the value format is: F1+F2+F3+F4).

#Duration is the time duration (in milliseconds) of ringing the tone. It ranges from 0 to 30000ms.

#At most eight tones can be configured for dial, ring, busy and so on, each tone is separated by a comma.

#For example, voice.tone.dial = 100/200,200/150,300+400+500+1200/1000,0/1200,500+900/800,5000+2000+3000/6000,0/1500,3600/1800

```
voice.tone.dial = 1
voice.tone.ring = 1
voice.tone.busy = 1
voice.tone.congestion = 1
voice.tone.callwaiting = 1
voice.tone.dialrecall = 1
voice.tone.info = 1
voice.tone.stutter = 1
voice.tone.message = 1
voice.tone.autoanswer = 1
```

```
#####
#####
##                               Volume Settings                               ##
#####
#####
#Configures the Ringer Volume.Integer from 1 to 15
voice.ring_vol=15
```



```
#####
#####
##                               Remote Phonebook Related Parameters##
#####
#####

#enables or disables the phone to perform a remote phone book search when
receiving an incoming call.0-Disabled(default),1-Enabled
features.remote_phonebook.enable =0
#configures the interval (in seconds)for the phone to update the data of the
remote phone book from the remote phone book server.Integer from 3600 to
2592000.The default value is 21600.
features.remote_phonebook.flash_time =21610
#The following parameter required to name the remote Phonebook.String within 99
characters.
remote_phonebook.data.1.name=Remote Contacts

#####
#####
##                               Receiving of Multicast Paging  ##
#####
#####

#configures the codec of multicast paging(Sending).Codecs
PCMU,PCMA,G729,G722(Default),G726-32,G723_53.
multicast.codec =PCMU
#enables or disables the phone to handle the incoming multicast paging calls when
there is a multicast paging call on the phone.0-Disabled,1-Enabled (default)
multicast.receive_priority.enable =1
#configures the priority of multicast paging calls.1 is the highest priority, 10
is the lowest priority.The default value is 10.
multicast.receive_priority.prority =10
#It configures the listening multicast IP address and port number for the
phone.The default value is blank.IP address:port
multicast.listen_address.1.ip_address= 224.5.6.20:10008
#configures the label displayed on the LCD screen when receiving the multicast
paging. The default value is blank.String within 99 characters.
```

```
multicast.listen_address.1.label =Announcement1
```

```
#####  
#####
```

```
##                      Miscellaneous                      ##
```

```
#####  
#####
```

```
#enables or disables recent call in dialing feature. If it is enabled, you can  
see the placed calls list when the phone is in the pre-dialing screen.0-
```

```
Disabled(default),1-Enabled
```

```
super_search.recent_call =0
```

```
#####  
#####
```

```
##                      Security                      ##
```

```
#####  
#####
```

```
#configures the password of the user or administrator.The valid value format is  
username:new password.The default value is blank.String within 32 characters.
```

```
#The following means setting the password of administrator (current user name is  
"admin") to password123.
```

```
security.user_password =admin:password123
```

```
security.user_password =user:password123
```

```
#enables or disables the phone to only accept the certificates in the Trusted  
Certificates list.It takes effect after a reboot.
```

```
security.trust_certificates =0
```

```
#enables or disables the phone to mandatorily validate the CommonName or  
SubjectAltName of the certificate received from the connecting server.0-
```

```
Disabled(default), 1-Enabled It takes effect after a reboot.
```

```
security.cn_validation =0
```

```
#configures the source certificates for the phone to authenticate for TLS  
connection.
```

```
#0-Default certificates,1-Custom certificates,2-All certificates(Default).It  
takes effect after a reboot.
```

```
security.ca_cert =2
```

```
#configures the access URL of the custom trusted certificate file.The default
```

value is blank.URL within 511 characters.

```
trusted_certificates.url =http://192.168.153.27:8081/matrixca.crt
```

#deletes all uploaded trusted certificate files.Here the parameter is intentionally kept commented with "#" below.URL within 511 characters.

```
#trusted_certificates.delete =http://localhost/all
```

#configures the device certificates for the phone to send for TLS authentication.0-Default certificates,1-Custom certificates.The default value is 0.

```
security.dev_cert =0
```

#configures the access URL of the custom server certificate file.The default value is blank.URL within 511 characters.

```
server_certificates.url =http://192.168.153.27:8081/matrixservercertificate.pem
```

#deletes the uploaded server certificate file.Here the parameter is intentionally kept commented with "#" below.URL within 511 characters.

```
#server_certificates.delete =http://localhost/all
```

Appendix F - 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

Declaration of Conformity

Matrix SPARSH VP110 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 VP110 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?BaseInfoCatId=293&NewsCatId=293&CatId=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 warrants 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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