Yealink

SIP-T3xG IP Phone Family Administrator Guide

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This device is compliant with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

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2. This device must accept any interferences 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.

About This Guide

The SIP-T3xG IP Phone Family Administrator Guide is considered to be an administration-level version, which is intended for administrators who need to properly configure, customize, manage, and troubleshoot the IP phone systems rather than the end-users of the IP phones. It provides details on the functionality and configuration of the IP phones.

Many of the features are described in this guide involving the network settings, which could affect the IP phone's performance in the network. So an understanding of IP networking and prior knowledge of IP telephony concepts are necessary.

Documentations

This guide covers the SIP-T38G and T32G IP phones. The following related documents for the SIP-T3xG IP phones are available:

- Quick Installation Guides, which describe how to assemble the IP phones.
- Quick Reference Guides, which describe the most basic features available on the IP phones.
- User Guides, which describe the basic and advanced features available on the IP phones.
- Yealink Auto Provisioning User Guide, which describes how to auto provision the IP phones using the configuration files.
- Yealink Configuration Conversion Tool User Guide, which describes how to encrypt the configuration files using the Configuration Conversion Tool.
- <y0000000000xx>.cfg and <MAC>.cfg template configuration files.
- Yealink IP Phones Deployment Guide for BroadWorks Environments, which describes how to configure the BroadSoft features on the BroadWorks web portal and the IP phones.

For support or service, please contact your Yealink reseller or go to Yealink Technical Support at http://www.yealink.com/Support.aspx.

In This Guide

The information detailed in this guide is applicable to the firmware version 70 or higher. The firmware format likes x.x.x.x.rom. The second x should be greater than or equal to 70. This administrator guide includes the following chapters:

- Chapter 1, "Product Overview" describes the SIP components and SIP IP phones.

- Chapter 2, "Getting Started" describes how to install and connect the IP phones and the IP phone interface methods.
- Chapter 3, "Configuring Basic Features" describes how to configure the basic features on the IP phones.
- Chapter 4, "Configuring Advanced Features" describes how to configure the advanced features on the IP phones.
- Chapter 5, "Configuring Audio Features" describes how to configure the audio features on the IP phones.
- Chapter 6, "Configuring Security Features" describes how to configure the security features on the IP phones.
- Chapter 7, "Upgrading the Firmware" describes how to upgrade the firmware of the IP phones.
- Chapter 8, "Resource Files" describes the resource files that can be downloaded by the IP phones.
- Chapter 9, "Troubleshooting" describes how to troubleshoot the IP phones and provides some common troubleshooting solutions.
- Chapter 10, "Appendix" provides the glossary, reference information about the IP phones compliant with RFC 3261, SIP call flows and the sample configuration files.

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Product Overview

This chapter contains the following information about the SIP-T3xG IP phones:

- VoIP Principle
- SIP Components
- SIP IP Phone Models

VoIP Principle

VoIP

VoIP (Voice over Internet Protocol) is a technology using the Internet Protocol instead of traditional Public Switch Telephone Network (PSTN) technology for voice communications.

It is a family of technologies, methodologies, communication protocols, and transmission techniques for the delivery of voice communications and multimedia sessions over IP networks. The H.323 and Session Initiation Protocol (SIP) are two popular VoIP protocols that are found in widespread implement.

H.323

H.323 is a recommendation from the ITU Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences.

It is widely implemented by voice and video conference equipment manufacturers, is used within various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over IP networks.

SIP

SIP (Session Initiation Protocol) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. It is an ASCII-based, application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints. Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP provides the capabilities to:

- Determine the location of the target endpoint -- SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target endpoint -- Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between the endpoints. Conferences are established using only the media capabilities that can be supported by all endpoints.
- Determine the availability of the target endpoint -- A call cannot be completed because the target endpoint is unavailable, SIP determines whether the called party is already on the IP phone or did not answer in the allotted number of rings. It then returns a message indicating why the target endpoint was unavailable.
- Establish a session between the origin and target endpoint -- The call can be completed, SIP establishes a session between the endpoints. SIP also supports mid-call changes, such as the addition of another endpoint to the conference or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls -- SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP simply establishes a session between the transferee and a new endpoint (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

SIP Components

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function as one of the following roles:

- User Agent Client (UAC) -- A client application that initiates the SIP request.
- User Agent Server (UAS) -- A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

User Agent Client (UAC)

The UAC is an application that initiates up to six feasible SIP requests to the UAS. The six requests issued by the UAC are: INVITE, ACK, OPTIONS, BYE, CANCEL and REGISTER. When the SIP session is being initiated by the UAC SIP component, the UAC determines the information essential for the request, which is the protocol, the port and the IP address of the UAS to which the request is being sent. This information can be dynamic and this will make it challenging to put through a firewall. For this reason it may be recommended to open the specific application type on the firewall. The UAC is also capable of using the information in the request URI to establish the course of the SIP request to its destination, as the request URI always specifies the host which is essential. The port and protocol are not always specified by the request URI. Thus if the request does not specify a port or protocol, a default port or protocol is contacted. Using this

method may be the preferred measure when not using an application layer firewall, application layer firewalls like to know what applications are flowing though which ports and it is possible using content types of other applications other than the one you are trying to let through which has been denied.

User agent server (UAS)

UAS is the server that hosts the application responsible for receiving the SIP requests from a UAC, and on reception returns a response to the request back to the UAC. The UAS may issue multiple responses to the UAC, not necessarily a single response. Communication between UAC and UAS is client/server and peer-to–peer.

Typically, a SIP endpoint is capable of functioning as both a UAC and a UAS, but it functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiates the request.

SIP IP Phone Models

This section introduces the SIP-T3xG IP phone family. The SIP-T3xG IP phones are endpoints in the overall network topology, which are designed to interoperate with other compatible equipments including application servers, media servers, internet-working gateways, voice bridges, and other endpoints. The SIP-T3xG IP phones are characterized by a large number of functions, which simplify business communication with a high standard of security and can work seamlessly with a large number of SIP PBXs.

The SIP-T3xG IP phones provide a powerful and flexible IP communication solution for Ethernet TCP/IP networks, delivering excellent voice quality. The high-resolution graphic display supplies content in multiple languages for system status, call history and directory access. The SIP-T3xG IP phones also support advanced functionalities, including LDAP, Busy Lamp Field, Shared Call Appearance and Network Conference.

The following IP phone models are described:

- SIP-T38G
- SIP-T32G

The SIP-T3xG IP phones comply with the SIP standard (RFC 3261), and they can only be used within a network that supports this type of phone.

For successfully operating as SIP endpoints in your network, the SIP-T3xG IP phones must meet the following requirements:

- A working IP network is established.
- Routers are configured for VoIP.
- VoIP gateways are configured for SIP.
- The latest (or compatible) firmware of the SIP-T3xG IP phones is available.

- A call server is active and configured to receive and send SIP messages.

Physical Features of the SIP-T3xG IP Phones

This section lists the available physical features of the SIP-T3xG IP phones.

SIP-T38G

Physical Features:

- TI Aries chipset and TI voice engine
- 4.3" TFT-LCD, 480 x 272 pixel, 16.7M colors
- 6 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 48 keys including 16 programmable keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100/1000Mbps Ethernet ports
- 1XRJ12 (6P6C) expansion module port
- 19 LEDs: 1xpower, 6xline, 1xmessage, 1xheadset, 10xmemory
- Power adapter: AC 100~240V input and DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)

SIP-T32G

Physical Features:

- TI Aries chipset and TI voice engine
- 3" TFT-LCD, 400 x 240 pixel, 262K colors
- 3 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 32 keys including 3 programmable keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100/1000Mbps Ethernet ports
- 5 LEDs: 1xpower, 3xline, 1xmessage
- Power adapter: AC 100~240V input and DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)

Key Features of the SIP-T3xG IP Phones

In addition to the physical features introduced above, the SIP-T3xG IP phones also support the following key features when running the latest firmware:

- **Phone Features**
  - **Call Options**: emergency call, call waiting, call hold, call mute, call forward, call transfer, call pickup, 3-way conference.
  - **Basic Features:** DND, phone lock, auto redial, live dialpad, dial plan, hotline, caller identity, auto answer.

- **Advanced Features:** BLF/BLF list, shared call appearance, distinctive ring tones, remote phonebook, LDAP, 802.1x authentication.

- **Codecs and Voice Features**
  - Wideband codec: G.722
  - Narrowband codec: G.711, G.723.1, G.726, G.729AB
  - VAD, CNG, AEC, PLC, AJB, AGC
  - Full-duplex speakerphone with AEC

- **Network Features**
  - SIP v1 (RFC2543), v2 (RFC3261)
  - NAT Traversal: STUN mode
  - DTMF: INBAND, RFC2833, SIP INFO
  - Proxy mode and peer-to-peer SIP link mode
  - IP assignment: Static/DHCP/PPPoE
  - Bridge/Router mode for PC port
  - TFTP/DHCP/PPPoE client
  - HTTP/HTTPS server
  - DNS client
  - NAT/DHCP server

- **Management**
  - FTP/TFTP/HTTP/PnP auto-provision
  - Configuration: browser/phone/auto-provision
  - Direct IP call without SIP proxy
  - Dial number via SIP server
  - Dial URL via SIP server

- **Security**
  - HTTPS (server/client)
  - SRTP (RFC3711)
  - Transport Layer Security (TLS)
  - VLAN (802.1q), QoS
  - Digest authentication using MD5/MD5-sess
  - Secure configuration file via AES encryption
  - Phone lock for personal privacy protection
  - Admin/User configuration mode

Getting Started

This chapter introduces the initialization of the SIP-T3xG IP phones, the installing and connecting process of the IP phones which you need to follow.

This chapter provides the following major sections:

- Connecting the IP Phones
- Initialization Process Overview
- Verifying Startup
- Configuration Interfaces
- Reading Icons
- Configuring Basic Network Parameters
- Creating Dial Plan

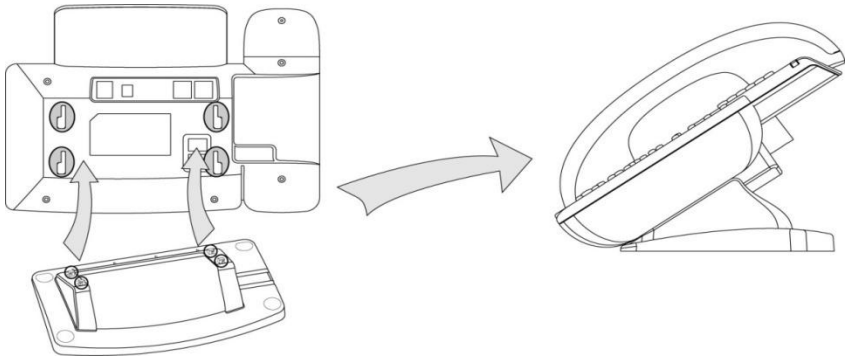
Connecting the IP Phones

This section introduces how to install SIP-T3xG IP phones with the components in the packing list.

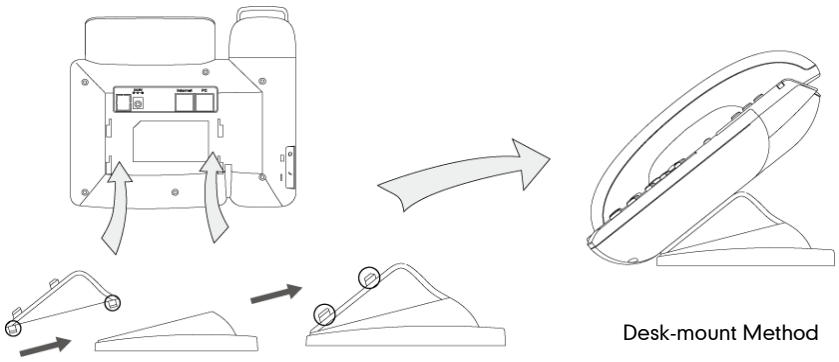
1. Attach the stand
2. Connect the handset and optional headset
3. Connect the network and power

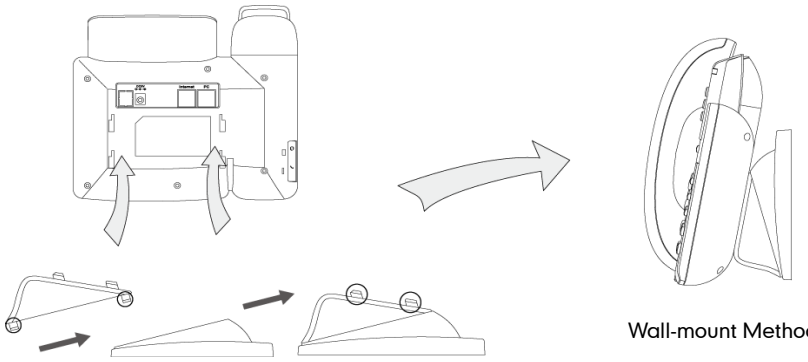
Note A headset is not provided in the packing list.

1) **Attach the stand:**



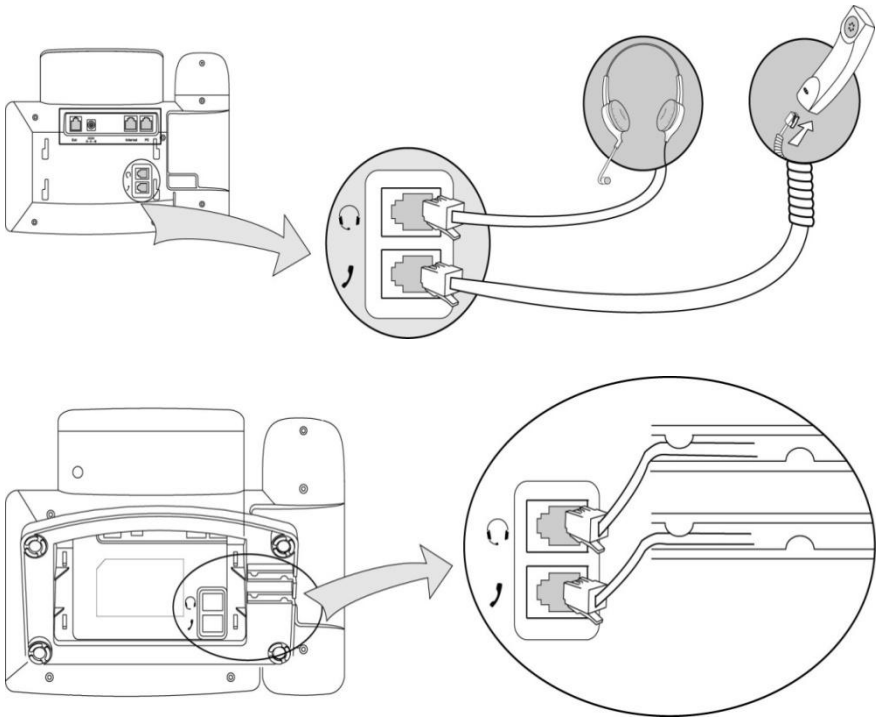
SIP-T38G



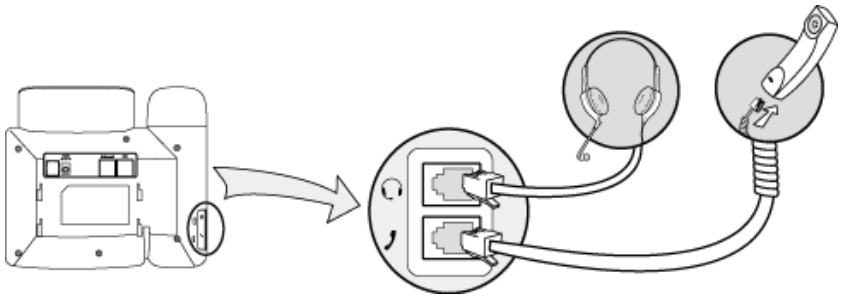


SIP-T32G

2) Connect the handset and optional headset:



SIP-T38G



SIP-T32G

3) Connect the network and power:

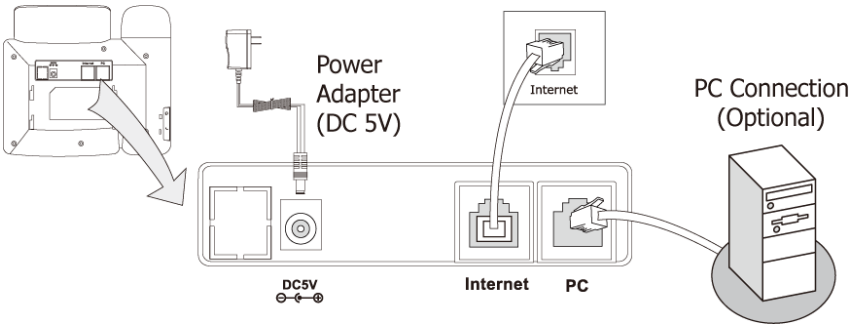
- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power and network:

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

2. Connect the supplied Ethernet cable between the Internet port on the IP phone and the Internet port in your network or switch/hub device port.

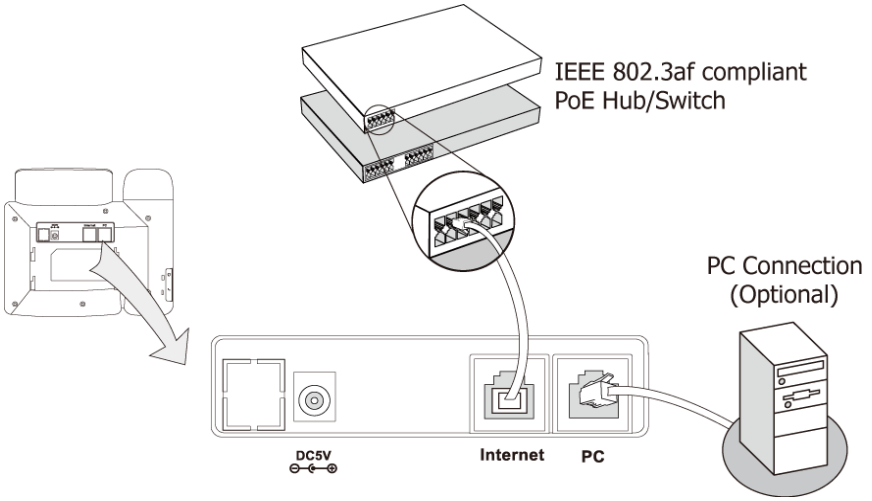


Power over Ethernet

Using a regular Ethernet cable, the IP phones can be powered from a PoE (IEEE 802.3af) compliant switch or hub.

To connect the PoE:

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



Note

If in-line power is provided, you do not need to connect the AC adapter. Make sure the Ethernet cable and switch/hub is PoE compliant.

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

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

Initialization Process Overview

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:

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 preloaded. 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). 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 following network parameters can be obtained from the DHCP server during initialization:

- IP Address
- Subnet Mask
- Gateway
- Primary DNS (Domain Name Server)
- Secondary DNS

You need to configure the 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 to Configuring Network Parameters Manually on page 19.

Contacting the TFTP server

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

Updating the firmware

If the access URL of the firmware has been defined in the configuration file, the IP phone will download the 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 performs 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 employed, these files are required.

The followings are examples of resource files:

- Language packs
- Ring tones
- Directories

Verifying Startup

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

1. The power indicator LED illuminates.
2. The message "Initializing, Please wait" appears as the IP phone starts up.
3. The main LCD screen displays the following:
   - Time and date
   - Soft key labels
4. Press the OK key to verify 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.

Configuration Interfaces

You can use the following methods to setup and configure the IP phones:

- Phone User Interface
- Web User Interface
- Configuration Files

The following sections describe how to configure the IP phones using each method above.

Phone User Interface

The phone user interface provides an easy way to configure and use the IP phones. Accessing specific features is restricted to the administrator. These specific features are password protected by default. The default password is "admin"(case-sensitive). Not all features are available for configuring via phone user interface.

Web User Interface

An administrator can configure the IP phones via web user interface. The default administrator's name and password for logging in the web user interface are both "admin" (case-sensitive). Almost all features are available for configuring via web user interface. The IP phones support both HTTP and HTTPS protocols for accessing the web user interface. For more information, refer to Web Server Type on page 109.

Configuration Files

You can configure the IP phones using the configuration files. There are two configuration files both of which are CFG formatted. We call them Common CFG file and MAC-Oriented CFG file. A Common CFG file will be effectual for all IP phones of the same model. However, a MAC-Oriented CFG file will only be effectual for a specific IP phone. The Common CFG file has a fixed name for each IP phone model, while the MAC-Oriented CFG file is named after the MAC address of the IP phone. For example, if the MAC address of the SIP-T38G IP phone is 001565113af8, the names of these two configuration files must be: y000000000038.cfg and 001565113af8.cfg.

The name of the Common CFG file for each SIP-T3xG IP phone model is:

- SIP-T38G: y000000000038.cfg
- SIP-T32G: y000000000032.cfg

In order to configure the IP phones using the configuration files (<y0000000000xx>.cfg and <MAC>.cfg), you need to use a text-based editing application to edit the configuration files, and store the configuration files to the root directory of a provisioning server. The IP phones support downloading the configuration files using any of the following protocols: FTP, TFTP, HTTP and HTTPS.

The IP phones can get the address of the provisioning server during startup through one of the following processes: Zero Touch, PnP, DHCP Option and Phone Flash. Then the IP phones download the configuration files from the provisioning server, resolve and apply the configurations written in the configuration files. This entire process is called auto provisioning. For more information on auto provisioning, refer to the document "Yealink Auto Provisioning User Guide".

When modifying parameters, remember the following:

- Parameters in the configuration files override those stored in the IP phone's flash memory.
- The .cfg extension of the configuration files must be in lowercase.
- 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 variable name and value with 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 separated line. Use the pound (#) delimiter to distinguish the comments.

The IP phones can accept two sources of configuration data:

- Downloaded from the configuration files
- Changed on the phone user interface or the web user interface

The latest values applied to the IP phones are the values that take effect.

Reading Icons

When you use or configure different features on the IP phones, a variety of icons may appear on the LCD screen. The following table lists and describes icons that you might see while using different IP phone models.

T38G/T32G	Description
	Network unavailable
	Registered successfully
	Registration failed
	Registering
	Hands-free speakerphone mode
	Handset mode

T38G/T32G	Description
	Headset mode
	Multi-lingual lowercase letters input mode
	Multi-lingual uppercase letters input mode
	Alphanumeric input mode
	Numeric input mode
	Voice Mail
	Text Message
	Auto Answer
	Do Not Disturb
	Call Forward
	Call Hold
	Call Mute
	Ringer volume is 0
	Phone Lock
	Missed Calls
	Received Calls
	Dialed Calls
	Missed Calls

T38G/T32G	Description
	Recording box is full
	A call cannot be recorded
	Recording starts successfully
	Recording cannot be started
	Recording cannot be stopped
	Open VPN
	Conference
	The default contact photo

Configuring Basic Network Parameters

This section describes how to configure the basic network parameters that are required for the IP phones to operate in the network.

DHCP

DHCP (Dynamic Host Configuration Protocol) is a network protocol that is used to dynamically allocate network parameters to hosts connected to a network. The automatic distribution of network parameters to hosts eases the administrative burden of maintaining IP networks. The IP phones comply with the DHCP specifications documented in RFC 2131. If using DHCP, the IP phones connected to the network become operational without having to be manually assigned IP addresses and additional network parameters. By default, DHCP is enabled on the IP phones.

DHCP Option

DHCP provides a framework for passing network information to devices on a TCP/IP network. 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.

When the IP phones are simply plugged into the network, the DHCP process begins. The IP phones broadcast DISCOVER messages to request the network information carried in DHCP options and the DHCP server responds with the specific values in the corresponding options.

The following table lists the common DHCP options supported by the 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 address 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 hostnames 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 the 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.
Bootfile Name	67	Identify a bootfile when the 'file' field in the DHCP header has been used for DHCP options.

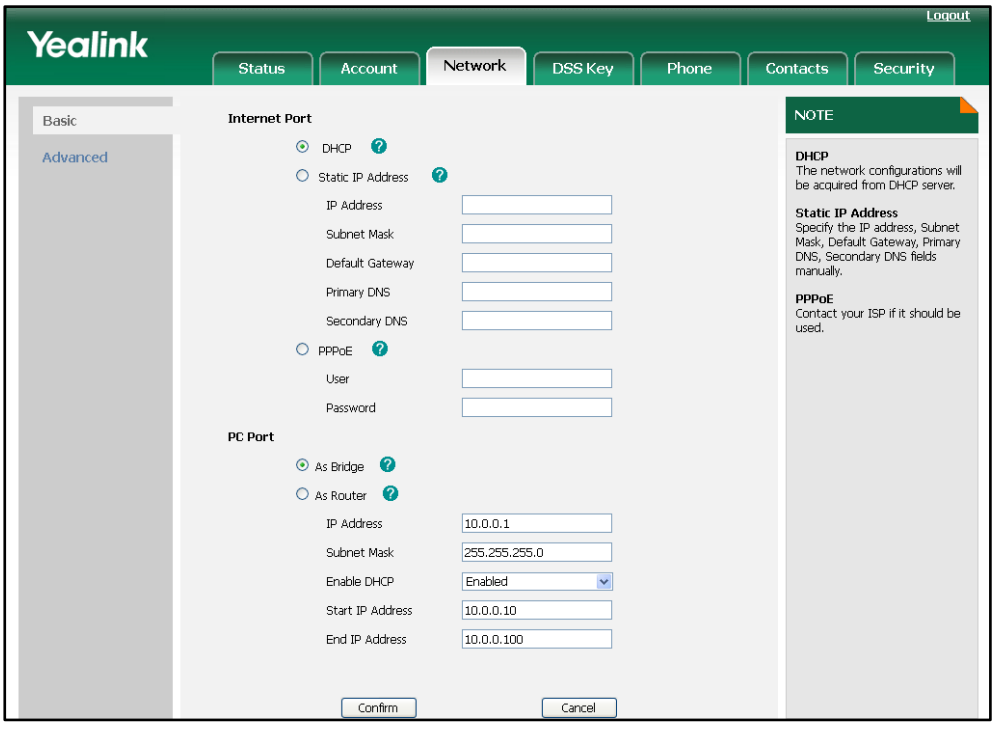
Procedure

DHCP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure DHCP on the IP phone. For more information, refer to DHCP on page 224.
Local	Web User Interface	Configure DHCP on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Basic.htm
	Phone User Interface	Configure DHCP on the IP phone.

To configure DHCP via web user interface:

1. Click on **Network->Basic**.
2. In the **Internet Port** field, mark the **DHCP** radio box.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.
4. Click **OK** to reboot the IP phone.

To configure DHCP via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**WAN Port**.
2. Press ▲ or ▼ to highlight the **DHCP IP Client** field, and then press the **Enter** soft key.

   The IP phone reboots automatically to make the settings effective after a period of time.

Configuring Network Parameters Manually

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

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

Procedure

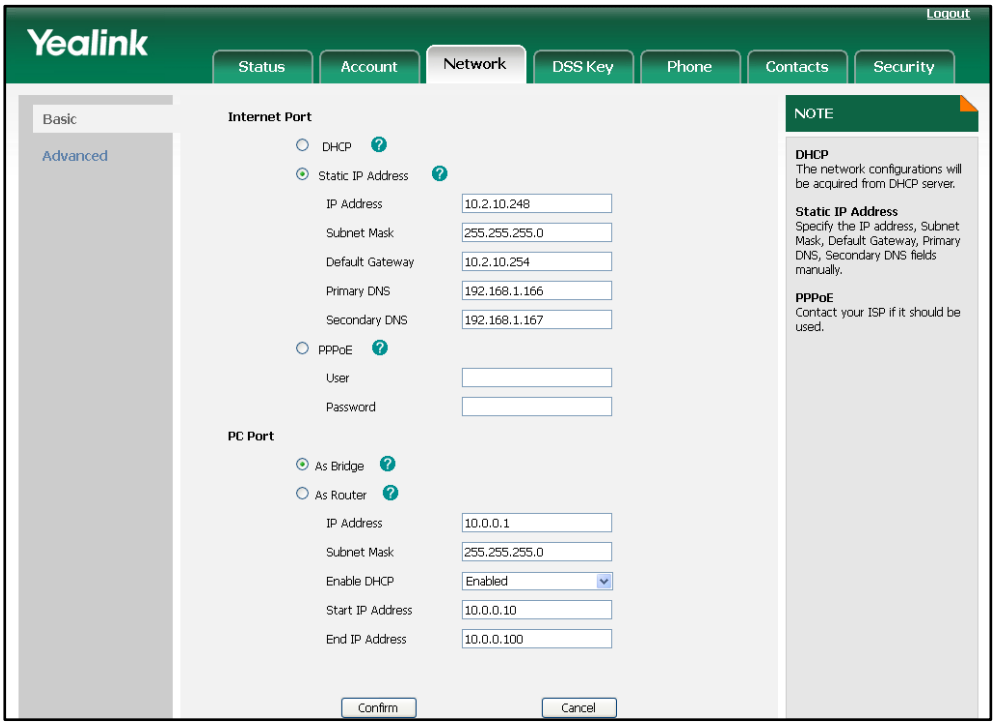
Network parameters can be configured manually using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure network parameters of the IP phone manually. For more information, refer to Static Network Settings on page 225.
Local	Web User Interface	Configure network parameters of the IP phone manually. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Basic.htm
	Phone User Interface	Configure network parameters of the IP phone manually.

To configure network parameters manually via web user interface:

1. Click on **Network**->**Basic**.
2. In the **Internet Port** field, mark the **Static IP Address** radio box.

3. Enter the IP address, subnet mask, default gateway, primary DNS and secondary DNS in the corresponding fields.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

To configure network parameters manually via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**WAN Port**.
2. Press ▲ or ▼ to highlight the **Static IP Client** field, and then press the **Enter** soft key.
3. Enter the IP address, subnet mask, default gateway, primary DNS and secondary DNS in the corresponding fields.
4. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Note 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 about these parameters, contact your network administrator.

PPPoE

PPPoE (Point-to-Point Protocol over Ethernet) is a network protocol that is used by Internet Service Providers (ISPs) to provision 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. The Internet port on the IP phone can be configured as a PPPoE port to connect to the Internet. Contact your ISP for the PPPoE username and password.

Procedure

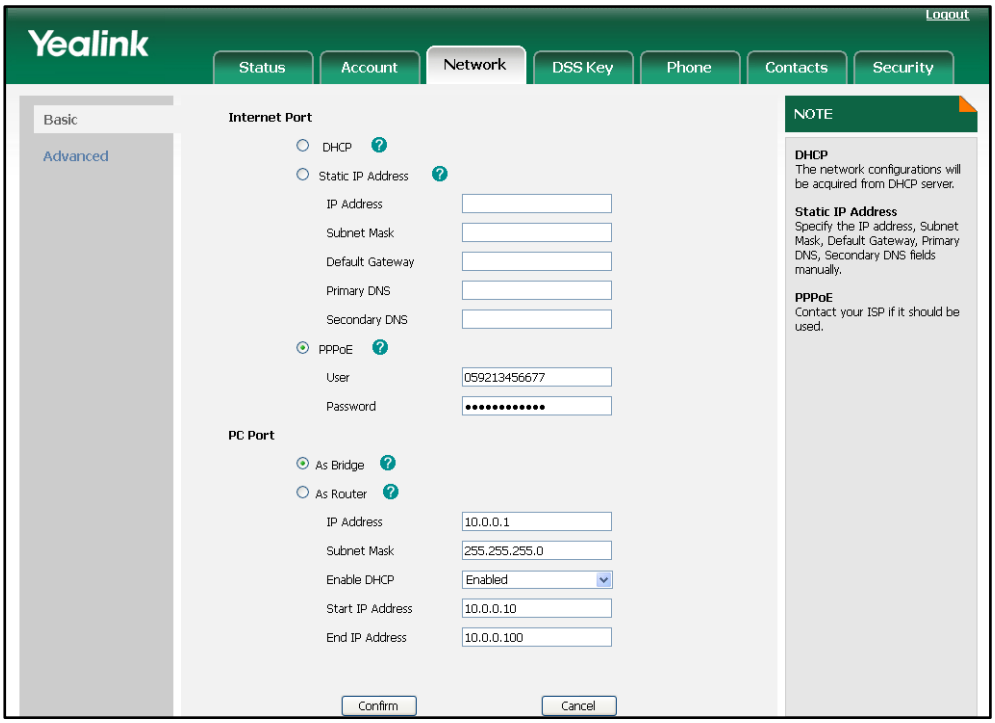
PPPoE can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure PPPoE on the IP phone. For more information, refer to PPPoE on page 227.
Local	Web User Interface	Configure PPPoE on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Basic.htm
	Phone User Interface	Configure PPPoE on the IP phone.

To configure PPPoE via web user interface:

1. Click on **Network**->**Basic**.
2. In the **Internet Port** field, mark the **PPPoE** radio box.

3. Enter the username and password in the corresponding fields.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

To configure PPPoE via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**WAN Port**.
2. Press ▲ or ▼ to highlight the **PPPoE IP Client** field, and then press the **Enter** soft key.
3. Enter the username and password in the corresponding fields.
4. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Configuring PC Port Mode

The PC port is used to connect a PC behind the IP phone, which can be configured as the following two modes:

- **Bridge**: In the bridge mode, the IP phone is considered as a bridge, the PC attached to the PC port appears on the network as a stand-alone device with its own IP address.
- **Router**: In the router mode, the IP phone is considered as a router, and provides a DHCP service to the PC attached to the PC port.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the PC port mode. For more information, refer to PC Port Mode on page 228.
Local	Web User Interface	Configure the PC port mode. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Basic.htm
	Phone User Interface	Configure the PC port mode.

To configure the PC port mode via web user interface:

1. Click on **Network**->**Basic**.
2. In the **PC Port** field, mark the desired radio box.

   If you select **As Router**, you can configure the IP address for the PC port and configure DHCP for the PC attached to the PC port.

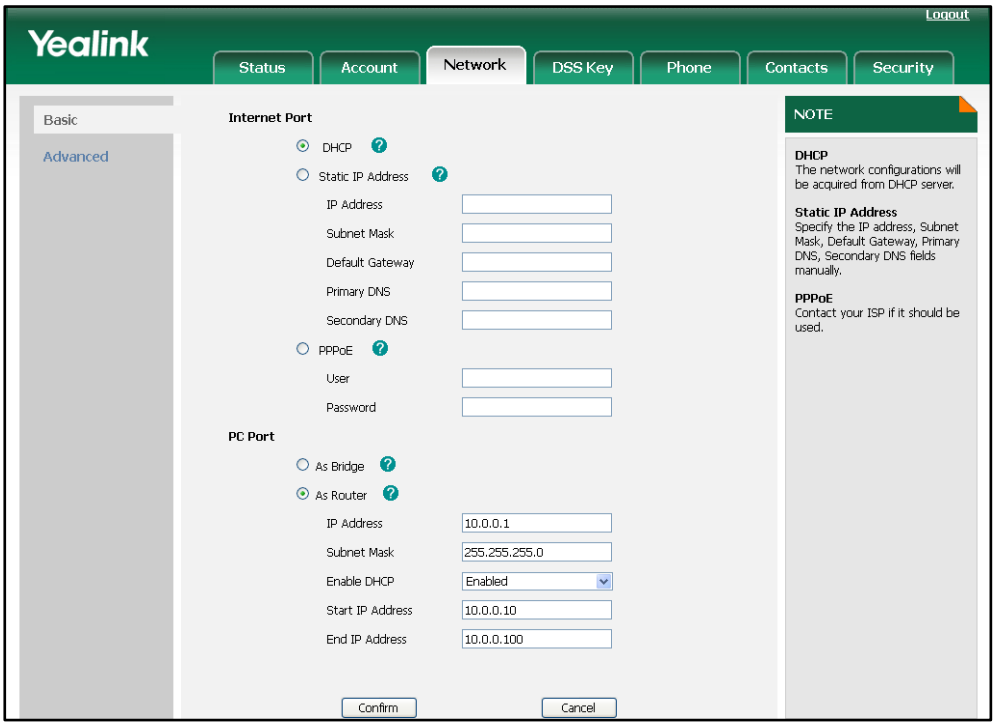
   1) Enter the IP address in the **IP Address** field.

   2) Enter subnet mask in the **Subnet Mask** field.

   3) Select the desired value from the pull-down list of **Enable DHCP**.

   4) (If enabled) Enter the start IP address in the **Start IP Address** field.

5) (If enabled) Enter the end IP address in the **End IP Address** field.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

To configure the PC port mode via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**PC Port**.
2. Select the desired mode.

   If you select **Router**, you can configure the IP address for the PC port and configure DHCP for the PC attached to the PC port.

   1) Enter the IP address in the **IP** field.
   2) Enter the subnet mask in the **Subnet Mask** field.
   3) Press (▲) or (▼) to highlight the **DHCP Server** field, and then press the **Enter** soft key to enter the DHCP Server screen.
   4) Select the desired value from the **Server Status** field.
   5) (If enabled) Enter the start IP address in the **Start IP** field.
   6) (If enabled) Enter the end IP address in the **End IP** field.

3. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Creating 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 dial plan for the IP phones. Dial plan is a string of characters that governs the way for the IP phones processing the inputs received from the IP phone keypads. The IP phones support the following dial plan features:

- Replace Rule
- Dial-now
- Area Code
- Block Out

The priority of matching dial plan is: Dial Now>Replace Rule>Area Code>Block Out.

You need to know the following basic regular expression syntax when creating dial plan:

.	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

Replace rule is an alternative string that replaces the numbers entered by the user. You can create up to 100 replace rules for the IP phone. The replace rules can be created either one by one or in batch using a replace rule template. For more information on the replace rule template, refer to Replace Rule Template on page 199.

Procedure

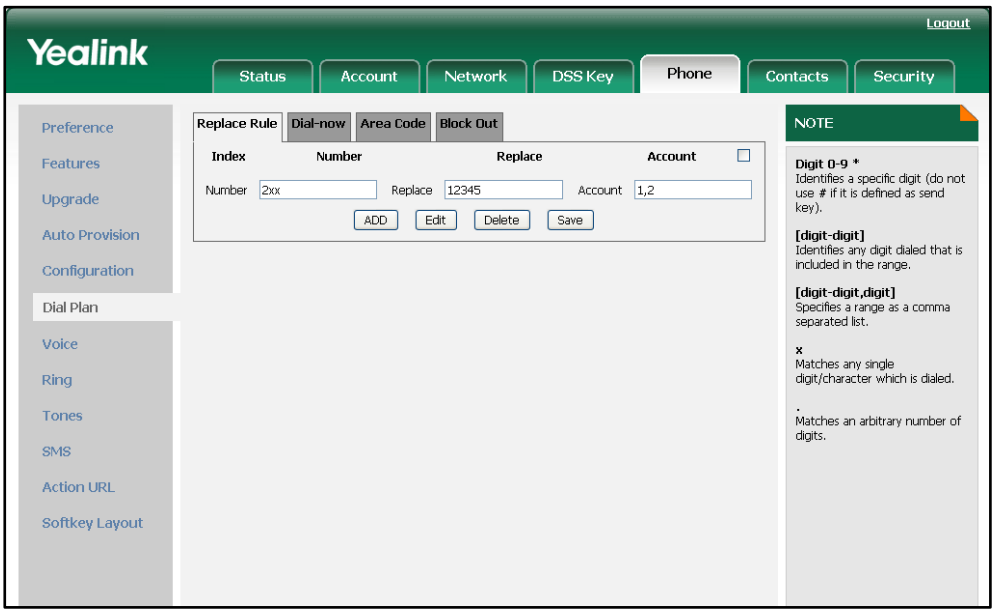
Replace rule can be created using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Create the replace rule for the IP phone. For more information, refer to Dial Plan on page 231.
Local	Web User Interface	Create the replace rule for the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Dialplan.htm

To create the replace rule via web user interface:

1. Click on **Phone**->**Dial Plan**->**Replace Rule**.
2. Enter the string in the **Number** field.
3. Enter the string in the **Replace** field.
4. Enter the desired line ID in the **Account** field or leave it blank.

If you leave the field blank, the replace rule applies to all accounts on the IP phone.



5. Click **Add** to add the replace rule.
6. Click **Save** to accept the change.

Dial-now

Dial-now is a string that is 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. You can create up to 20 dial-now rules for the IP phone. The dial-now rules can be created either one by one or in batch using a dial-now rule template. For more information on the dial-now template, refer to Dial-now Template on page 200.

Delay Time for Dial-now Rule

The IP phone will automatically dial out the entered number, which matches the dial-now rule, after the configurable delay time.

Procedure

Dial-now rule can be created using the configuration files or locally.

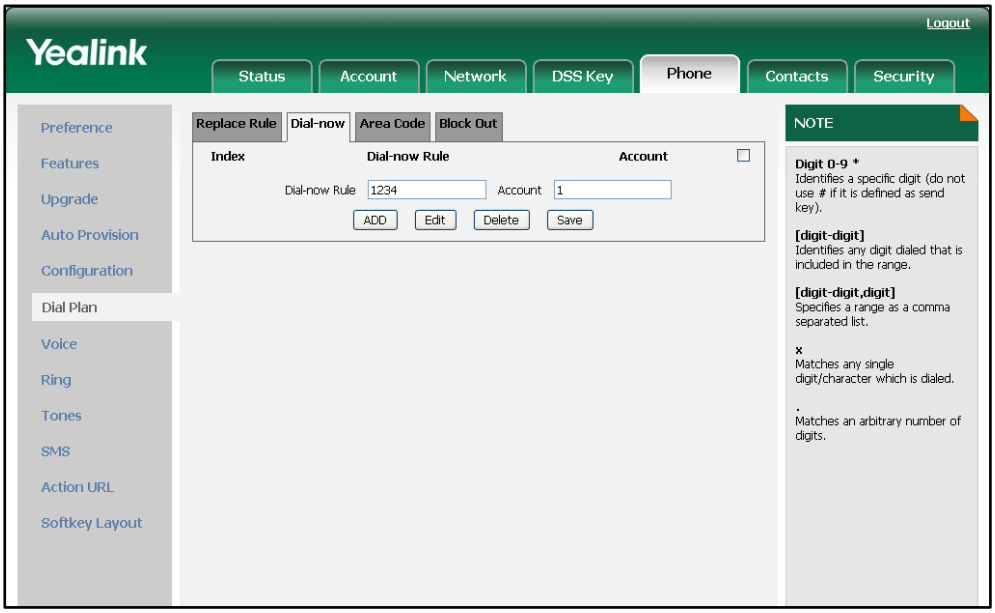
Configuration File	<y0000000000xx>.cfg	Create the dial-now rule for the IP phone. For more information, refer to Dial Plan on page 231. Configure the delay time for the dial-now rule. For more information, refer to Dial

		Plan on page 231.
Local	Web User Interface	Create the dial-now rule for the IP phone. **Navigate to:** http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-DialNow.htm Configure the delay time for the dial-now rule. **Navigate to:** http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To create the dial-now rule via web user interface:

1. Click on **Phone->Dial Plan->Dial-now**.
2. Enter the desired value in the **Dial-now Rule** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

   If you leave the field blank, the dial-now rule applies to all accounts on the IP phone.

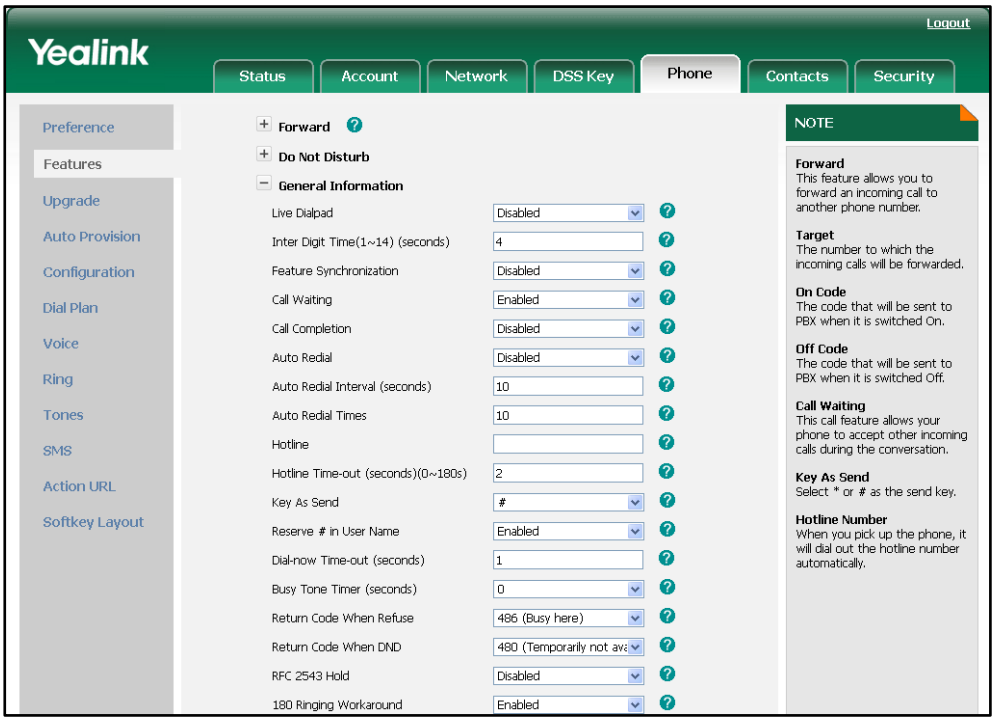


4. Click **Add** to add the dial-now rule.

To configure the delay time for the dial-now rule via web user interface:

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

2. Enter the desired time in the **Dial-now Time-out (seconds)** field.



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

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in one country. When entered numbers match the predefined area code rule, the IP phone will automatically add the area code to the beginning of the dialed numbers. The IP phones only support one area code rule.

Procedure

Area code rule can be configured using the configuration files or locally.

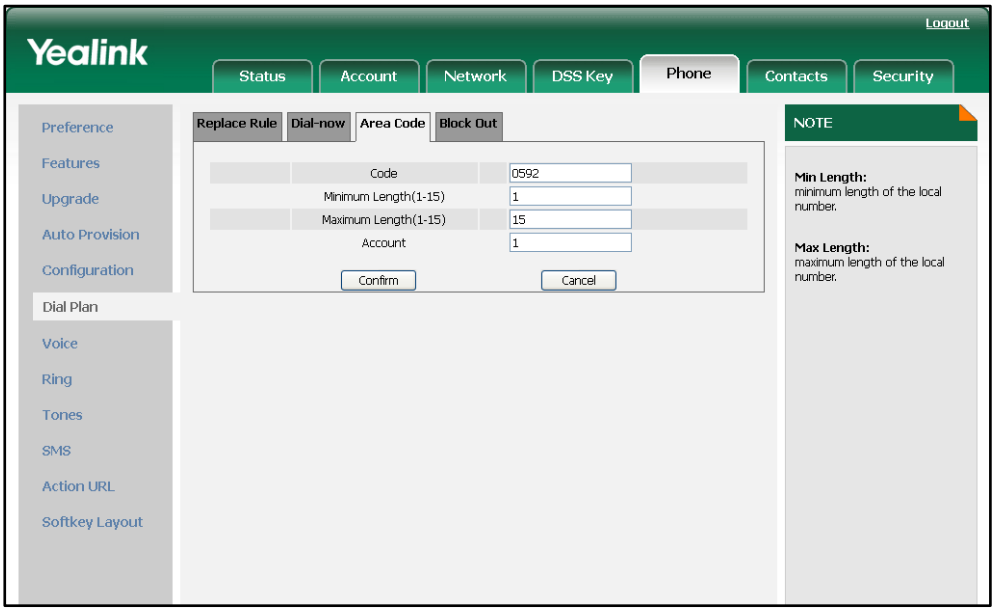
Configuration File	<y0000000000xx>.cfg	Create the area code rule and specify the maximum and minimum lengths of the entered numbers. For more information, refer to Dial Plan on page 231.
Local	Web User Interface	Create the area code rule and specify the maximum and minimum lengths of the entered numbers. **Navigate to**:

		http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-AreaCode.htm

To configure an area code rule via web user interface:

1. Click on **Phone**->**Dial Plan**->**Area Code**.
2. Enter the desired values in the **Code**, **Minimum Length (1-15)** and **Maximum Length (1-15)** fields.
3. Enter the desired line ID in the **Account** field or leave it blank.

   If you leave the field blank, the area code rule applies to all accounts on the IP phone.



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

Block Out

Block out rule can prevent users from dialing out some specific numbers. When entered numbers match the predefined block out rule, the phone LCD screen prompts "Forbidden Number". You can create up to 10 block out rules.

Procedure

Block out rule can be created using the configuration files or locally.

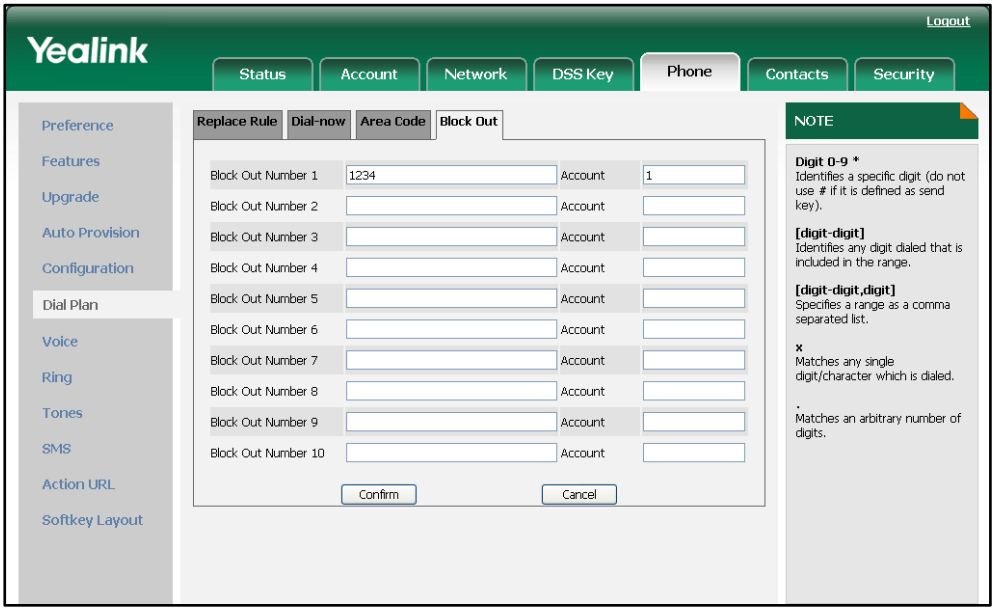
Configuration File	<y0000000000xx>.cfg	Create the block out rule for the IP phone. For more information, refer to Dial Plan on page 231.

Local	Web User Interface	Create the block out rule for the desired line. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-BlockOut.htm

To create the block out rule via web user interface:

1. Click on **Phone**->**Dial Plan**->**Block Out**.
2. Enter the desired value in the **Block Out Number** field.
3. Enter the desired line ID in the **Account** field or leave it blank.

   If you leave the field blank, the block out rule applies to all accounts on the IP phone.



4. Click **Confirm** to add the block out rule.

Configuring Basic Features

This chapter provides information for making configuration changes for the following basic features:

- Wallpaper
- Screensaver
- User Password
- Administrator Password
- Phone Lock
- Time and Date
- Language
- Softkey Layout
- Key as Send
- Hotline
- Call Log
- Missed Call Log
- Local Directory
- Live Dialpad
- Call Waiting
- Auto Redial
- Auto Answer
- Call Completion
- Anonymous Call
- Anonymous Call Rejection
- Do Not Disturb
- Busy Tone
- Return Code When Refuse
- 180 Ring Workaround
- Use Outbound Proxy in Dialog
- SIP Session Timer
- Session Timer
- Call Hold
- Call Forward

- Call Transfer
- Network Conference
- Transfer on Conference Hang Up
- Direct Pickup
- Group Pickup
- Dialog-Info Call Pickup
- Call Return
- Call Park
- Web Server Type
- Calling Line Identification Presentation
- Connected Line Identification Presentation
- DTMF
- Intercom

Wallpaper

Wallpaper is the image that fills the background of the phone idle screen. Some users choose one of the default backgrounds provided by the IP phone system. But some users prefer to make customized wallpaper from personal pictures. For using customized wallpaper, you need to upload the customized wallpaper in advanced.

The following table lists the wallpaper image format and resolution for each phone model:

Phone Model	Wallpaper Image Format	Resolution
SIP-T38G	.jpg/.png/.bmp	**<=480*272**
SIP-T32G	.jpg/.png/.bmp	**<=480*272**

Procedure

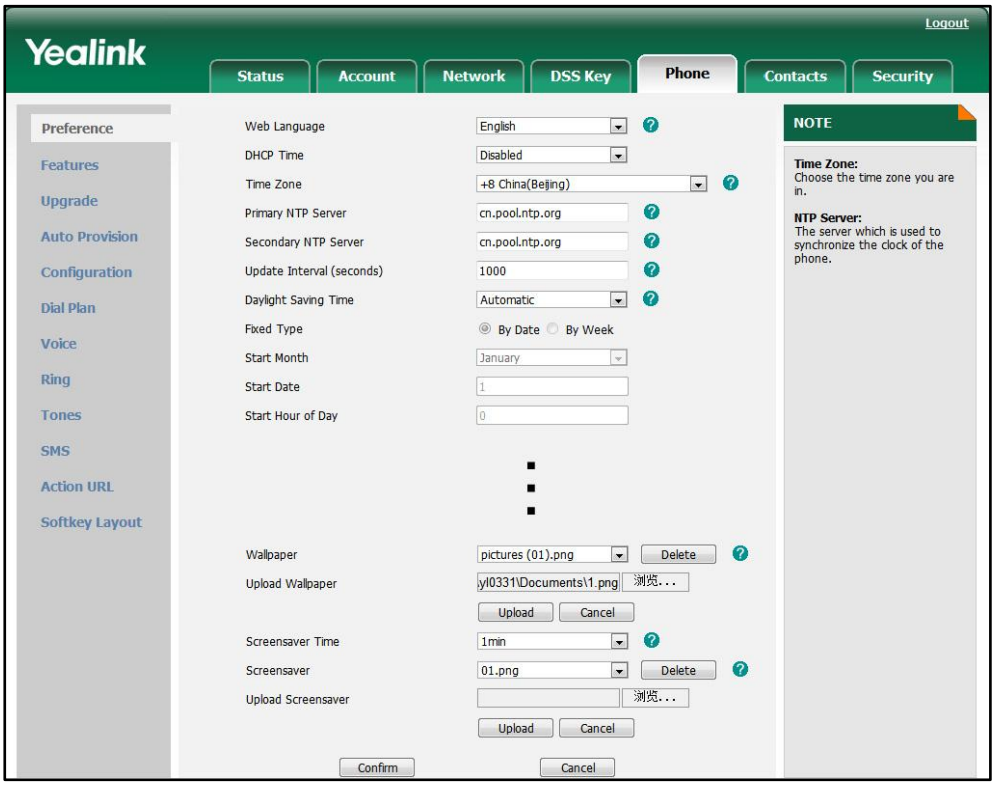
The wallpaper shown on the idle screen can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the customized wallpaper. For more information, refer to Access URL of Wallpaper Image on page 312.
Local	Web User Interface	Upload the customized wallpaper.

		Change the wallpaper shown on the idle screen via web user interface. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Preference.htm
		Change the wallpaper shown on the idle screen via phone user interface.

To upload a customized wallpaper via web user interface:

1. Click on **Phone**->**Preference**.
2. In the **Upload Wallpaper** field, click **Browse** to select the wallpaper image from your local system.
3. Click **Upload** to upload the file.



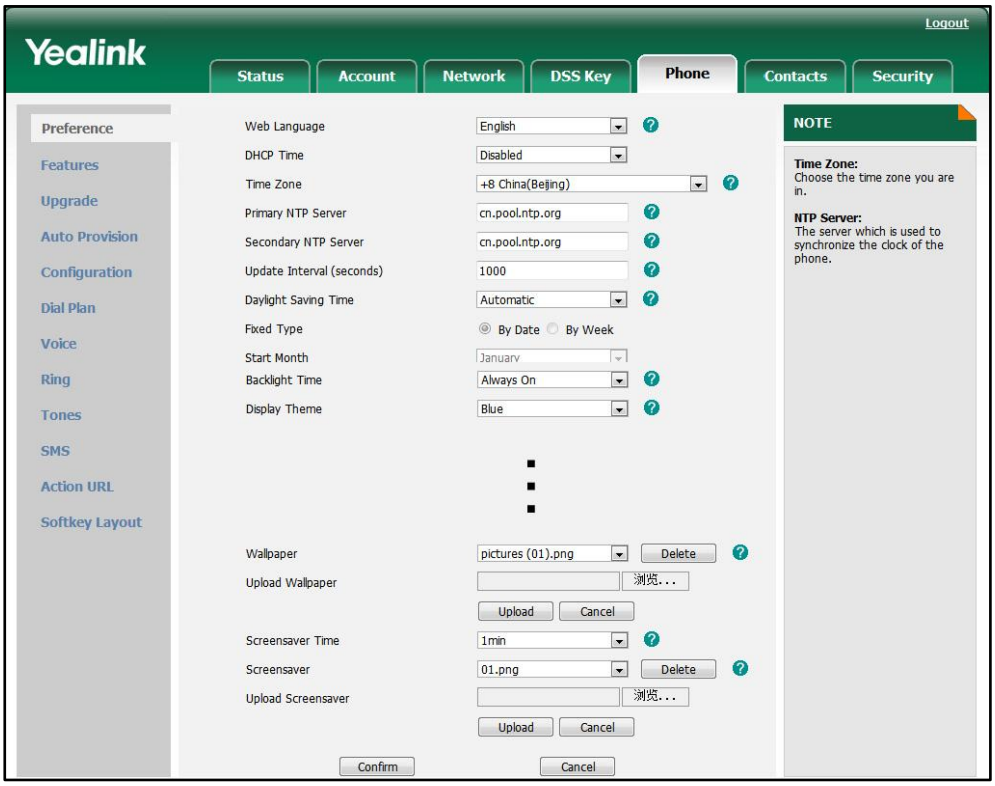
4. Click **Confirm** to accept the change.

   The customized wallpaper appears in the pull-down list of **Wallpaper**.

To change the wallpaper via web user interface:

1. Click on **Phone**->**Preference**.

2. Select the desired wallpaper from the pull-down list of **Wallpaper**.



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

To change the wallpaper via phone user interface:

1. Press **Menu->Display->Wallpaper**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired wallpaper image.
3. Press the **Save** soft key to accept the change

Screensaver

A screen saver is an animated image that is activated on the IP phone display after periods of user inactivity.

The following table lists the screensaver image format and resolution for each phone model:

Phone Model	Screensaver Image Format	Resolution
SIP-T38G	.jpg/.png/.bmp	**<=480*272**
SIP-T32G	.jpg/.png/.bmp	**<=480*272**

Procedure

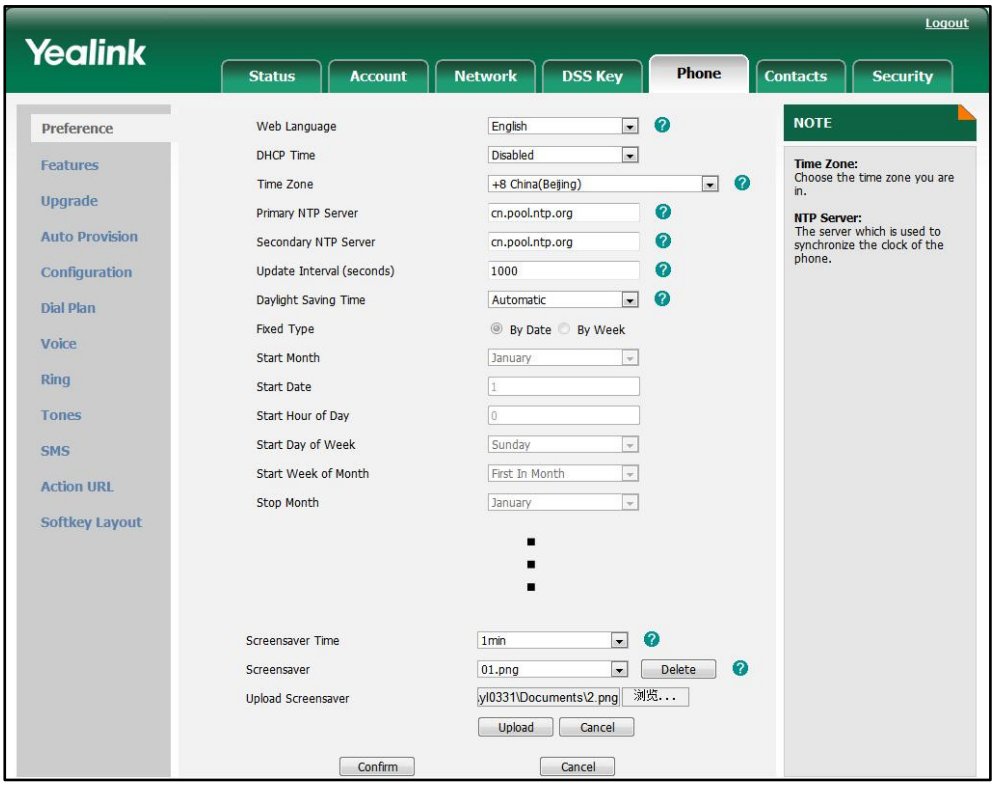
The screensaver can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the customized screensaver. For more information, refer to Access URL of Screensaver Image on page 312.
Local	Web User Interface	Upload the customized screensaver. Configure the screensaver via web user interface. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Preference.htm
		Configure the screensaver via phone user interface.

To upload a customized screensaver via web user interface:

1. Click on **Phone**->**Preference**.
2. In the **Upload Screensaver** field, click **Browse** to select the screensaver image from your local system.

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



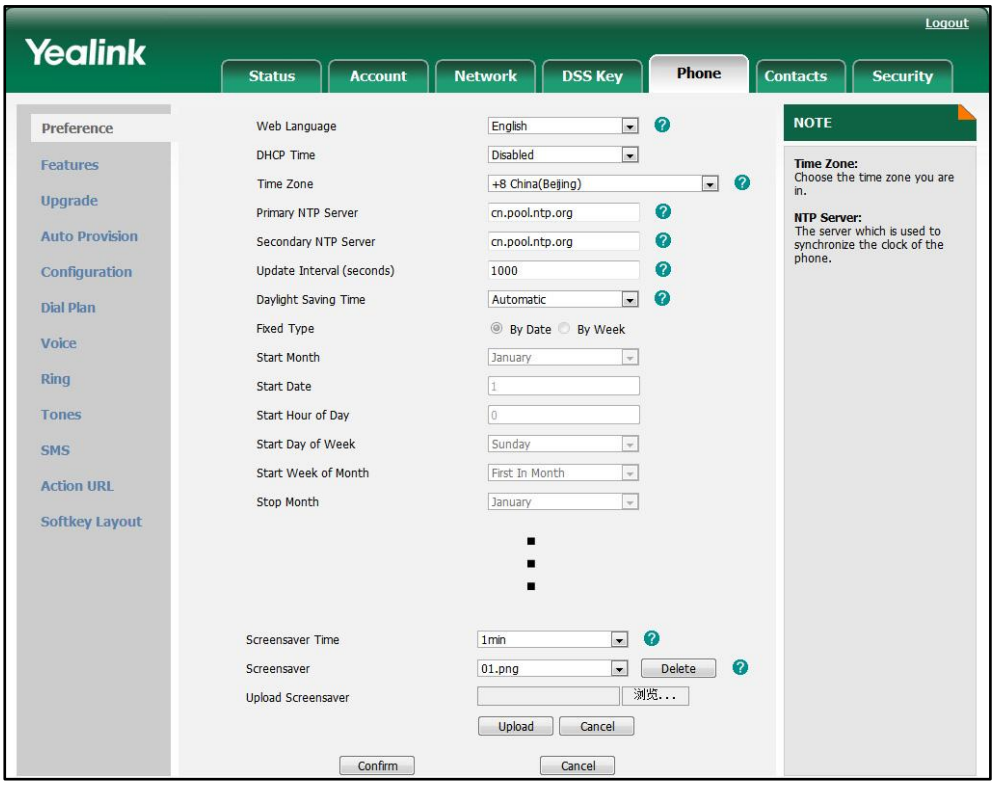
4. Click **Confirm** to accept the change.

   The customized screensaver appears in the pull-down list of **Screensaver**.

To configure the screensaver via web user interface:

1. Click on **Phone**->**Preference**.

2. Select the desired time from the pull-down list of **Screensaver Time**.



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

To configure the screensaver via phone user interface:

1. Press **Menu->Display->Screensaver**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired time in the **Time-out** field.

   After the time you specified in the Time-out field, your phone will display the screensaver.
3. Press ◄ to ► scroll to the **Preview Screensaver Pictures** field, press the **Enter** soft key to preview the screensaver pictures and then press the **Exit** soft key to back to the previous interface.
4. Press the **Save** soft key to accept the change

Backlight

Backlight provides the brightness necessary for making the phone LCD screen readable in darkened environment. Backlight time specifies the delay time to turn off the backlight when the IP phone is inactive. Shorter backlight time is annoying if the backlight is turned off quickly which does not give users enough time to read messages. Backlight level is used to adjust the backlight intensity of the LCD screen. Inactive backlight level defines whether the IP phone completely turns off the backlight of the LCD screen after a period of inactivity.

You can configure the backlight time as one of the following types:

- **Always On:** Backlight is turned on permanently.
- **1min, 2min, 5min, 10min or 30min:** Backlight is turned off when the IP phone is inactive after a preset period of time (in minutes), but it is automatically turned on if the status of the IP phone changes or any key is pressed.

Procedure

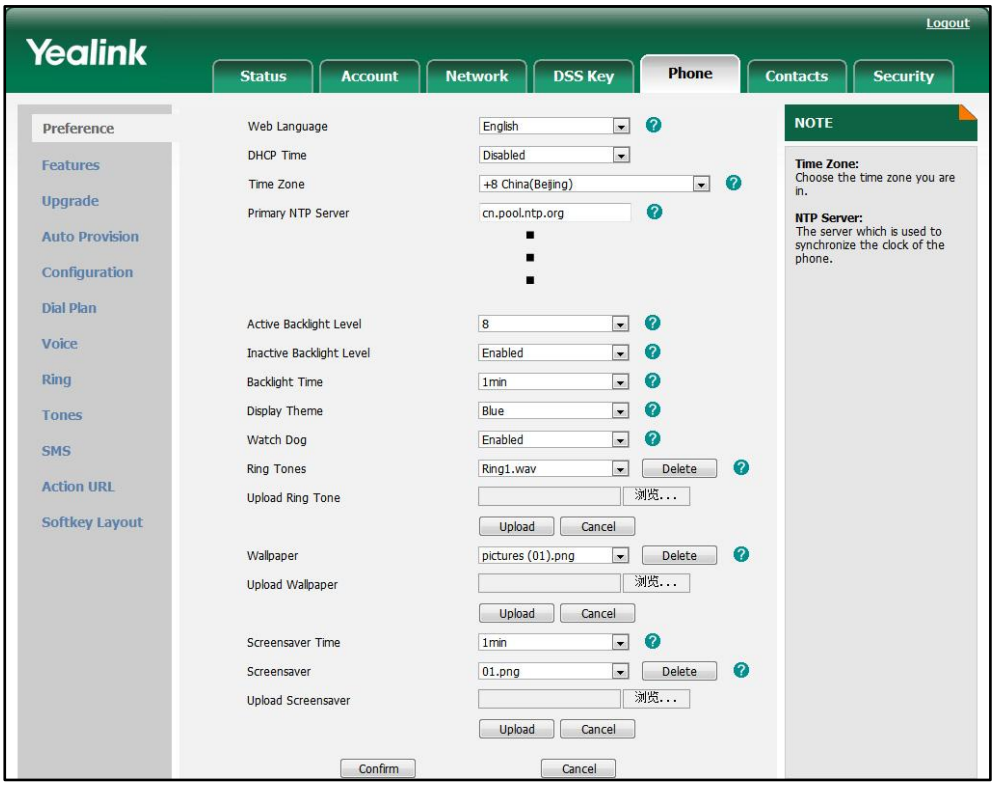
Backlight can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the backlight of the LCD screen. For more information, refer to Backlight on page 235.
Local	Web User Interface	Configure the backlight of the LCD screen. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Preference.htm
	Phone User Interface	Configure the backlight of the LCD screen.

To configure the backlight via web user interface:

1. Click on **Phone**->**Preference**.
2. Select the desired value from the pull-down list of **Active Backlight Level**.
3. Select the desired value from the pull-down list of **Inactive Backlight Level**.

4. Select the desired value from the pull-down list of **Backlight Time**.



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

To configure the backlight via phone user interface:

1. Press **Menu->Display->Backlight**.
2. Press or , or the **Switch** soft key to select the desired level from the **Active Level** field.
3. Press or , or the **Switch** soft key to select the desired level from the **Inactive Level** field.
4. Press or , or the **Switch** soft key to select the desired time from the **Backlight Time** field.
5. Press the **Save** soft key to accept the change.

User Password

Several setting menus are protected with two privilege levels, user and administrator, each with its own password. When logging in the web user interface, you need to enter the username and password for granting access to various menu options.

A user or an administrator can change the user password. The IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.

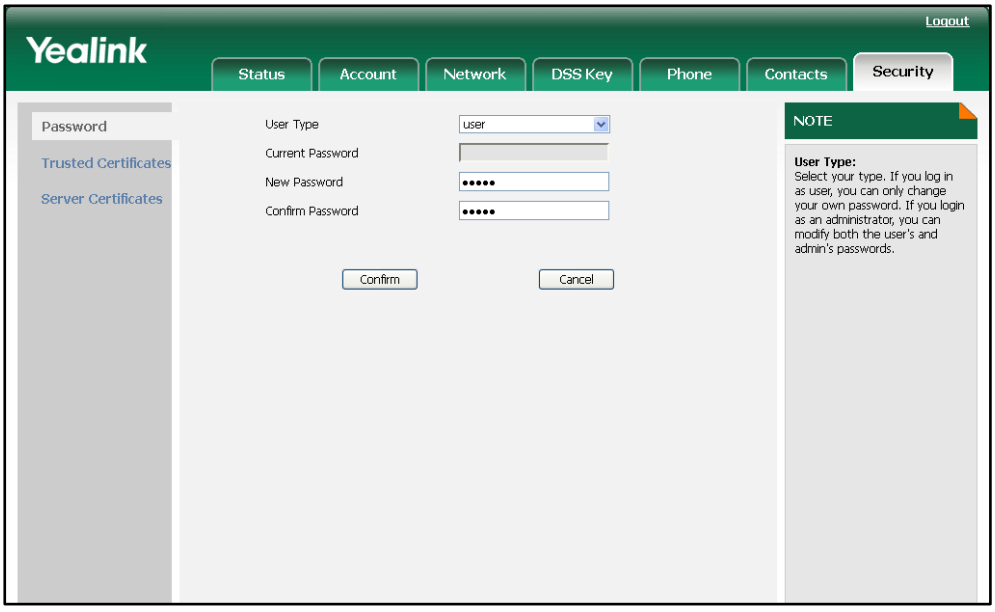
Procedure

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

Configuration File	<y0000000000xx>.cfg	Change the user password of the IP phone. For more information, refer to on User Password page 236.
Local	Web User Interface	Change the user password of the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Security.htm

To change the user password via web user interface:

1. Click on **Security->Password**.
2. Select **user** from the pull-down list of **User Type**.
3. Enter a new password in the **New Password** and **Confirm Password** fields.



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

Note If an administrator changes the user password via web user interface, the Current Password field is grayed out.

Administrator Password

Advanced menu options are restricted to an administrator. You can configure them only

if having administrator privileges. The administrator password can be only changed by the administrator. The IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.

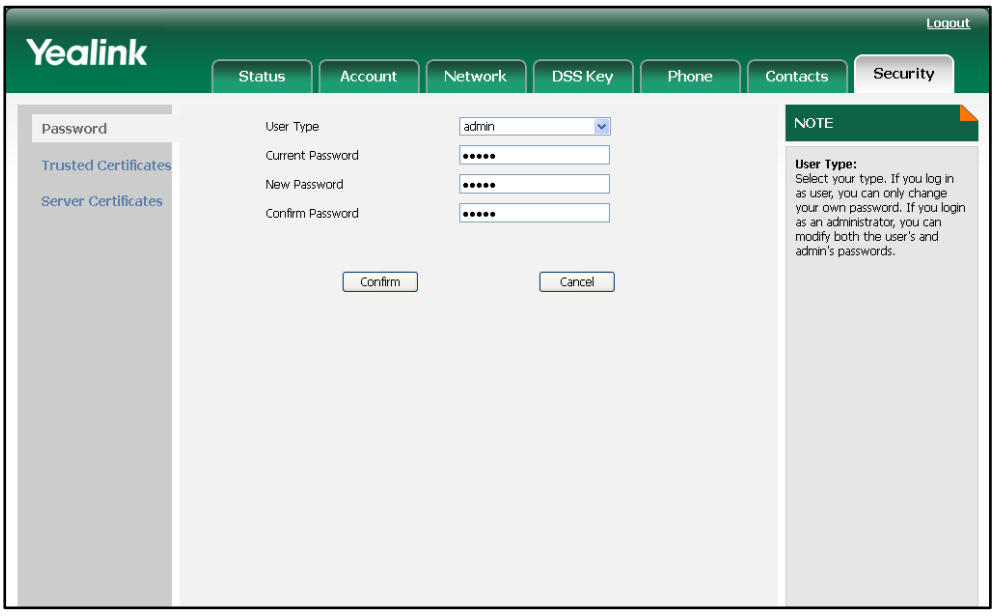
Procedure

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

Configuration File	<y0000000000xx>.cfg	Change the administrator password of the IP phone. For more information, refer to Administrator Password on page 237.
Local	Web User Interface	Change the administrator password. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Security.htm
	Phone User Interface	Change the administrator password of the IP phone.

To change the administrator password via web user interface:

1. Click on **Security**.
2. Select **admin** from the pull-down list of **User Type**.
3. Enter the current administrator password in the **Current Password** field.
4. Enter a new password in the **New Password** and **Confirm Password** fields.



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

To change the administrator password via phone user interface:

1. Press **Menu**->**Setting**->**Advanced Settings** (password: admin) ->**Set Password**.
2. Enter the old password in the **Current PWD** field.
3. Enter the new password in the **New PWD** field.
4. Enter the new password again in the **Confirm PWD** field.
5. Press the **Save** soft key to accept the change.

Phone Lock

Phone lock is used to lock the IP phones to prevent it from unauthorized use. Once the IP phone is locked, a user needs to enter the password to unlock it. The IP phone offers four types of phone lock: Menu Key, Function Keys, All Keys and Answer call only. The IP phone lock feature cannot take effect immediately after the IP phone lock type is configured. One of the following steps is also needed by the user:

- 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 steps above, you can configure the IP phone to automatically lock the keypad after a time interval.

Procedure

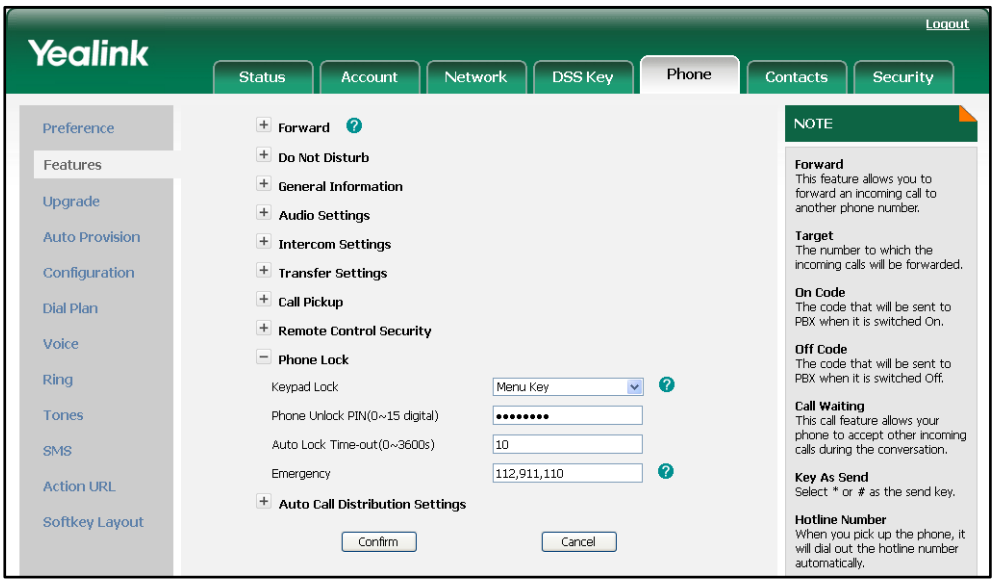
Phone lock can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the type of phone lock. Change the unlock password. Configure the IP phone to automatically lock the keypad after a time interval. For more information, refer to Phone Lock on page 237. Assign a keypad lock key. For more information, refer to Keypad Lock Key on page 322.
Local	Web User Interface	Configure the type of phone lock. Change the unlock password. Configure the IP phone to automatically lock the keypad after a time interval. **Navigate to**:

		http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm Assign a keypad lock key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Configure the type of phone lock. Assign a keypad lock key.

To configure phone lock via web user interface:

1. Click on **Phone->Features->Phone Lock**.
2. Select the desired type from the pull-down list of **Keypad Lock**.
3. Enter the unlock password (numeric characters) in the **Phone Unlock PIN (0~15 digital)** field.
4. Enter the desired time in the **Auto Lock Time-out (0~3600s)** field.

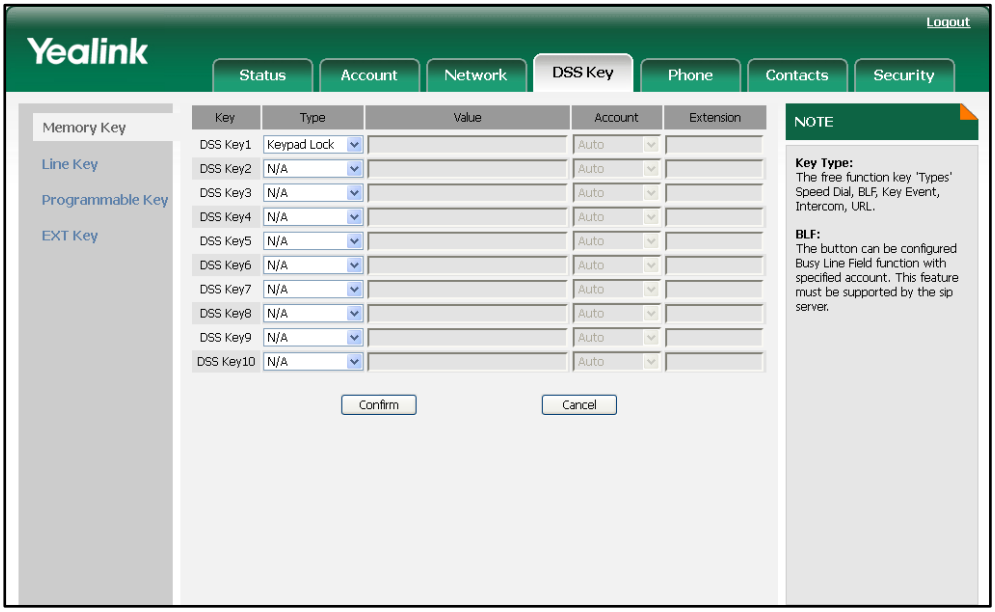


5. Click **Confirm** to accept the change.

To configure a keypad lock key via web user interface:

1. Click on **DSS Key->Memory Key** (or **Line Key**).

2. In the desired DSS key field, select **Keypad Lock** from the pull-down list of **Type**.



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

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

1. Press **Menu**->**Setting**->**Advanced Settings** (password: admin) ->**Phone Setting**->**Lock**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired type from the **Keypad Lock** field.
3. Press the **Save** soft key to accept the change.

To configure a keypad lock key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ► , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ► , or the **Switch** soft key to select **Keypad Lock** from the **Key Type** field.
5. Press the **Save** soft key to accept the change.

Time and Date

The IP phone maintains a local clock and calendar. Time and date can be displayed on the idle screen of the IP phone. The IP phone obtains the time and date automatically from the NTP server by default. If the IP phone cannot obtain the time and date from the NTP server, you can manually configure them. The time and date display can use one of several different formats.

Time Zone

A time zone is a region on the 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 need to 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 near the start of spring and are adjusted 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. Usually there is no need to change this setting.

The following table lists the available methods for each feature:

Feature	Method of Configuration
Set Time Zone	Configuration Files Web User Interface Phone User Interface
Set Time	Web User Interface Phone User Interface
Set Time Format	Configuration Files Web User Interface Phone User Interface
Set Date	Web User Interface Phone User Interface
Set Date Format	Configuration Files Web User Interface Phone User Interface
Set Daylight Saving Time	Configuration Files Web User Interface

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the NTP server, time zone and DST. Configure the time and date formats. For more information, refer to Time and Date on page 239.
Local	Web User Interface	Configure the NTP server, time zone and DST. Configure the time and date manually. Configure the time and date formats. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Preference.htm
	Phone User Interface	Configure the NTP server and time zone. Configure the time and date manually. Configure the time and date formats.

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

1. Click on **Phone**->**Preference**.
2. Select **Disabled** from the pull-down list of **Manual Time**.
3. Select the desired time zone from the pull-down list of **Time Zone**.
4. Enter the domain names or IP addresses in the **Primary NTP Server** and **Second NTP Server** fields respectively.
5. Enter the desired time interval in the **Update Interval (seconds)** field.
6. Select the desired value from the pull-down list of **Daylight Saving Time**.

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

   - Mark the **By Date** radio box in the **Fixed Type** field.

     Select the start month from the pull-down list of **Start Month**.

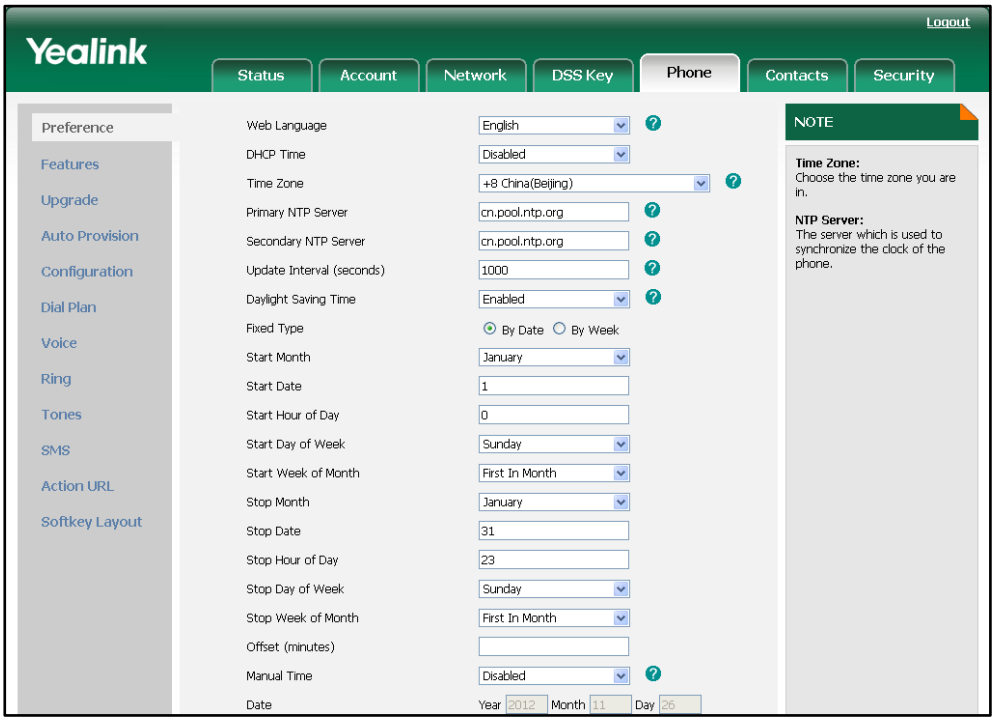
Enter the start date in the **Start Date** field.

Enter the start time in the **Start Hour of Day** field.

Select the end month from the pull-down list of **Stop Month**.

Enter the end date in the **Stop Date** field.

Enter the end time in the **Stop Hour of Day** field.



- Mark the **By Week** radio box in the **Fixed Type** field.

  Select the start month from the pull-down list of **Start Month**.

  Enter the start time in the **Start Hour of Day** field.

  Select the start day from the pull-down list of **Start Day of Week**.

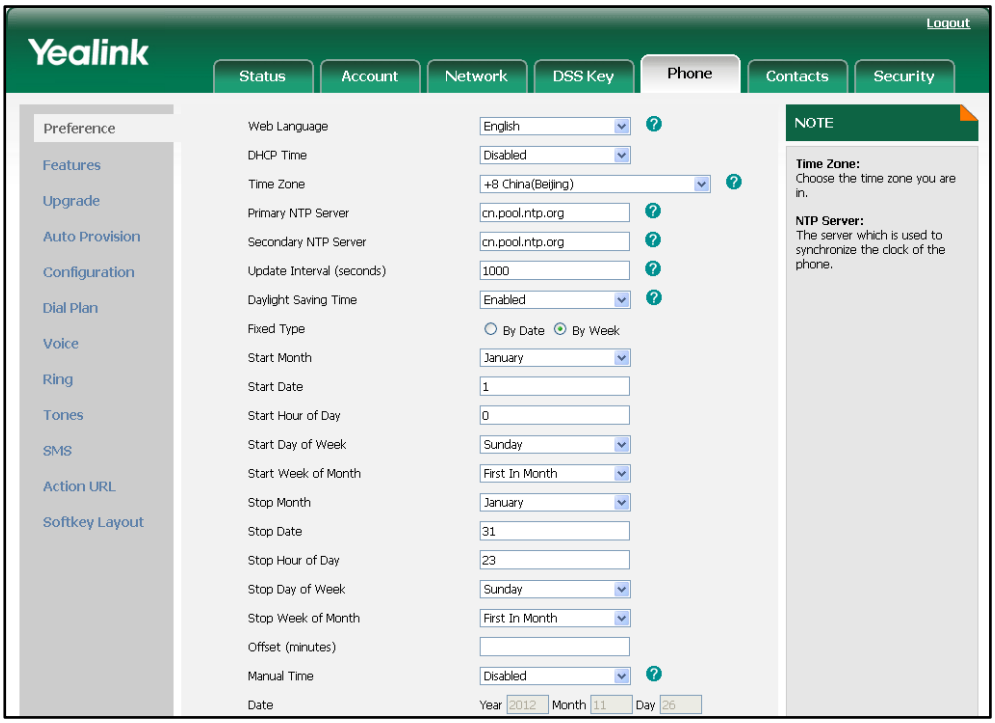
  Select the start week from the pull-down list of **Start Week of Month**.

  Select the end month from the pull-down list of **Stop Month**.

  Enter the end time in the **Stop Hour of Day** field.

  Select the end day from the pull-down list of **Stop Day of Week**.

Select the end week from the pull-down list of **Stop Week of Month**.

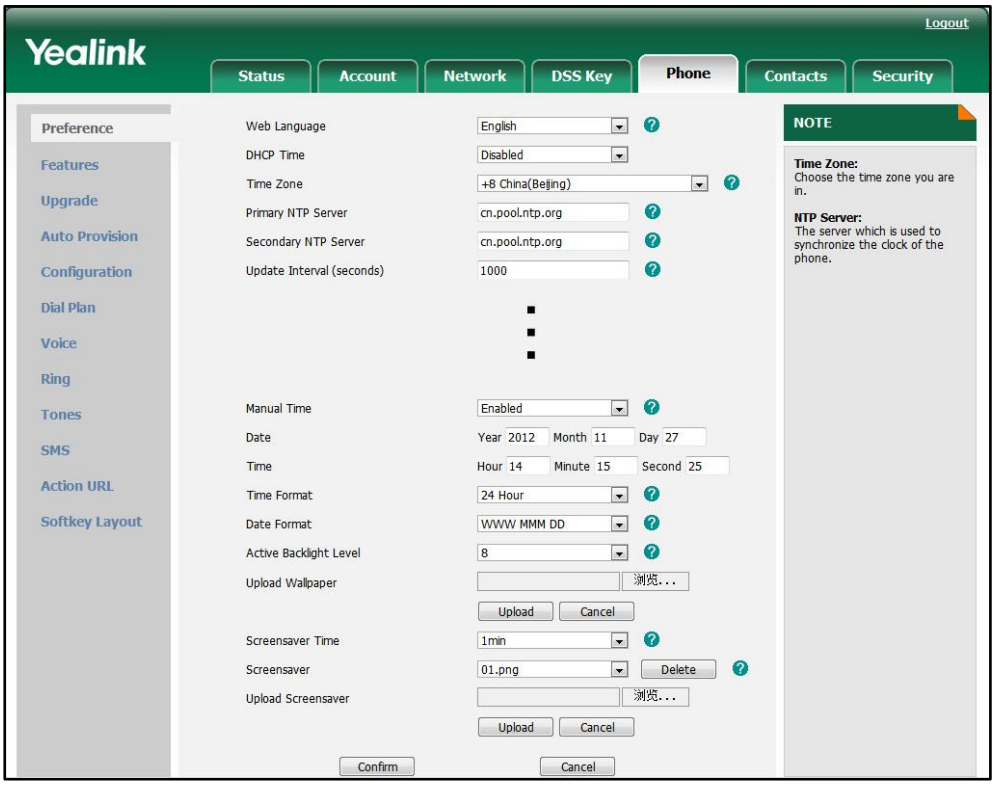


7. Enter the desired offset in the **Offset (Minutes)** field.
8. Click **Confirm** to accept the change.

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

1. Click on **Phone**->**Preference**.
2. Select **Enabled** from the pull-down list of **Manual Time**.

3. Enter the date and time in the corresponding fields.

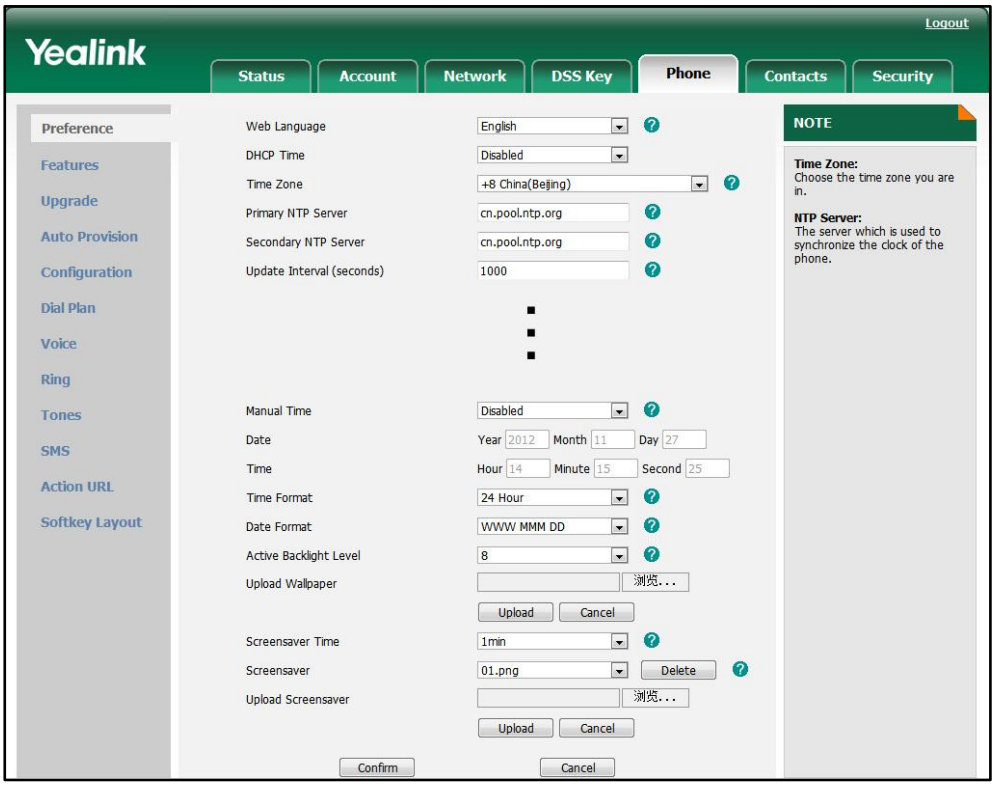


4. Click **Confirm** to accept the change.

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

1. Click on **Phone**->**Preference**.
2. Select the desired value from the pull-down list of **Time Format**.

3. Select the desired value from the pull-down list of **Date Format**.



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

To configure the NTP server and time zone via phone user interface:

1. Press **Menu->Setting->Basic Settings->Time & Date->SNTP Settings**.
2. Press ◄ or ► , 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 "+8 China(Beijing)".
3. Enter the domain names or IP addresses in the **NTP Server1** and **NTP Server2** fields, respectively.
4. Press the **Save** soft key to accept the change.

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

1. Press **Menu->Setting->Basic Settings->Time & Date->Manual Setting**.
2. Enter the specific date and time.
3. Press the **Save** soft key to accept the change.

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

1. Press **Menu->Setting->Basic Settings->Time & Date Format**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired time format (12 Hour or 24 Hour) from the **Clock** field.
3. Press ◄ or ► , or the **Switch** soft key to select the desired date format from the **Date Format** field.

4. Press the **Save** soft key to accept the change.

Language

The IP phones support multiple languages. The languages used on the phone user interface and web user interface can be specified respectively as required.

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

Phone User Interface	Web User Interface
English	English
Chinese_S	Chinese_S
Chinese_T	Deutsch
German	French
French	Italian
Italian	Portuguese
Portuguese	Spanish
Polish	Turkish
Spanish	
Turkish	

Loading Language Packs

All supported languages may not be available for selection. The languages available for selection depend on the language packs currently loaded on the IP phone. You can make languages available to use on the phone user interface by loading language packs to the IP phone. You can only load language packs to the IP phone using the configuration files.

The following table lists the available language and the associated language packs:

Available	Associated Language Pack
English	Lang+English.txt
Chinese_S	lang-Chinese_S.txt
Chinese_T	lang-Chinese_T.txt
Deutsch	lang-German.txt
French	lang-French.txt
Italian	lang-Italian.txt
Portuguese	lang-Portuguese.txt

Available	Associated Language Pack
Polish	lang-Polish.txt
Spanish	lang-Spanish.txt
Turkish	lang-Turkish.txt

Procedure

Loading language pack can be only performed using the configuration files.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the language pack. For more information, refer to Language on page 244.

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 web user interface respectively.

Procedure

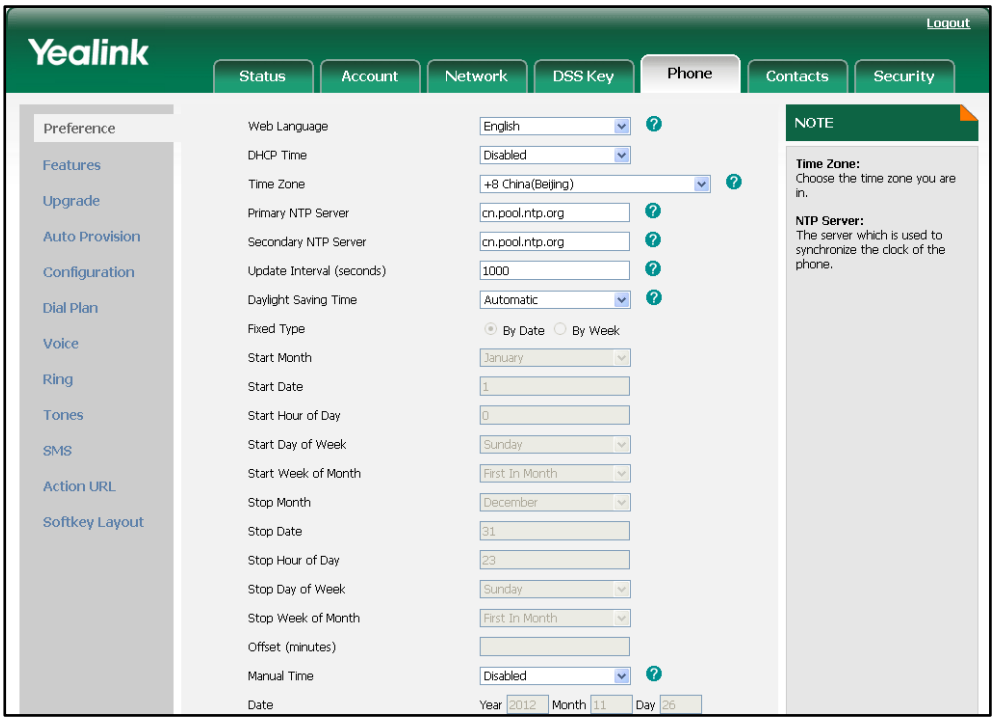
Specify the language for the web user interface or the phone user interface using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the languages for the phone user interface and the web user interface. For more information, refer to Language on page 244.
Local	Web User Interface	Specify the language for the web user interface. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Preference.htm
	Phone User Interface	Specify the language for the phone user interface.

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

1. Click on **Phone**->**Preference**.

2. Select the desired language from the pull-down list of **Web Language**.



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

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

1. Press **Menu->Settings->Basic Settings->Language**.
2. Press (▲) or (▼) to select the desired language.
3. Press the **Save** soft key to accept the change.

Softkey Layout

Softkey layout is used to customize the soft keys at the bottom of the IP phone to best suit the needs of users. It can be controlled based on the call states. In addition to specifying which soft keys to display, you can determine the display order of the soft keys. You can create a template about the soft key layout of the different call states. For more information on the soft key layout template, refer to Softkey Layout Template on page 201.

The following table lists the soft keys available for the IP phone in different states:

Call State		Default Soft Key	Optional Soft Key
CallFailed		NewCall Empty Empty Cancel	Empty Switch
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 Directory Switch Line Pool GPickup DPickup
RingBack	RingBack	Empty Empty	Empty Switch

Call State		Default Soft Key	Optional Soft Key
		Empty Cancel	
	SemiAttendTransBack	Transfer Empty Empty Cancel	Empty Switch
Talking	Talk	Transfer Hold Conference Cancel	Empty Mute SWAP NewCall Switch Answer Reject
	Hold	Transfer Resume Empty Cancel	Empty Switch Answer Reject NewCall
	Held	Empty Empty Empty Cancel	Empty Switch Answer Reject NewCall
	PreConf	Send IME Delete Cancel	Empty Directory Switch
	PreTrans	Transfer Send Delete Cancel	Empty Directory Switch IME
	InConference	Empty Empty Empty	Empty Switch

Call State		Default Soft Key	Optional Soft Key
		Cancel	
	InConferenceTalk	Empty Empty Conference Cancel	Empty Switch
	Conferenced	Empty Hold Split Cancel	Empty Switch Answer Reject Mute

Procedure

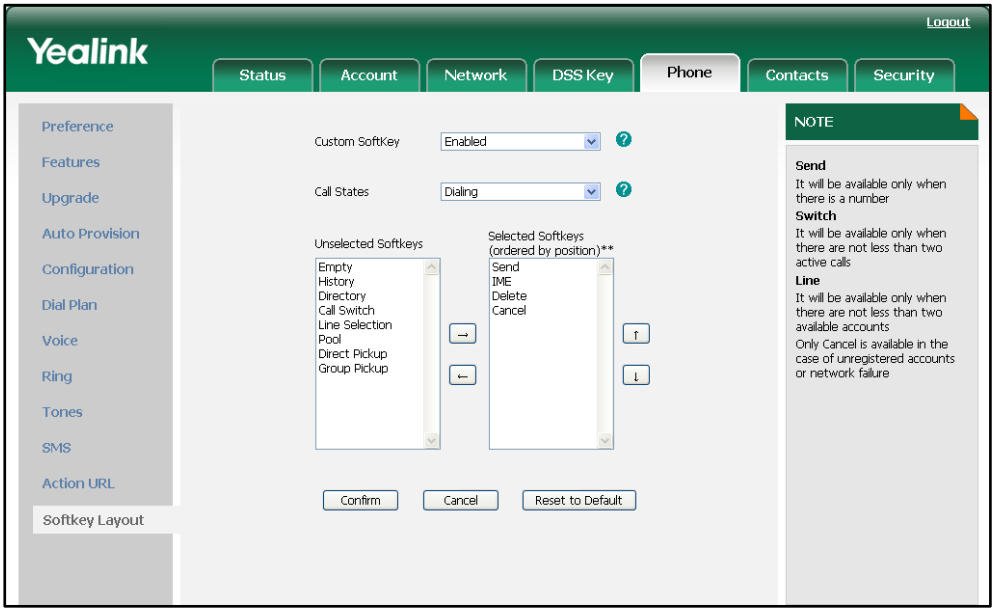
Softkey layout can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the softkey layout template. For more information, refer to Access URL of Softkey Layout Template on page 313.
Local	Web User Interface	Configure the softkey layout. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Softkey.htm

To configure soft key layout via web user interface:

1. Click on **Phone**->**Softkey Layout**.
2. Select the desired value from the pull-down list of **Custom SoftKey**.
3. Select the desired state from the pull-down list of **Call States**.
4. In the **Unselected Softkeys** box, select the desired soft key and click → to move to the **Selected Softkeys** box.
5. In the **Selected Softkeys** box, select the undesired soft key and click ← to move to the **Unselected Softkeys** box.

6. Click ↑ or ↓ to move up or down the soft key.



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

Key as Send

The key as send feature allows assigning the pound key or star key as a send key. The send tone feature determines whether the IP phone plays a key tone when a user presses the send key.

Procedure

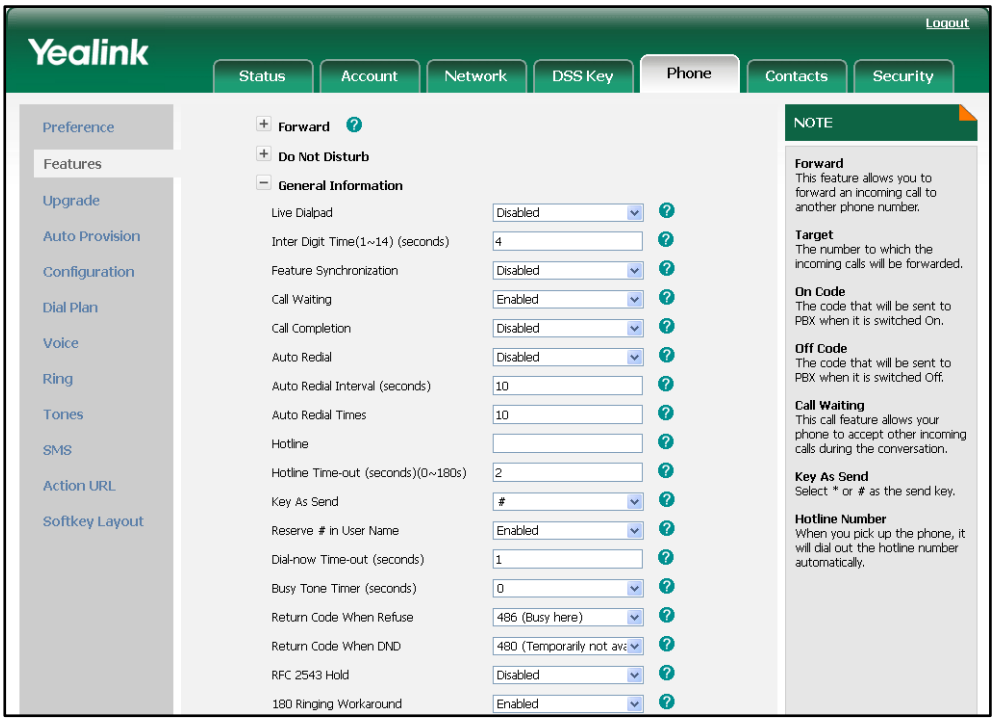
Key as send can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the send key. Configure the send tone feature. For more information, refer to Key as Send on page 246.
Local	Web User Interface	Configure the send key. Configure the send tone feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the send key.

To configure the send key via web user interface:

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

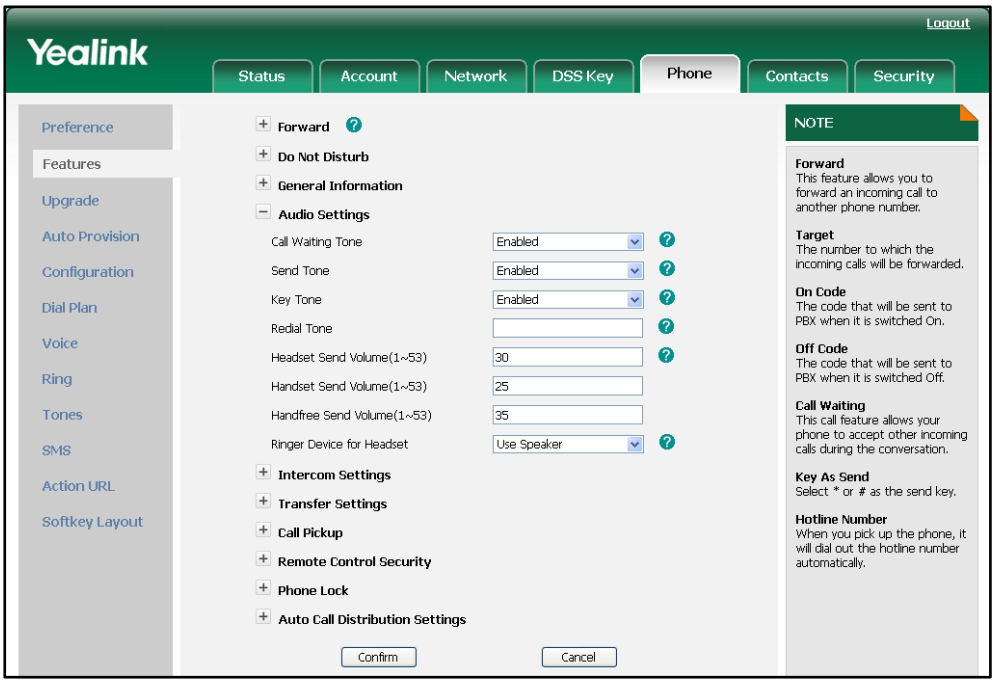
2. Select the desired value from the pull-down list of **Key As Send**.



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

To configure the send tone via web user interface:

1. Click on **Phone->Features->Audio Settings**.
2. Select the desired value from the pull-down list of **Send Tone**.



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

To configure the send key via phone user interface:

1. Press **Menu**->**Features**->**Key as Send**.
2. Press ◀ or ▶, or the **Switch** soft key to select "#" or "*" from the **Key as Send** field, or select **Disable** to disable this feature.
3. Press the **Save** soft key to accept the change.

Note The send tone feature works only if the key tone feature is enabled. The key tone feature is enabled by default.

Hotline

A hotline is a point-to-point communications link in which a call is automatically directed to the preset hotline number. The IP phone automatically dials out the hotline number using the first available line after a time interval when the IP phone is off-hook. The IP phone only supports one hotline number.

Procedure

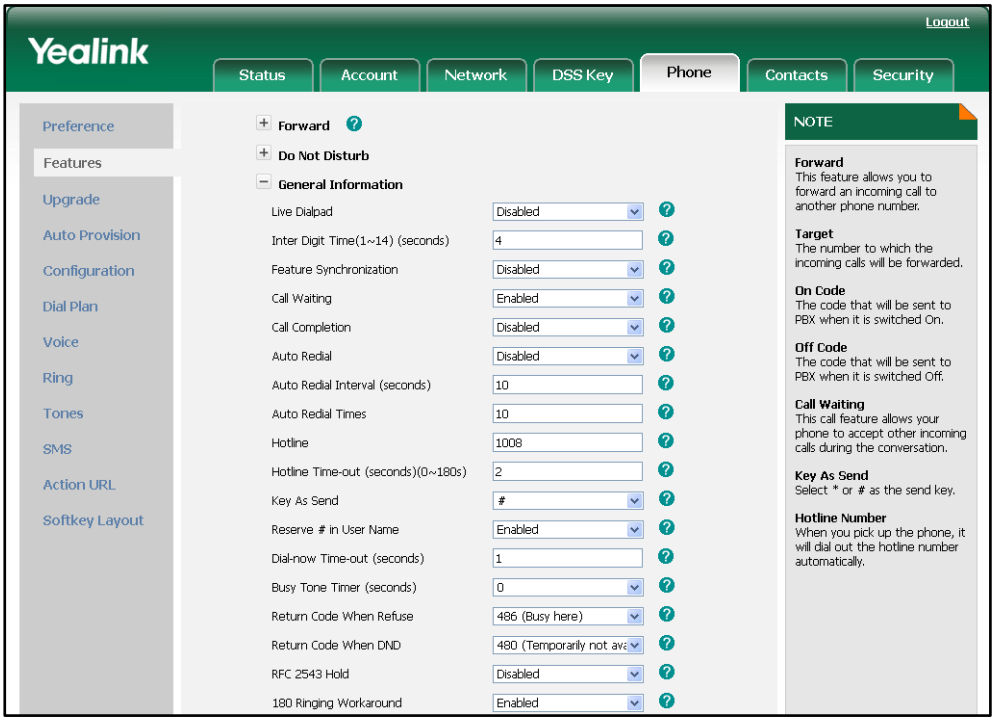
Hotline can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the hotline number. Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number. For more information, refer to Hotline on page 247.
Local	Web User Interface	Configure the hotline number. Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the hotline number. Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number.

To configure hotline via web user interface:

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

2. Enter the hotline number in the **Hotline** field.
3. Enter the delay time in the **Hotline Time-out (seconds) (0~180s)** field.



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

To configure hotline via phone user interface:

1. Press **Menu->Features->Hot Line**.
2. Enter the hotline number in the **Hotline** field.
3. Enter the delay time (in seconds) in the **Hotline Time-out** field.
4. Press the **Save** soft key to accept the change.

Call Log

The IP phone maintains a local call log. The call log contains call information such as remote party identification, time and date, and call duration. The IP phone maintains four call log lists: Dialed Calls, Received Calls, Missed Calls and Forwarded Calls. All call log lists support to store 100 entries in all. To manage the entries of the call log lists, you should enable the IP phone to save call log in advance.

Procedure

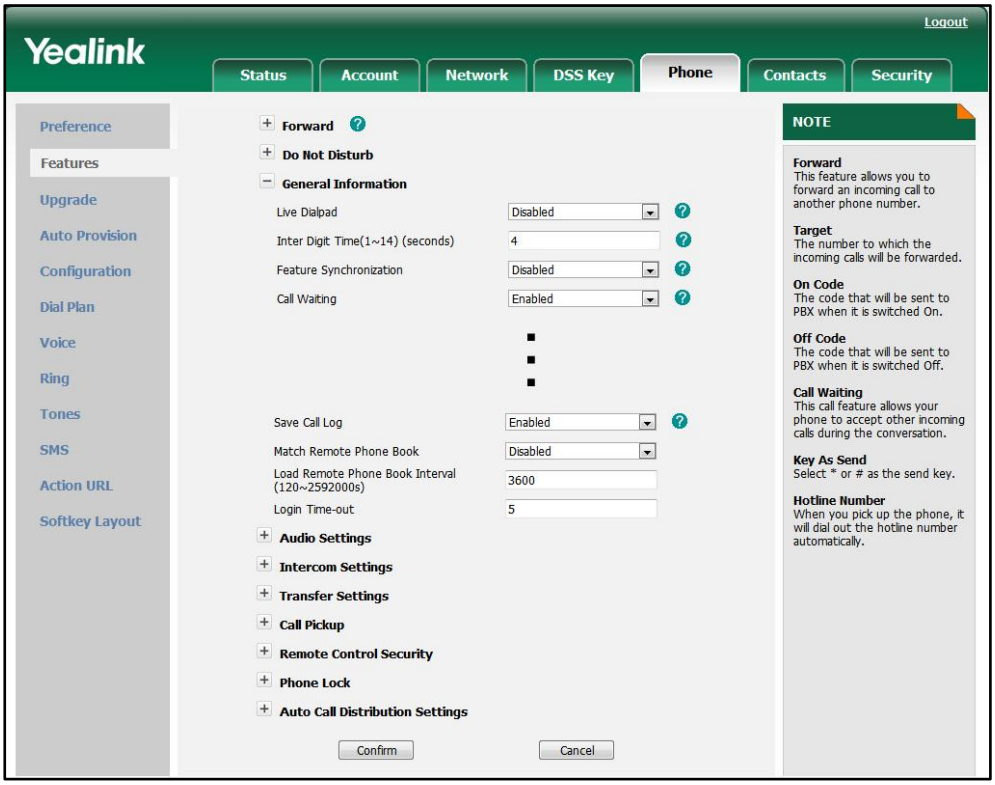
Call log can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the call log. For more information, refer to Call Log on page 248.

Local	Web User Interface	Configure the call log. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the call log.

To configure the call log via web user interface:

1. Click on **Phone**->**Features**->**General Information**.
2. Select the desired value from the pull-down list of **Save Call Log**.



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

To configure the call log via phone user interface:

1. Press **Menu**->**Features**->**History Setting**.
2. Press (◄) or (►), or the **Switch** soft key to select the desired value from the **History Record** field.
3. Press the **Save** soft key to accept the change.

Missed Call Log

When the IP phone misses calls, the missed call log feature allows the IP phone to display the number of the missed calls and indicator icon on the idle screen, and to log the missed calls in the Missed Calls list. The missed call log feature is configurable on a per-account basis. Once the user accesses the Missed Calls list, the prompt message and the indicator icon on the idle screen are cleared.

Procedure

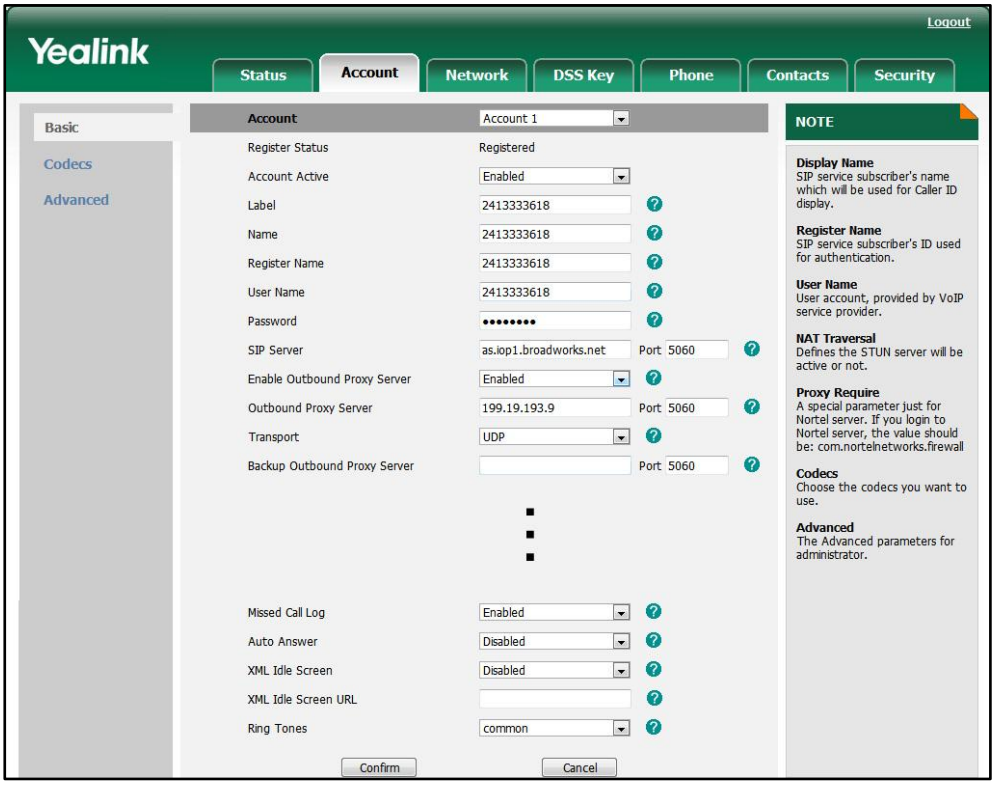
Missed call log can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the missed call log feature. For more information, refer to Missed Call Log on page 249.
Local	Web User Interface	Configure the missed call log feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure missed call log via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Missed Call Log**.



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

Local Directory

The IP phone maintains a local directory. The directory can be used to store the frequently used contacts. When adding a contact to the local directory, you can specify the account, ring tone and group for the contact in addition to name and phone numbers. The local directory can store up to 1000 contacts. The contacts can be created either one by one or in batch using a contact file. For more information on the contact file, refer to Local Contact File on page 203.

Procedure

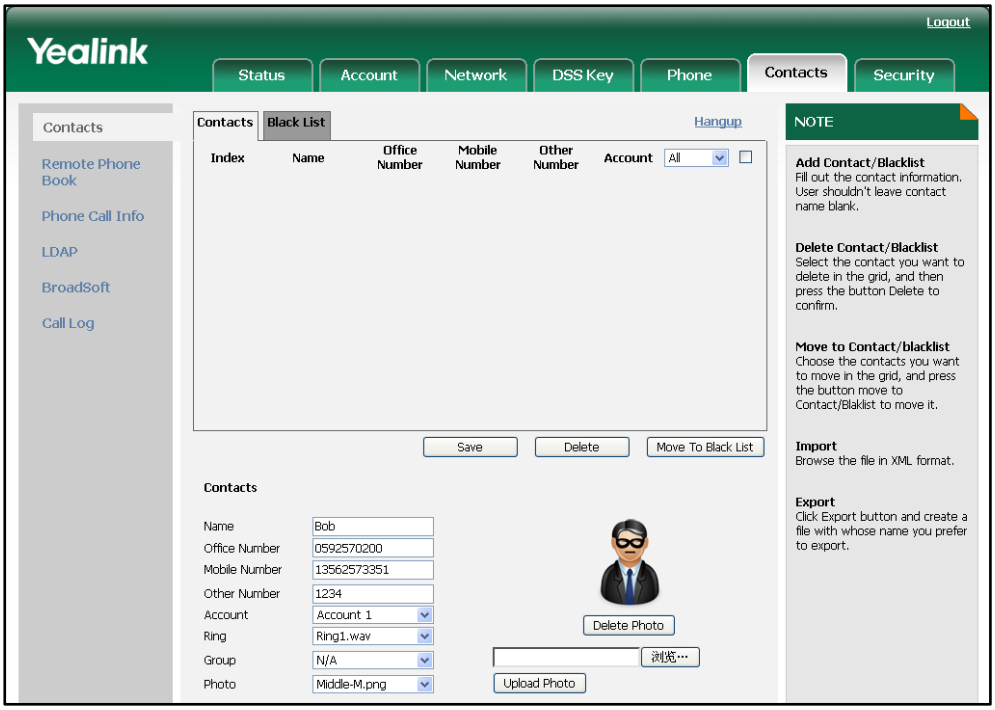
Configuration changes can be performed using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the local contact file. For more information, refer to Access URL of Local Contact File on page 315.
Local	Web User Interface	Add the contact to the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/

		cgiServer.exx?page=Contacts.htm
	Phone User Interface	Add the contact to the local directory directly.

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

1. Click on **Contacts**->**Contacts**->**Contacts**.
2. Enter the name and the office, mobile or other numbers in the corresponding fields.
3. Select the desired account from the pull-down list of **Account**.
4. Select the desired ring tone from the pull-down list of **Ring**.
5. Select the desired group from the pull-down list of **Group**.
6. Select the desired photo from the pull-down list of **Photo**.



7. Click **Add** to add the contact.
8. Click **Save** to accept the change.

To add the contact to the local directory via phone user interface:

1. Press **Directory**->**Local Directory**.
2. Select the desired contact group (For example, select **Contacts**).
3. Press the **Add** soft key.
4. Enter the name and the office, mobile or other numbers in the corresponding fields.
5. Press ◄ or ►, or the **Switch** soft key to select the desired account from the **Account** field. If **Auto** is selected, the IP phone will use the first available account when placing calls to the contact from the local directory.

6. Press ◄ or ►, or the **Switch** soft key to select the desired ring tone from the **Ring** field.
7. Press the **Save** soft key to accept the change.

Live Dialpad

Commonly, a user dials a number while the IP phone is on-hook, he needs to lift the handset or press the speakerphone key to initiate the call. Live dialpad enables the IP phone to automatically dial out the entered phone number after a time interval.

Procedure

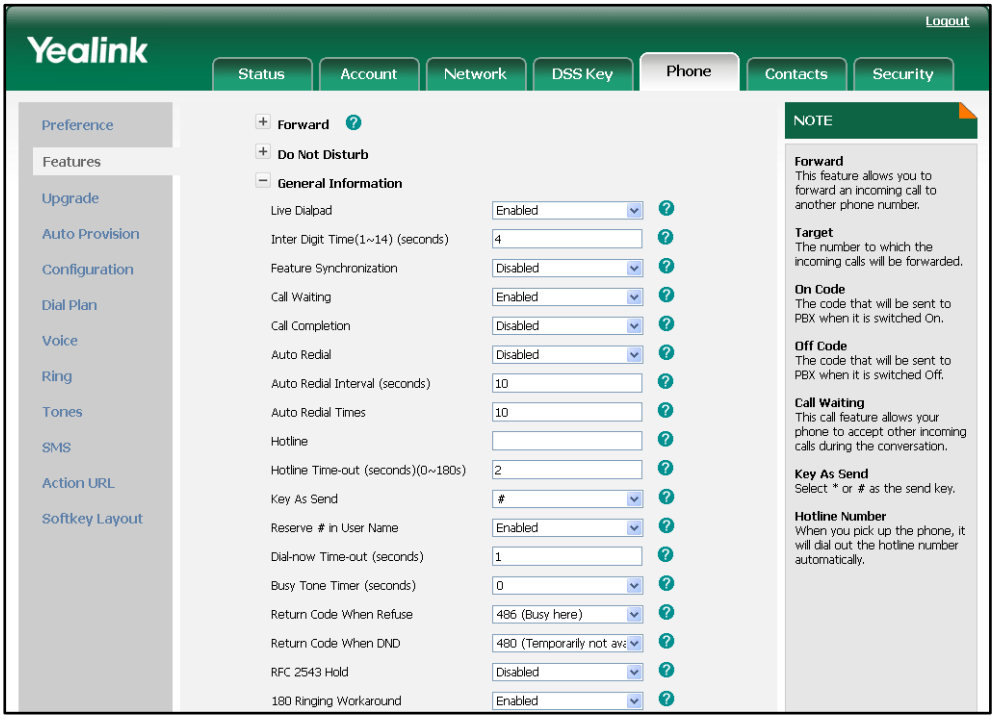
Live dialpad can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the live dialpad feature. For more information, refer to Live Dialpad on page 249.
Local	Web User Interface	Configure the live dialpad feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure live dialpad via web user interface:

1. Click on **Phone**->**Features**->**General Information**.
2. Select the desired value from the pull-down list of **Live Dialpad**.

3. (If enabled) Enter the desired delay time in the **Inter Digit Time (1~14) (seconds)** field.



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

Call Waiting

The Call waiting feature enables the IP phone to receive a new call when there is an active call. The new call is presented to the user visually on the LCD screen. The call waiting tone feature enables the IP phone to play a short tone when receiving a new incoming call during a conversation. The tone is audible to remind the user of the new incoming call. The call waiting tone feature works only if the call waiting is enabled.

Procedure

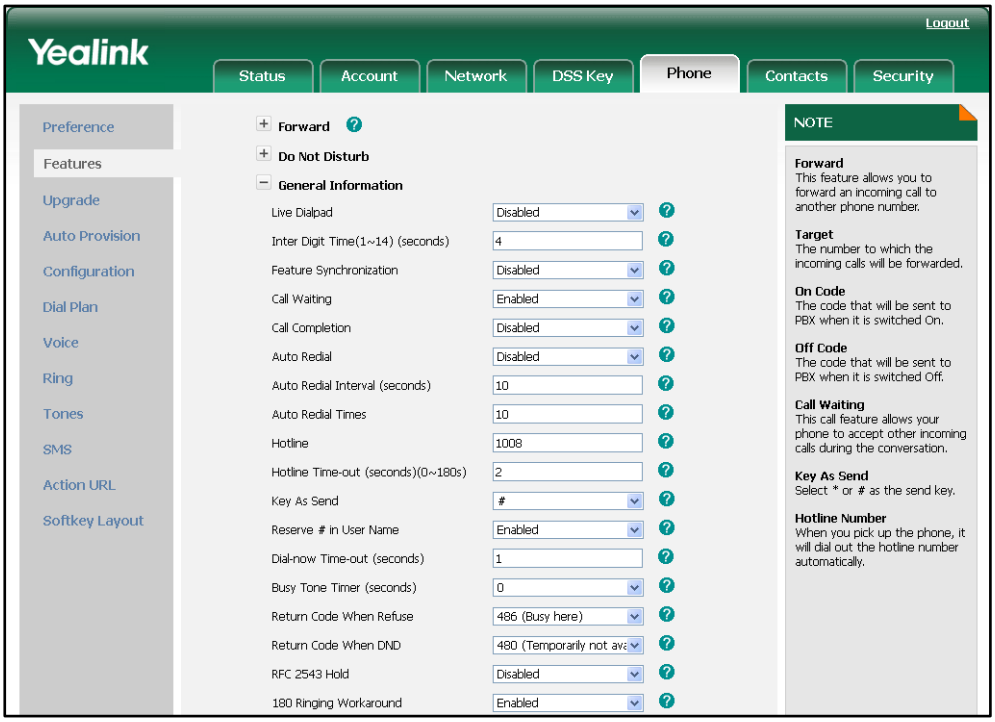
Call waiting and call waiting tone can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the call waiting feature. For more information, refer to Call Waiting on page 250.
Local	Web User Interface	Configure the call waiting feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/

		cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the call waiting feature.

To configure call waiting via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Call Waiting**.

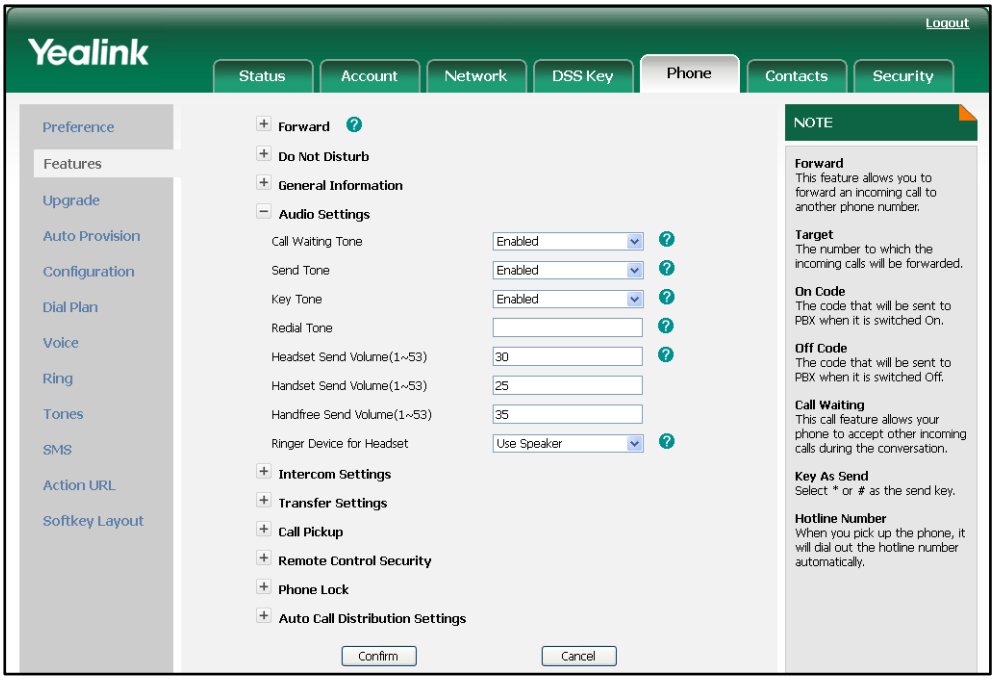


3. Click **Confirm** to accept the change.

To configure the call waiting tone via web user interface:

1. Click on **Phone->Features->Audio Settings**.

2. Select the desired value from the pull-down list of **Call Waiting Tone**.



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

To configure call waiting and call waiting tone via phone user interface:

1. Press **Menu**->**Features**->**Call Waiting**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired value from the **Call Waiting** field.
3. Press ◄ or ►, or the **Switch** soft key to select the desired value from the **Call Waiting Tone** field.
4. Press the **Save** soft key to accept the change.

Auto Redial

Auto redial allows the IP phone to redial a busy number after the first attempt. Both the number of attempts and delay between redials are configurable.

Procedure

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

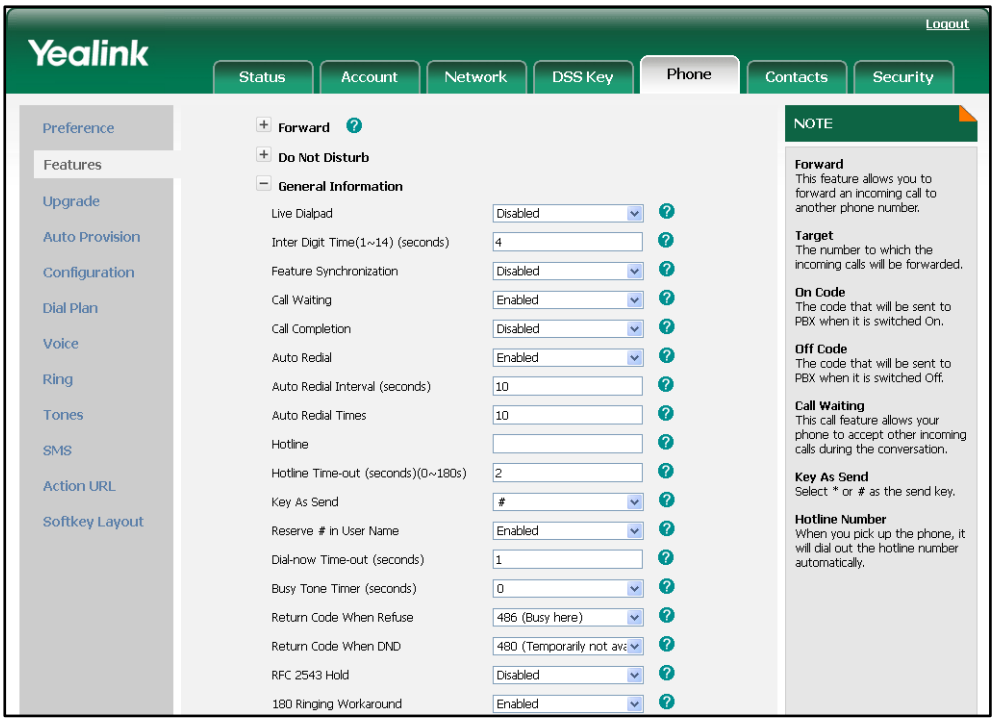
Configuration File	<y0000000000xx>.cfg	Configure the auto redial feature. For more information, refer to Auto Redial on page 251.
Local	Web User Interface	Configure the auto redial feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the auto redial feature.

To configure auto redial via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Auto Redial**.
3. (If enabled) Enter the desired time interval in the **Auto Redial Interval (seconds)** field.

   The default time interval is 10s.
4. (If enabled) Enter the desired times in the **Auto Redial Times** field.

   The default times are 10.



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

To configure auto redial via phone user interface:

1. Press **Menu**->**Features**->**Auto Redial**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired value from the **Auto Redial** field.
3. Enter the desired time in the **Auto Redial Interval** field.
4. Enter the desired times in the **Auto redial times** field.
5. Press the **Save** soft key to accept the change.

Auto Answer

Auto answer allows the IP phone to automatically answer an incoming call. The IP phone cannot automatically answer the incoming call during a call even if auto answer is enabled. Auto answer is configurable on a per-account basis.

Procedure

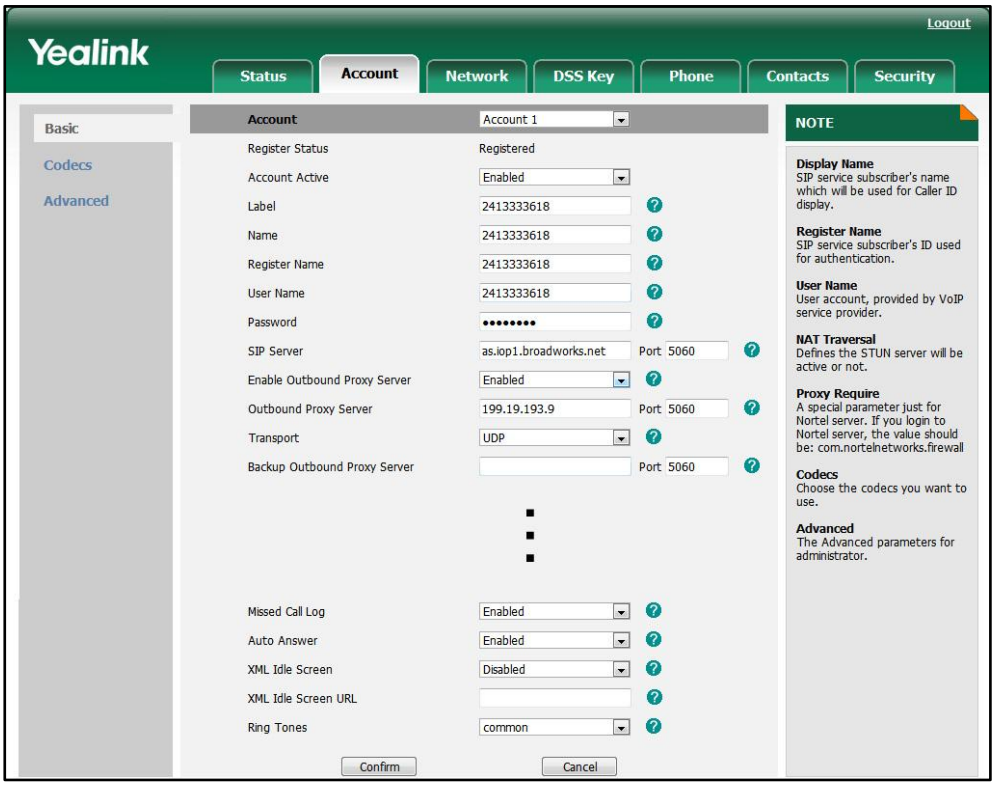
Auto answer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the auto answer feature. For more information, refer to Auto Answer on page 252.
Local	Web User Interface	Configure the auto answer feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.
	Phone User Interface	Configure the auto answer feature.

To configure auto answer via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Select the desired value from the pull-down list of **Auto Answer**.



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

To configure auto answer via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Accounts**.
2. Select the desired account and then press the **Enter** soft key.
3. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Auto Answer** field.
4. Press the **Save** soft key to accept the change.

Call Completion

When a call fails, the call completion feature allows notifying the caller when the callee becomes available to receive a call. There are several possible factors which can prevent a call from connecting successfully.

- Callee does not answer
- Callee actively rejects the incoming call before answering

The IP phones support call completion using the SUBSCRIBE/NOTIFY method, which is specified in draft-poetzl-sipping-call-completion-00, to subscribe to and manage a call completion call and to receive notifications of status changes of the call.

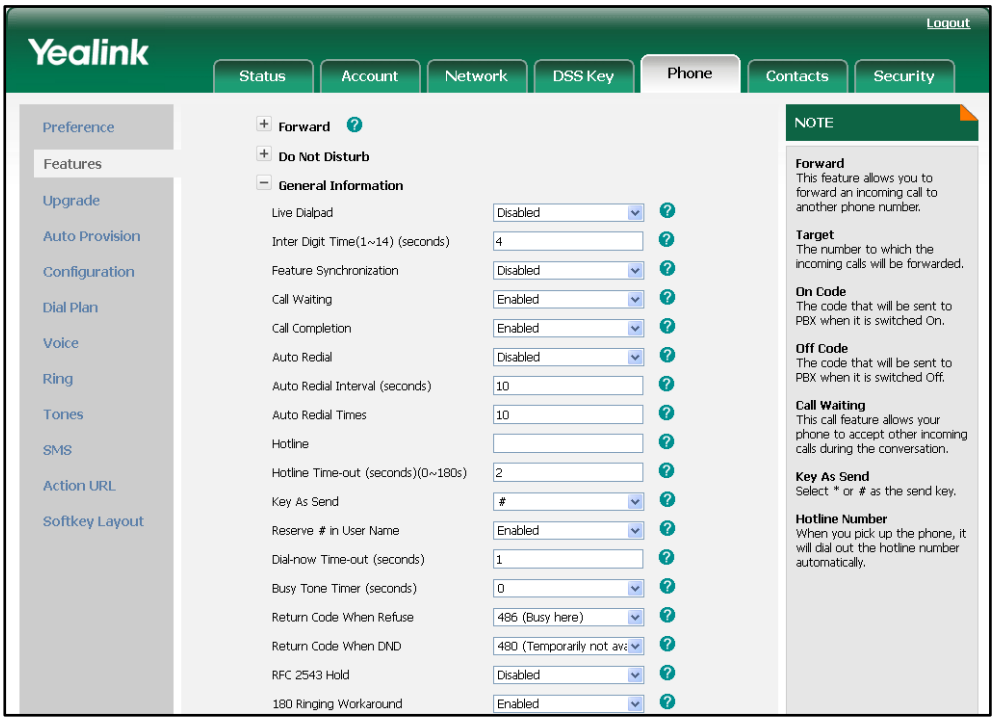
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the call completion feature. For more information, refer to Call Completion on page 252.
Local	Web User Interface	Configure the call completion feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the call completion feature.

To configure call completion via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Call Completion**.



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

To configure call completion via phone user interface:

1. Press **Menu->Features->Call Completion**.

2. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Call Completion** field.
3. Press the **Save** soft key to accept the change.

Anonymous Call

The anonymous call feature allows the caller to block the 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.

The example of the SIP header for anonymity for reference:

```
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: Yealink SIP-T38G 38.70.0.100
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 or anonymous call off code configured on the IP phone is used to inform the server of activating or deactivating the anonymous call feature. The anonymous call on code and anonymous call off code may vary on different servers.

Procedure

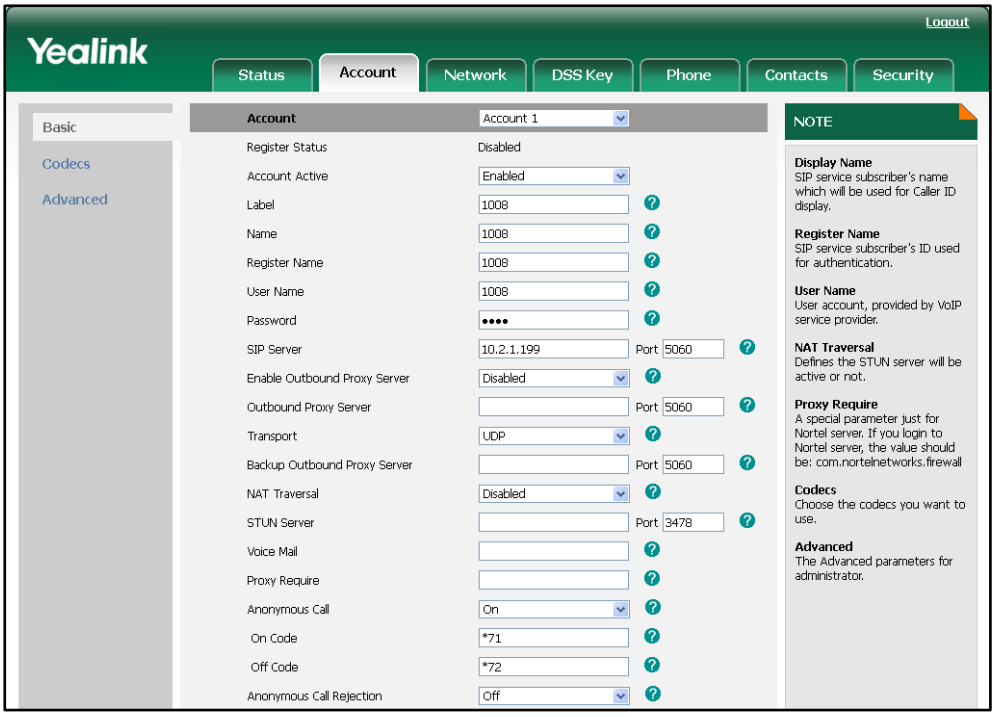
Anonymous call can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the anonymous call feature. For more information, refer to Anonymous Call on page 253.
Local	Web User Interface	Configure the anonymous call feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.ht

		m&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.
	Phone User Interface	Configure the anonymous call feature.

To configure the anonymous call via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Anonymous Call**.
4. (Optional.) Enter the anonymous call on code in the **On Code** field.
5. (Optional.) Enter the anonymous call off code in the **Off Code** field.



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

To configure the anonymous call via phone user interface:

1. Press **Menu->Features->Anonymous Call**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired line from the **Line ID** field.
3. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Anonymous Call** field.
4. (Optional.) Enter the anonymous call on code in the **Call On Code** field.
5. (Optional.) Enter the anonymous call off code in the **Call Off Code** field.

6. Press the **Save** soft key to accept the change.

Anonymous Call Rejection

The anonymous call rejection feature allows the IP phone to automatically reject incoming calls from callers who deliberately block their identities from showing up. The anonymous caller's phone LCD screen presents "Anonymity Disallowed".

The anonymous call rejection on code or anonymous call rejection off code configured on the IP phone is used to inform the server of activating or deactivating the anonymous call rejection feature. The anonymous call rejection on code and anonymous call rejection off code may vary on different servers.

Procedure

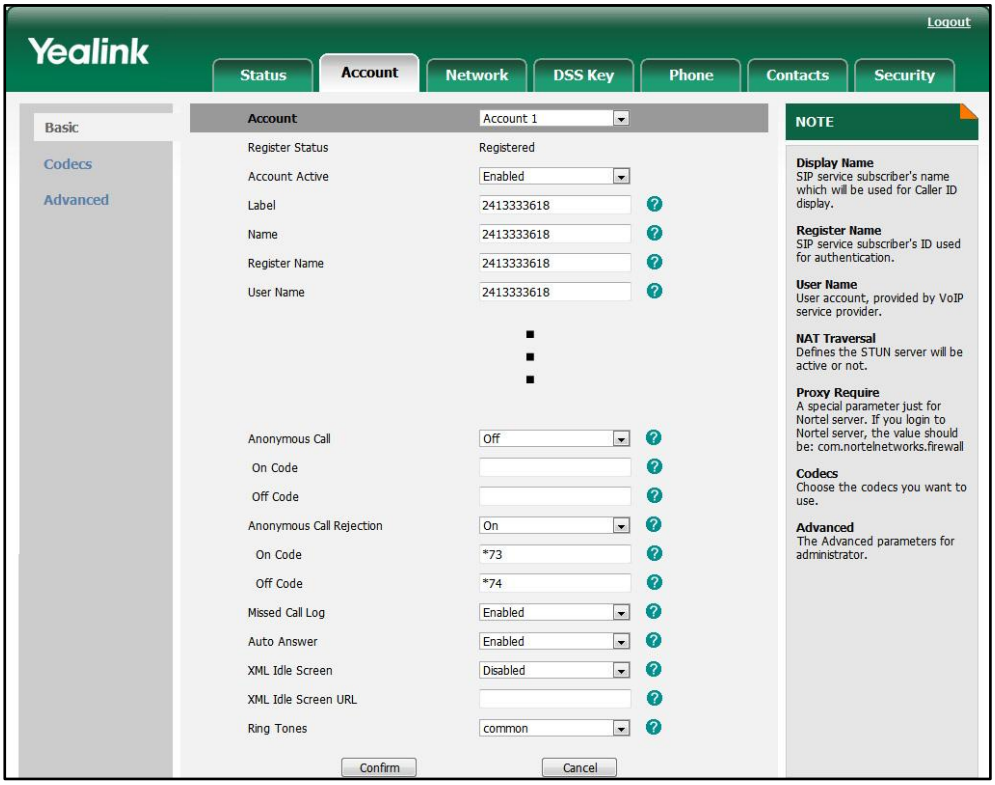
Anonymous call rejection can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the anonymous call rejection feature. For more information, refer to Anonymous Call Rejection on page 254.
Local	Web User Interface	Configure the anonymous call rejection feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.
	Phone User Interface	Configure the anonymous call rejection feature.

To configure anonymous call rejection via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select the desired value from the pull-down list of **Anonymous Call Rejection**.
4. (Optional.) Enter the anonymous call rejection on code in the **On Code** field.

5. (Optional.) Enter the anonymous call rejection off code in the **Off Code** field.



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

To configure anonymous call rejection via phone user interface:

1. Press **Menu**->**Features**->**Anonymous Call**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired line from the **Line ID** field.
3. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **Rejection** field.
4. (Optional.) Enter the anonymous call rejection on code in the **Reject On Code** field.
5. (Optional.) Enter the anonymous call rejection off code in the **Reject Off Code** field.
6. Press the **Save** soft key to accept the change.

Do Not Disturb

Do Not Disturb (DND) allows the IP phone to ignore incoming calls. A user can activate or deactivate the DND feature using a DND soft key or DND key. DND activated on the IP phone disables the local call forward settings. The DND configurations on the IP phone may be overridden by the server settings.

The DND on code or DND off code configured on the IP phone is used to inform the server of activating or deactivating the DND feature. The DND on code and DND off code may vary on different servers.

Return Message When DND

This feature defines the return code and the reason of the SIP response message when the IP phone rejects an incoming call for DND. The caller's phone LCD screen display the reason according to the return code received.

Procedure

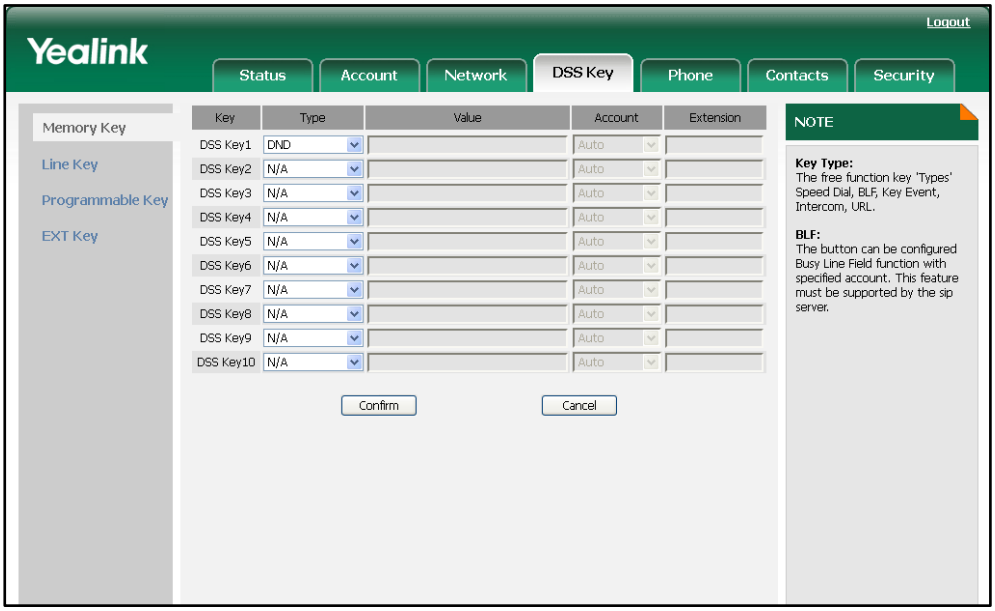
DND can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a DND key. For more information, refer to DND Key on page 323. Configure the DND on code and DND off code. Specify the DND authorized numbers. Specify the return code and the reason of the SIP response message. For more information, refer to Do Not Disturb on page 255.
Local	Web User Interface	Assign a DND key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm Configure the DND on code and DND off code. Specify the DND authorized numbers. Specify the return code and the reason of the SIP response message.

		Navigate to: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Assign a DND key.

To configure a DND key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **DND** from the pull-down list of **Type**.

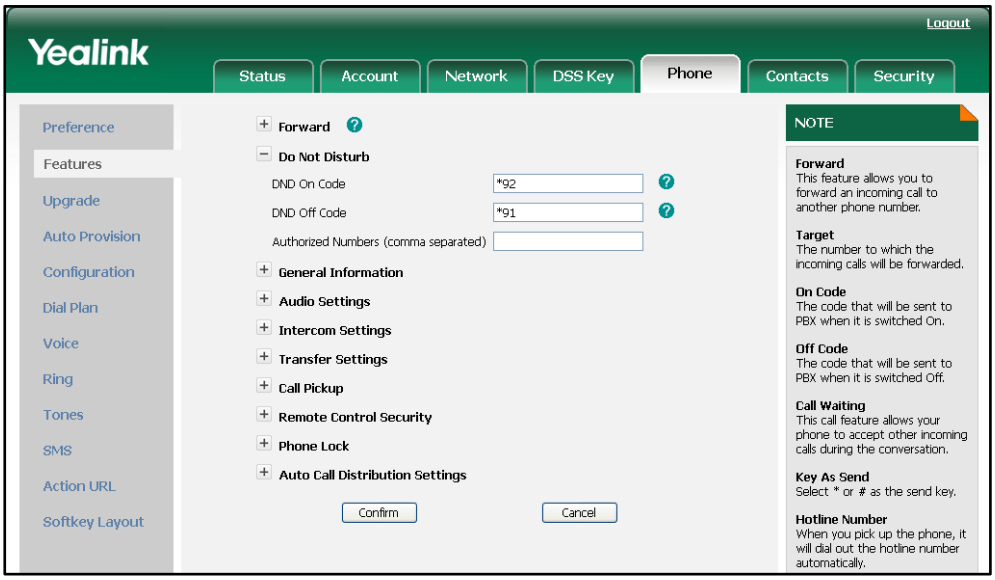


3. Click **Confirm** to accept the change.

To configure the DND on code and DND off code via web user interface:

1. Click on **Phone**->**Features**->**Do Not Disturb**.
2. Enter the DND on code in the **DND On Code** field.

3. Enter the DND off code in the **DND Off Code** field.

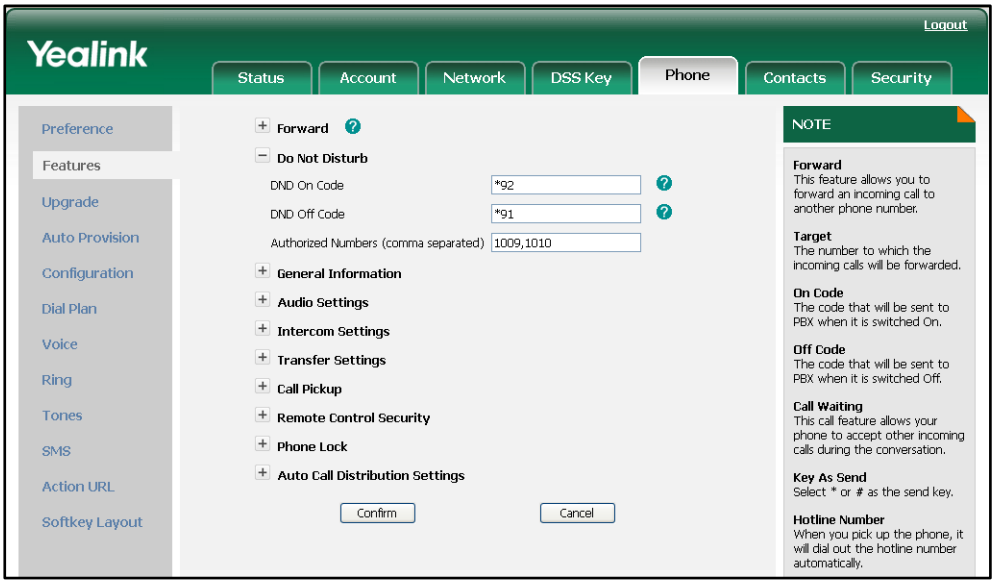


4. Click **Confirm** to accept the change.

To specify the DND authorized numbers via web user interface:

1. Click on **Phone->Features> Do Not Disturb**.
2. Enter the numbers in the **Authorized Numbers (comma separated)** field.

   Multiple numbers are separated by comma.

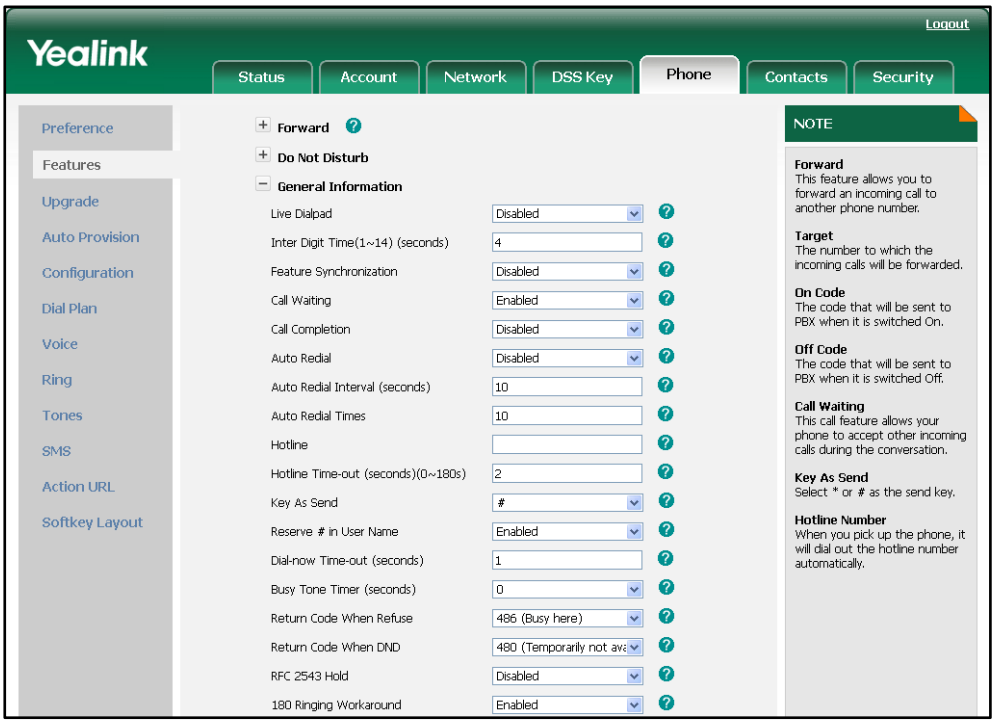


3. Click **Confirm** to accept the change.

   If DND mode is activated on the phone, the phone can still receive the incoming call from the numbers specified in the Authorized Numbers (comma separated) field.

To specify the return code and the reason via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Return Code When DND**.



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

To configure a DND key via phone user interface:

1. Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **DND** from the **Key Type** field.
5. Press the **Save** soft key to accept the change.

Busy Tone Delay

When one party releases the call, a busy tone is audible to the other party indicating that the call connection breaks. Busy tone delay defines a period of time for which the busy tone is audible.

Procedure

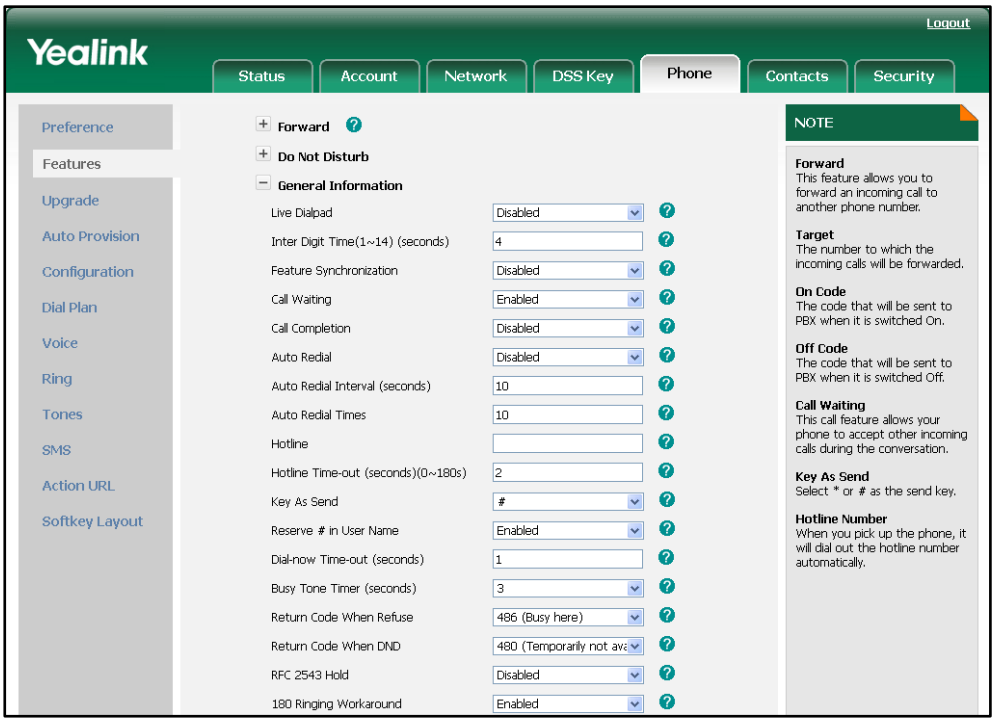
Busy tone delay can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the busy tone delay feature.

		For more information, refer to Busy Tone Delay on page 257.
Local	Web User Interface	Configure the busy tone delay feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure busy tone delay via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Busy Tone Timer (seconds)**.



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

Return Code When Refuse

Return Code When Refuse defines the return code and reason of the SIP response message when refusing an incoming call. The caller's phone LCD screen displays the reason according to the return code received. The following types of return code and reason are available:

- 404 (Not found)
- 480 (Temporarily not available)
- 486 (Busy here)

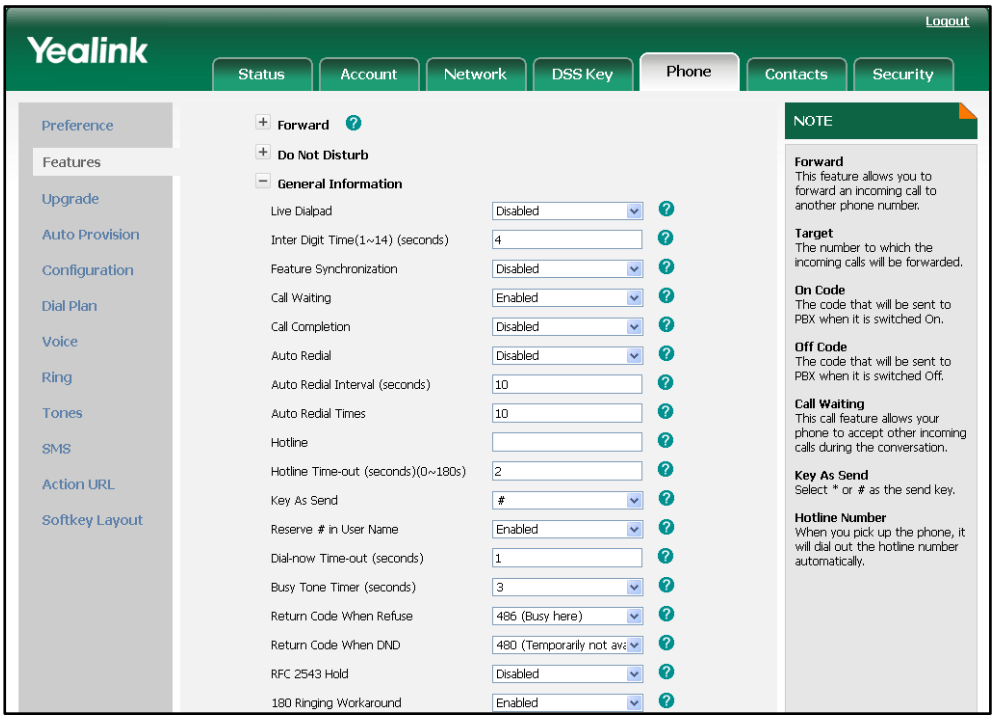
Procedure

Return code when refusing a call can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the return code when refusing a call. For more information, refer to Return Code When Refuse on page 257.
Local	Web User Interface	Configure the return code when refusing a call. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure the return code when refusing a call via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Return Code When Refuse**.



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

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 ringback tone and begins to play the media received. 180 ring workaround allows the IP phone to resume and play the local ringback tone upon a subsequent 180 message received.

Procedure

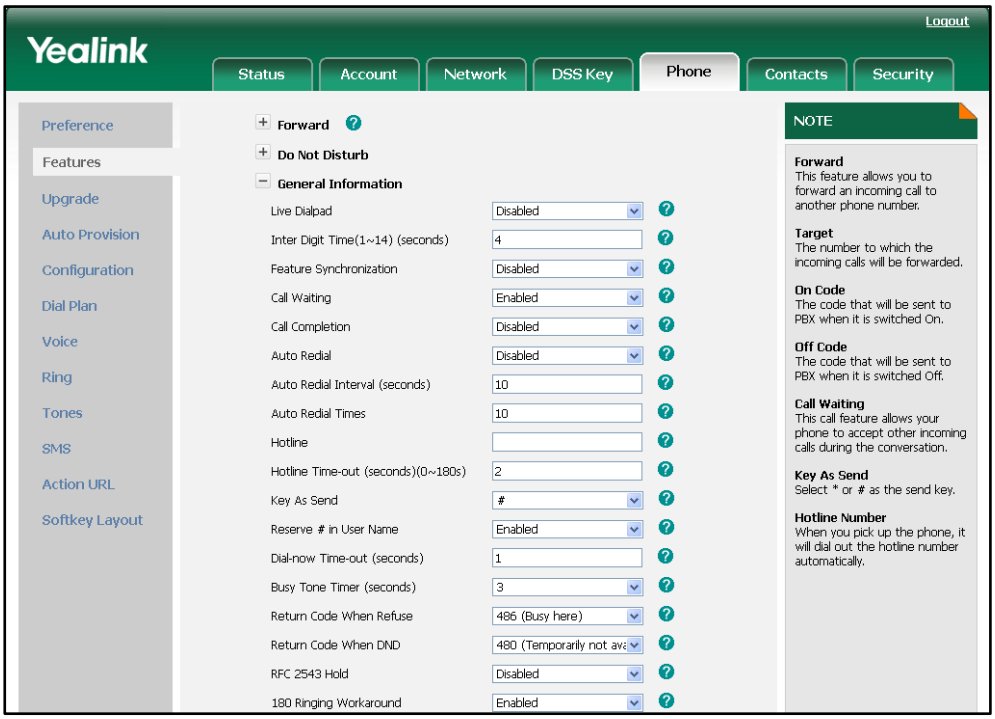
180 ring workaround can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the 180 ring workaround feature. For more information, refer to 180 Ring Workaround on page 258.
Local	Web User Interface	Configure the 180 ring workaround feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure 180 ring workaround via web user interface:

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

2. Select the desired value from the pull-down list of **180 Ring Workaround**.



3. Click **Confirm** to accept 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 the SIP request messages from the IP phone will be forced to send to the outbound proxy server.

Procedure

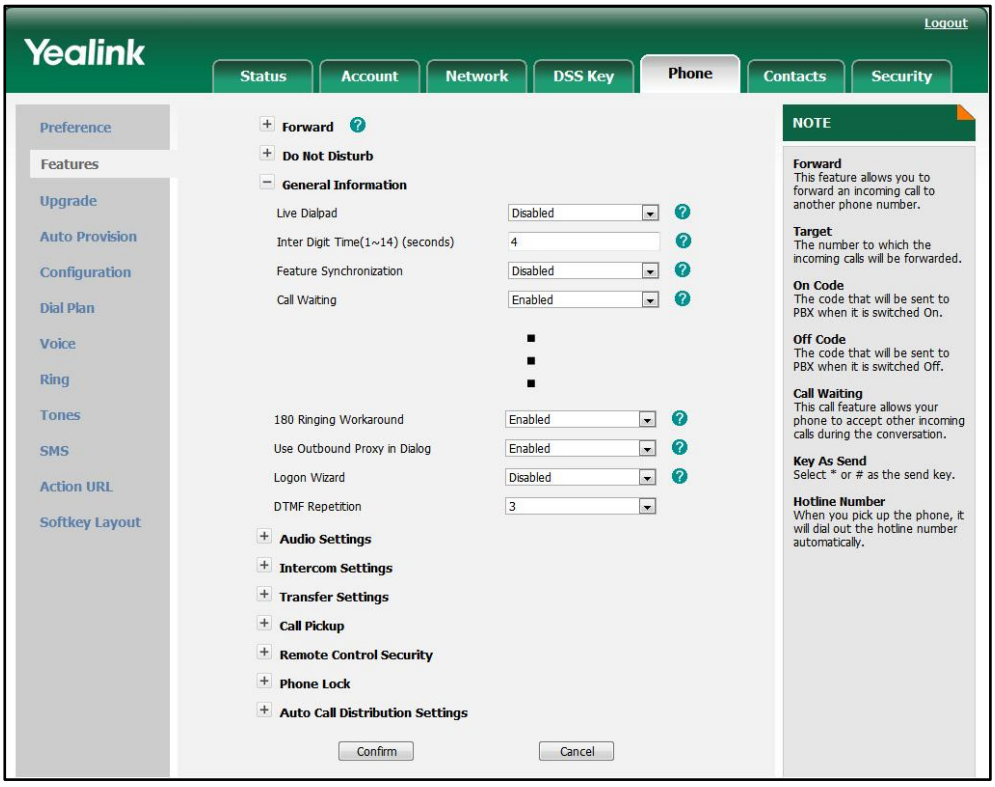
Use outbound proxy in dialog can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify whether to use outbound proxy in a dialog. For more information, refer to Use Outbound Proxy in Dialog on page 258.
Local	Web User Interface	Specify whether to use outbound proxy in a dialog. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Feat

		ures.htm

To specify whether to use outbound proxy server in a dialog via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Use Outbound Proxy in Dialog**.



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

SIP Session Timer

The IP phones support to configure SIP session timers T1, T2 and T4. These timers 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 retransmitting time of any SIP request message. The re-transmitting and doubling of T1 continues until the retransmitting time reaches the T2 value. Timer T4 represents the time the network will take to clear messages between the SIP Client and SIP Server.

Procedure

SIP session timer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the SIP session timer feature. For more information, refer to SIP

		Session Timer on page 259.
Local	Web User Interface	Configure the SIP session timer feature. **Navigate to:** http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

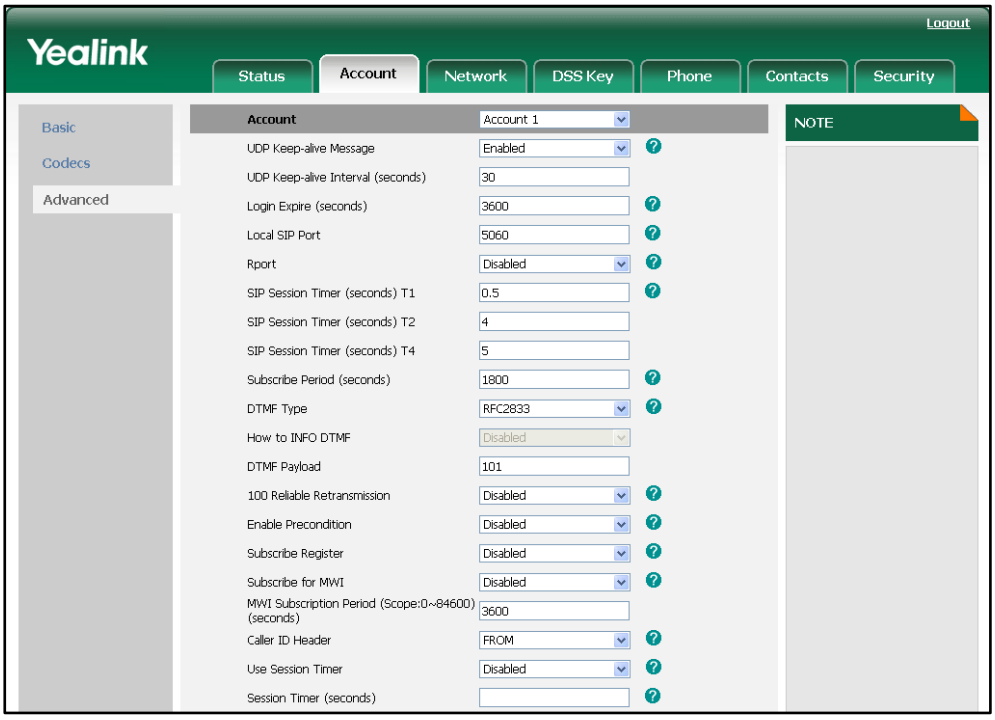
To configure the session timer via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Enter the desired value in the **SIP Session Timer (seconds) T1** field.

   The default value is 0.5s.
5. Enter the desired value in the **SIP Session Timer (seconds) T2** field.

   The default value is 4s.
6. Enter the desired value in the **SIP Session Timer (seconds) T4** Field.

   The default value is 5s.



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

Session Timer

The IP phones support to use session timer to send periodic re-INVITE requests to refresh the session during a call. The session timer is defined in RFC 4082. The 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 message at or before the negotiated session expiration.

Procedure

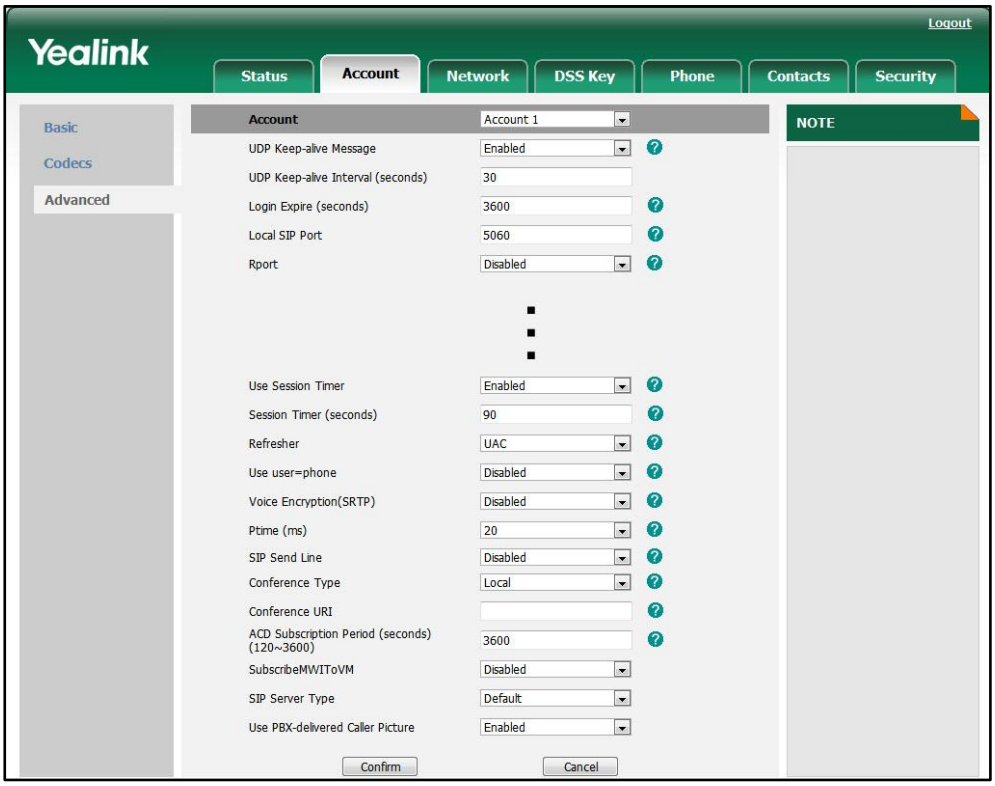
Session timer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the session timer feature. For more information, refer to Session Timer on page 260.
Local	Web User Interface	Configure the session timer feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the session timer via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **Use Session Timer**.
5. Enter the desired time interval in the **Session Timer (seconds)** field.

6. Select the desired refresher from the pull-down list of **Refresher**.



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

Call Hold

Call hold feature provides a service of putting an active call on hold. When a call is placed on hold, the IP phone sends an INVITE request with a HOLD SDP to the server. The IP phones support two call hold methods, one is RFC 3264, it is used to set the "a" media attribute in the SDP to sendonly, recvonly or inactive, for example: a=sendonly. The other is RFC 2543, it is used to set the "c" connection addresses for the media streams in the SDP to zero, for example: c=0.0.0.0.

Procedure

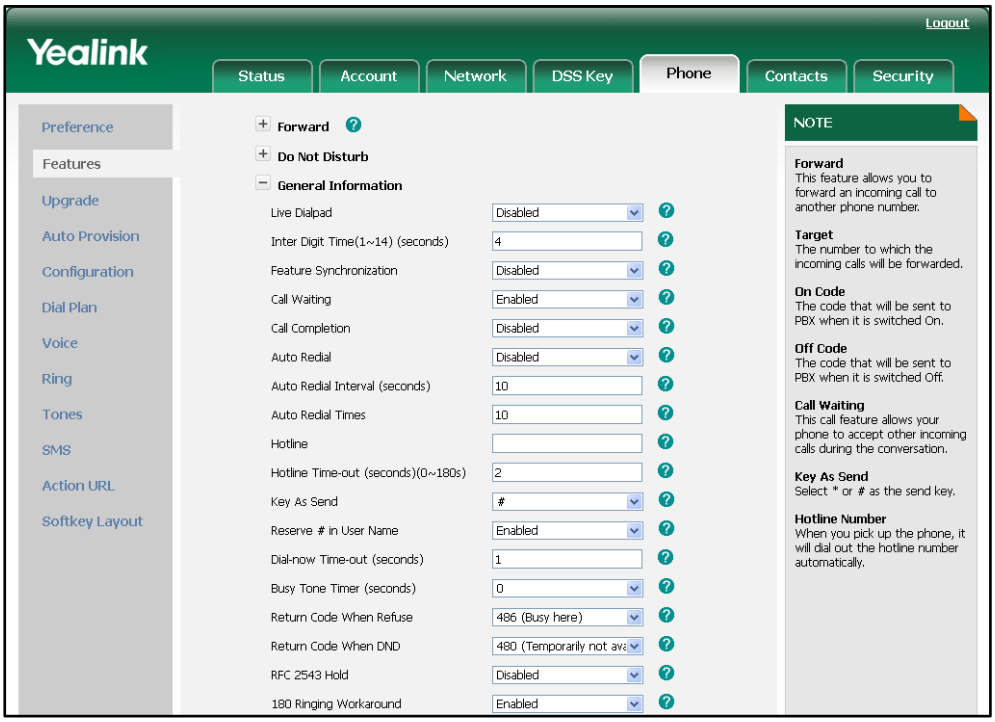
Call hold can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. For more information, refer to Call Hold on page 261.
Local	Web User Interface	Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.

		Navigate to: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure the call hold method via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **RFC 2543 Hold**.



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

Call Forward

Call forward allows redirecting an incoming call to a third party. The IP phones support to redirect an incoming INVITE message by responding with a 302 Moved Temporarily message. This response contains a Contact header with a new URI that should be tried. The IP phones offer three types of forward:

- **Always Forward** -- Forward the incoming calls immediately.
- **Busy Forward** -- Forward the incoming call when the callee is busy.
- **No Answer Forward** -- Forward the incoming call after a period of ring time.

The call forward on code or call forward off code configured on the IP phone is used to inform the server of activating or deactivating the call forward feature. The call forward on code and call forward off code may vary on different servers.

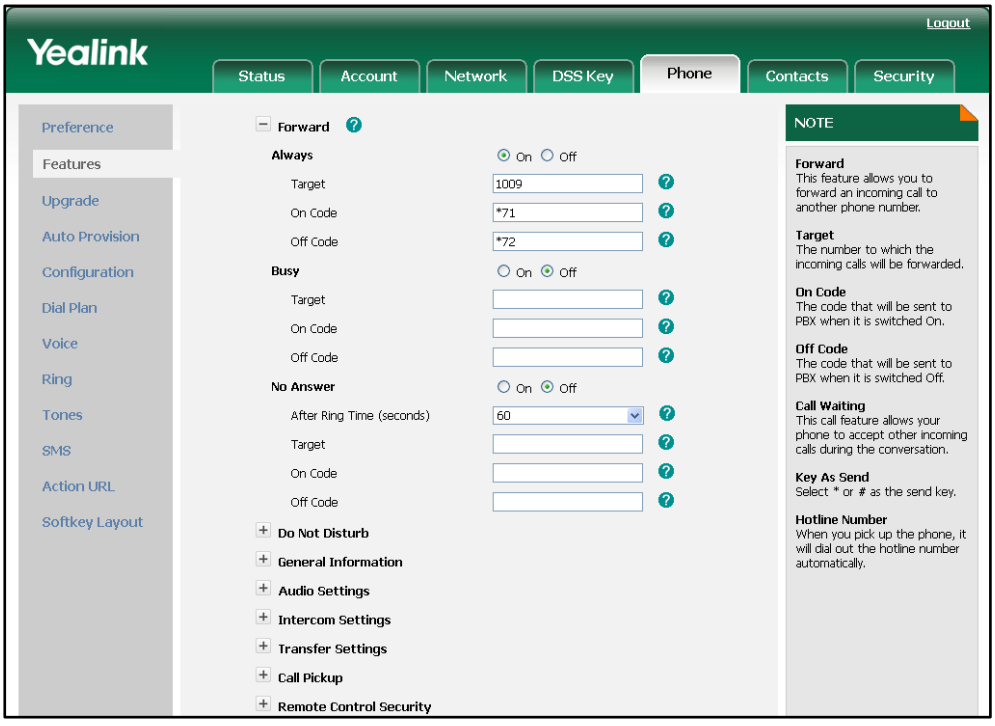
Procedure

Call forward can be configured locally.

Local	Web User Interface	Configure the call forward feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the call forward feature.

To configure always forward via web user interface:

1. Click on **Phone->Features->Forward**.
2. Mark the desired radio box in the **Always** field.
3. Enter the destination number you want to forward in the **Target** field.
4. (Optional.) Enter the on code or off code in **On Code** or **Off Code** field.

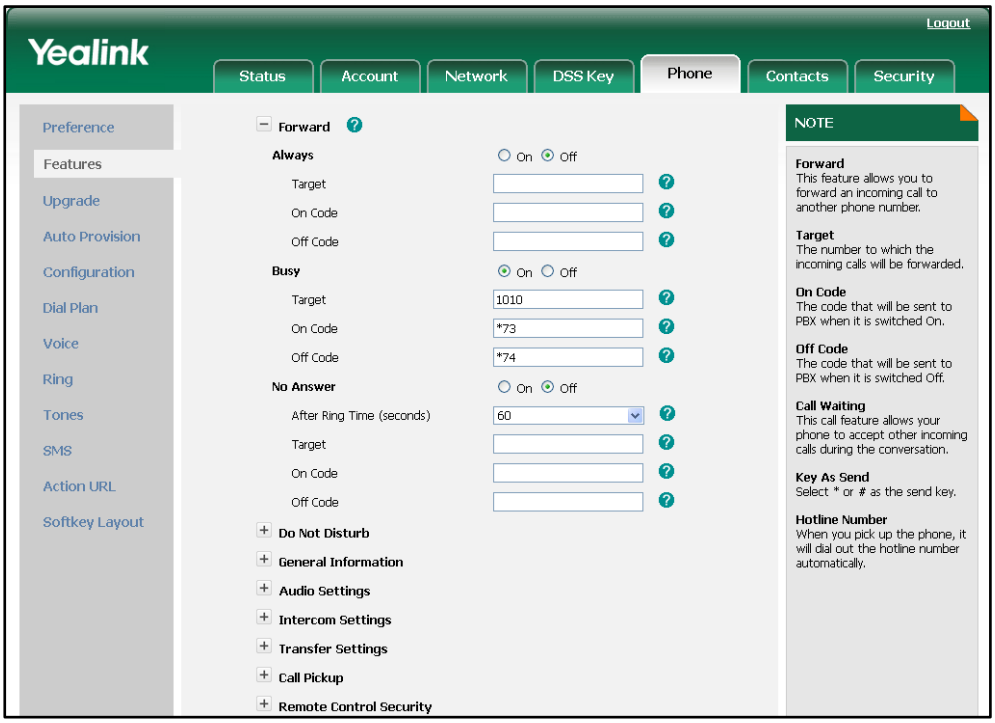


5. Click **Confirm** to accept the change.

To configure busy forward via web user interface:

1. Click on **Phone->Features->Forward**.
2. Mark the desired radio box in the **Busy** field.
3. Enter the destination number you want to forward in the **Target** field.

4. (Optional.) Enter the on code or off code in **On Code** or **Off Code** field.

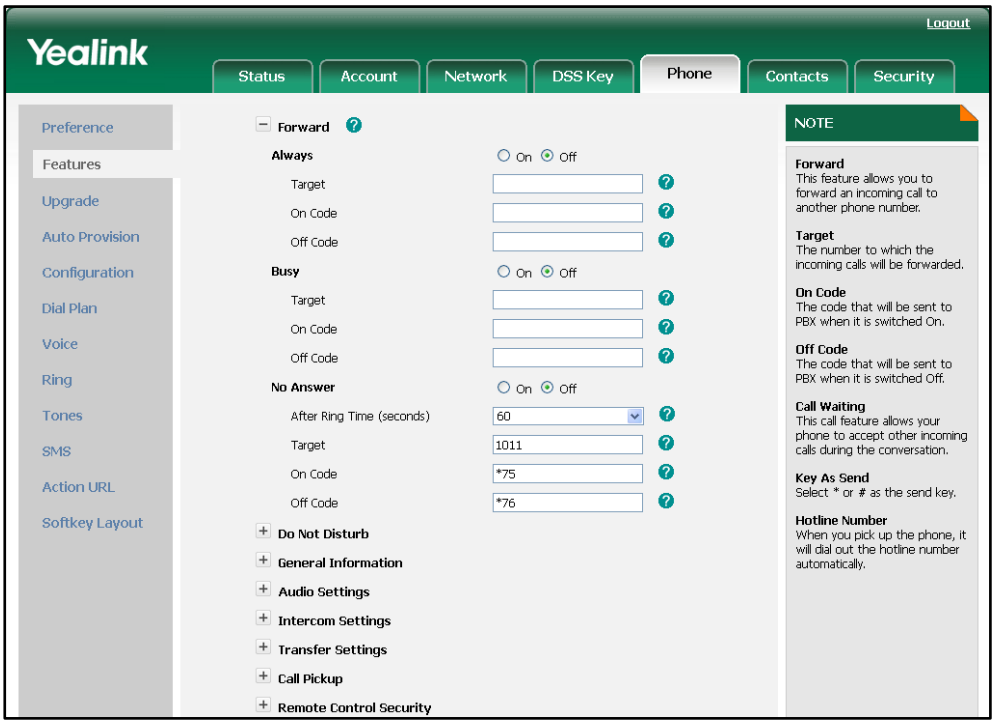


5. Click **Confirm** to accept the change.

To configure no answer forward via web user interface:

1. Click on **Phone->Features->Forward**.
2. Mark the desired radio box in the **No Answer** field.
3. Select the ring time to wait before forwarding from the pull-down list of **After Ring Time (seconds)**.
4. Enter the destination number you want to forward in the **Target** field.

5. (Optional.) Enter the on code or off code in **On Code** or **Off Code** field.



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

To configure call forward via phone user interface:

1. Press **Menu->Features->Call Forward**.
2. Press ( ) or ( ) to select the desired forwarding type, and then press the **Enter** soft key.

   a) If you select **Always Forward**:

      1) Press ( ) or ( ), or the **Switch** soft key to select **Enable** from the **Always** field.

      2) Enter the destination number you want to forward all incoming calls to in the **Forward to** field.

      3) (Optional.) Enter the always forward on code or off code respectively in the **On Code** or **Off Code** field.

   b) If you select **Busy Forward:**

      1) Press ( ) or ( ), or the **Switch** soft key to select **Enable** from the **Busy** field.

      2) Enter the destination number you want to forward all incoming calls to when the phone is busy in the **Forward to** field.

      3) (Optional.) Enter the busy forward on code or off code respectively in the **On Code** or **Off Code** field.

   c) If you select **No Answer Forward**:

      1) Press ( ) or ( ), or the **Switch** soft key to select **Enable** from the **No Answer** field.

2) Enter the destination number you want to forward all unanswered incoming calls to in the **Forward to** field.

3) Press (◂) or (▸), or the **Switch** soft key to select the ring time to wait before forwarding from the **After Ring Times** field.

The default ring time is 60 seconds.

4) (Optional.) Enter the no answer forward on code or off code respectively in the **On Code** or **Off Code** field.

3. Press the **Save** soft key to accept the change.

Call Transfer

Call transfer enables the IP phone to transfer an existing call to another party. The IP phones support call transfer using the REFER method specified in RFC 3515. The IP phones 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. The 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 the semi-attended transfer, the semi-attended transfer feature determines whether to display the prompt "1 New Missed Call(s)" on the destination party's phone LCD screen.

Procedure

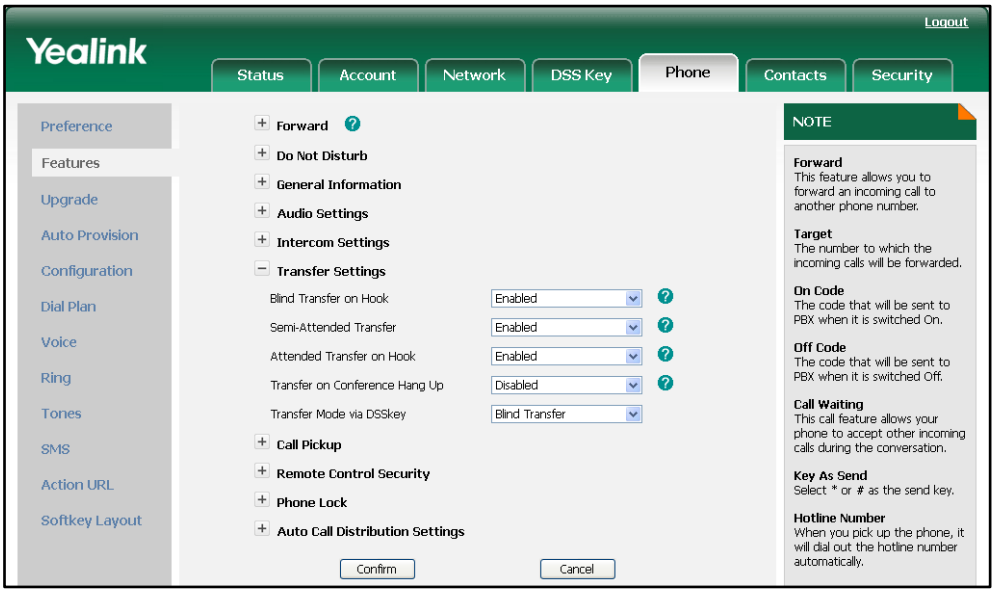
Call transfer can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify whether to complete the transfer through on-hook. Configure the semi-attended transfer feature. For more information, refer to Call Transfer on page 262.
Local	Web User Interface	Specify whether to complete the transfer through on-hook.

		Configure the semi-attended transfer feature. **Navigate to:** http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure call transfer via web user interface:

1. Click on **Phone->Features->Transfer Settings**.
2. Select the desired values from the pull-down lists of **Semi-Attended Transfer**, **Blind Transfer on Hook** and **Attended Transfer on Hook**.



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

Network Conference

Network conference, also known as centralized conference, provides users with flexibility of call with multiple participants (more than three). Once a network conference is commenced, the media server holds the conference, therefore, even if the initiator drops the call or puts the call on hold, the conference will continue with the remaining participants. The IP phones implement network conference using the REFER method specified in RFC 4579. This feature depends on support from a SIP server.

Procedure

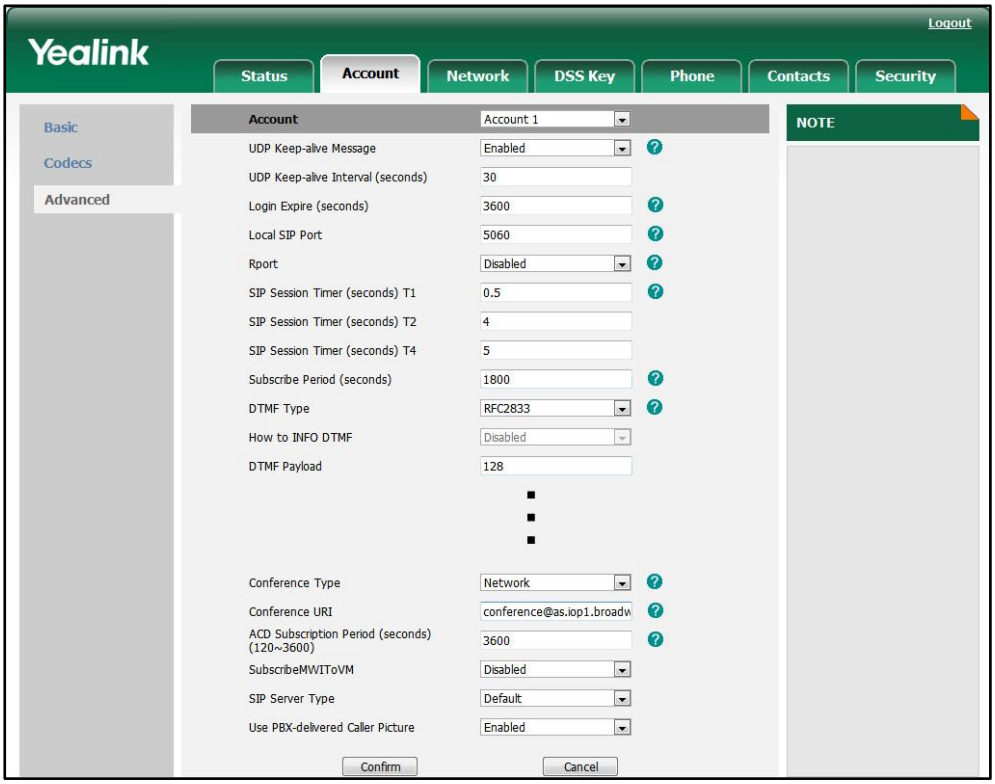
Network conference can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the network conference.

		For more information, refer to Network Conference on page 263.
Local	Web User Interface	Configure the network conference. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the network conference via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select **Network** from the pull-down list of **Conference Type**.
5. Enter the conference URI in the **Conference URI** field.



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

Transfer on Conference Hang Up

For local conference, all parties release the call when the conference initiator drops the conference call. The transfer on conference hang up feature allows the other two parties remain connected when the conference initiator drops the conference call.

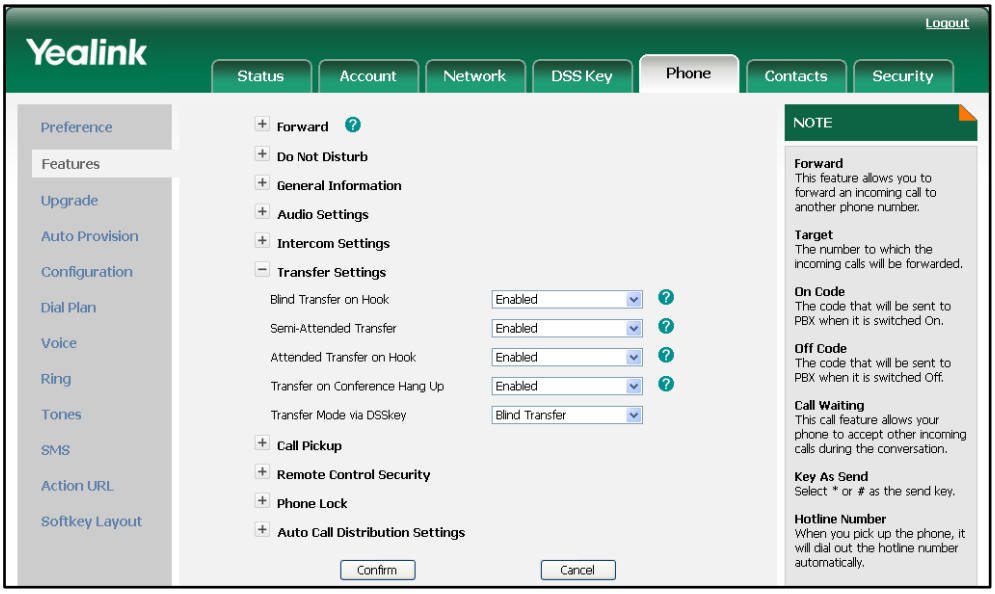
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the transfer on conference hang up feature. For more information, refer to Transfer on Conference Hang Up on page 264.
Local	Web User Interface	Configure the transfer on conference hang up feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure Transfer on Conference Hang up via web user interface:

1. Click on **Phone->Features->Transfer Settings**.
2. Select the desired value from the pull-down list of **Transfer on Conference Hang Up**.



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

Direct Pickup

Direct pickup is used for picking up an incoming call on a specific extension. A user can pick up the incoming call using a direct pickup key or DPickup soft key. This feature depends on support from a SIP server. For many SIP servers, direct pickup is implemented requiring a direct pickup code. The direct pickup code can be configured on a phone or per-account basis.

Note We recommend that you should not configure two types of the keys introduced above simultaneously. If you do, the direct pickup key will not be used correctly.

Procedure

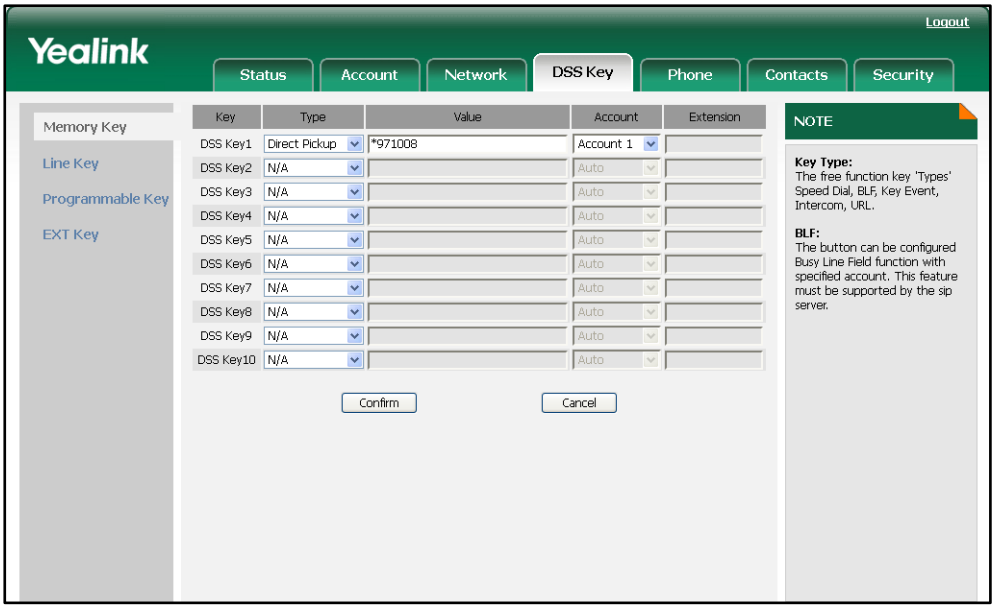
Direct pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the direct pickup code on a per-account basis. For more information, refer to Direct Pickup on page 265.
	<y0000000000xx>.cfg	Assign a direct pickup key. For more information, refer to Direct Pickup Key on page 323. Configure the direct pickup feature on a phone basis. For more information, refer to Direct Pickup on page 264.
Local	Web User Interface	Assign a direct pickup key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm Configure the direct call pickup feature on a phone basis. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm Configure the direct pickup code on a per-account basis. **Navigate to**:

		http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.
	Phone User Interface	Assign a direct pickup key.

To configure a direct pickup key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Direct Pickup** from the pull-down list of **Type**.
3. Enter the direct call pickup code followed by the specific extension in the **Value** field.
4. Select the desired line from the pull-down list of **Account**.

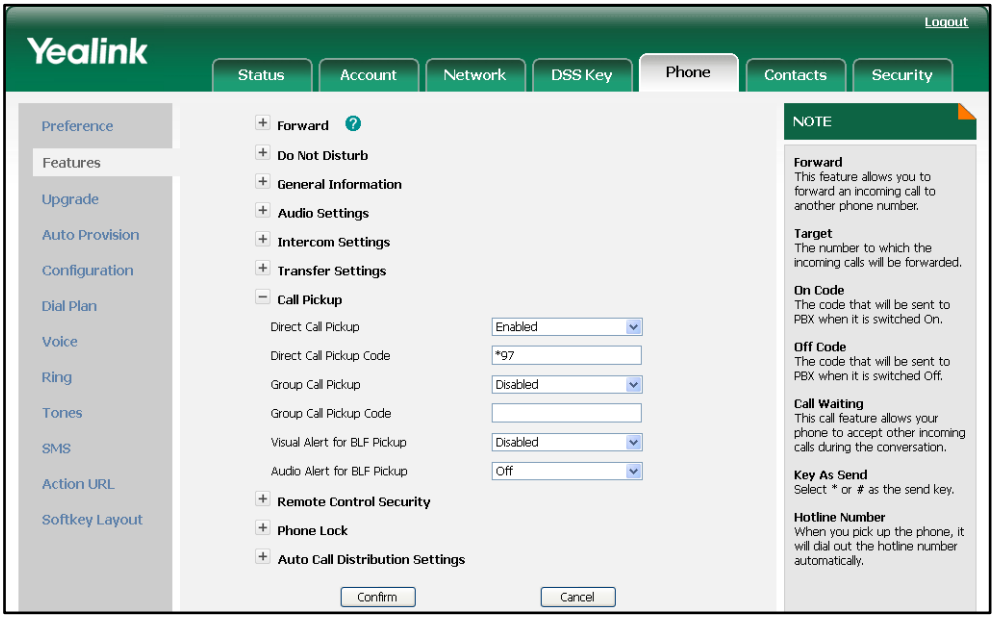


5. Click **Confirm** to accept the change.

To configure the direct call pickup feature on a phone basis via web user interface:

1. Click on **Phone**->**Features**->**Call Pickup**.
2. Select the desired value from the pull-down list of **Direct Call Pickup**.

3. Enter the direct call pick up code in the **Direct Call Pickup Code** field.

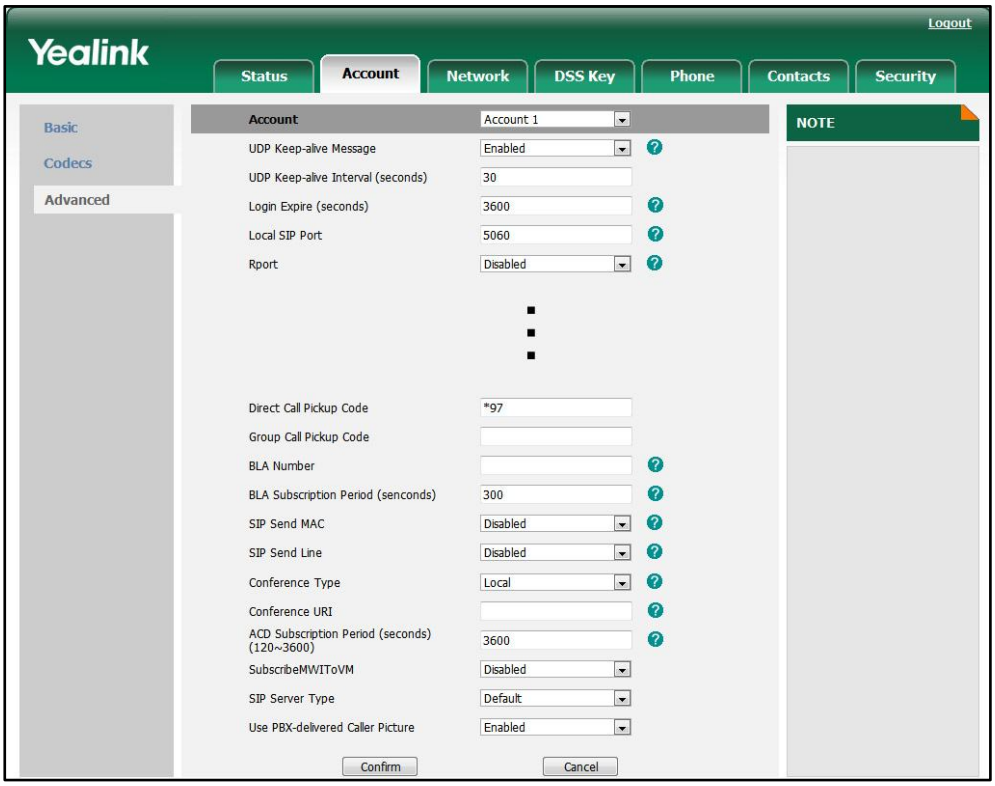


4. Click **Confirm** to accept the change.

To configure the direct call pickup code on a per-account basis via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.

4. Enter the direct pickup feature code in the **Direct Call Pickup Code** field.



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

To configure the direct pickup key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **Direct Pickup** from the **Key Type** field.
5. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
6. Enter the direct call pickup code followed by the specific extension in the **Value** field.
7. Press the **Save** soft key to accept the change.

Group Pickup

Group pickup is used for picking up incoming calls within a pre-defined group. If there are many incoming calls at the same time, the user will pick up the call that rang first. The user can pick up the incoming call using a group pickup key or GPickup soft key. This feature depends on support from a SIP server. For many SIP servers, group pickup is implemented requiring a group pickup code. The group pickup code can be configured on a phone or per-account basis.

Procedure

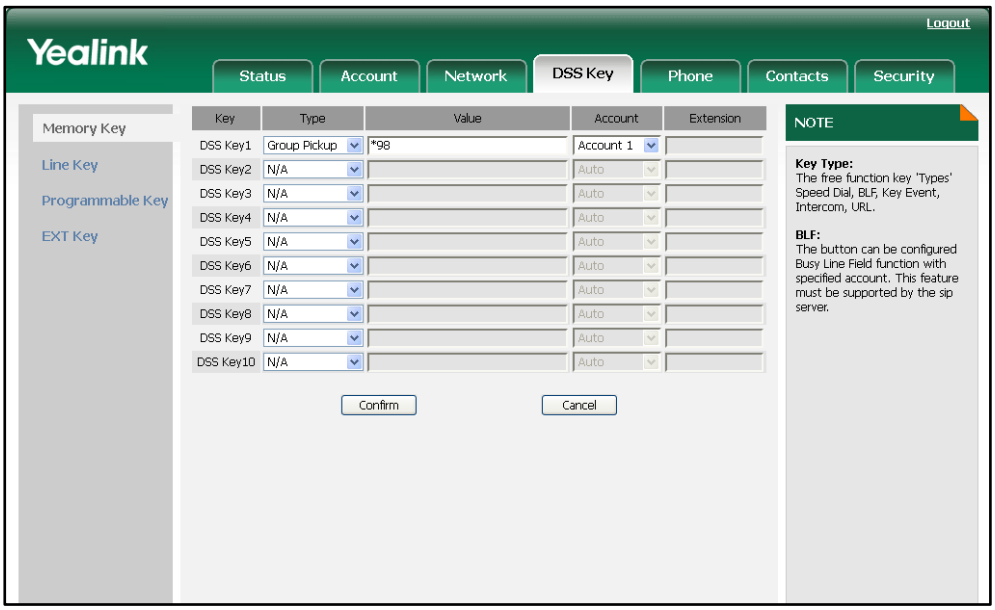
Group pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the group pickup code on a per-account basis. For more information, refer to Group Pickup on page 266.
	<y0000000000xx>.cfg	Assign a group pickup key. For more information, refer to Group Pickup Key on page 324. Configure the group pickup feature on a phone basis. For more information, refer to Group Pickup on page 265.
Local	Web User Interface	Assign a group pickup key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm Configure the group pickup feature on a phone basis. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm Configure the group pickup code on a per-account basis. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.
	Phone User Interface	Assign a group pickup key.

To configure a group pickup key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Group Pickup** from the pull-down list of **Type**.
3. Enter the group call pickup code in the **Value** field.

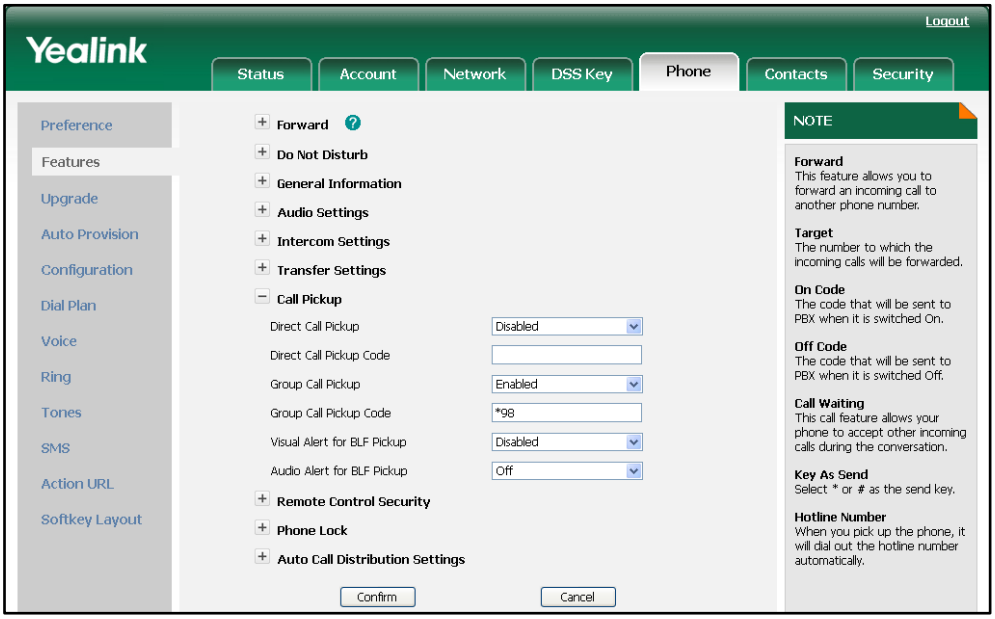
4. Select the desired line from the pull-down list of **Account**.



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

To configure the group call pickup feature on a phone basis via web user interface:

1. Click on **Phone->Features->Call Pickup**.
2. Select the desired value from the pull-down list of **Group Call Pickup**.
3. Enter the group call pickup code in the **Group Call Pickup Code** field.

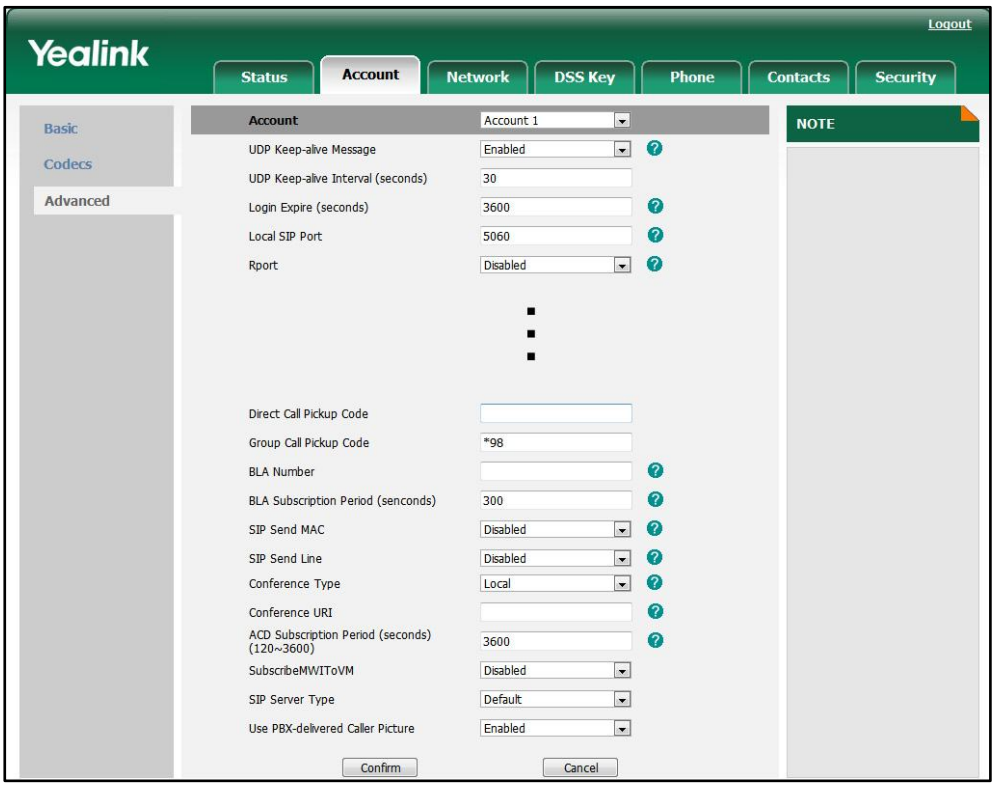


4. Click **Confirm** to accept the change.

To configure the group call pickup code on a per-line basis via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Click on **Advanced**.
4. Enter the group call pickup code in the **Group Call Pickup Code** field.



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

To configure a group pickup key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **Group Pickup** from the **Key Type** field.
5. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
6. Enter the group call pickup code in the **Value** field.
7. Press the **Save** soft key to accept the change.

Dialog-Info Call Pickup

On some specific servers, call pickup is implemented through SIP signals. The IP phones support to pick up incoming calls via a NOTIFY message with dialog-info event. A user can pick up an incoming call by pressing a DSS key used to monitor a specific extension (such as a BLF key).

The example of the dialog-info message carried in NOTIFY message for reference:

```
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="6" state="full"
entity="sip:1013@10.2.1.199">
<dialog id="706655206@10.2.8.213" call-id="706655206@10.2.8.213" local-tag="827932784"
remote-tag="1887460740" direction="recipient">
<state>early</state>
<local>
<identity>sip:1013@10.2.1.199</identity>
<target uri="sip:1013@10.2.1.199">
</target>
</local>
<remote>
<identity>sip:1011@10.2.1.199</identity>
<target uri="sip:1011@10.2.8.213:5063">
</target>
</remote>
</dialog>
</dialog-info>
```

Procedure

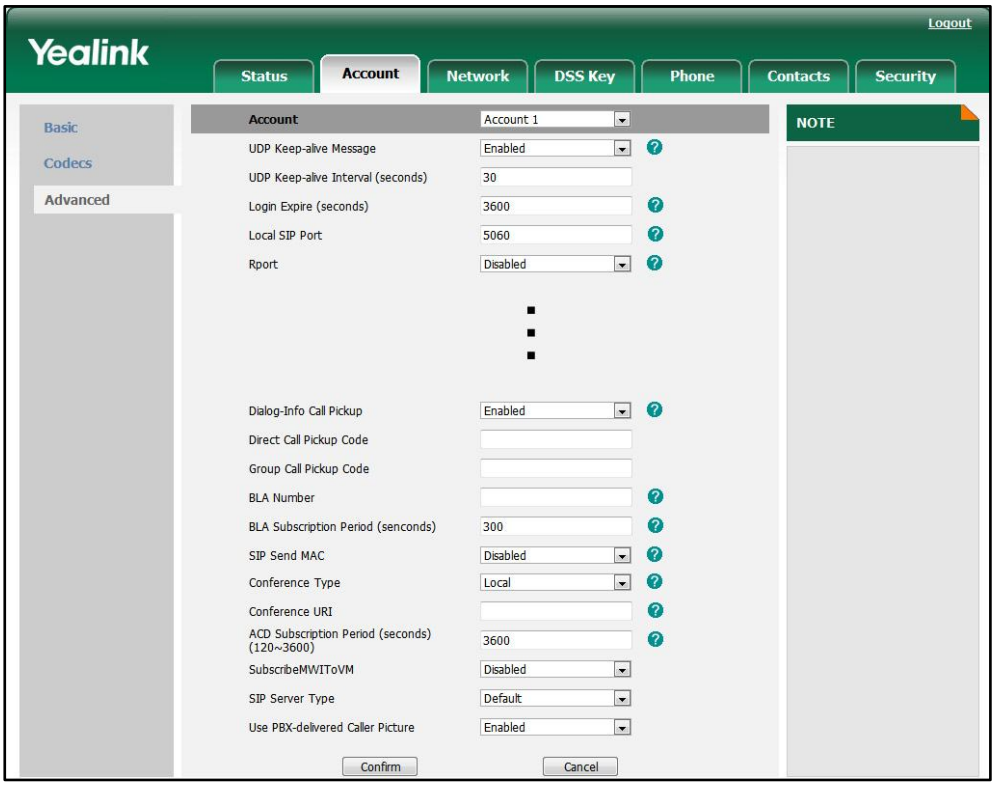
Dialog-Info Call Pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the Dialog-Info Call Pickup feature on the IP phone. For more information, refer to Dialog-Info Call Pickup on page 267.
Local	Web User Interface	Configure the Dialog-Info Call Pickup feature on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure Dialog-Info Call Pickup via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.

4. Select the desired value from the pull-list of **Dialog-Info Call Pickup**.



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

Call Return

Call return, also known as last call return, provides convenience for a user to place a call back to the caller of the last incoming call. The IP phones implement call return using a call return key.

Procedure

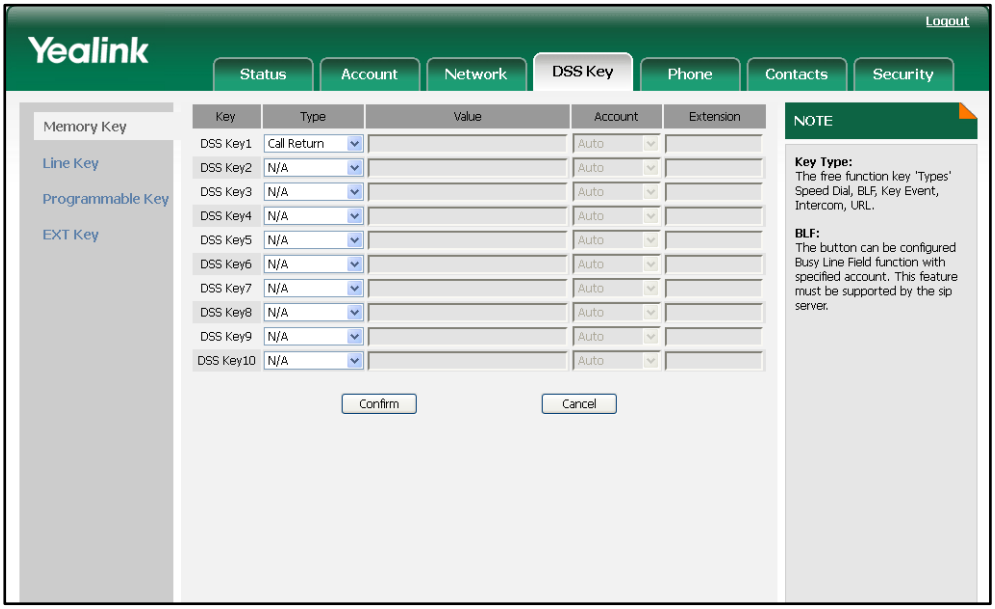
Call return key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a call return key. For more information, refer to Call Return Key on page 325.
Local	Web User Interface	Assign a call return key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign a call return key.

To configure a call return key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).

2. In the desired DSS key field, select **Call Return** from the pull-down list of **Type**.



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

To configure a call return key via phone user interface:

1. Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ► , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ► , or the **Switch** soft key to select **Call Return** from the **Key Type** field.
5. Press the **Save** soft key to accept the change.

Call Park

Call park allows a user to park a call at a special extension and then retrieve it on any other phone in the system. The user can park a call at an extension, known as call park orbit, by pressing a call park key. The current call is put on hold and can be retrieved on another IP phone. This feature depends on support from a SIP server.

Procedure

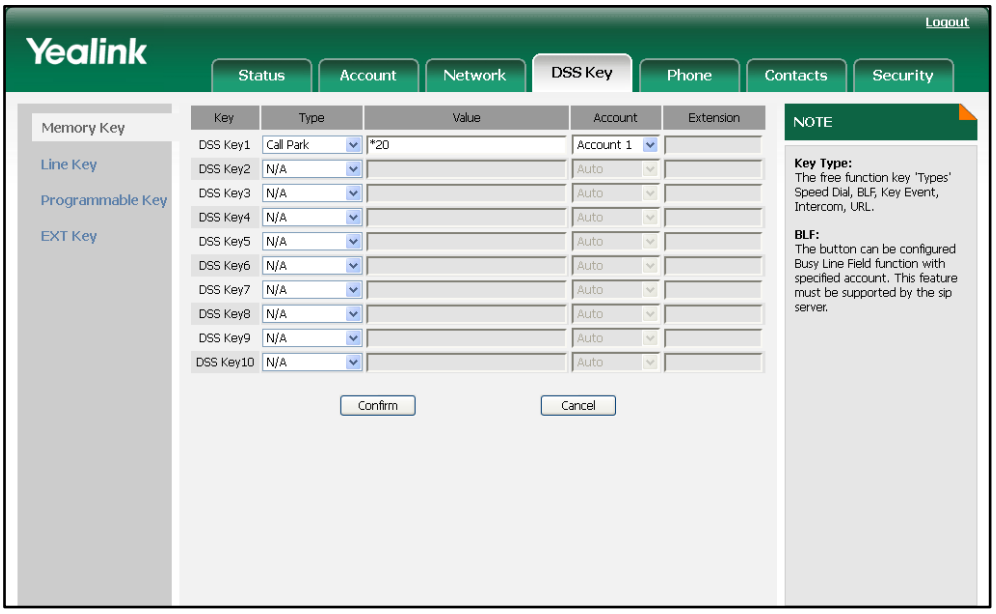
Call park key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a call park key. For more information, refer to Call Park Key on page 325.
Local	Web User Interface	Assign a call park key. **Navigate to**:

		http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign a call park key.

To configure a call park key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Call Park** from the pull-down list of **Type**.
3. Enter the desired value (e.g., call park feature code) in the **Value** field.
4. Select the desired line from the pull-down list of **Account**.



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

To configure a call park key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **Call Park** from the **Key Type** field.
5. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
6. Enter the desired value (e.g., call park feature code) in the **Value** field.
7. Press the **Save** soft key to accept the change.

Web Server Type

The web server type feature determines access permission of the IP phone's web user

interface. The 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 the pages returned by the web server. Both the HTTP and HTTPS port numbers are configurable.

Procedure

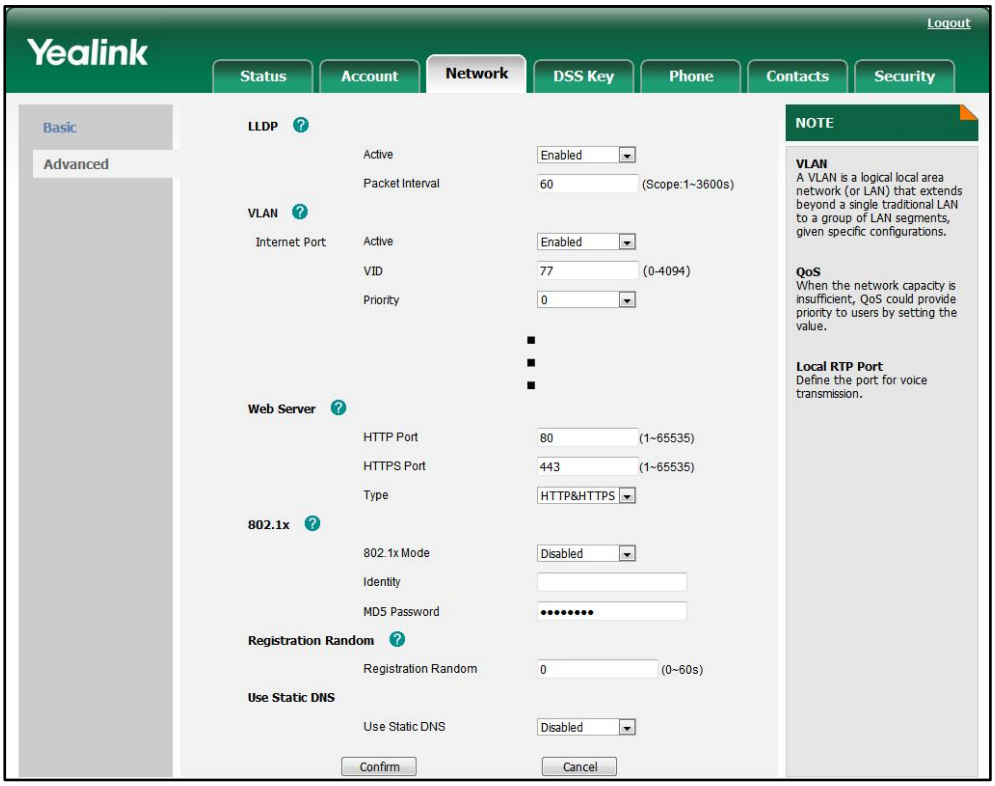
Web server type can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the web access type, HTTP port and HTTPS port. For more information, refer to Web Server Type on page 267.
Local	Web User Interface	Specify the web access type, HTTP port and HTTPS port. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm
	Phone User Interface	Specify the web access type.

To configure the web server type via web user interface:

1. Click on **Network**->**Advanced**.
2. In the **Web Server** field, enter the HTTP port in the **HTTP Port** field.
   The default HTTP port is 80.
3. Enter the HTTPS port in the **HTTPS Port** field.
   The default HTTPS port is 443.

4. Select the desired type from the pull-down list of **Type**.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

To configure the web server type via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**Webserver Type**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired type from the **Webserver Type** field.
3. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Calling Line Identification Presentation

Calling Line Identification Presentation (CLIP) allows the IP phone to display the caller's identity, derived from a SIP header contained in the INVITE message, when receiving an incoming call. The IP phones support three types of SIP headers: 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 name assigned to the caller

should be preferentially displayed.

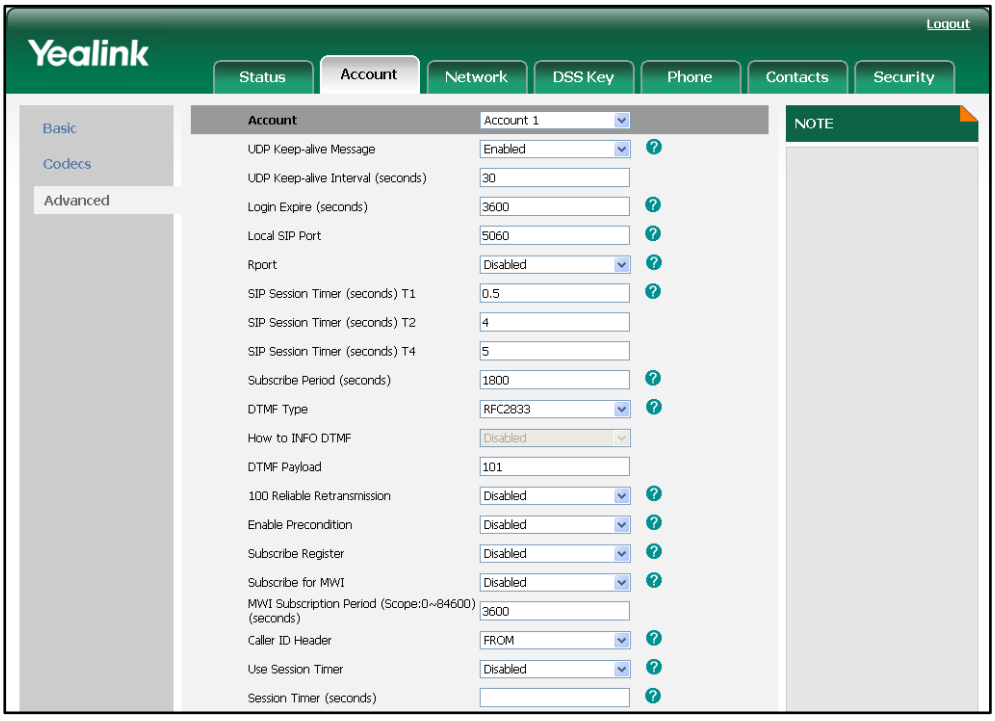
Procedure

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 to Calling Line Identification Presentation on page 269.
Local	Web User Interface	Configure the presentation of the caller identity. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the presentation of the caller identity via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of the **Caller ID Header**.



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

Connected Line Identification Presentation

Connected Line Identification Presentation (COLP) allows the IP phone to display the identity of the callee specified for outgoing calls. The IP phone 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.

If the callee has existed in the directory, the local 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 identity. For more information, refer to Connected Line Identification Presentation on page 269.

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

There are 3 common methods of transmitting DTMF digits on SIP calls:

- **RFC 2833** – DTMF digits are transmitted by RTP Events compliant to RFC 2833.
- **INBAND** – DTMF digits are transmitted in the voice band.
- **SIP INFO** – DTMF digits are transmitted by the SIP INFO messages.

The method of transmitting DTMF digits is configurable on a per-account basis.

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 the RTP Event packets is configurable. The IP phone often defaults to 101 for the payload type, which uses your 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 number of times for the IP phone to send the RTP Event packet with End bit set to 1. The number of times is 3 by default.

INBAND

DTMF digits are transmitted within the audio of the IP phone conversation. It uses the same VoIP codec as your voice and is audible to the 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 configured using the configuration files or locally.

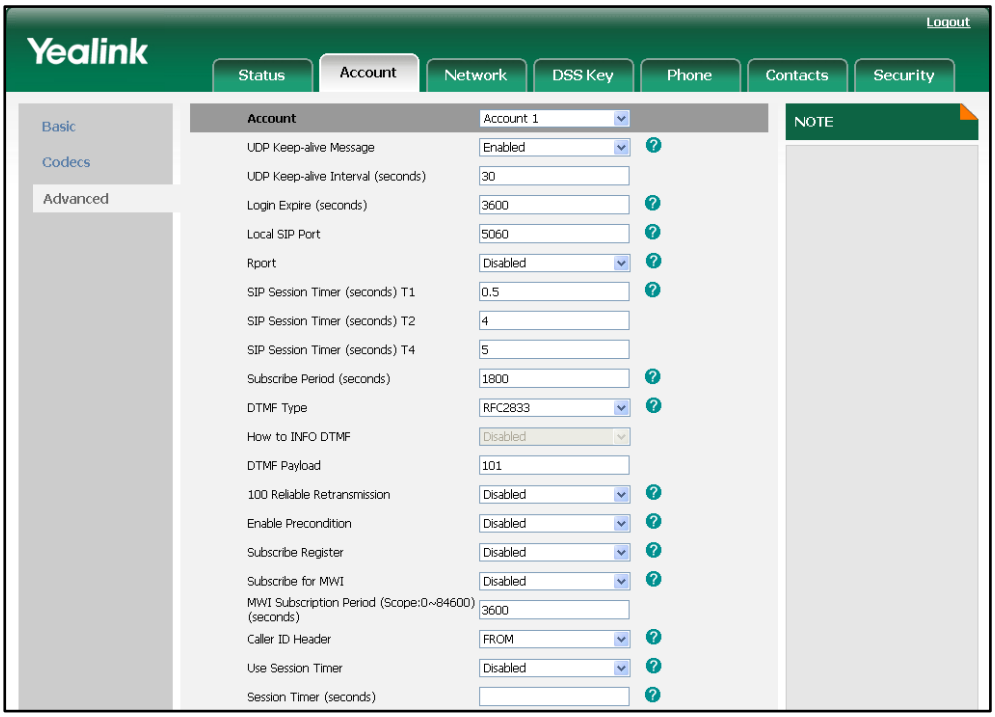
Configuration File	<MAC>.cfg	Configure the method of transmitting DTMF digit and the payload type. For more information, refer to DTMF on page 270.
	<y0000000000xx>.cfg	Configure the number of times for the IP phone to send the end RTP Event packet. For more information, refer to

		DTMF on page 270.
Local	Web User Interface	Configure the method of transmitting DTMF digits and the payload type. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Configure the number of times for the IP phone to send the end RTP Event packet. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

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

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **DTMF Type**.
5. (If SIP INFO or AUTO+SIP INFO is selected.) Select the desired value from the pull-down list of **How to INFO DTMF**.

6. Enter the desired value in the **DTMF Payload** field.

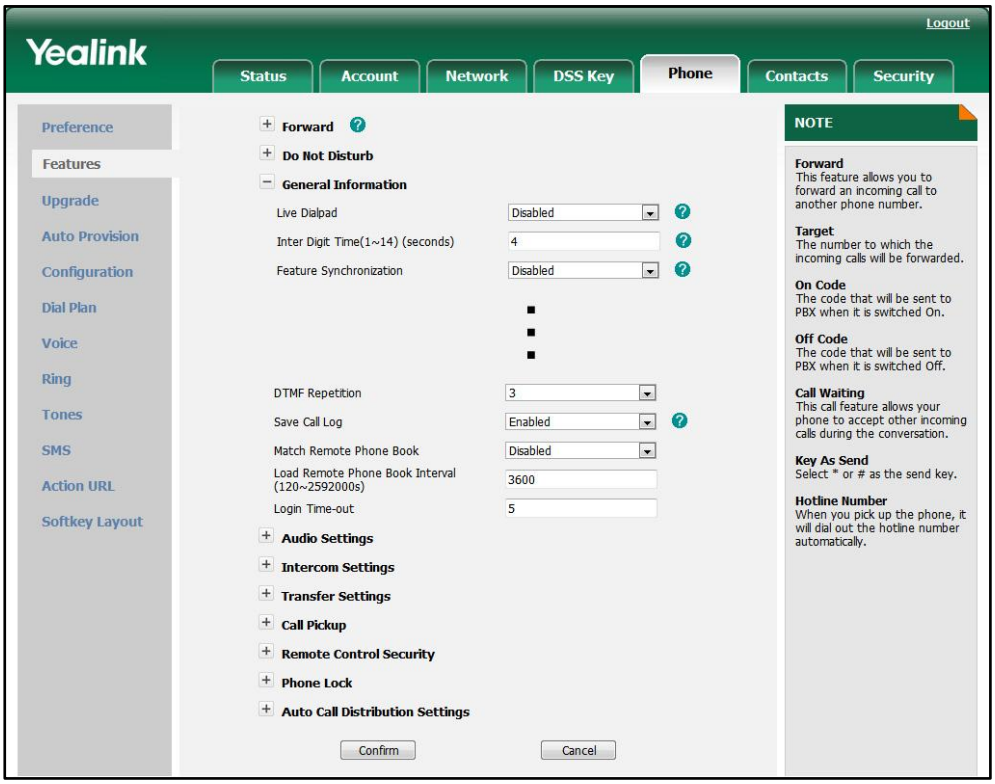


7. Click **Confirm** to accept the change.

To configure the number of times to send the end RTP Event packet via web user interface:

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

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



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

Intercom

Intercom allows establishing a two-way audio conversation directly. The called phone picks up intercom calls automatically and establishes intercom conversations. This feature depends on support from a SIP server.

Outgoing Intercom Calls

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

Procedure

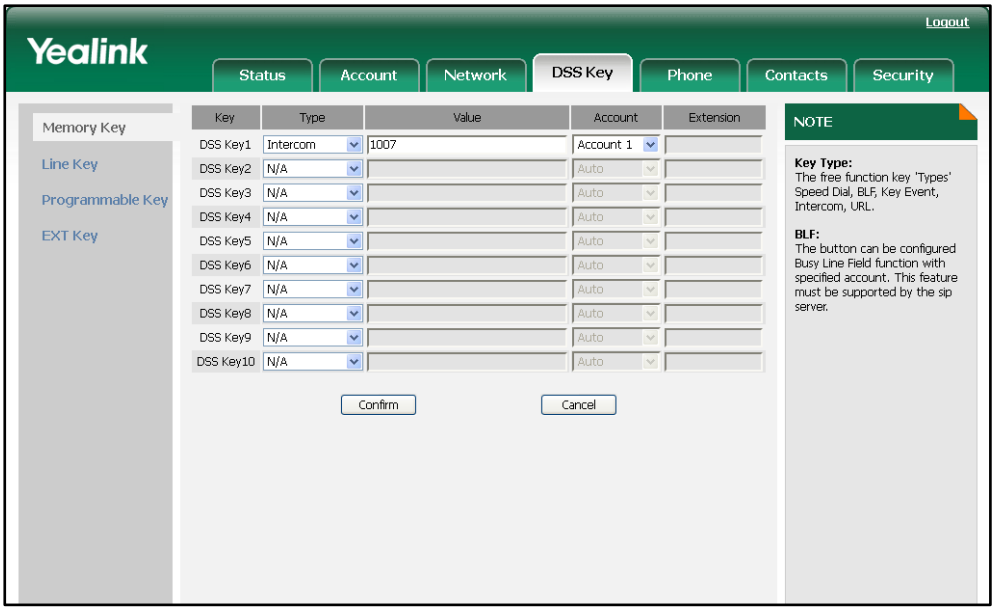
Intercom key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign an intercom key. For more information, refer to Intercom Key on page 326.
Local	Web User Interface	Assign an intercom key. **Navigate to**:

		http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign an intercom key.

To configure an intercom key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Intercom** from the pull-down list of **Type**.
3. Enter the remote extension number in the **Value** field.
4. Select the desired line from the pull-down list of **Account**.



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

To configure an intercom key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Intercom** from the **Type** field.
4. Select the desired line from the **Account ID** field.
5. Enter the remote extension number in the **Value** field.
6. Press the **Save** soft key to accept the change.

Incoming Intercom Calls

The way in which the IP phone handles the incoming intercom calls depends on the incoming intercom call configurations. The following describes each configuration parameter for incoming intercom calls.

Accept Intercom

Accept Intercom allows the IP phone to automatically answer an incoming intercom call.

Intercom Mute

Intercom Mute allows the IP phone to mute the microphone for incoming intercom calls.

Warning Tone

Warning Tone allows the IP phone to play a warning tone before answering an intercom call.

Intercom Barge

Intercom Barge allows the IP phone to automatically answer an incoming intercom call while there is already an active call on the IP phone. The active call is put on hold.

Procedure

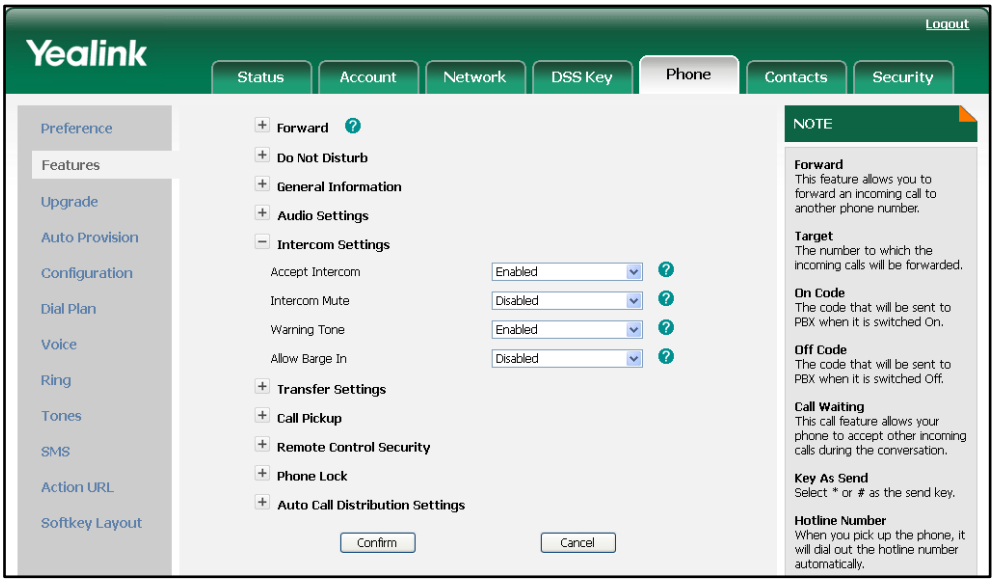
Incoming intercom calls can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the incoming intercom call feature. For more information, refer to Incoming Intercom calls on page 272.
Local	Web User Interface	Configure the incoming intercom call feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Configure the incoming intercom call feature.

To configure intercom via web user interface:

1. Click on **Phone**->**Features**->**Intercom Settings**.

2. Select the desired values from the pull-down lists of **Accept Intercom**, **Intercom Mute**, **Warning Tone** and **Intercom Barge**.



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

To configure intercom via phone user interface:

1. Press **Menu->Features->Intercom**.
2. Press ◄ or ► , or the **Switch** soft key to select the desired values from the **Accept Intercom**, **Intercom Mute**, **Warning Tone** and **Intercom Barge** fields.
3. Press the **Save** soft key to accept the change.

Configuring Advanced Features

This chapter provides information for making configuration changes for the following advanced features:

- Distinctive Ring Tones
- Tones
- Remote Phonebook
- LDAP
- Busy Lamp Field
- BLF List
- Shared Call Appearance
- As-Feature-Event
- Automatic Call Distribution
- Message Waiting Indicator
- Call Recording
- Hot Desking
- Action URL
- Action URI
- Server Redundancy
- LLDP
- VLAN
- VPN
- Quality of Service
- Network Address Translation
- 802.1X Authentication

Distinctive Ring Tones

The Distinctive Ring Tones feature allows specific incoming calls to trigger the IP phone to play distinctive ring tones. The IP phone inspects the "Alert-Info" header in the INVITE request 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 it to the appropriate ring tone. The followings are two examples of "Alert-Info" headers and the italicized text is a placeholder for the actual value:

Alert-Info: http://127.0.0.1/Bellcore-dr*3*

Alert-Info: <http://*192.168.0.12:8080/ring.wav*>;info=*Family*;x-line-id=0

- If the "Alter-Info" header contains the keywords "Bellcore-drN" or "MyMelodyN", the IP phone will map the index "N" to the relevant ring tone.

Value of N	Ring Tone
1	Ring1.wav
2	Ring2.wav
3	Ring3.wav
4	Ring4.wav
5	Ring5.wav
6	Ring6.wav
7	Ring7.wav
8	Ring8.wav

- If the "Alert-Info" header contains a remote URL, the IP phone will try to download and play the ring tone from the URL. If failing to download, the IP phone will match the keyword (e.g., *Family*) with the internal ringer text configured on the IP phone, and then play the specified ring tone. If there is no text matched, the IP phone will play the ring tone configured on the IP phone in about ten seconds.

Procedure

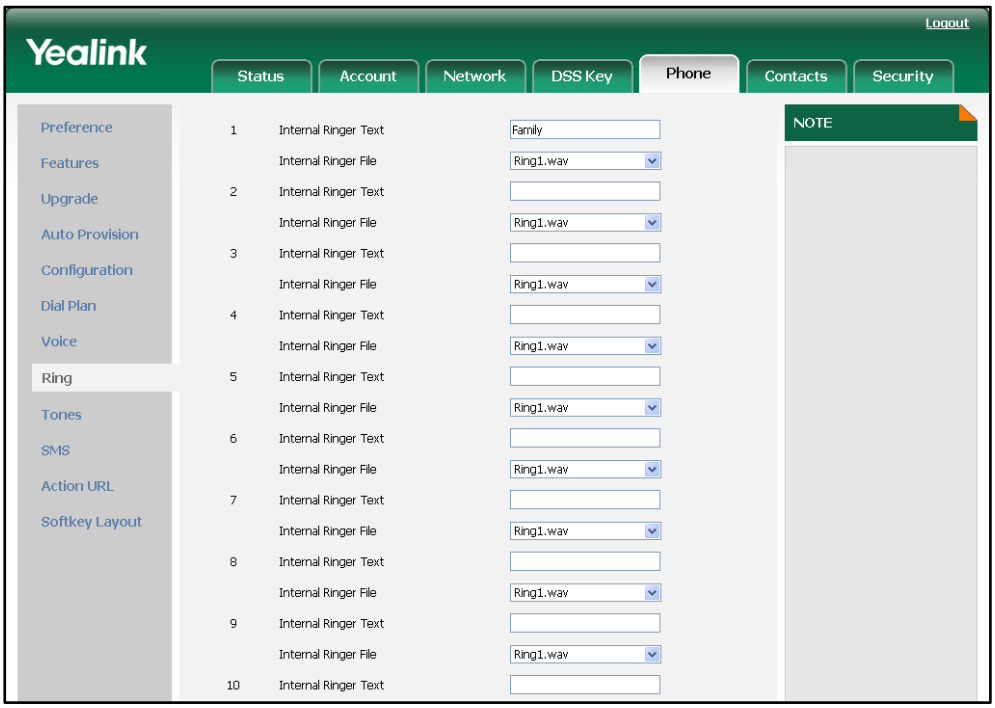
Distinctive ring tones can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the distinctive ring tones feature. For more information, refer to Distinctive Ring Tones on page 274.
	<y0000000000xx>.cfg	Configure the internal ringer text and internal ringer file. For more information, refer to Distinctive Ring Tones on page 274.
Local	Web User Interface	Configure the internal ringer text and internal ringer file.

		Navigate to: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Ring.htm

To configure the internal ringer text and internal ringer file via web user interface:

1. Click on **Phone**->**Ring**.
2. Enter the keywords in the **Internal Ringer Text** fields.
3. Select the desired ring tones for each text from the pull-down lists of **Internal Ringer File**.



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

Tones

When receiving a message or recording a call, the IP phone will play a warning tone. You can customize tones or select the tones customized for a specific country to indicate different conditions of the IP phone. Tone sets vary from country to country. The default tones used on the IP phone are the tone sets of US. The available tone sets are:

- 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

Configured tones can be heard on the IP phone 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	Call hold tone
Record	When recording a call
Info	When receiving a special message

Condition	Description
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.

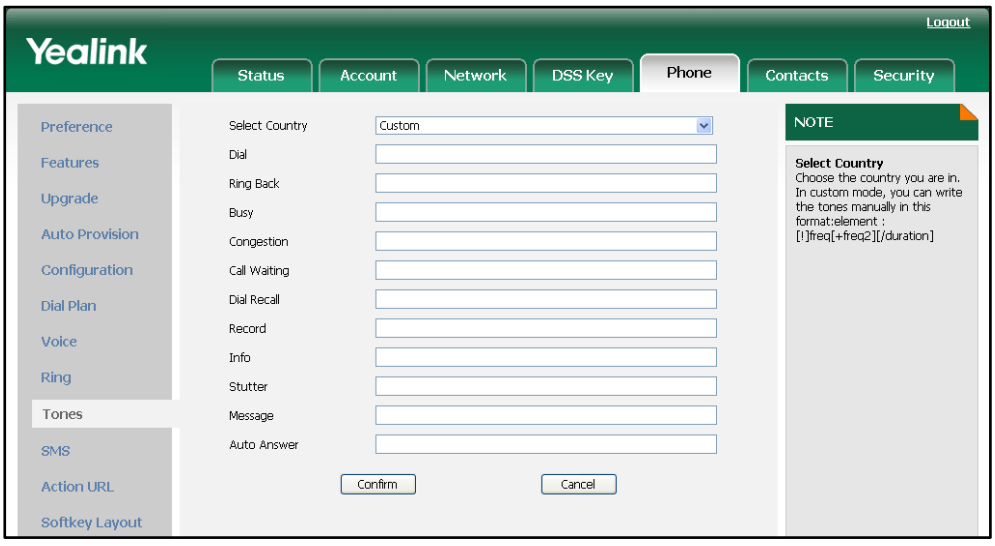
Configuration File	<y0000000000xx>.cfg	Configure the tones for the IP phone. For more information, refer to Tones on page 275.
Local	Web User Interface	Configure the tones for the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Tones.htm

To configure tones via web user interface:

1. Click on **Phone->Tones**.
2. Select the desired type from the pull-down list of **Select Country**.

   If you select **Custom**, you can customize the tone for indicating each condition of the IP phone.

   1) Enter the desired tones in the corresponding fields.



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

Remote Phonebook

Remote phonebook is the IP phone book maintained centrally, which is stored on the remote server. Users just need the access URL of the remote phonebook. The IP phone can establish a connection with the remote server and download the entries, and then display the entries on the phone user interface. The IP phone supports up to 5 remote phonebooks. All remote phonebooks support to store 500 entries in all. The remote phonebook can be customized. For more information, refer to Remote XML Phonebook on page 204.

The Match Remote Phone Book feature allows the IP phone to query the entry names from the remote phonebook when receiving incoming calls. The Load Remote Phone Book Interval feature defines how often the IP phones refresh the local cache of the remote phonebook.

Procedure

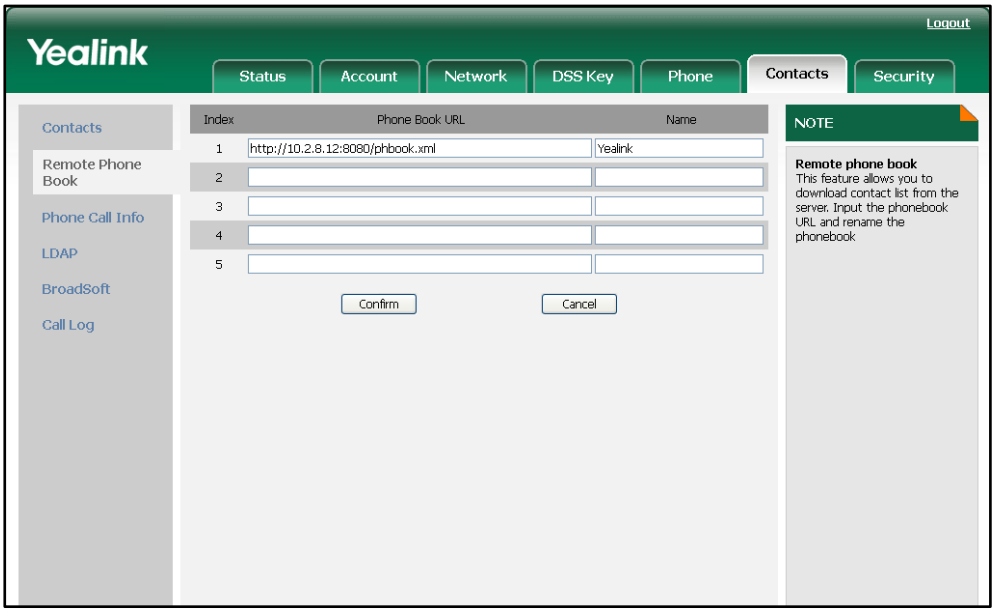
Remote phonebook can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the remote phonebook. For more information, refer to Remote XML Phonebook on page 204. Specify whether to query the entry names from the remote phonebook when the IP phone receives incoming calls. Specify how often the IP phones refresh the local cache of the remote phonebook. For more information, refer to Remote Phonebook on page 277.
Local	Web User Interface	Specify the access URL of the remote phonebook. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Contacts-Remote.htm Specify whether to query the contact names from the remote phonebook when the IP phone receives incoming calls.

		Specify how often the IP phones refresh the local cache of the remote phonebook. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To specify the access URL of the remote phonebook via web user interface:

1. Click on **Contacts**->**Remote Phone Book**.
2. Enter the access URL in the **Phone Book URL** field.
3. Enter the name in the **Name** field.

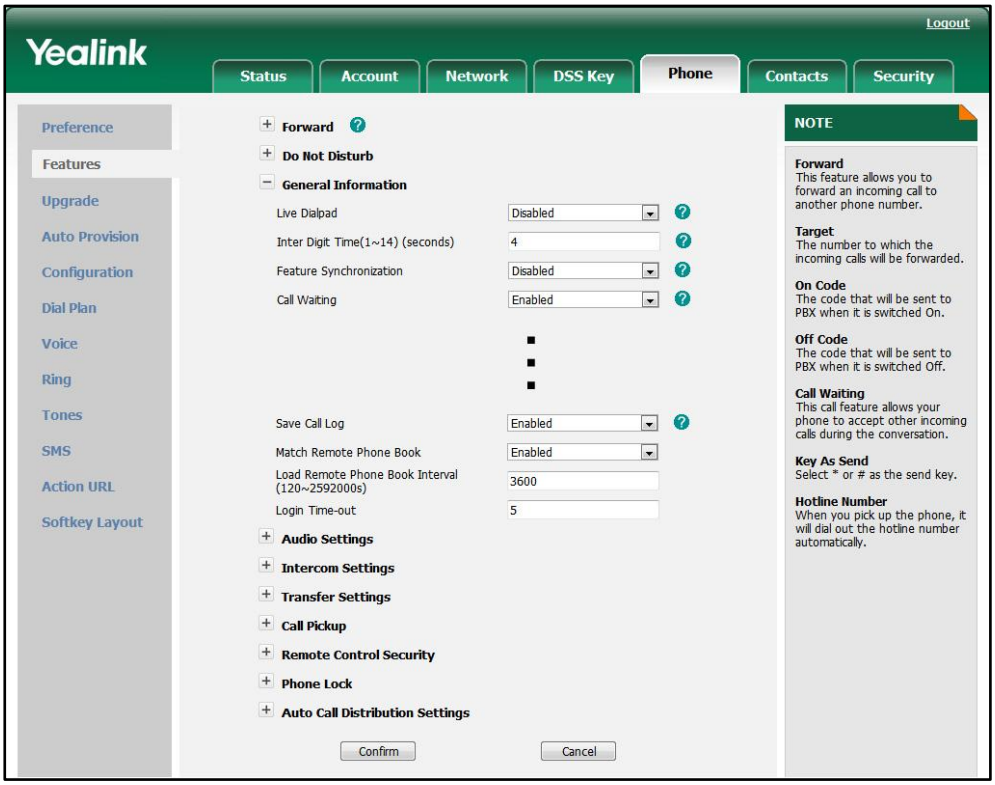


4. Click **Confirm** to accept the change

To configure the remote phonebook via web user interface:

1. Click on **Phone**->**Features**->**General Information**.
2. Select the desired value from the pull-down list of **Match Remote Phone Book**.

3. Enter the desired time in the **Load Remote Phone Book Interval (120~2592000s)** field.



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

LDAP

LDAP (Lightweight Directory Access Protocol) is an application protocol for accessing and maintaining information services of the distributed directory over an IP network. The IP phone can be configured to interface with a corporate directory server that supports LDAP version 2 or 3 (Microsoft's Active Directory is included).

The biggest plus for LDAP is that users can access the central LDAP directory of your corporate using the IP phone, so they do not need to maintain the local directory. Users can search and dial from the LDAP directory and save the LDAP entries to the local directory. The LDAP entries displayed on the IP phone are read only. Users can not add, edit or delete the LDAP entries. When the LDAP server is properly configured, the IP phone can look up entries from the LDAP server in a wide variety of ways. The LDAP server indexes all the data in its entries, and "filters" may be used to select just the desired contact or group, and return just the desired information.

The configurations on the IP phone limit the amount of displayed entries when querying from the LDAP server, and decide how the attributes are displayed and sorted.

There are two ways to perform an LDAP search on the IP phone:

- Simply start a search against LDAP by entering a number. All suitable entries will

be shown according to your query setup.

- Assign a DSS key to be an LDAP key, and press the LDAP key to enter the LDAP Search interface when the IP phone is idle.

LDAP Attributes

The following table lists the most common attributes used to configure the LDAP lookup on the IP phone:

Abbreviation	Name	Description
gn	givenName	First name
cn	commonName	LDAP attribute being made up from given name joined to surname.
sn	surname	Last name or family name
dn	distinguishedName	Unique identifier for each entry
dc	dc	Domain component
-	company	Company or organization name
-	telephoneNumber	Office phone number
mobile	mobilephoneNumber	Mobile or cellular phone number
ipPhone	IPphoneNumber	Home phone number

Procedure

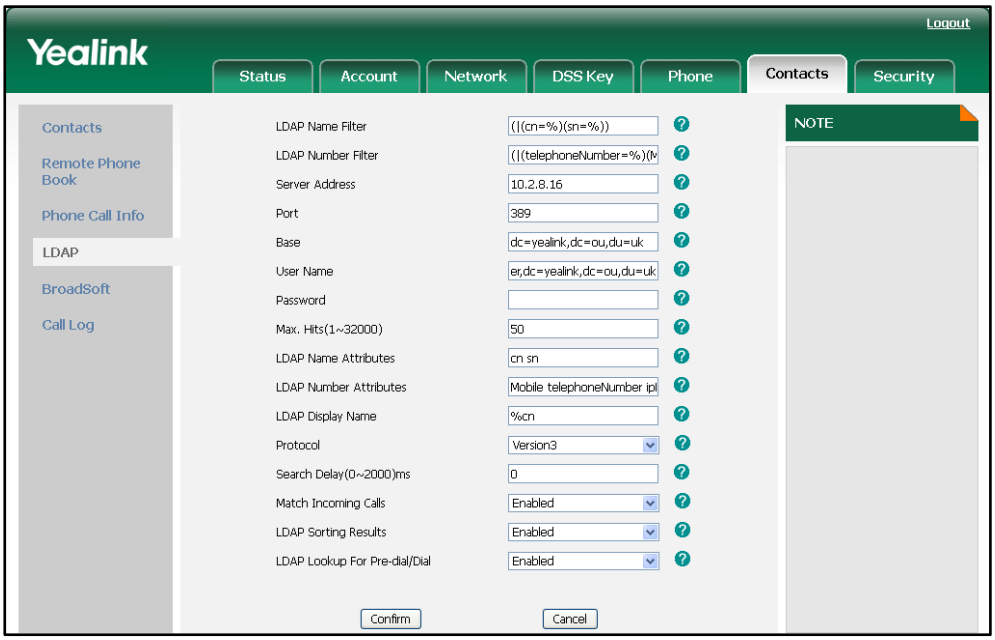
LDAP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the LDAP feature. For more information, refer to LDAP on page 277. Assign an LDAP key. For more information, refer to LDAP Key on page 327.
Local	Web User Interface	Configure the LDAP feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm Assign an LDAP key. **Navigate to**: http://<phoneIPAddress>/cgi-bi

		n/cgiServer.exx?page=Contacts-LDAP.htm
	Phone User Interface	Assign an LDAP key.

To configure LDAP via web user interface:

1. Click on **Contacts**->**LDAP**.
2. Enter the values in the corresponding fields.
3. Select the desired values from the corresponding pull-down lists.

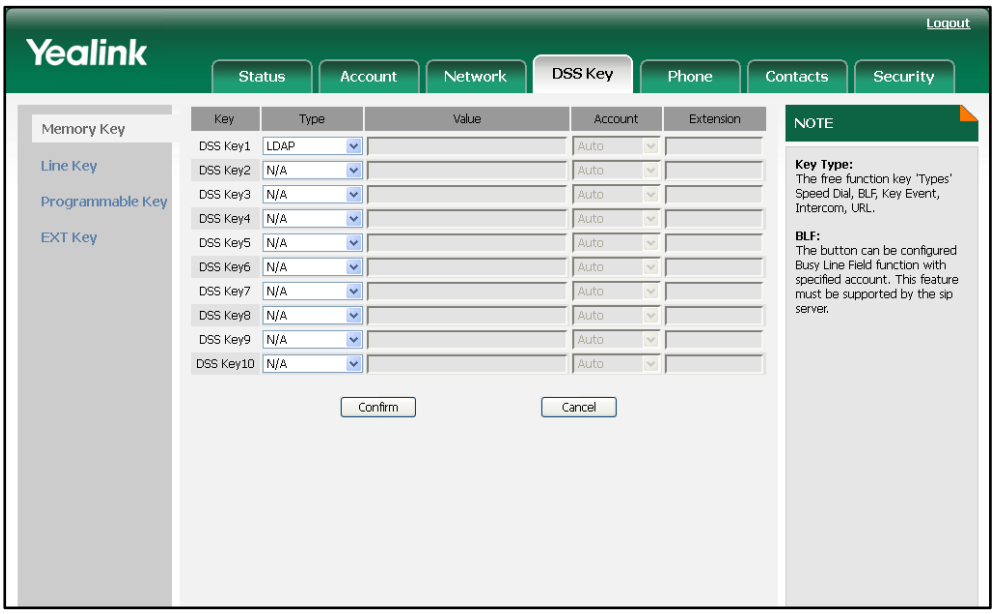


4. Click **Confirm** to accept the change.

To configure an LDAP key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).

2. In the desired DSS key field, select **LDAP** from the pull-down list of **Type**.



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

To configure an LDAP key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ► , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ► , or the **Switch** soft key to select **LDAP** from the **Key Type** field.
5. Press the **Save** soft key to accept the change.

Busy Lamp Field

Busy Lamp Field (BLF) is used to monitor a specific user for status changes on the IP phone. For example, you can configure a BLF key on a supervisor's phone for monitoring the status of a user's phone (busy or idle). When the user picks up his phone to make a call, a busy indicator on the supervisor's phone shows that the user's phone is in use and busy.

Visual and Audio Alert for BLF Pickup

The visual and audio alert for BLF pickup features allow the supervisor's phone to play an alert tone and display a visual prompt (e.g. "6001<-6002", 6001 is the monitored extension) when the monitored user receives an incoming call. The visual alert for BLF pickup feature also enables the supervisor to pick up the incoming call of the monitored user by pressing the Pickup soft key directly. The direct pickup code must be configured in advance, for more information on how to configure the direct pickup code, refer to Direct Pickup on page 99.

Procedure

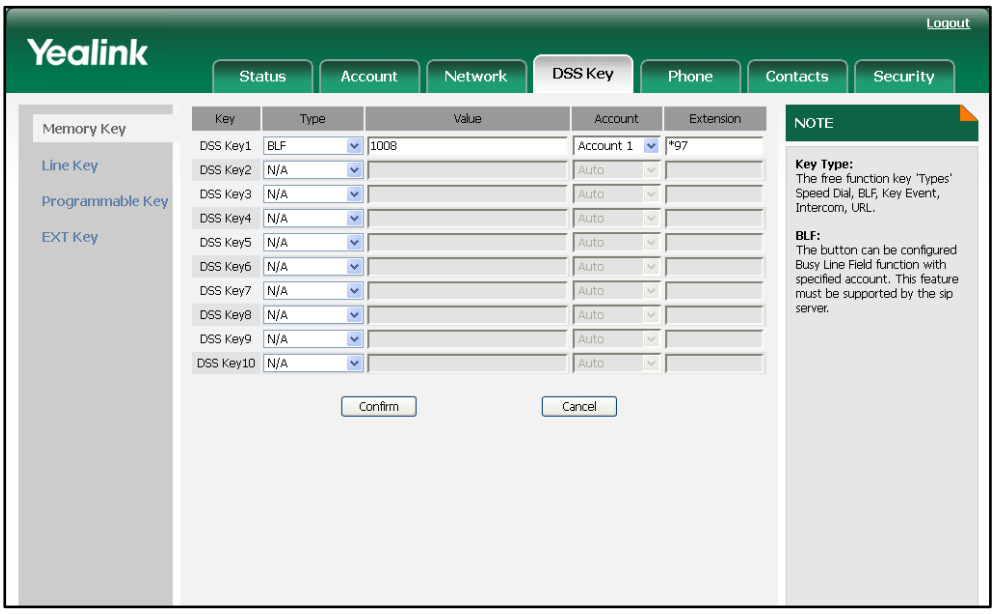
BLF can be configured using the configuration files or locally.

Configuration File	y0000000000xx.cfg	Assign a BLF key. For more information, refer to BLF Key on page 328. Specify whether to use visual and audio alert for BLF pickup. For more information, refer to BLF on page 283.
Local	Web User Interface	Assign a BLF key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm Specify whether to use visual and audio alert for BLF pickup. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm
	Phone User Interface	Assign a BLF key.

To configure a BLF key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **BLF** from the pull-down list of **Type**.
3. Enter the phone number or extension you want to monitor in the **Value** field.
4. Select the desired line from the pull-down list of **Account**.

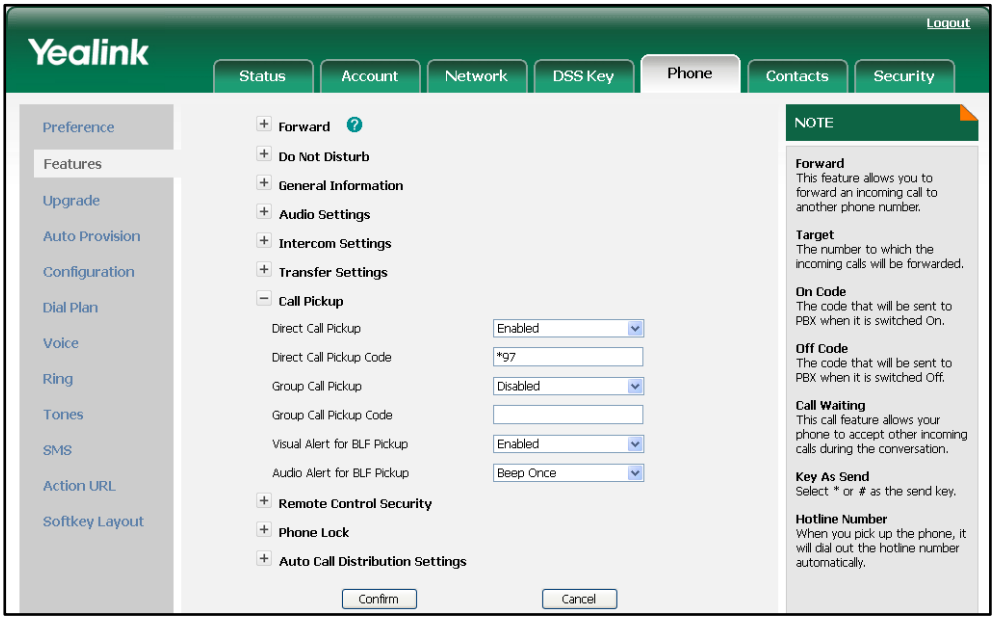
5. (Optional.) Enter the pickup code in the **Extension** field.



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

To configure the visual and audio alert feature via web user interface:

1. Click on **Phone->Features->Call Pickup**.
2. Select the desired value from the pull-down list of **Visual Alert for BLF Pickup**.
3. Select the desired value from the pull-down list of **Audio Alert for BLF Pickup**.



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

To configure a BLF key via phone user interface:

1. Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.

3. Press ◄ or ►, or the **Switch** soft key to select **BLF** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
5. Enter the phone number or extension you want to monitor in the **Value** field.
6. (Optional.) Enter the pickup code in the **Extension** field.
7. Press the **Save** soft key to accept the change.

BLF List

The BLF list feature is used to monitor a list of specific users for status changes on the IP phone. This feature enables the supervisor's phone to subscribe to a list of users, and receive notifications of the status of the monitored users. You need to specify the BLF list URI on the supervisor's phone to monitor the list of users. The BLF list URI is configurable on a per-account basis. The BLF list keys on the IP phone can present the status of the list of users.

When the monitored user is idle, the user presses the BLF list key to dial out the phone number. When the monitored user receives an incoming call, the user presses the BLF list key to pick up the call directly.

Procedure

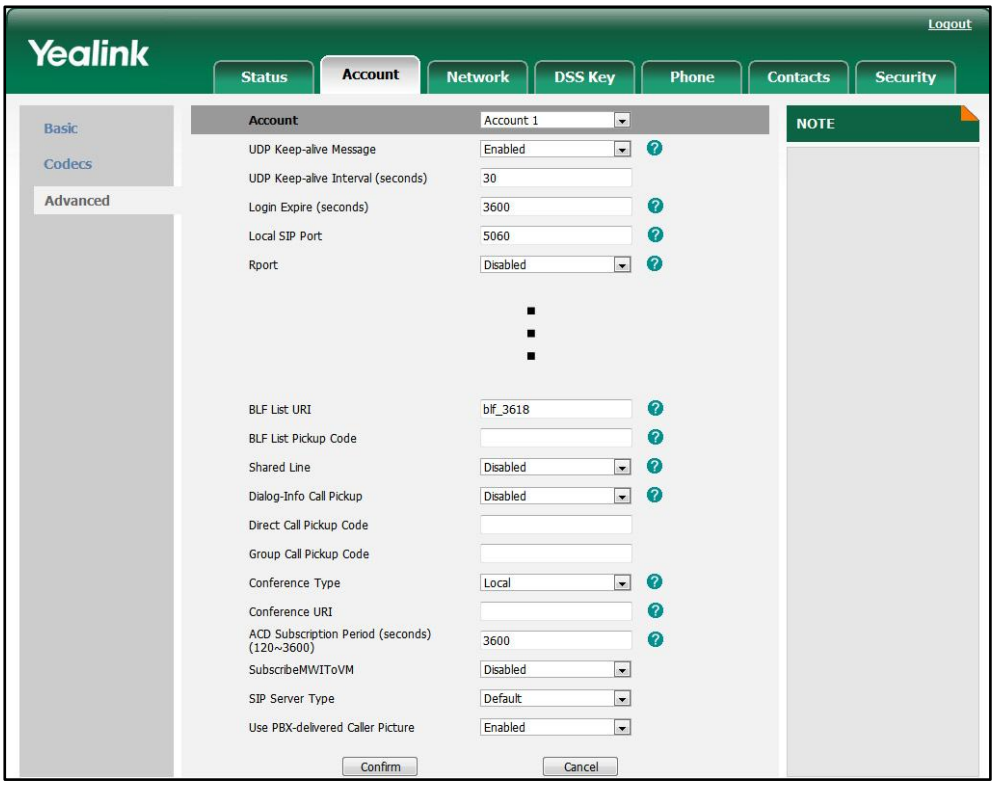
BLF list can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the BLF list URI and BLF list pickup code. For more information, refer to BLF List on page 284.
	y0000000000xx.cfg	Assign a BLF list key. For more information, refer to BLF List Key on page 329.
Local	Web User Interface	Configure the BLF list URI and BLF list pickup code. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Assign BLF list keys. **Navigate to**: http://<phoneIPAddress>/cgi-bi

		n/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign BLF list keys.

To configure BLF list via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the **Account** field.
3. Click on **Advanced**.
4. Enter the BLF List URI in the **BLF List URI** field.
5. (Optional.) Enter the BLF pickup code in the **BLF List Pickup Code** field.

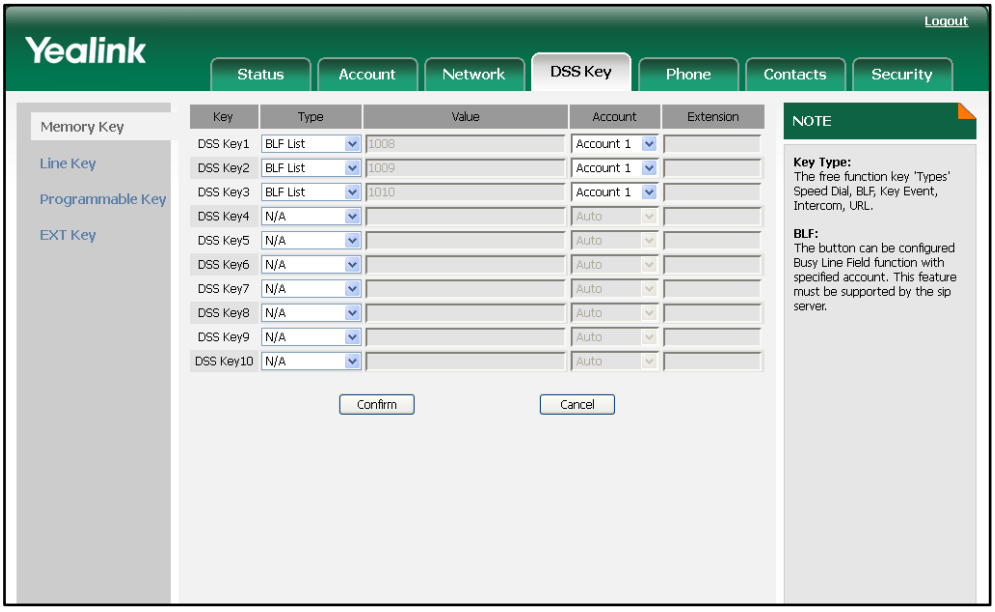


6. Click **Confirm** to accept the change.

To assign BLF list keys via web user interface:

1. Click on **DSS Key->Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **BLF List** from the pull-down list of **Type**.
3. Select the desired line from the pull-down list of **Account**.

4. Repeat steps 2 to 3 to configure more BLF List keys.



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

After the above configurations, according to the response message from the BLF List server, the IP phone will automatically assign the phone number of the BLF List users to the BLF List keys in order.

To assign BLF List keys via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **BLF List** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
5. Press the **Save** soft key to accept the change.
6. Repeat steps 2 to 5 to configure more BLF List keys.

Shared Call Appearance

Shared Call Appearance (SCA) allows users to share a SIP line on several IP phones and also provides status monitoring of the shared line. The IP phones support SCA using the SUBSCRIBE-NOTIFY method as specified in RFC 3265. The events used are:

- "call-info" for call appearance state notification
- "line-seize" for the IP phone to ask to seize the line

When a user places an outgoing call using the registered shared line, all users sharing this line will receive notify of this usage. The LEDs available on the IP phones indicate the status of the shared line. Incoming calls to this line will cause all phones sharing this line to ring simultaneously. The incoming call can be answered on one of the IP phones but

not all of them. An SCA user can retrieve a public hold call on the shared line. If the SCA bridging feature is enabled, SCA users can barge in an existing call on the shared line.

Procedure

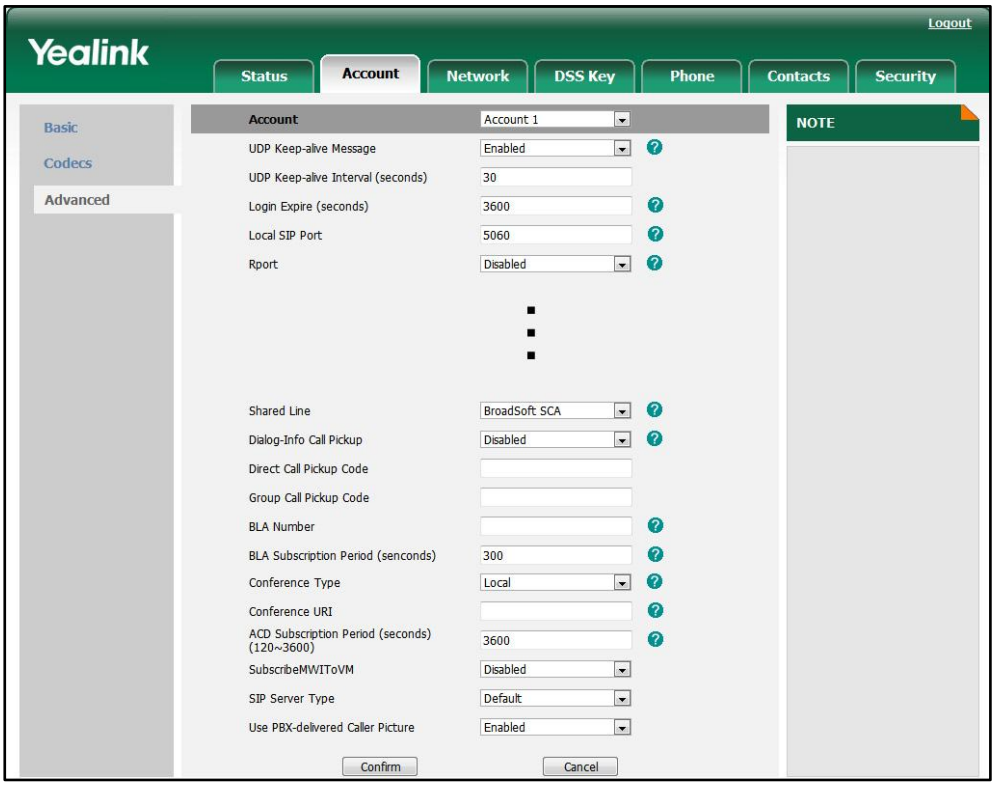
Register the primary and secondary lines on two IP phones using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the shared line on the IP phone. For more information, refer to Shared Call Appearance on page 136.
	y0000000000xx.cfg	Assign a shared line key. For more information, refer to Shared Line Key on page 330.
Local	Web User Interface	Configure the shared line on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Assign a shared line key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign a shared line key.

To register the primary and secondary lines via web user interface:

1. Click on **Account**->**Basic**.
2. Register the line as usual (entering the register name of the primary line in the **Register Name** field when registering the secondary line).
3. Click on **Advanced**.

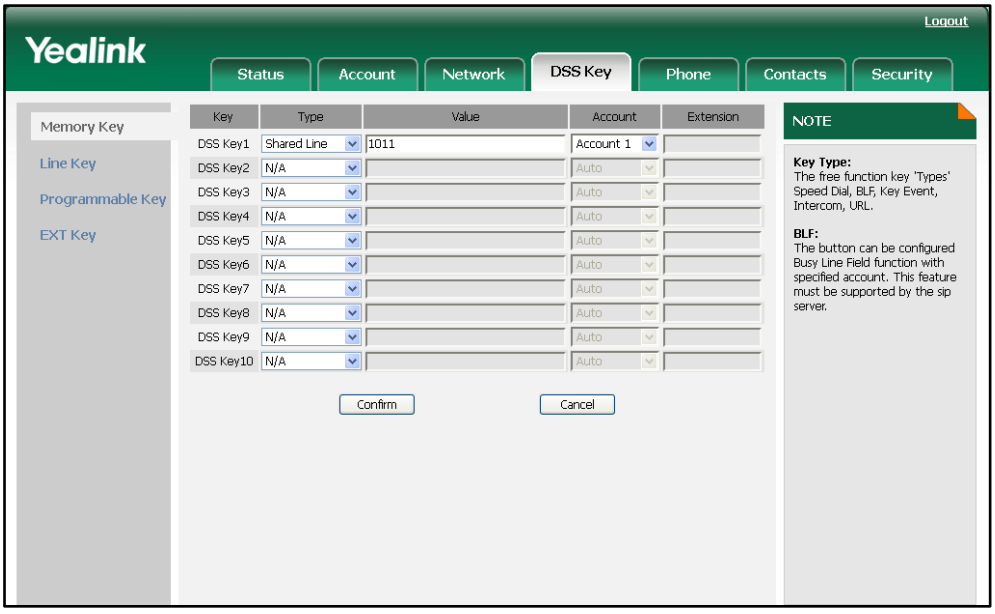
4. Select **BroadSoft SCA** from the pull-down list of **Shared Line**.



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

To assign a shared line key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Shared Line** from the pull-down list of **Type**.
3. Enter the primary account in the **Value** field.
4. Select the desired line from the pull-down list of **Account**.



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

To assign a shared line key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **Shared Line** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
5. Enter the primary account in the **Value** field.
6. Press the **Save** soft key to accept the change.

As-Feature-Event

The IP phones support server-side Do Not Disturb (DND), Call Forward (CFWD) and Automatic Call Distribution (ACD) features. The as-feature-event feature allows the IP phones and the server to synchronize the status of the following features with each other:

- Do Not Disturb
- Call Forwarding Always (CFA)
- Call Forwarding Busy (CFB)
- Call Forwarding No Answer (CFNA)
- ACD

If a user changes the status of one of these features on the IP phone, the IP phone notifies the server of synchronizing the status. Conversely, if the status of one of these features is changed on the server, the server notifies the IP phone of synchronizing the status.

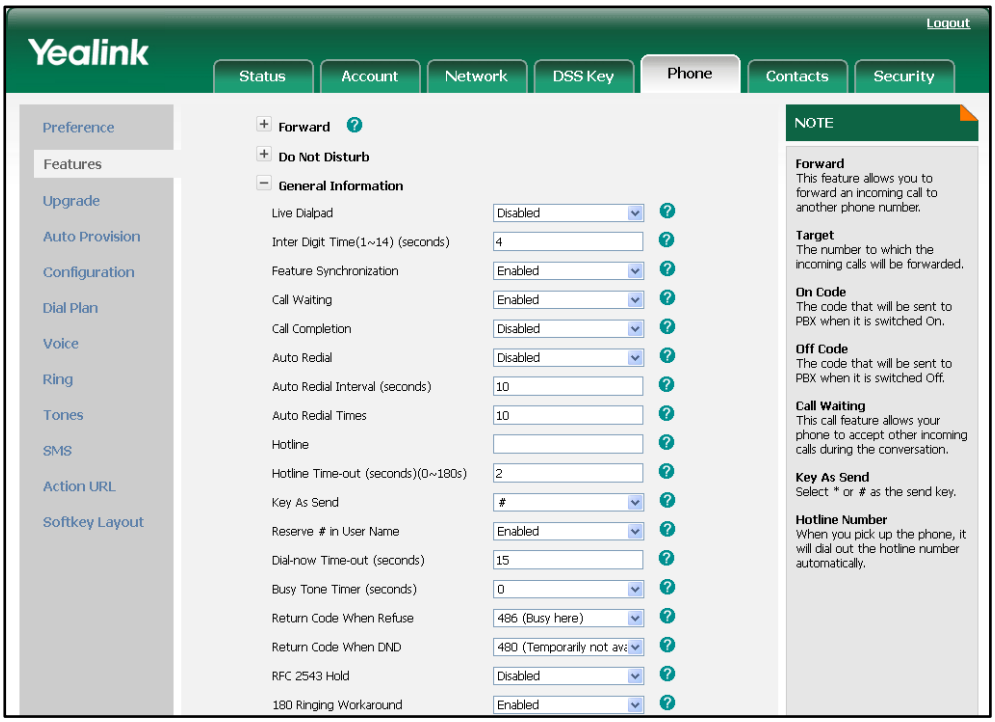
Procedure

As-feature-event feature can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the as-feature-event. For more information, refer to As-Feature-Event on page 288.
Local	Web User Interface	Configure the as-feature-event. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure the as-feature-event via web user interface:

1. Click on **Phone->Features->General Information**.
2. Select the desired value from the pull-down list of **Feature Synchronization**.



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

Automatic Call Distribution

Automatic Call Distribution (ACD) enables organizations to manage a large number of phone calls on an individual basis. ACD enables use of the IP phones in a call-center role by automatically directing incoming calls to available persons, or agents. The ACD feature depends on support from a SIP server.

A user needs to press an ACD key to log in the ACD system. The ACD system monitors the ACD status on the user's phone and then decides whether to assign an incoming call to it. The user can change the ACD status on the IP phone. The ACD key LED on the IP phone indicates the ACD status.

Procedure

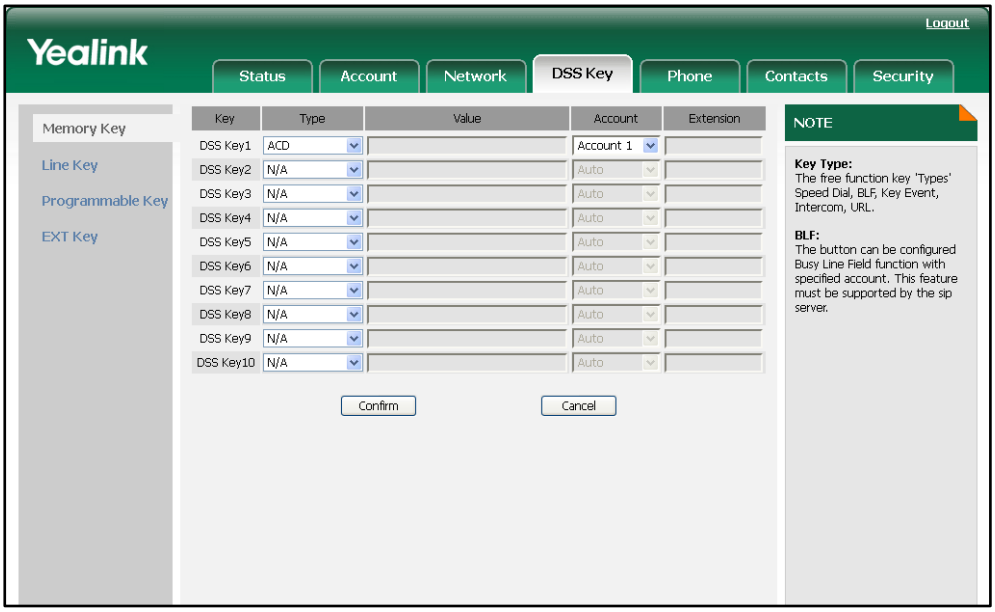
ACD key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign an ACD key. For more information, refer to ACD Key on page 331.

Local	Web User Interface	Assign an ACD key. **Navigate to:** http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign an ACD key.

To configure an ACD key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **ACD** from the pull-down list of **Type**.
3. Select the desired line from the pull-down list of **Account**.



4. Press the **Save** soft key to accept the change.

To configure an ACD key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **ACD** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select the desired line from the **Account ID** field.
5. Press the **Save** soft key to accept the change.

Message Waiting Indicator

Message Waiting Indicator (MWI) is a feature that informs users that they have messages waiting in their mailboxes. This feature indicates how many messages are waiting without the users having to call their mailboxes. The IP phones support both

audio and visual MWI when receiving new voice messages.

The IP phones support both solicited and unsolicited MWI. Unsolicited MWI is a server related feature.

Solicited MWI: MWI notification is subscription-based. 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 the MWI subscription feature on the IP phone.

Unsolicited MWI: MWI notification is not subscription-based. The IP phone does 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.

Procedure

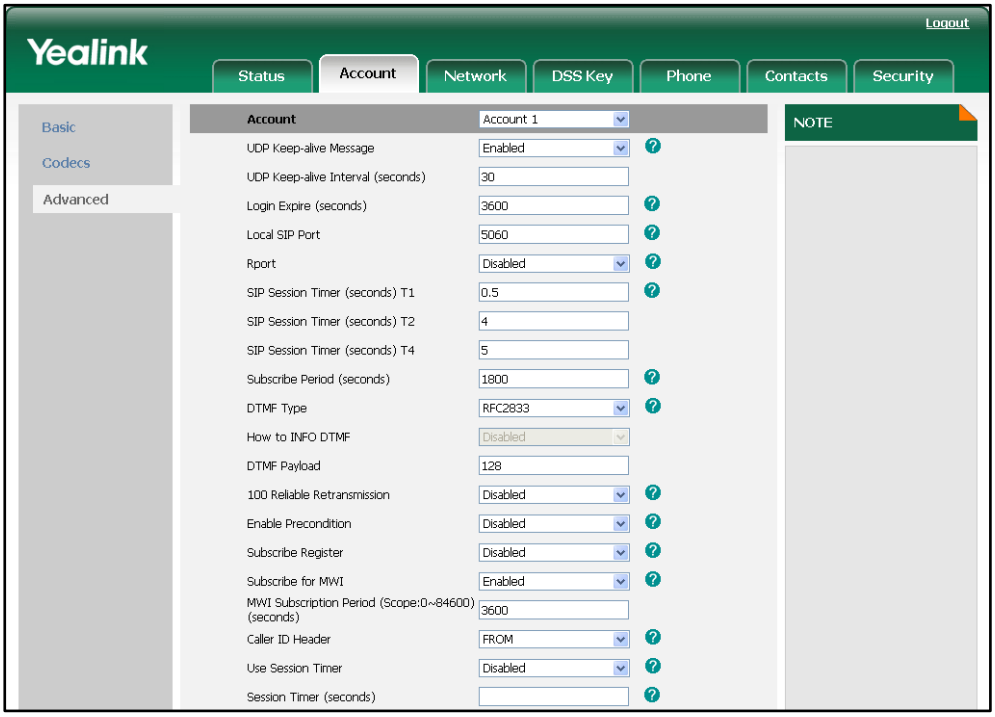
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the MWI subscription feature on the IP phone. For more information, refer to Message Waiting Indicator on page 289.
Local	Web User Interface	Configure the MWI subscription feature on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the MWI subscription feature via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **Subscribe for MWI**.

5. Enter the period time in the **MWI Subscription Period (Scope: 0~84600) (seconds)** field.



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

Call Recording

Call recording enables a user to record a call. It depends on support from a SIP server. When the user presses a call record key, the IP phone sends a record request to the server. The IP phones themselves do not have memory to store the recording, what they can do is only to trigger the recording and indicate the recording status.

Normally, there are 2 main methods to trigger a recording on a certain server. We call them record and URL record. Record is for the IP phone to send the server a SIP INFO message containing a specific header. URL record is for the IP phone to send an HTTP URL to the server. The server processes these messages and decides to start or stop a recording.

Record

When a user presses a record key for the first time during a call, the IP phone sends a SIP INFO message to the server with a specific header "Record: on", and then the recording starts.

The example of a SIP INFO message for reference:

```
Via: SIP/2.0/UDP 10.1.4.148:5063;branch=z9hG4bK1139980711
From: "827" <sip:827@192.168.1.199>;tag=2066430997
```

```
To:<sip:614@192.168.1.199>;tag=371745247
Call-ID: 1895019940@10.1.4.148
CSeq: 2 INFO
Contact: <sip:827@10.1.4.148:5063>
Max-Forwards: 70
User-Agent: Yealink SIP-T38G 38.70.0.100
Record: on
Content-Length: 0
```

When the user presses the record key for the second time, the IP phone sends a SIP INFO message to the server with a specific header "Record: off", and then the recording stops.

The example of a SIP INFO message for reference:

```
Via: SIP/2.0/UDP 10.1.4.148:5063;branch=z9hG4bK1619489730
From: "827" <sip:827@192.168.1.199>;tag=1831694891
To:<sip:614@192.168.1.199>;tag=2228378244
Call-ID: 1051886688@10.1.4.148
CSeq: 3 INFO
Contact: <sip:827@10.1.4.148:5063>
Max-Forwards: 70
User-Agent: Yealink SIP-T38G 38.70.0.100
Record: off
Content-Length: 0
```

URL Record

When a user presses a URL record key for the first time during the call, the IP phone sends an HTTP GET message to the server.

The example of an HTTP GET message for reference:

```
Get /phonerecording.cgi?model=yealink HTTP/1.0\r\n
    Request Method: GET
    Request URI: /phonerecording.cgi?model=yealink
    Request version: HTTP/1.0
Host: 10.1.2.224\r\n
User-agent: yealink SIP-T38G 38.70.0.100 00:16:65:11:30:68\r\n
```

If the recording is successfully started, the server will respond with 200 OK as below:

```
<YealinkIPPhoneText>
 <Title>
    </Title>
 <Text>
    The recording session is successfully started.
    </Text>
```

```
<YealinkIPPhoneText>
```

If the recording fails for some reasons, for example, the recording box is full, the server will respond with 200 OK as below:

```
<YealinkIPPhoneText>
 <Title>
    </Title>
 <Text>
    Probably the recording box is full.
    </Text>
<YealinkIPPhoneText>
```

When the user presses the URL record key for the second time, the IP phone sends an HTTP GET message to the server, and then the server will respond with the following 200 OK message:

```
<YealinkIPPhoneText>
 <Title>
    </Title>
 <Text>
    The recording session is successfully stopped.
    </Text>
<YealinkIPPhoneText>
```

Procedure

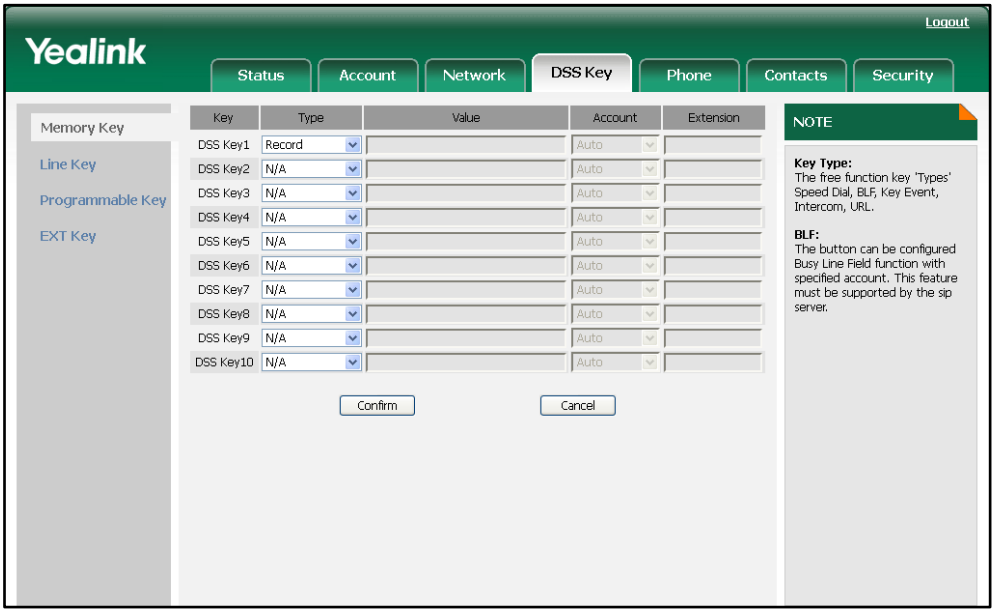
Call recording key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a record key. For more information, refer to Record Key on page 331. Assign a URL record key. For more information, refer to URL Record Key on page 332.
Local	Web User Interface	Assign a record key. Assign a URL record key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign a record key. Assign a URL record key.

To configure a Record key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).

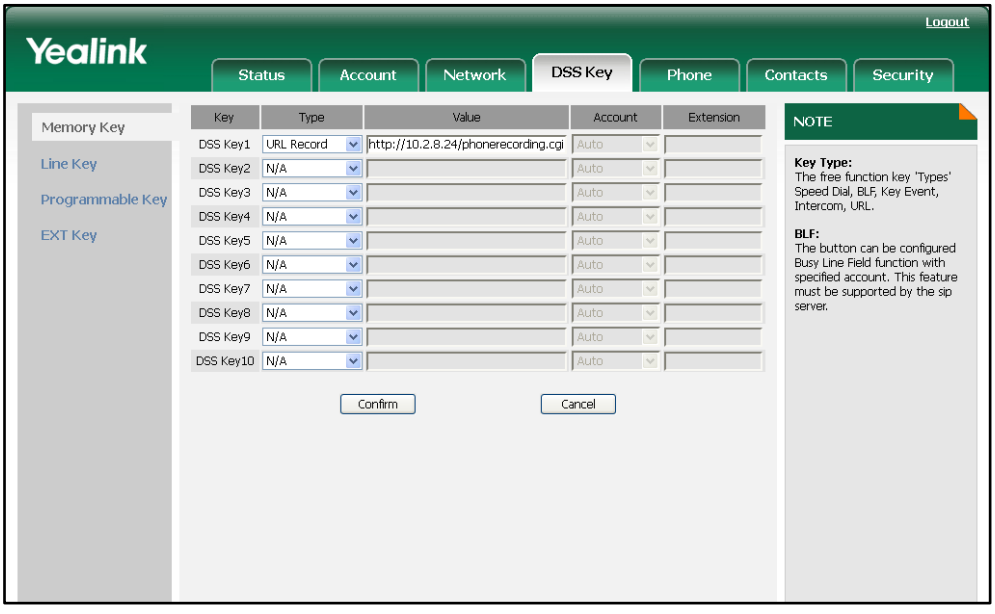
2. In the desired DSS key field, select **Record** from the pull-down list of **Type**.



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

To configure a URL Record key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **URL Record** from the pull-down list of **Type**.
3. Enter the URL in the **Value** field.



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

To configure a Record key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.

3. Press ◄ or ►, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ►, or the **Switch** soft key to select **Record** from the **Key Type** field.
5. Press the **Save** soft key to accept the change.

To configure a URL Record key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ►, or the **Switch** soft key to select **URL Record** from the **Type** field.
4. Enter the URL in the **URL Record** field.
5. Press the **Save** soft key to accept the change.

Hot Desking

Hot desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee. A primary motivation for hot desking is cost reduction. Hot desking is regularly used in places where not all the employees are in the office at the same time, or not in the office for long periods at a time, which means actual personal offices would often be vacant, consuming valuable space and resources.

The hot desking feature allows a user to delete all accounts on the IP phone, register his account on line 1. In order to use this feature, you need to assign a hot desking key.

Procedure

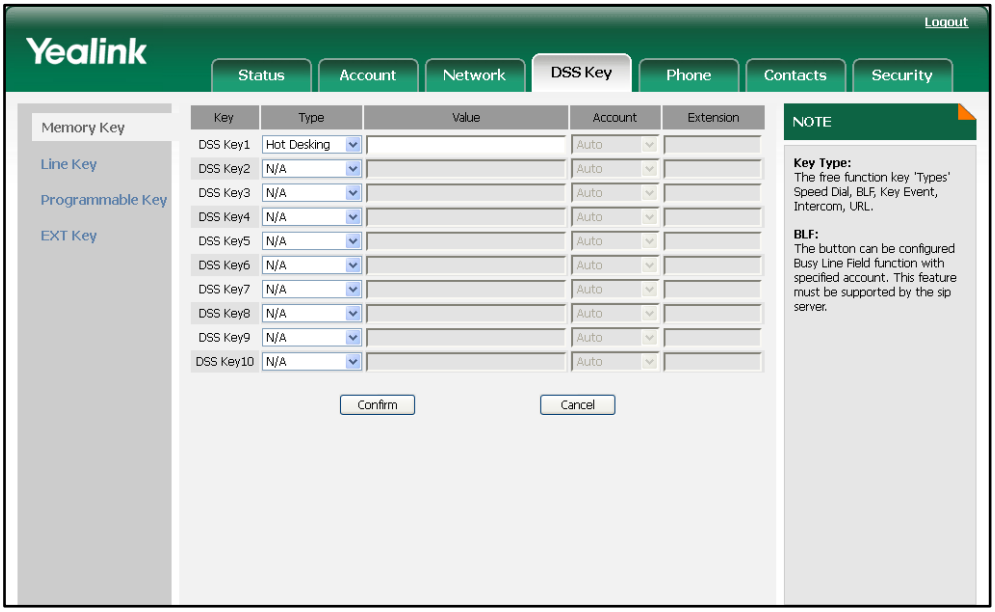
Hot desking key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a hot desking key. For more information, refer to Hot Desking Key on page 332.
Local	Web User Interface	Assign a hot desking key. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Dsskey.htm
	Phone User Interface	Assign a hot desking key.

To configure a hot desking key via web user interface:

1. Click on **DSS Key**->**Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Hot Desking** from the pull-down list of **Type**.

3. Leave the **Value** field blank.



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

To configure a hot desking key via phone user interface:

1. Press **Menu**->**Features**->**DSS Keys**->**Memory Keys** (or **Line Keys**).
2. Select the desired DSS key.
3. Press ◄ or ► , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press ◄ or ► , or the **Switch** soft key to select **Hot Desking** from the **Key Type** field.
5. Leave the **Value** field blank.
6. Press the **Save** soft key to accept the change.

Action URL

Action URL is an HTTP GET request allowing the IP phone to interact with web server applications. You can specify a URL that triggers a GET request when certain events occur. An HTTP GET request may contain variable name and variable value, which are separated by "=". Each variable value starts with $ in the query part of the URL. The URL format is: http://IP address of server/help.xml? variable name=variable value (e.g. http://192.168.1.10/help.xml?mac=$mac). Action URL can be only triggered by the predefined events (e.g., Log on).

The following table lists the predefined events for Action URL:

Event	Description
Setup Completed	When the IP phone completes startup.

Event	Description
Log On	When the IP phone successfully registers an account.
Log Off	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.
Call Established	When the IP phone establishes a call.
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 Call	When the IP phone blind transfers a call.
Attended Transfer Call	When the IP phone performs the attended transfer.
Hold	When the IP phone places a call on hold.
Unhold	When the IP phone retrieves a hold call.
Mute	When the IP phone mutes a call.
Unmute	When the IP phone unmutes a call.
Missed Call	When the IP phone misses a call.
Forward Incoming Call	When the IP phone forwards an incoming call.
Reject Incoming Call	When the IP phone rejects an incoming call.
Answer New Incoming Call	When the IP phones answers a new call.
Transfer Finished	When the IP phone completes to forward a call.
Transfer Failed	When the IP phone fails to transfer a call.

Event	Description
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.

The following table lists the variable values used when specifying a URL:

Variable	Description
$mac	MAC address of the IP phone
$ip	The current IP address of the IP phone
$model	Phone model
$firmware	Phone firmware version
$active_url	The SIP URI of the current account when the IP phone is in the incoming, outgoing or connecting state.
$active_user	The username of the current account when the IP phone is in the incoming, outgoing or connecting state.
$active_host	The host name of the current account when the IP phone is in the incoming, outgoing or connecting state.
$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 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 caller ID when in incoming, outgoing or connecting state.

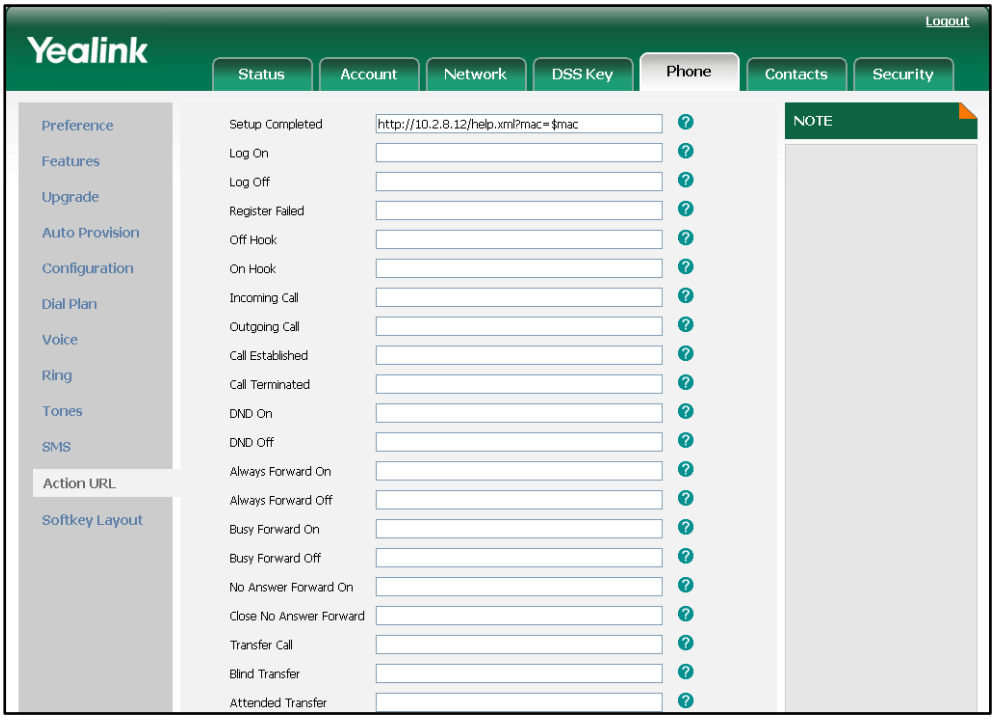
Procedure

Action URL can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the action URL on the IP phone. For more information, refer to Action URL on page 290.
Local	Web User Interface	Configure the action URL on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-ActionURL.htm

To configure the Action URL via web user interface:

1. Click on **Phone->Action URL**.
2. Enter the action URLs in the corresponding fields.



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

Action URI

Opposite to Action URL, Action URI allows the IP phone to interact with web server application by receiving and handling HTTP GET requests. When receiving a URI, the IP

phone will perform the specified action and respond with a 200 OK message. The HTTP GET request may contain variable named as "key" and variable value, which are separated by "=". The URI format is: http://IP address of phone/cgi-bin/cgiServer.exx?key= variable value (e.g. http://192.168.1.5/cgi-bin/cgiServer.exx?key=MUTE). Entering the URI in the web browser triggers the IP phone to perform the predefined action (e.g. mute the call).

The following table lists the variable values may be used when specifying a URI:

Variable	Phone Action
key=OK/key=ENTER	Press the OK key or the Enter soft key.
key=SPEAKER	Press the Speaker key.
key=F_TRANSFER	Press the Transfer key.
key=VOLUME_UP	Increase the volume.
key=VOLUME_DOWN	Decrease the volume.
key=MUTE	Mute the call.
key=F_HOLD	Press the Hold key.
key=X	Press the X key.
key=0-9/*/POUND	Send the DTMF digit (0-9, * or #).
key=L1-L6	Press the Line key.
key=D1-D10	Press the DSS key.
key=F_CONFERENCE	Press the Conference key.
key=F1-F4	Press the Soft key.
key=MSG	Press the MESSAGE key.
key=HEADSET	Press the HEADSET key.
key=RD	Press the Redial key.
key=UP/DOWN/LEFT/RIGHT	Press the Navigation keys.
key=Reboot	Reboot the IP phone.
key=AutoP	Let the IP phone do auto provisioning.
key=DNDOn	Activate the DND mode.
key=DNDOff	Deactivate the DND mode.

Note The variable does not work with all events. For example, the variable "key=MUTE" is only applicable when the IP phone is during a call.

For security reasons, the IP phone does not receive and handle the HTTP GET request by default. You need to specify the trusted IP address for Action URI. When receiving the

HTTP GET request from the specified IP address, the phone LCD screen prompts the message "Allow Remote Control?". You can specify one or more trusted IP addresses on the IP phone. You can also configure the IP phone to receive and handle the URI from any IP address.

Procedure

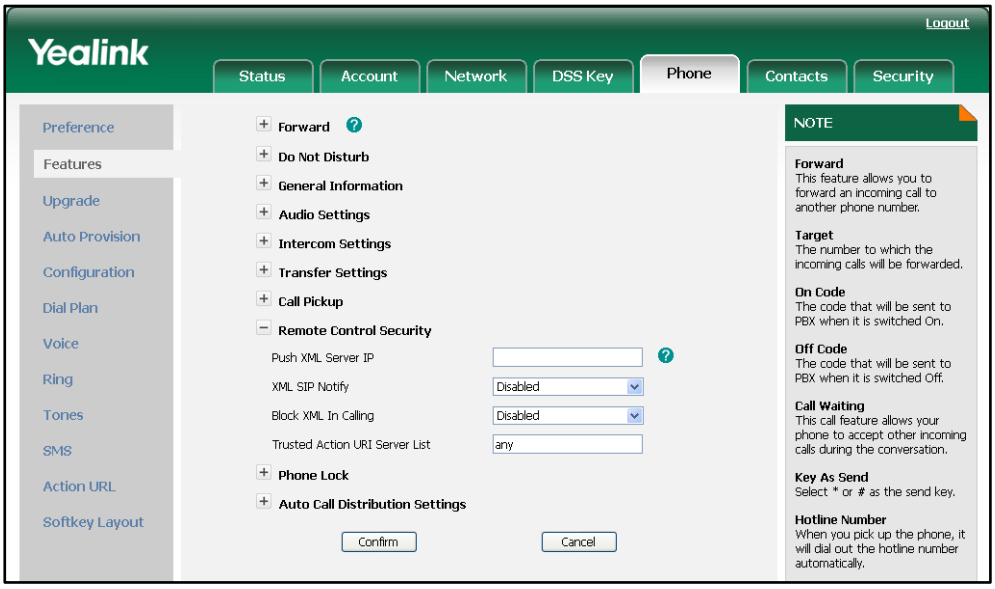
Specify the trusted IP address for Action URI using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the trusted IP address(es) for sending the Action URI to the IP phone. For more information, refer to Action URI on page 291.
Local	Web User Interface	Specify the trusted IP address(es) for sending the Action URI to the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Features.htm

To configure the trusted IP address(es) for Action URI via web user interface:

1. Click on **Phone->Features->Remote Control Security**.
2. Enter the IP address or any in the **Trusted Action URI Server List** field.

   Multiple IP addresses are separated by comma. If you set the field to "any", the IP phone receives and handles HTTP GET requests from any IP address. If you leave the field blank, the IP phone cannot receive or handle any HTTP GET request.



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

Server Redundancy

Many SIP servers are deployed in redundant pairs, designated as primary and secondary servers. The IP phone must always contact the primary server except in failover conditions. 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. The IP phone is able to route to a secondary (or alternate) server in a failure situation, which requires the use of DNS SRV query for the resolution of proxy address as specified by RFC 3263.

Before connecting a network through the domain name of the server, the IP phone performs a DNS SRV query. It sends out a DNS SRV query to the server to look up the IP address and port, and then waits for a response from the server. The DNS SRV query involves NAPTR, SRV and A queries, which allows the IP phone to adapt to various deployment environments. The DNS SRV query is configurable on a per-account basis.

NAPTR (Naming Authority Pointer)

First, the IP phone sends the NAPTR query to get the SRV pointer and service type. As an example, consider the IP phone wishes to resolve "sip:user@example.com". The IP phone performs a NAPTR query for the domain name. The sample of the NAPTR records for reference:

	order	pref	flags	service	regexp	replacement
IN NAPTR	90	50	"s"	"SIP+D2T"	""	_sip._tcp.example.com
IN NAPTR	100	50	"s"	"SIP+D2U"	""	_sip._udp.example.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 to process multiple NAPTR records with the same order value. Lower value is MORE preferred.
flags	The flag "s" means to do an SRV lookup.
service	Specify the service available for SIP by the following rules: 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 DNS name to be used 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.example.com". If the flag of the NAPTR record returned is empty, the IP phone will use "sip:user@example.com" for the next NAPTR query.

SRV (Service Location Record)

The IP phone performs a SRV query on the record returned from the NAPTR for the host name and the port number. The sample of the SRV records for reference:

	Priority	Weight	Port	Target
IN SRV	0	1	5060	server1.example.com
IN SRV	0	2	5060	server2.example.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. Again, keep the same to load balance.
Port	Identify the port number to be used.
Target	Identify the actual hosts for an A query.

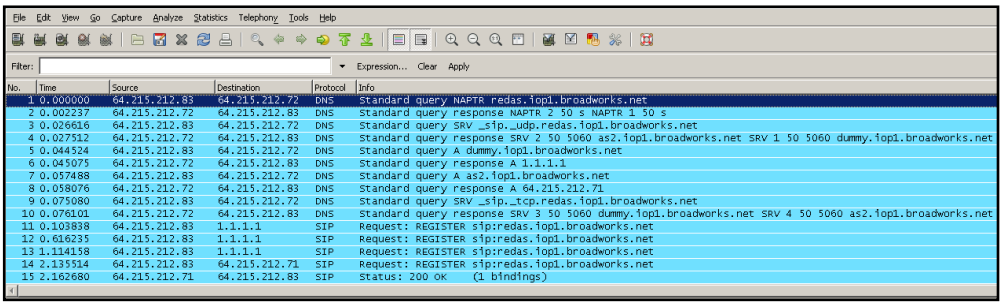
The two SRV records point to different hosts. The two records have the same priority 0. The weight of the second record is higher than the first one, so the second record is picked first. If there is no IP address returned in the response, the IP phone sends out an A query to look up the IP address. So in this case, the IP phone will use "server1.example.com" and "server2.example.com" for the A query.

A (Host IP Address)

The IP phone performs an A query for the IP address of the target host name. The sample of an A record for reference:

IN A 62.10.1.10

The following figure illustrates the IP phone has the availability of performing DNS SRV query, and fails over the request to the secondary server when there is no response from the primary server.



Procedure

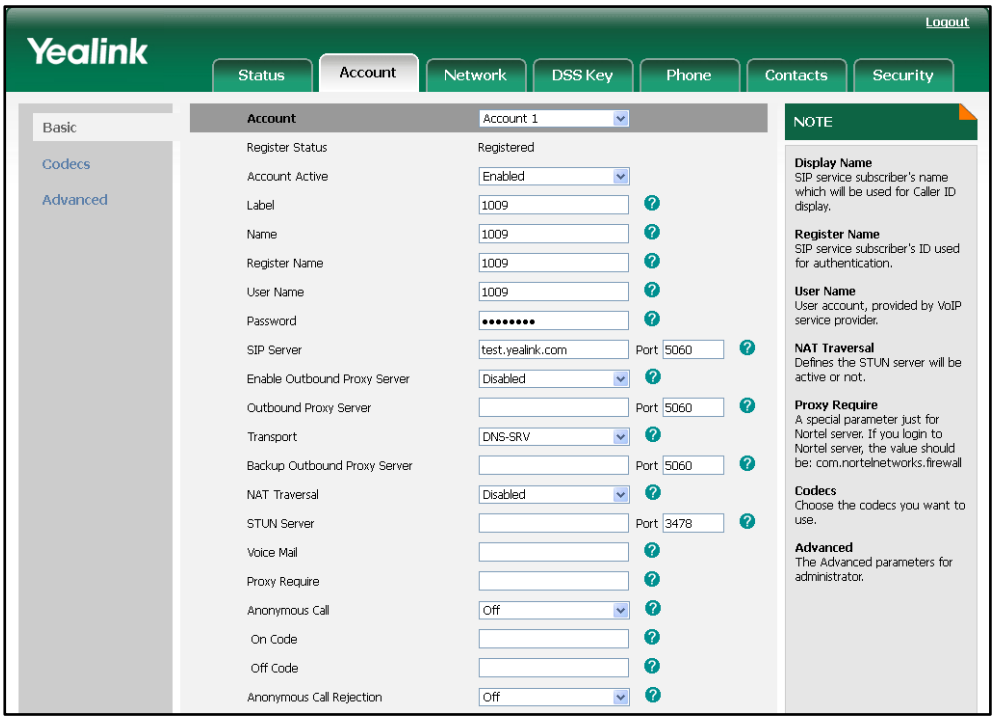
DNS SRV query can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the DNS SRV query on the IP phone. For more information, refer to Server Redundancy on page 292.
Local	Web User Interface	Configure the DNS SRV query on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the DNS SRV query via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Select **DNS-SRV** from the pull-down list of **Transport**.



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

LLDP

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol. It allows IP phones to receive and/or transmit device-related information to directly connected devices on the network that are also using the protocol, and store the information that is learned 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 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 provides the following capabilities for the IP phone:

- Capabilities Discovery — allows LLDP-MED endpoints to determine the capabilities that the connected device supports and has enabled.
- Network Policy — provides voice VLAN configuration to notify a device which VLAN to use and QoS-related configuration for voice data. It provides a "plug and play" network environment.
- Power Management — provides information related to how the device is powered,

power priority, and how much power the device needs.

- Inventory Management — provides a means to effectively manage device and the attributes of the device such as model number, serial number and software revision.

TLVs supported by the IP phone 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. The default value is 120s.
	End of LLDPDU	Marks end of LLDPDU.
Optional TLVs	System Name	Name assigned to the IP phone. The default value is "yealink".
	System Description	Description of the IP phone. The default value is "yealink".
	System Capabilities	The supported and enabled capabilities of phone. The supported capabilities are Bridge, Telephone and Router. The enabled capabilities are Bridge and Telephone by default.
	Port Description	Description of port that sent data unit. The default value is "WAN PORT".
IEEE Std 802.3 Organizationally Specific TLV	MAC/PHY Configuration/Status	Duplex and bit rate settings of the IP phone. The Auto Negotiation is supported and enabled by default. The advertised capabilities of PMD. Auto-Negotiation are: 100BASE-TX (full duplex mode), 100BASE-TX (half duplex mode), 10BASE-T (full duplex mode), 10BASE-T (half duplex mode).
TIA Organizationally Specific TLVs	Media Capabilities	The MED device type of the IP phone and the supported LLDP-MED TLV type can be encapsulated in LLDPDU. The supported LLDP-MED TLV types are:

TLV Type	TLV Name	Description
		LLDP-MED Capabilities, Network Policy, Extended Power via MDI-PD, 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 phone.
	Inventory – Firmware Revision	Firmware revision of phone.
	Inventory – Software Revision	Software revision of phone.
	Inventory – Serial Number	Serial number of phone.
	Inventory – Manufacturer Name	Manufacturer name of phone. The default value is "yealink".
	Inventory – Model Name	Model name of phone.
	Asset ID	Assertion identifier of phone. The default value is "asset".

Procedure

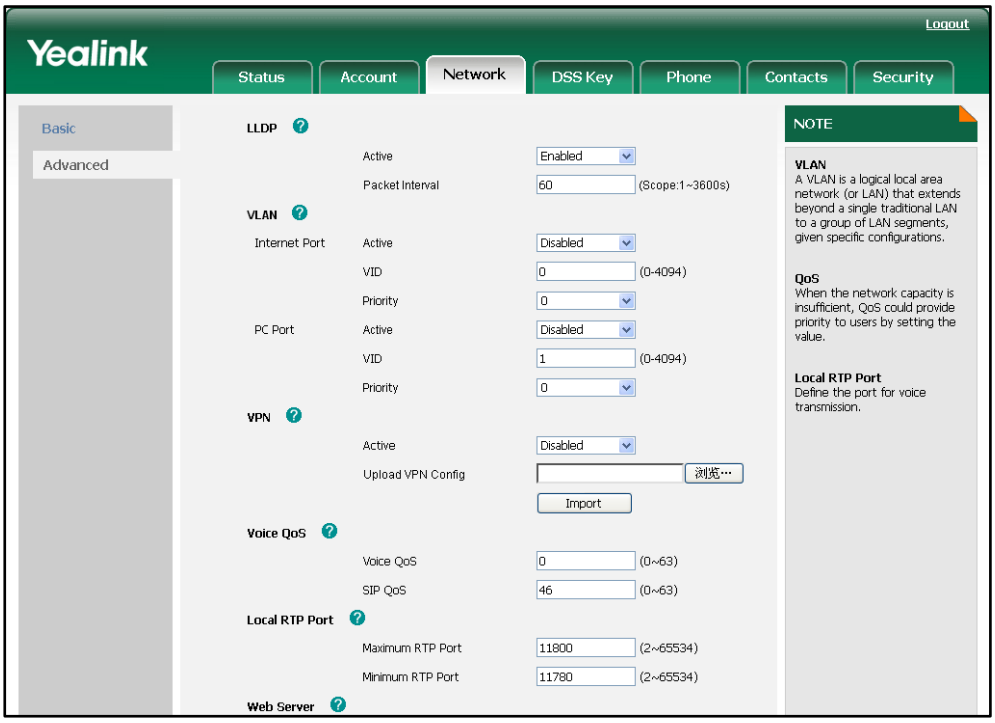
LLDP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the LLDP feature. For more information, refer to LLDP on page 292.
Local	Web User Interface	Configure the LLDP feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm

To configure LLDP via web user interface:

1. Click on **Network**->**Advanced**.
2. In the **LLDP** field, select the desired value from the pull-down list of **Active**.
3. Enter the desired time interval (in seconds) in the **Packet Interval** field.

The valid values range from 1 to 3600.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

5. 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 on the IP phone, the IP phone will tag all packets from these ports with the VLAN identifier. 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.

The IP phones support to configure the VLAN information either manually or dynamically using the LLDP feature. For more information on LLDP, refer to LLDP on page 157.

Note The VLAN information in the received LLDP packets will override the manual configuration.

Procedure

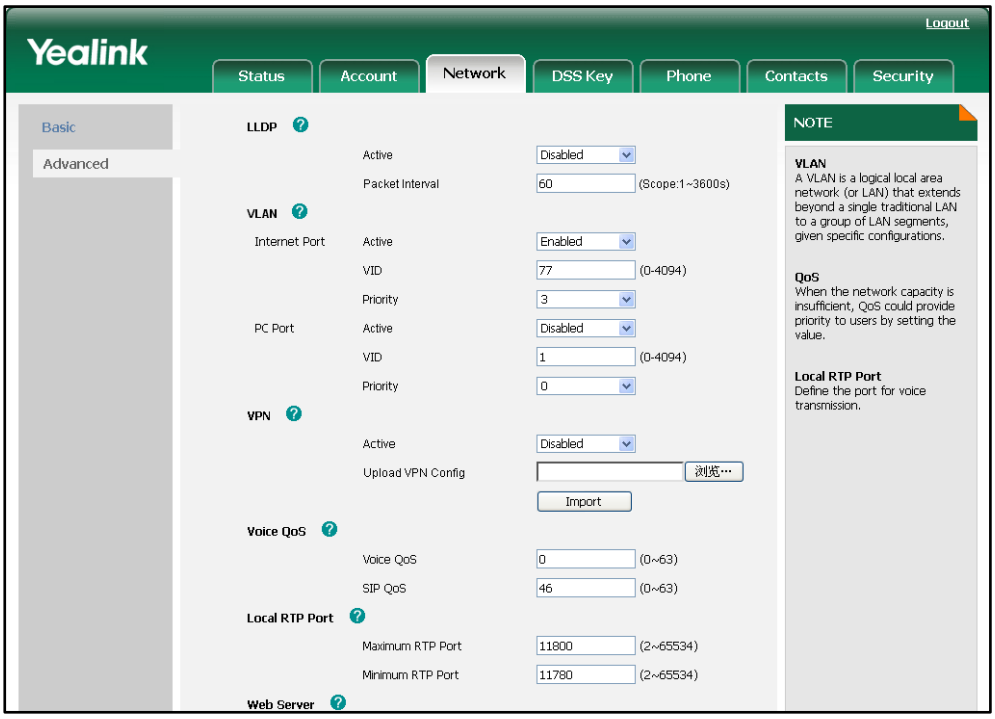
VLAN can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure VLAN for the Internet port. For more information, refer to VLAN on page 293. Configure VLAN for the PC port. For more information, refer to VLAN on page 293.
Local	Web User Interface	Configure VLAN for the Internet port and PC port. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm
	Phone User Interface	Configure VLAN for the Internet port and PC port.

To configure VLAN for Internet port via web user interface:

1. Click on **Network**->**Advanced**.
2. Select the desired value from the pull-down list of **Internet Port Active**.
3. Enter the VLAN ID (0-4094) in the **VID** field.

4. Select the desired value (0-7) from the pull-down list of **Priority**.



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

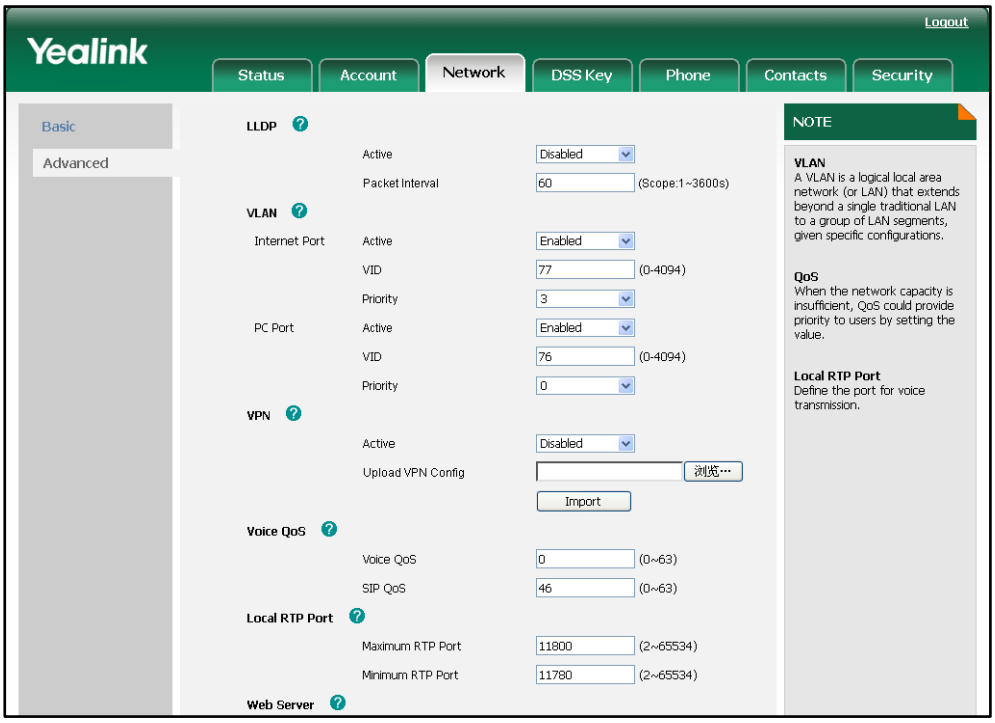
   The web user interface pops up a dialog box to prompt reboot to make the settings effective.

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

To configure VLAN for PC port via web user interface:

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **PC Port Active**.
3. Enter the VLAN ID (0-4094) in the **VID** field.

4. Select the desired value (0-7) from the pull-down list of **Priority**.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

To configure VLAN for Internet port (or PC port) via phone user interface:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**VLAN**->**WAN Port** (or **PC Port**).
2. Press ◄ or ► , or the **Switch** soft key to select the desired value from the **VLAN Status** field.
3. Enter the VLAN ID (0-4094) in the **VID Number** field.
4. Enter the priority value (0-7) in the **Priority** field.
5. Press the **Save** soft key to accept the change

   The IP phone reboots automatically to make the settings effective after a period of time.

VPN

VPN (Virtual Private Network) is a secured private network connection built on top of public telecommunication infrastructure, such as the Internet. It provides remote offices or individual users with secure access to their organization's network. VPN has become more prevalent due to the benefits: scalability, reliability, convenience and security.

There are two types of VPN: remote-access VPN (connecting an individual device to a network) and site-to-site VPN (connecting two networks together). Remote-access VPNs allow employees to access their company's intranet from home or outside the office, and site-to-site VPNs allow employees in geographically separated offices to share one cohesive virtual network. VPN systems can be also classified by the protocols used to tunnel the traffic. VPNs provide security through tunneling protocols: IPSec, SSL, L2TP and PPTP.

The IP phones support SSL VPN. SSL VPN provides remote access VPN capabilities through SSL. OpenVPN is a full featured SSL VPN software solution that creates secure connections in remote access facilities. It is designed to work with the TUN/TAP virtual networking interface. TUN and TAP are virtual network kernel devices. TAP simulates a link layer device and provides a virtual point-to-point connection. TUN simulates a network layer device and provides a virtual network segment. The IP phones support using OpenVPN to achieve the VPN feature. To prevent disclosure of private information, tunnel endpoints must authenticate each other before secure VPN tunnel is established. After the VPN feature is configured properly on the IP phone, the IP phone acts as a VPN client and uses the certificates to authenticate the VPN server.

To use the VPN feature on the IP phone, the compressed package of VPN-related files should be uploaded to the IP phone in advance. The file format of the compressed package must be .tar. The VPN-related files are: certificates (ca.crt, client.crt and client.key) and configuration file (vpn.cnf) of VPN client. Ask your network administrator for the tar package.

Procedure

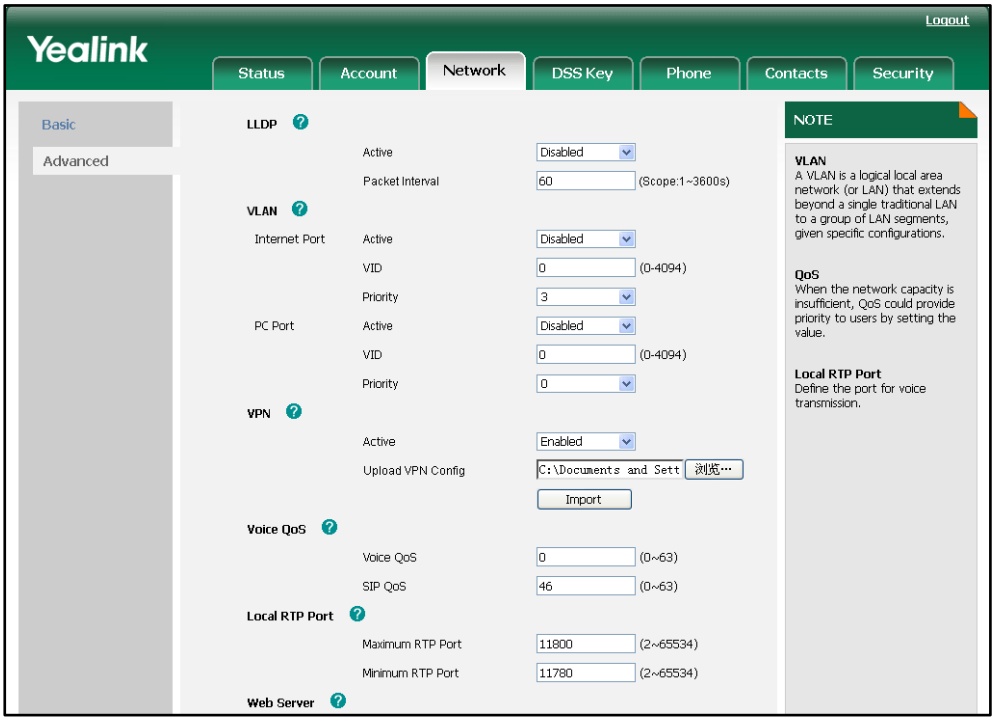
VPN can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the OpenVPN feature and upload the tar package to the IP phone. For more information, refer to VPN on page 295.
Local	Web User Interface	Configure the OpenVPN feature and upload the tar package to the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm
	Phone User Interface	Configure the OpenVPN feature under Advanced Settings.

To upload the tar package to the IP phone and configure VPN via web user interface:

1. Click on **Network**->**Advanced**.

2. Click **Browse** to locate the tar package from the local system.
3. Click **Import** to import the tar package.



4. Select the desired value from the pull-down list of **VPN Active** after importing.
5. Click **Confirm** to accept the change.

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.
6. Click **OK** to reboot the IP phone.

To configure VPN via phone user interface after uploading the tar package:

1. Press **Menu**->**Settings**->**Advanced Settings** (password: admin) ->**Network**->**VPN**.
2. Press ◄ or ►, or the **Switch** soft key to select the desired value from the **VPN Active** field.
3. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Quality of Service

Quality of Service (QoS) is the ability to provide different priorities to different packets in the network that allows 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

consider when configuring a modern QoS implementation, these include: bandwidth, delay, jitter and loss.

QoS provides better network service by providing 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 the 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 supported 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. Each DSCP specifies a particular per-hop behavior (PHB) that is applied to a packet. A PHB refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet.

There are four standard PHBs available to construct a DiffServ-enabled network and achieve QoS:

- **Class Selector PHB** – is backwards compatible with IP precedence. Class Selector code points are of the form "xxx000". The first three bits are the IP precedence bits. These PHBs retain almost the same forwarding behavior as nodes that implement IP-precedence based classification and forwarding.
- **Expedited Forwarding PHB** – is 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 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. The issue is how to guarantee that packet traffic will 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. The IP phones support the DiffServ model of QoS.

Voice QoS

For VoIP transmissions to be intelligible to the receiver, voice packets should not be dropped, excessively delayed, or suffer varying delay. DiffServ model can guarantee high-quality voice transmission when the voice packets are configured 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, the SIP packets emanating from the IP phone should be configured a high transmission priority.

You can specify DSCPs for voice packets and SIP packets respectively.

Note The DSCP value of voice traffic in the received LLDP packet will override the manual configuration.

Procedure

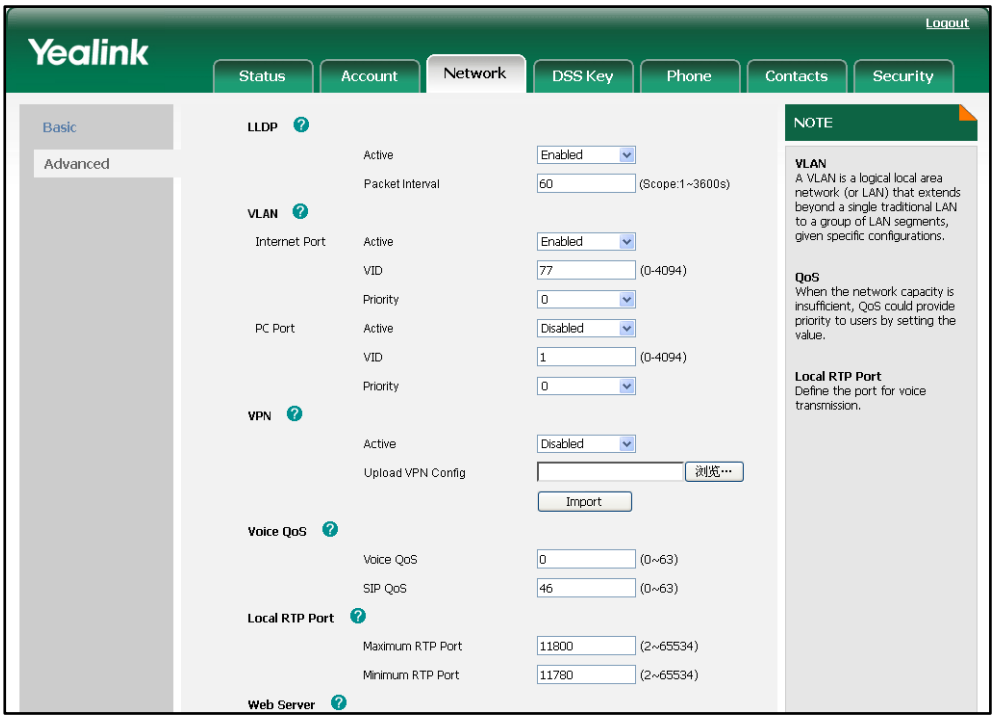
DSCPs for voice packets and SIP packets can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the DSCPs for voice packets and SIP packets. For more information, refer to QOS on page 296.
Local	Web User Interface	Configure the DSCPs for voice packets and SIP packets. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm

To configure DSCPs for voice packets and SIP packets via web user interface:

1. Click on **Network**->**Advanced**.
2. Enter the desired value (0-63) in the **Voice QoS** field.

3. Enter the desired value (0-63) in the **SIP QoS** field.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.

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

Network Address Translation

Network Address Translation (NAT) is essentially a translation table that maps public IP address and port combinations to private IP address and port combinations. This reduces the need for a large amount of public IP addresses. The NAT feature ensures security since each outgoing or incoming request must 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. It is typically required for client-to-client networking applications, especially for VoIP deployments. STUN is one of the NAT traversal techniques supported by the IP phones.

STUN (Simple Traversal of UDP over NATs)

STUN is a network protocol, which is 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 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, which sends exploratory STUN messages to the STUN server. The server uses those messages to determine the public IP address and port used, and then informs the client.

The NAT traversal and STUN server are configurable on a per-account basis.

Procedure

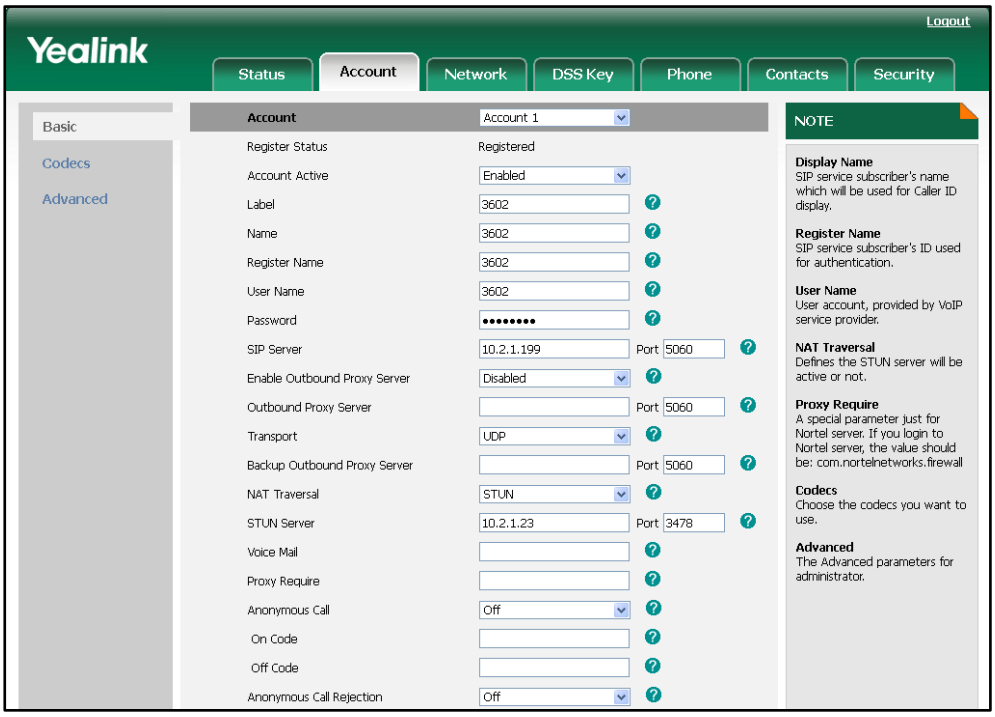
NAT traversal and STUN server can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the NAT traversal and STUN server on the IP phone. For more information, refer to Network Address Translation on page 297.
Local	Web User Interface	Configure the NAT traversal and STUN server on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the NAT traversal and STUN server via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **STUN** from the pull-down list of **NAT Traversal**.

4. Enter the IP address or the domain name in the **STUN Server** field.



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

802.1X Authentication

IEEE 802.1X authentication is an IEEE Standard for Port-based Network Access Control (PNAC). It is 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. 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, to 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.

The IP phone only supports using the EAP-MD5 for 802.1X authentication.

Procedure

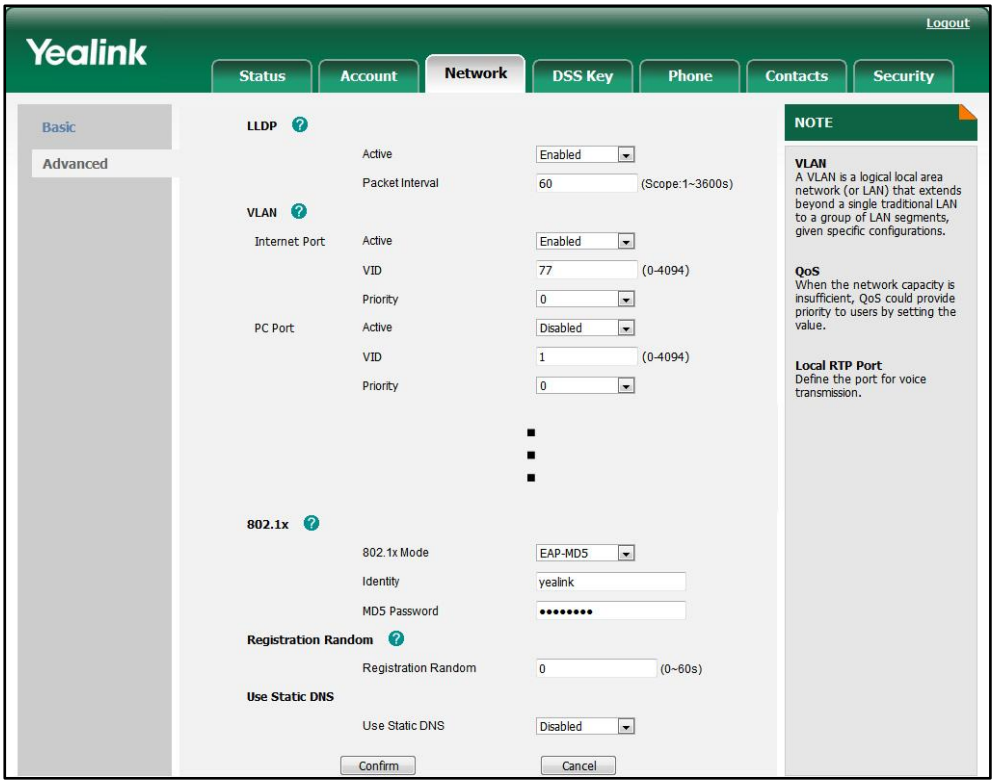
802.1X authentication can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the 802.1X authentication on the IP phone. For more information, refer to 802.1X on page 298.

Local	Web User Interface	Configure the 802.1X authentication on the IP phone. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Network-Adv.htm
	Phone User Interface	Configure the 802.1X authentication on the IP phone.

To configure the 802.1X via web user interface:

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **802.1x Mode**.
3. Enter the username for authentication in the **Identity** field.
4. Enter the password for authentication in the **MD5 Password** field.



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

   The web user interface pops up a dialog box to prompt that the settings will take effect after reboot.
6. Click **OK** to reboot the IP phone.

To configure the 802.1X via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) **->Network->802.1x**.

2. Press ◄ or ►, or the **Switch** soft key to select the desired value from the **802.1x Mode** field.
3. (If EAP-MD5 is selected) Enter the username for authentication in the **Identity** field.
4. (If EAP-MD5 is selected) Enter the password for authentication in the **MD5 Password** field.
5. Press the **Save** soft key to accept the change.

   The IP phone reboots automatically to make the settings effective after a period of time.

Configuring Audio Features

This chapter provides information for making configuration changes for the following audio features:

- Audio Codecs
- Acoustic Clarity Technology

Audio Codecs

CODEC is an abbreviation of COmpress-DECompress. It is 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 transmission of the audio.

The default codecs used on the IP phone 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 u-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, the IP phone also supports the codecs: G723_53, G723_63, G726_16, G726_24, G726_32, G726_40, iLBC_13_3 and iLBC_15_2. You can configure the preferred codecs to use on a per-account basis instead of using the default codecs. You can also configure the priorities for the enabled codecs. 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 Method	Priority	RTPmap
PCMU	Configuration Files Web User Interface	1	0
PCMA	Configuration Files Web User Interface	2	8
G729	Configuration Files Web User Interface	3	18

Codec	Configuration Method	Priority	RTPmap
G722	Configuration Files Web User Interface	4	9
G723_53	Configuration Files Web User Interface	0	4
G723_63	Configuration Files Web User Interface	0	4
G726_16	Configuration Files Web User Interface	0	112
G726_24	Configuration Files Web User Interface	0	102
G726_32	Configuration Files Web User Interface	0	2
G726_40	Configuration Files Web User Interface	0	104
iLBC_13_3	Configuration Files	0	97
iLBC_15_2	Configuration Files	0	97

Packetization Time

Ptime (Packetization Time) is measurement of the duration (in milliseconds) of the audio data in each RTP packet sent to the destination, and hence it defines how much network bandwidth is used for transfer of the RTP stream. 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.

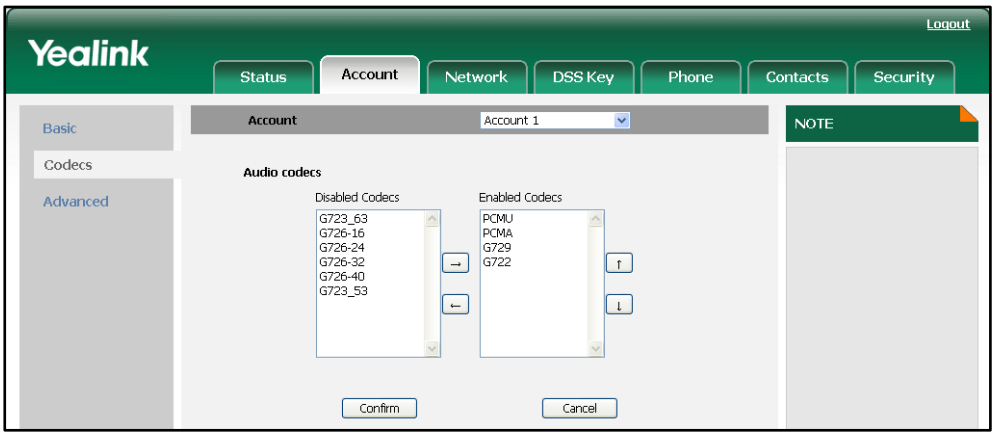
Configuration File	<MAC>.cfg	Configure the codecs to use on a per-account basis. Configure the priority and rtpmap for the enabled codec. For more information, refer to Audio Codecs on page 299. Configure the ptime. For more information, refer to

		Audio Codecs on page 299.
Local	Web User Interface	Configure the codecs and adjust the priority of the enabled codecs on a per-account basis. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Codec.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Configure the ptime. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the codecs and adjust the priority of the enabled codecs on a per-account basis via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Codecs**.
4. In the **Disabled Codecs** box, select the desired codec and click → to move to the **Enabled Codecs** box.
5. In the **Enabled Codecs** box, select the undesired codec and click ← to move to the **Disabled Codecs** box.

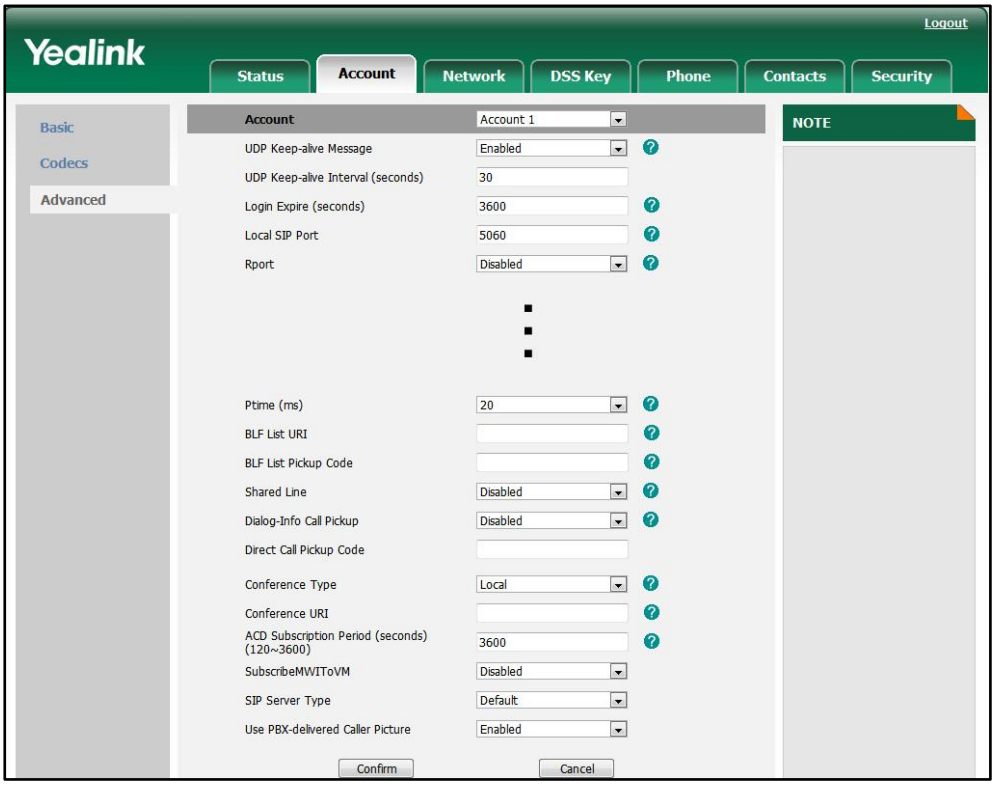
6. Click [↑] or [↓] to adjust the priorities of the enabled codecs.



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

To configure the Ptime on a per-account basis via web user interface:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **Ptime (ms)**.



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

Acoustic Clarity Technology

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. The IP phone employs advanced AEC for hands-free operation. Echo cancellation is done using the echo canceller.

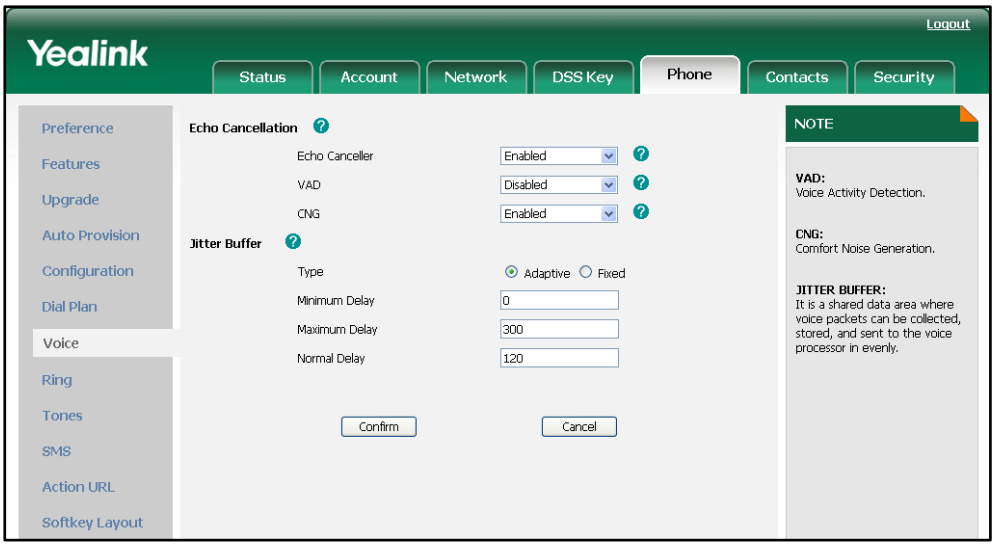
Procedure

AEC can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the AEC feature. For more information, refer to Acoustic Echo Cancellation on page 302.
Local	Web User Interface	Configure the AEC feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Voice.htm

To configure AEC via web user interface:

1. Click on **Phone->Voice**.
2. Select the desired value from the pull-down list of **Echo Canceller**.



3. Click **Confirm** to accept 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 can also be used to 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 on network bandwidth.

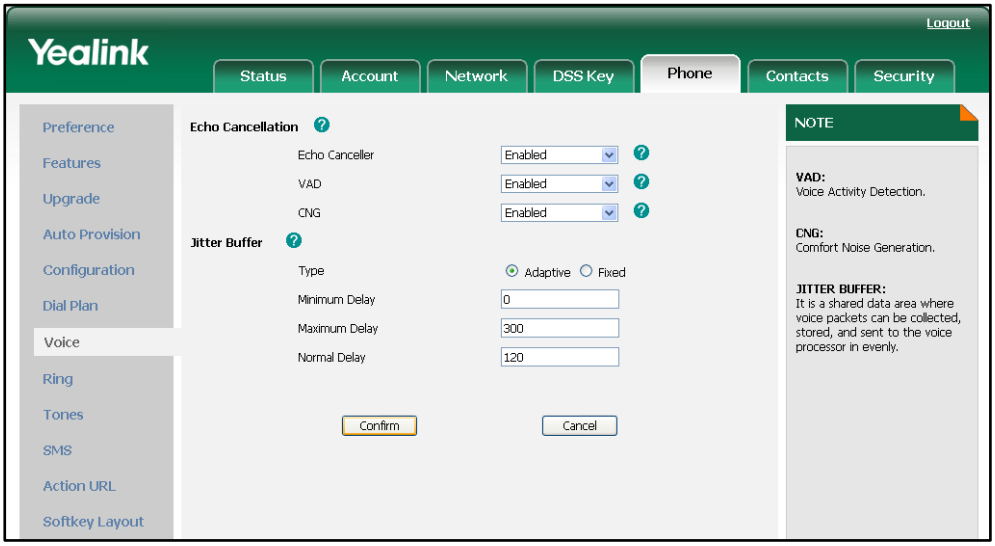
Procedure

VAD can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the VAD feature. For more information, refer to Voice Activity Detection on page 303.
Local	Web User Interface	Configure the VAD feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Voice.htm

To configure VAD via web user interface:

1. Click on **Phone->Voice**.
2. Select the desired value from the pull-down list of **VAD**.



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

Comfort Noise Generation

Comfort Noise Generation (CNG) is used to generate background noise for voice communications during periods of silence that occur during the conversation. It is part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly determines 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.

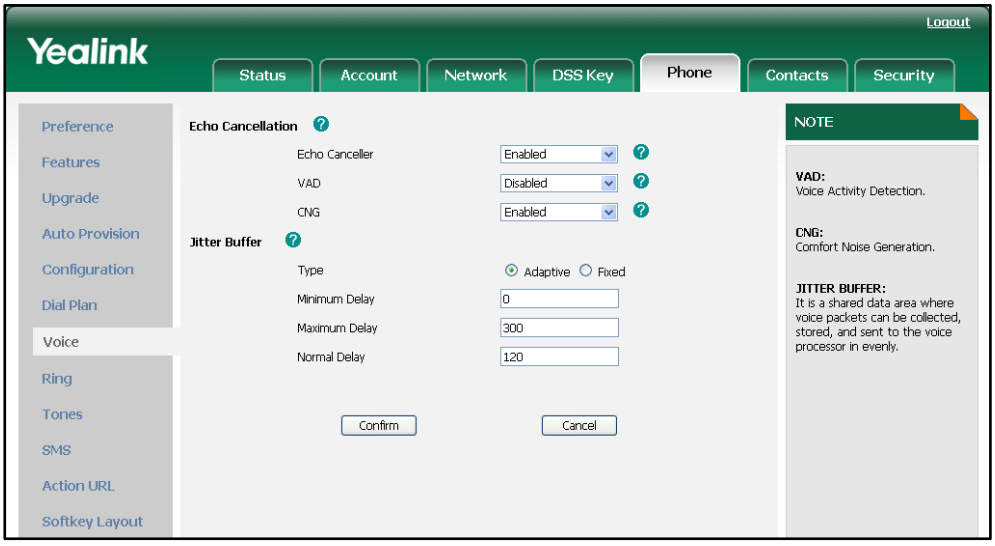
Procedure

CNG can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the CNG feature. For more information, refer to Comfort Noise Generation on page 303.
Local	Web User Interface	Configure the CNG feature. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Voice.htm

To configure CNG via web user interface:

1. Click on **Phone->Voice**.
2. Select the desired value from the pull-down list of **CNG**.



3. Click **Confirm** to accept 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 evenly spaced intervals. Jitter is variations in packet arrival time, can occur because of network congestion, timing drift or route changes. The jitter buffer, which is 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. There are two types of jitter buffers: static and dynamic. The IP phones support these two types of jitter buffer. A static jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on the IP phone. The default delay time is 120ms. A dynamic jitter buffer is capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can also be configured on the IP phone. The dynamic jitter buffer is enabled on the IP phone by default and the valid delay time ranges from is 0 to 300ms.

Procedure

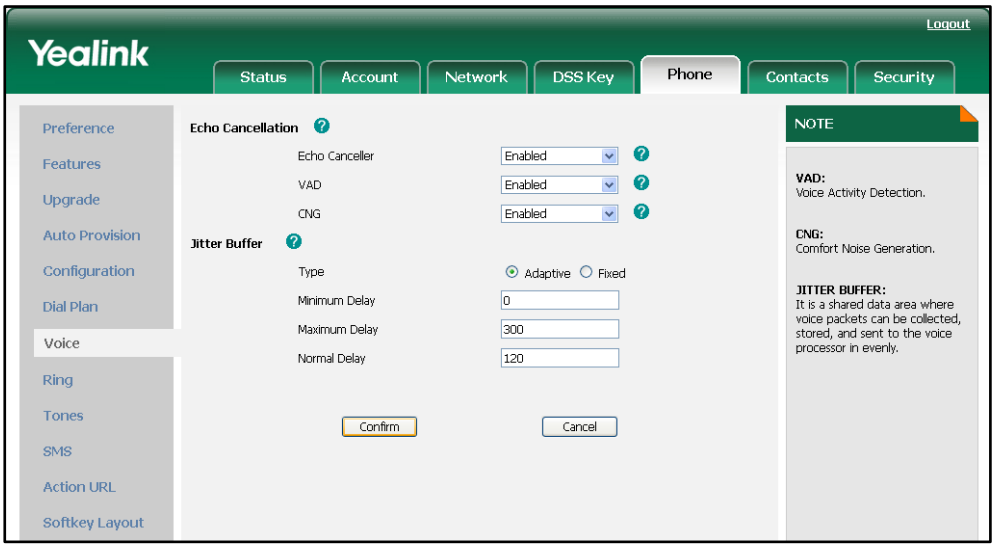
Jitter buffer can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the mode of jitter buffer and the delay time for jitter buffer. For more information, refer to Jitter Buffer on page 303.
Local	Web User Interface	Configure the mode of jitter buffer and the delay time for jitter buffer. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-Voice.htm

To configure Jitter Buffer via web user interface:

1. Click on **Phone**->**Voice**.
2. Mark the desired radio box in the **Type** field.
3. Enter the minimum delay time for adaptive jitter buffer in the **Minimum Delay** field.
4. Enter the maximum delay time for adaptive jitter buffer in the **Maximum Delay** field.

5. Enter the fixed delay time for fixed jitter buffer in the **Normal Delay** field.



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

Configuring Security Features

This chapter provides information for making configuration changes for the following security-related features:

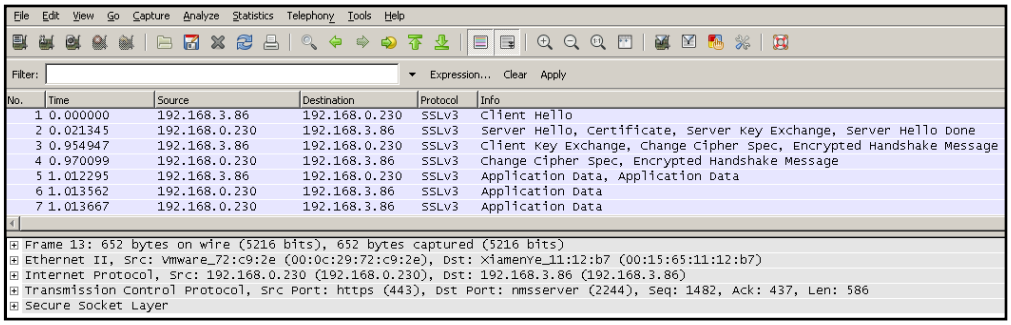
- Transport Layer Security
- Secure Real-Time Transport Protocol
- Encrypting Configuration Files

Transport Layer Security

The TLS protocol is a commonly-used protocol for providing communications privacy and managing the security of message transmission. The TLS allows the IP phone 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.

The TLS protocol is composed of two layers: the TLS Record Protocol and the 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 following figure illustrates the TLS messages exchanged between the IP phone and TLS server to establish an encrypted communication channel:



Step1: IP phone sends "Client Hello" message proposing SSL options.

Step2: Server responds with "Server Hello" message selecting the SSL options, sends its public key information in "Server Key Exchange" message and concludes its part of the negotiation with "Server Hello Done" message.

Step3: IP phone sends session key information (encrypted with 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.

The IP phone can encrypt SIP with TLS, which is called SIPS. When TLS is enabled for an account, the SIP message of this account will be encrypted, and a lock icon appears on the phone LCD screen after the successful TLS negotiation. You can specify the IP phone to encrypt the SIP signal using the RC4 encryption algorithm.

In order to use the TLS on the IP phone, you need to perform the following steps:

- Uploading certificates to the IP phone
- Configuring the IP phone to use the TLS

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 the server is trusted based on the trusted certificates list. You can upload up to 10 trusted certificates to the IP phone.
- **Server Certificate**: When the other clients request a TLS connection with the IP phone, the IP phone sends the server certificate to the clients for authentication. You can only upload one server certificate to the IP phone. The old server certificate will be overwritten by the new one.

You can configure the "Only Accepted Trusted Certificates" feature on the IP phone. If enabled, the IP phone will check the certificate sent by the server and only accept the certificates listed in the Trusted Certificates list. You can configure the TLS on a per-account basis.

Procedure

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

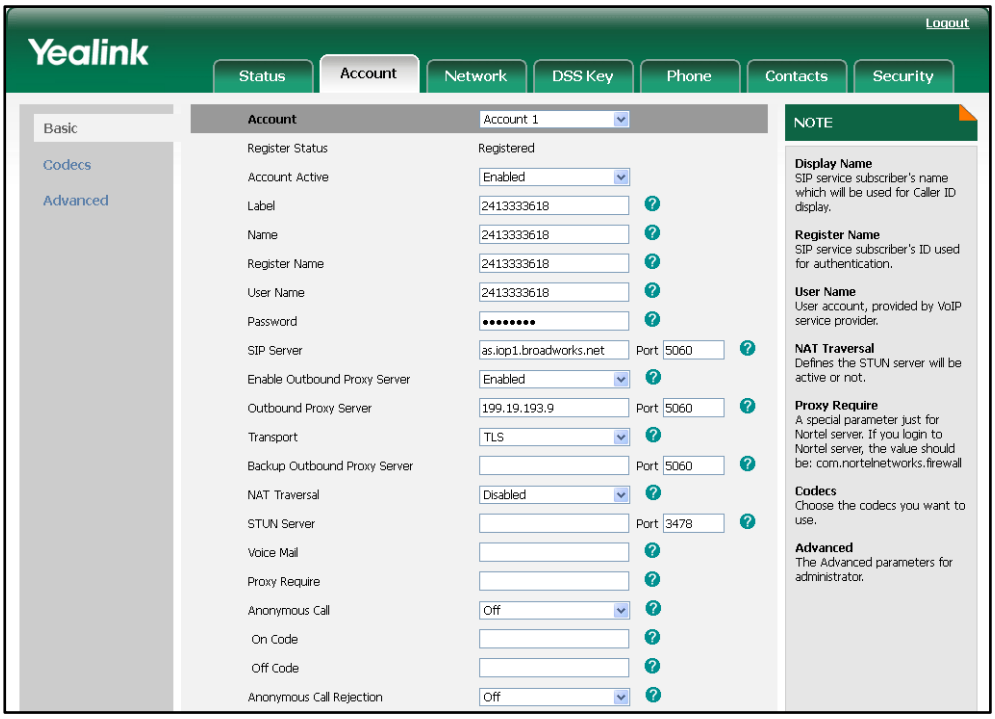
Configuration File	<MAC>.cfg	Configure the IP phone to use TLS and authenticate the connected server. For more information, refer to TLS on page 305. Specify the IP phone to encrypt the SIP signal using RC4 encryption algorithm. For more information, refer to TLS on page 305.

	<y0000000000xx>.cfg	Upload certificates to the IP phone. For more information, refer to Uploading Certificates on page 306.
Local	Web User Interface	Configure the IP phone to use TLS. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Specify the IP phone to encrypt the SIP signal using RC4 encryption algorithm. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2. Upload the trusted certificate. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=TrustCerts.htm Upload the server certificate. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=ServerCerts.htm

To configure TLS via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.

3. Select **TLS** from the pull-down list of the **Transport**.

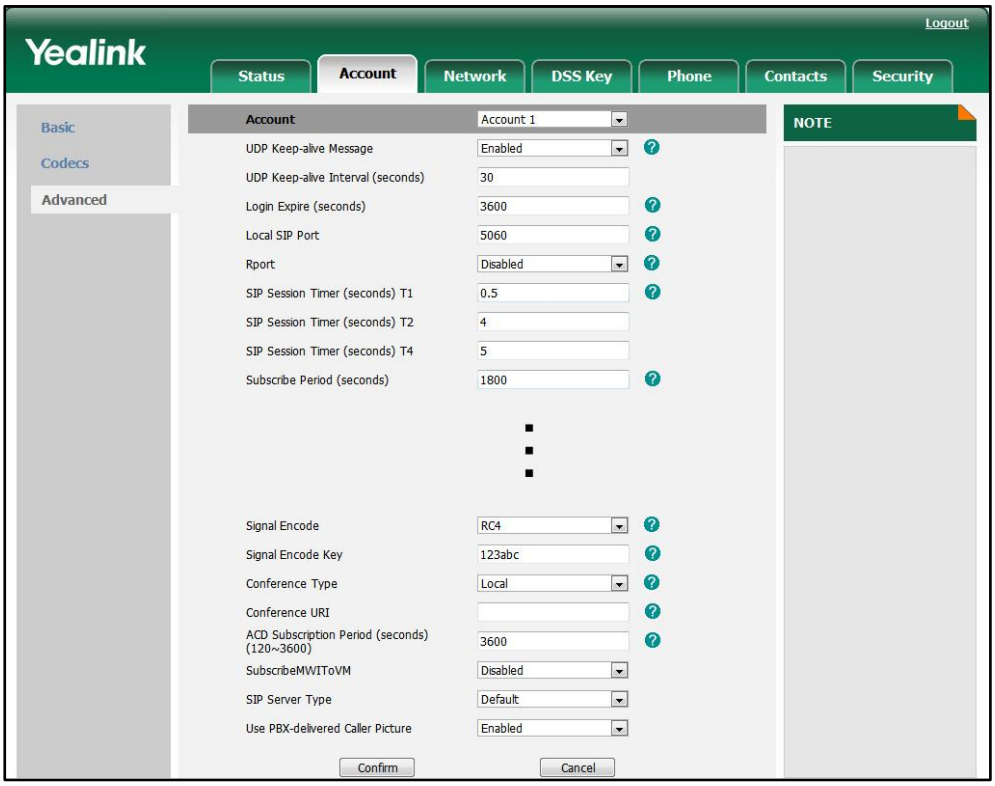


4. Click **Confirm** to accept the change.

To Specify the IP phone to encrypt the SIP signal using RC4 encryption algorithm:

1. Click on **Account->Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select **RC4** from the pull-down list of **Signal Encode**.

5. Enter the desired key in the **Signal Encode Key** field.

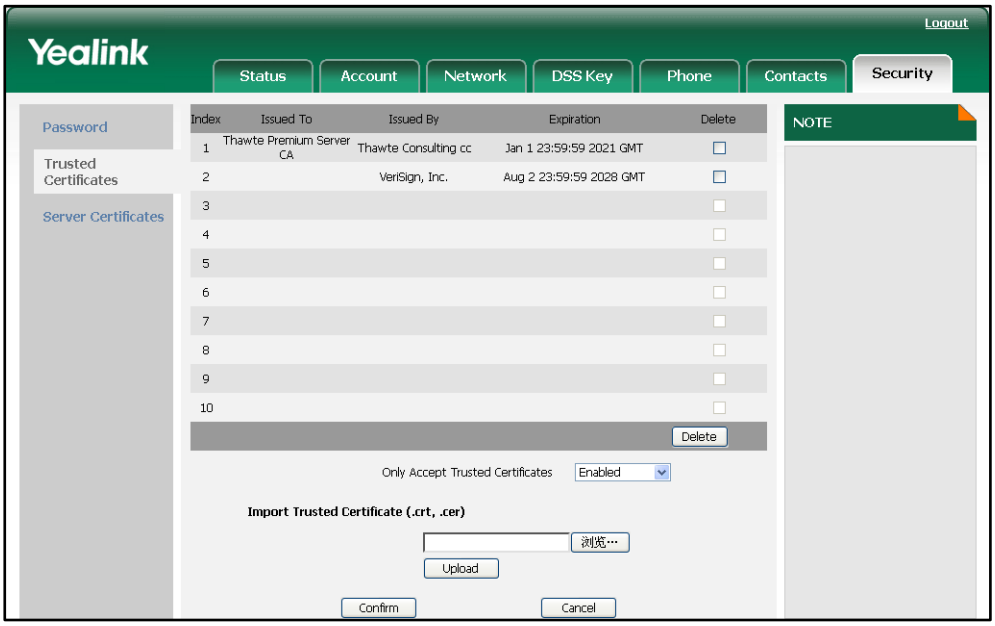


6. Click **Confirm** to accept the change.

To configure Only Accepted Trusted Certificates via web user interface:

1. Click on **Security->Trusted Certificates**.

2. Select the desired value from the pull-down list of **Only Accept Trusted Certificates**.



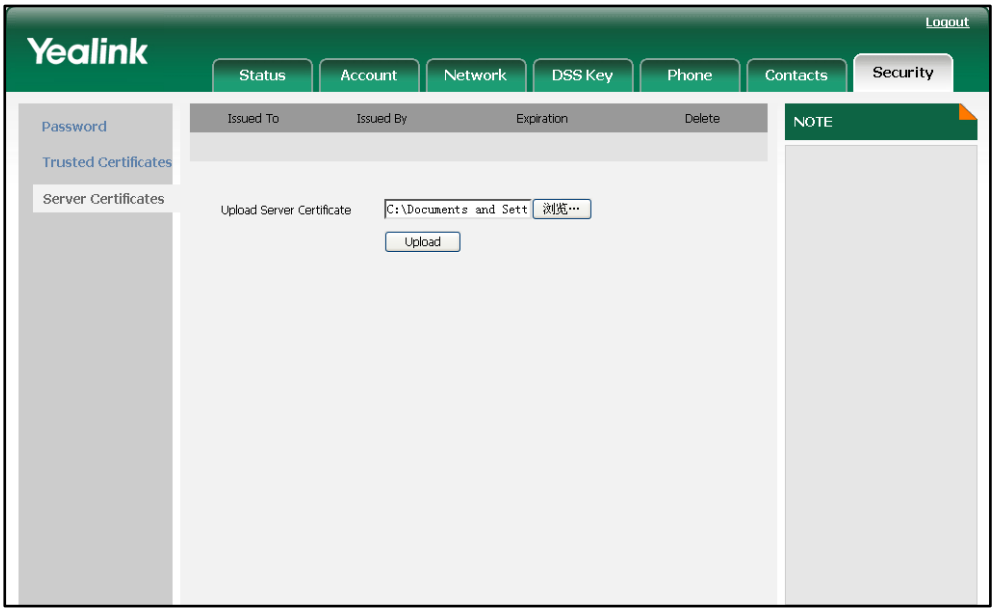
3. Click **Confirm** to accept the change.

To upload the trusted certificate via web user interface:

1. Click on **Security->Trusted Certificates**.
2. Click **Browse** to select the trusted certificate (*.crt or *.cer) from your local system.
3. Click **Upload** to upload the trusted certificate.

To upload the server certificate via web user interface:

1. Click on **Security->Server Certificates**.
2. Click **Browse** to select the server certificate (*.pem) from your local system.



3. Click **Upload** to upload the server certificate.

The web user interface pops up the dialog box to prompt "Rebooting, please wait...".

Secure Real-Time Transport Protocol

Secure Real-Time Transport Protocol (SRTP) provides means of encrypting the RTP streams during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call should enable the SRTP feature simultaneously. When this feature is enabled on both phones, the IP phone will negotiate with the other phone what type of encryption to utilize for the session. 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.

The sample of the RTP encryption algorithm carried in the SDP of the INVITE message for reference:

```
m=audio 11780 RTP/SAVP 0 8 18 9 101
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NzFlNTUwZDk2OGVlOTc3YzNkYTkwZWVkMTM1YWFj
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:101 0-15
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

The callee receives the INVITE message with the RTP encryption algorithm. The callee answers the call and responses with a 200 OK message carrying the negotiated RTP encryption algorithm.

The sample of the RTP encryption algorithm carried in the SDP of the 200 OK message for reference:

```
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
```

You can configure the SRTP feature on a per-account basis. When SRTP is enabled on both phones, the RTP streams will be encrypted, and a lock icon appears on the LCD screen of each IP phone after the successful negotiation.

Note If you enable SRTP, then you should also enable TLS. This ensures the security of SRTP encryption. For more information on TLS, refer to Transport Layer Security on page 183 .

Procedure

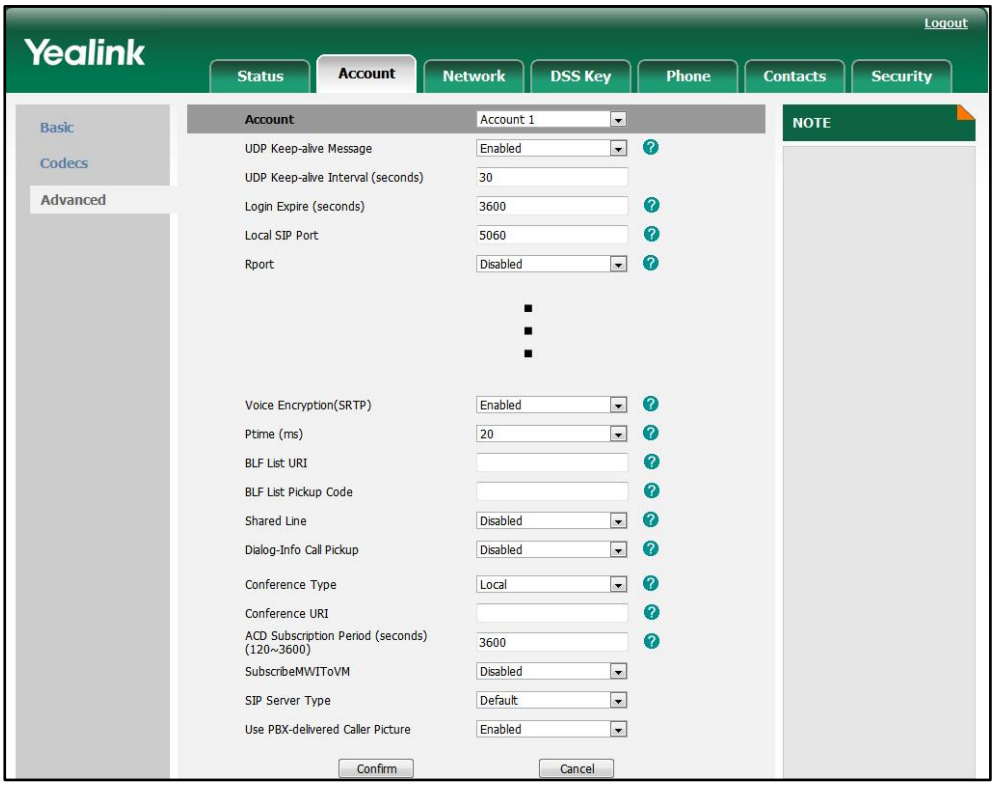
SRTP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the SRTP feature on a per-account basis. For more information, refer to SRTP on page 307.
Local	Web User Interface	Configure the SRTP feature on a per-account basis. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Account-Adv.htm&acc=<x> For T38G, x ranges from 0 to 5. For T32G, x ranges from 0 to 2.

To configure the SRTP feature via web user interface:

1. Click on **Account**->**Basic**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.

4. Select the desired value from the pull-down list of **Voice Encryption (SRTP)**.



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

Encrypting Configuration Files

The IP phone can download the encrypted configuration files from the provisioning server to protect against unauthorized access and tampering of sensitive information (i.e., login passwords, registration information). Configuration files can be encrypted using a command line tool. The encryption algorithm is AES 128. From a Microsoft Windows command line, you can use the Yealink-supplied encryption tool called "EncryptUtilityWindows.exe" to encrypt the <y0000000000xx>.cfg and <MAC>.cfg files respectively.

Note

Yealink also supplies an encryption tool (EncryptUtilityLinux.exe) to support Linux platforms if required.

You can also encrypt the configuration files using the Yealink Configuration Conversion Tool. For more information, refer to the document "Yealink Configuration Conversion Tool User Guide".

The filename extension of the encrypted configuration files must be .cfg. The Common AES key is used to encrypt and decrypt the <y0000000000xx>.cfg file and the MAC-Oriented AES key is used to encrypt and decrypt the <MAC>.cfg file. The AES keys must be 16 characters. The AES key should be configured on the IP phone for

decrypting before provisioning.

Procedure to Encrypt Configuration Files

To encrypt the <y0000000000xx>.cfg file:

1. Place the "EncryptUtilityWindows.exe" tool and <y0000000000xx>.cfg file to the same directory (i.e., D:\).
2. Open a command line window application (i.e., DOS window).
3. Enter the following command, and then press the <Enter> key.

```
D:EncryptUtilityWindows.exe 123456789abcdef0 e F:\y000000000038.cfg
D:\y000000000038.cfg
```

```
#D:EncryptUtilityWindows.exe <a 16-character secret key> e <a new
directory and file name of the encrypted configuration file> <the
directory and file name of the original configuration file>
```

4. Place the encrypted configuration file to the root directory of the provisioning server.

The way for encrypting the <MAC>.cfg file is the same as the <y0000000000xx>.cfg file. After encrypting the configuration files, you need to configure the AES keys on the IP phone.

Procedure

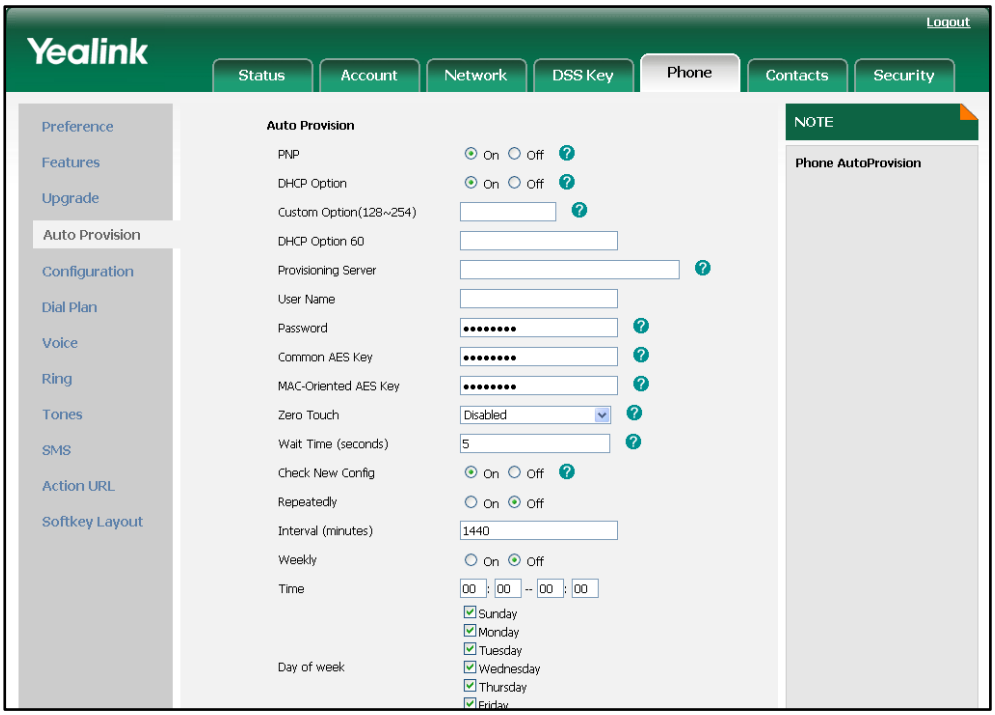
AES keys can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the AES keys. For more information, refer to Configuring AES Keys on page 308.
Local	Web User Interface	Configure the AES keys. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-AutoProvision.htm

To configure the AES keys via web user interface:

1. Click on **Phone->Auto Provision**.

2. Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.



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

Upgrading the Firmware

This chapter provides information about upgrading the IP phone firmware. There are two methods used to upgrade the firmware on the IP phone:

- Upgrade the firmware manually from the local system
- Upgrade the firmware from the provisioning server automatically.

The following table lists the associated firmware for each IP phone model:

IP Phone Model	Associated Firmware
SIP-T38G	38.x.x.x.rom
SIP-T32G	32.x.x.x.rom

Note You can download the latest firmware at: http://www.yealink.com/Support.aspx.

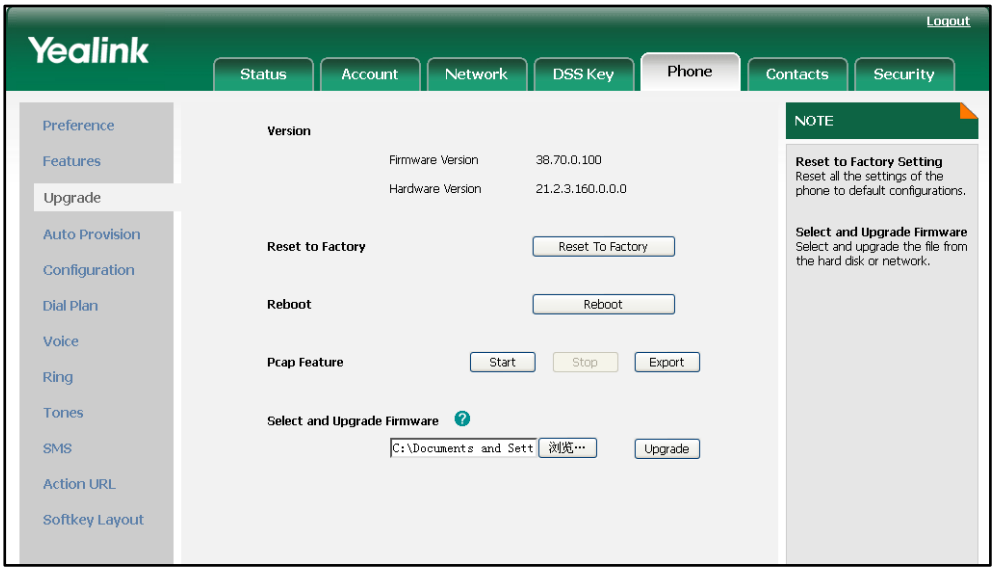
Upgrade via Web User Interface

To manually upgrade firmware via web user interface, you need to store the firmware to your local system in advance.

To upgrade the firmware manually via web user interface:

1. Click on **Phone**->**Upgrade**.
2. Click **Browse**.
3. Select the firmware from the local system.
4. Click **Upgrade**.

The web user interface pops up the dialog box to prompt "Firmware of the SIP PHONE will be updated. It will take several minutes to complete. So, please don't power off!".



5. Click **OK** to confirm the upgrading.

Note

Do not unplug the network and power cables when the IP phone is upgrading the firmware.

Do not close the browser when the IP phone is upgrading the firmware via web user interface.

Upgrade Firmware from the provisioning server

The IP phones support to use the FTP, TFTP, HTTP, and HTTPS protocols to download the configuration files and firmware from the provisioning server, and then upgrade the firmware automatically.

The IP phones can download the firmware stored on the provisioning server in one of two ways:

- The IP phones check for both configuration files and firmware stored on the provisioning server during booting up.
- The IP phones automatically check for configuration files and firmware at a fixed interval or at specific time.

You can configure the way for the IP phones to check for configuration files and firmware.

Procedure

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

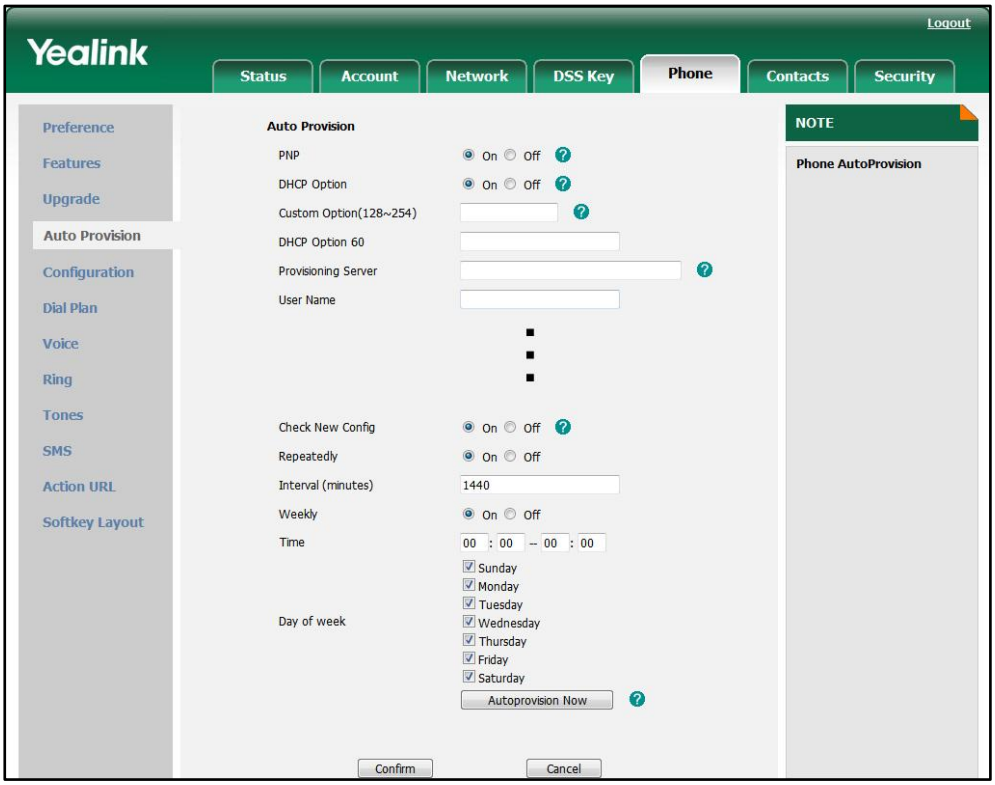
Configuration File	<y0000000000xx>.cfg	Configure the way for the IP phone to check for

		configuration files. Specify the access URL of the firmware. For more information, refer to Upgrading the Firmware on page 308.
Local	Web User Interface	Configure the way for the IP phone to check for configuration files. **Navigate to**: http://<phoneIPAddress>/cgi-bin/cgiServer.exx?page=Phone-AutoProvision.htm

To configure the way for the IP phone to check for new configuration files via web user interface:

1. Click on **Phone**->**Auto Provision**.
2. Mark the desired radio box in **Check New Config** field.
3. Mark the desired radio box in **Repeatedly** field.
4. (If the **Repeatedly On** radio box is marked) Enter the time interval (in minutes) in the **Interval (minutes)** field.
5. Mark the desired radio box in **Weekly** field.
6. (If the **Weekly On** radio box is marked) Enter the desired time in the **Time** field.

7. (If the **Weekly On** radio box is marked) Check the desired checkbox in the **Day of week** field.



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

When the "Check New Config" is set to **On**, the IP phone will check for both firmware and configuration files stored on the provisioning server during booting up.

Resource Files

When configuring some features, you may need to upload resource files to the IP phone. The resources files can be local contact directory, remote phonebook and so on. If the resource file is to be used for all IP phones of the same model, the access URL of the resource file is best specified in the <y0000000000xx>.cfg file. However, if you want to specify the desired phone to use the resource file, the access URL of the resource file should be specified in the <MAC>.cfg file.

This chapter provides the detailed information on how to work with the following resource files and specify the access URL:

- Replace Rule Template
- Dial-now Template
- Softkey Layout Template
- Local Contact File
- Remote XML Phonebook
- Specifying the Access URL of Resource Files

Replace Rule Template

You can create multiple replace rules using the replace rule template. After preparing the replace rule template, you need to place the replace rule template to the root directory of the provisioning server and specify the access URL in the configuration files.

When editing a replace rule template, remember the following:

- <DialRule> indicates the start of a template and </DialRule> indicates the end of a template.
- Create replace rules between <DialRule> and </DialRule>.
- When specifying the desired line(s) to apply the replace rule, the valid values are 0 and line IDs. The digit 0 stands for all lines, multiple line IDs are separated by comma.
- Do not modify the file name.
- The expression syntax in the replace rule template is the same as introduced in the section Creating Dial Plan on page 25.

Procedure

Use the following procedures to customize a replace rule template.

Customizing a replace rule template:

1. Open the template file using an ASCII editor.
2. 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. When leaving it blank, this replace rule will apply to all lines.
3. Specify the values within double quotes.
4. Place this file to the root directory of the provisioning server.

The following is 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="1,2,3"/>
  <Data Prefix="0(.)" Replace="9$1" LineID="2"/>
  <Data Prefix="1009" Replace="05921009" LineID="1"/>
</DialRule>
```

Dial-now Template

You can create multiple dial-now rules using the dial-now template. After preparing the dial-now template, you need to place the dial-now template to the root directory of the provisioning server and specify the access URL in the configuration files.

When editing a dial-now template, remember 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>.
- When specifying the desired line(s) for the dial-now rule, the valid values are 0 and line ID. 0 stands for all lines, multiple line IDs are separated by comma.
- At most 20 rules can be added to the IP phone.
- The expression syntax in the dial-now rule template is the same as introduced in the section Creating Dial Plan on page 25.

Procedure

Use the following procedures to customize dial-now template.

Customizing a dial-now template:

1. Open the template file using an ASCII editor.
2. 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 leaving it blank, the IP phone will apply to all lines.
3. Specify the values within double quotes.
4. Place this file to the root directory of the provisioning server.

The following is an example of a dial-now template:

```
<DialNow>
  <Data DialNowRule="1234" LineID="1"/>
  <Data DialNowRule="52[0-6]" LineID="1"/>
  <Data DialNowRule="xxxxxx" LineID=""/>
</DialNow>
```

Softkey Layout Template

You can create the soft key layout of different call states respectively using the softkey layout templates. The call states are CallFailed, CallIn, Connecting, Dialing, RingBack and Talking. After preparing the templates, place the templates to the root directory of the provisioning server and specify the access URL in the configuration files.

When editing a soft key layout template, remember 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 phone LCD screen.
- Create the 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 phone LCD screen.
- Create the 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 phone LCD screen by default.

Procedure

Use the following procedures to customize a soft key layout template.

Customizing a soft key layout template:

1. Open the template file using an ASCII editor.
2. For each soft key that you want to enable, add the following string to the file, each starting 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.
3. Specify the values within double quotes.
4. Place this file to the root directory of the provisioning server.

The following is an example of a CallFailed template:

```
<CallFailed>
  <Disable>
    <Key Type="Empty"/>
    <Key Type="Switch"/>
  </Disable>
  <Enable>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Cancel"/>
  </Enable>
  <Default>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Cancel"/>
  </Default>
</CallFailed>
```

Local Contact File

You can add contact one by one on the IP phone directly. In some cases, you may want to add multiple contacts to the IP phone at the same time or share the contacts on many IP phones. You can create a local contact file, and then place the local contact file to the root directory of the provisioning server, specify the access URL of the contact file in the configuration files.

When editing a local contact file, remember the following:

- <contactData> indicates the start of a contact file and </contactData> indicates the end of a contact file.
- <group> indicates the start of a contact list and </group> indicates the end of a contact list.
- <groupinfo> indicates the start of a group list and </groupinfo> indicates the end of a group list.
- When specifying a ring tone for the contact or the group, the format of the value must be Auto, Resource:RingN.wav (for the default system ring tone) or Custom:Name.wav (for the customized ring tone).
- When specifying the desired line for the contact, the valid values are 0 and line ID, 0 stands for all lines, multiple line IDs are separated by comma.

Procedure

Use the following procedures to customize a local contact file.

Customizing a local contact file:

1. Open the template file using an ASCII editor.
2. For each contact that you want to add, add the following string to the file, each starting on a separate line:

   <contact sDisplayName="" sOfficeNumber="" sMobilNumber="" sOtherNumber="" sLine="" sRing="" group=""/>

   Where:

   contact sDisplayName="" specifies the name of the contact (This value cannot be blank or duplicated).

   OfficeNumber="" specifies the office number of the contact.

   sMobilNumber="" specifies the mobile number of the contact.

   sOtherNumber="" specifies the other number of the contact.

   sLine="" specifies the line you want to add this contact to.

   sRing="" specifies the ring tone for this contact.

   group="" specifies the existing group you want to add the contact to.
3. For each group that you want to add, add the following string to the file, each

starting on a separate line:

<group name="" ring=""/>

Where:

group name="" specifies the name of the group.

ring="" specifies the desired ring tone for this group.

4. Specify the values within double quotes.
5. Place this file to the root directory of the provisioning server.

The following is an example of a local contact file:

```
<contactData>
  <group>
    <contact sDisplayName="Alice" sOfficeNumber="2215"
sMobilNumber="" sOtherNumber="" sLine="0" sRing="Auto"
group="Friend"/>
    <contact sDisplayName="Bob" sOfficeNumber="2216" sMobilNumber=""
sOtherNumber="" sLine="2" sRing="Resource:Ring2.wav"
group="Family"/>
  </group>
  <groupinfo>
    <group name="Friend" Ring="Auto"/>
    <group name="Family" ring="Custom:family.wav"/>
  </groupinfo>
</contactData>
```

Remote XML Phonebook

The IP phone can access 5 remote phonebooks. You can customize the remote XML phonebook for the IP phone as required. Before specifying the access URL of the remote phonebook in the configuration files, you need to create a remote XML phonebook and then place it to the provisioning server.

When creating an XML phonebook, remember the following:

- <YealinkIPPhoneDirectory> indicates the start of a phonebook and </YealinkIPPhoneDirectory> indicates the end of a phonebook.
- <DirectoryEntry> indicates the start of a contact and </DirectoryEntry> indicates the end of a contact.

Procedure

Use the following procedures to customize an XML phonebook.

Customizing an XML phonebook:

1. Open the template file using an ASCII editor.
2. For each contact that you want to add, add the following strings to the IP phonebook, each starting 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>.
3. Specify the values within double quotes.
4. Place this file to the root directory of the provisioning server.

The following is an example of an XML phonebook:

```
<YealinkIPPhoneDirectory>
    <DirectoryEntry>
        <Name>Jack</Name>
        <Telephone>1003</Telephone>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>John</Name>
        <Telephone>1004</Telephone>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Marry</Name>
        <Telephone>1005</Telephone>
    </DirectoryEntry>
</YealinkIPPhoneDirectory>
```

Specifying the Access URL of Resource Files

Access URL of the resource file can be configured in the configuration files:

Configuration File	<y0000000000xx>.cfg	Configure the access URL of the replace rule template. For more information, refer to Access URL of Replace Rule

		Template on page 311.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the dial-now rule template. For more information, refer to Access URL of Dial-now Template on page 312.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the softkey layout template. For more information, refer to Access URL of Softkey Layout Template on page 313312.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the local contact file. For more information, refer to Access URL of Local Contact File on page 315.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the remote XML phonebook. For more information, refer to Access URL of Remote XML Phonebook on page 316.

Troubleshooting

This chapter provides an administrator with general information for troubleshooting some most common problems that may encounter while using the SIP-T3xG IP phones.

Troubleshooting Methods

The IP phone can provide feedback in a variety of forms such as log files, packets, status indicators and so on, which helps an administrator quickly find out the cause of failure and do the troubleshooting more easily.

The following are some methods for you to learn more about the working status of your IP phone and quickly find out the cause of failure.

- Viewing Log Files
- Capturing Packets
- Enabling the Watch Dog Feature
- Getting Information from Status Indicators
- Analyzing Configuration Files

Viewing Log Files

The IP phone can log various events to log files. So if your IP phone encounters some problems, commonly the log files are used. You can export the log files to a syslog server or the local system. You can specify the location for which to save log files for troubleshooting purposes using the configuration files or the web user interface. You can also set the system log level to specify the severity level of the logs to be reported to a log file. The system log level is 3 by default (Changes to this parameter via web user interface require a reboot).

In the configuration files, you can use the following parameters to configure log settings:

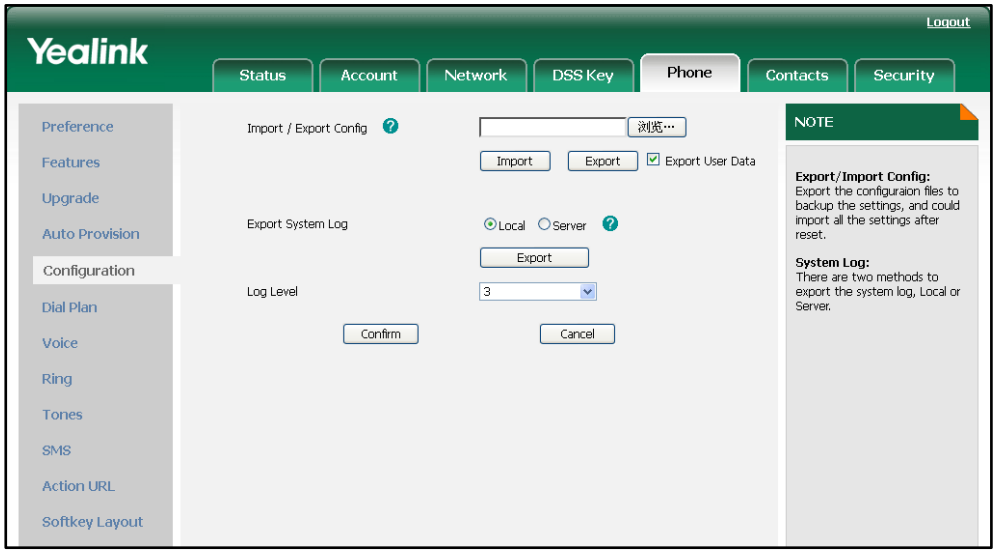
- **syslog.server**--Specify the IP address of the syslog server where to export the log files.
- **syslog.log_level**--Specify the severity level of the logs to be reported to a log file.

For more information about the log setting configuration parameters, refer to Log Settings on page 316.

To configure the level of the log files via web user interface:

1. Click on **Phone**->**Configuration**.

2. Select the desired level from the pull-down list of **Log Level**.



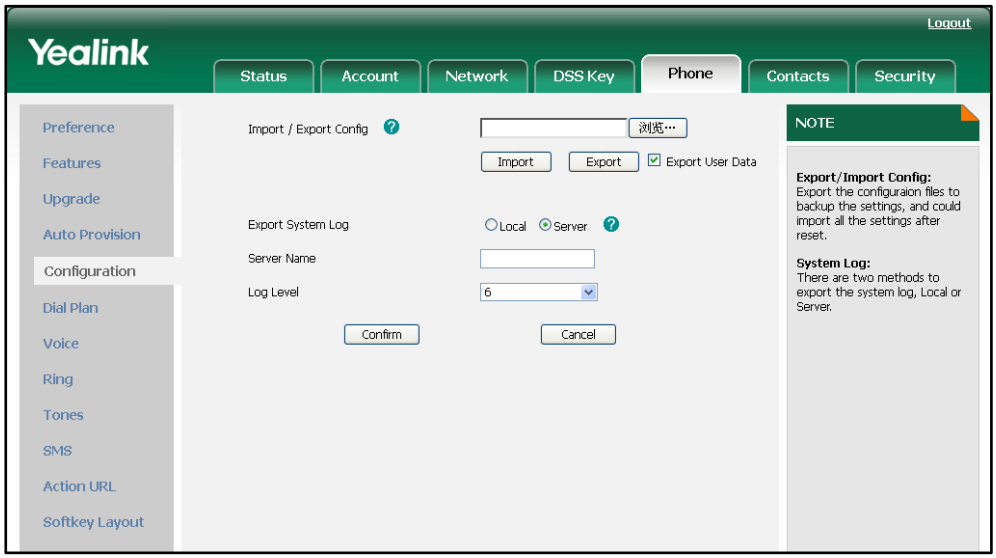
3. Click **Confirm** to accept the change.

   The web user interface pops up a dialog box to prompt "Do you want to restart your machine?"

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

To export log files to a syslog server via web user interface:

1. Click on **Phone->Configuration**.
2. Mark the **Server** radio box in the **Export System Log** field.
3. Enter the address of the syslog server in the **Server Name** field.

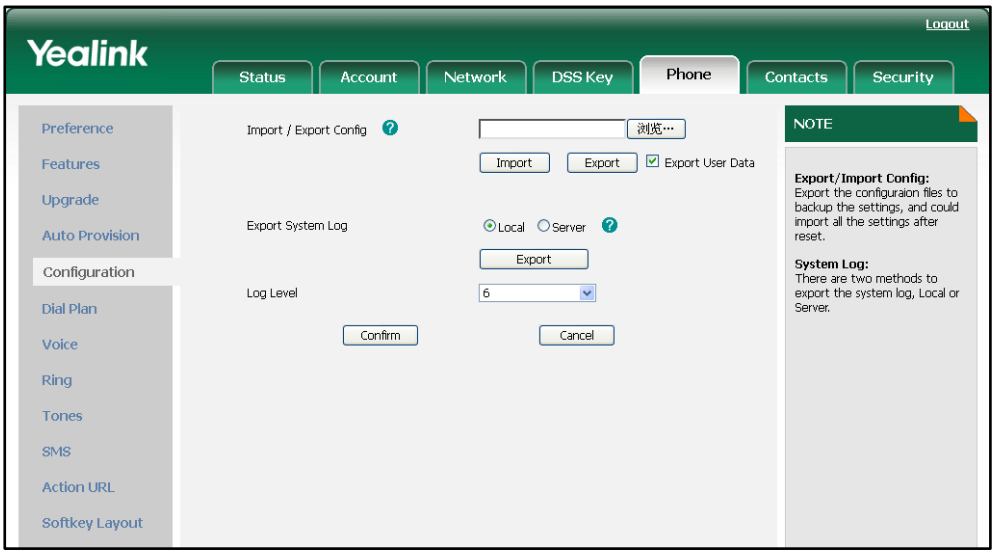


4. Click **Confirm** to accept the change.

To export log files to the local system via web user interface:

1. Click on **Phone->Configuration**.

2. Mark the **Local** radio box in the **Export System Log** field.
3. Click **Export** to open file download window, and then save the file to your local system.



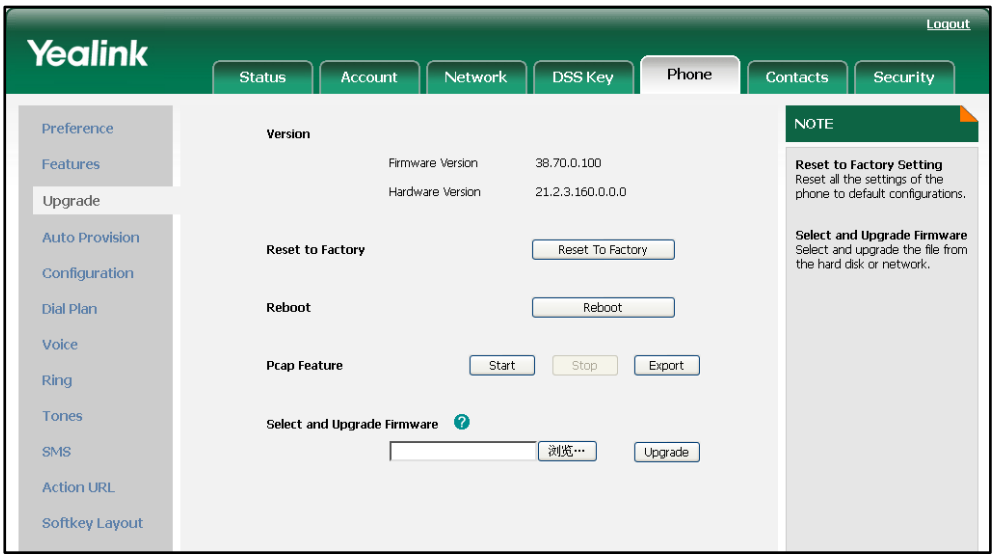
Capturing Packets

You can capture packets in two ways: capturing the packets via web user interface or using the Ethernet software. You can analyze the packets captured for troubleshooting purposes.

To capture packets via web user interface:

1. Click on **Phone->Upgrade**.
2. Click **Start** to begin capturing signal traffic.
3. Reproduce the issue to get stack traces.
4. Click **Stop** to end capturing.

5. Click **Export** to open file download window, and then save the file to your local system.



To capture packets using the Ethernet software:

Connect the IP phone's Internet port with the PC to the same HUB, and then use Sniffer, Ethereal or Wireshark software to capture the packets. You can also set a mirror port in the switch to monitor the port of the connected IP phone.

Enabling the Watch Dog Feature

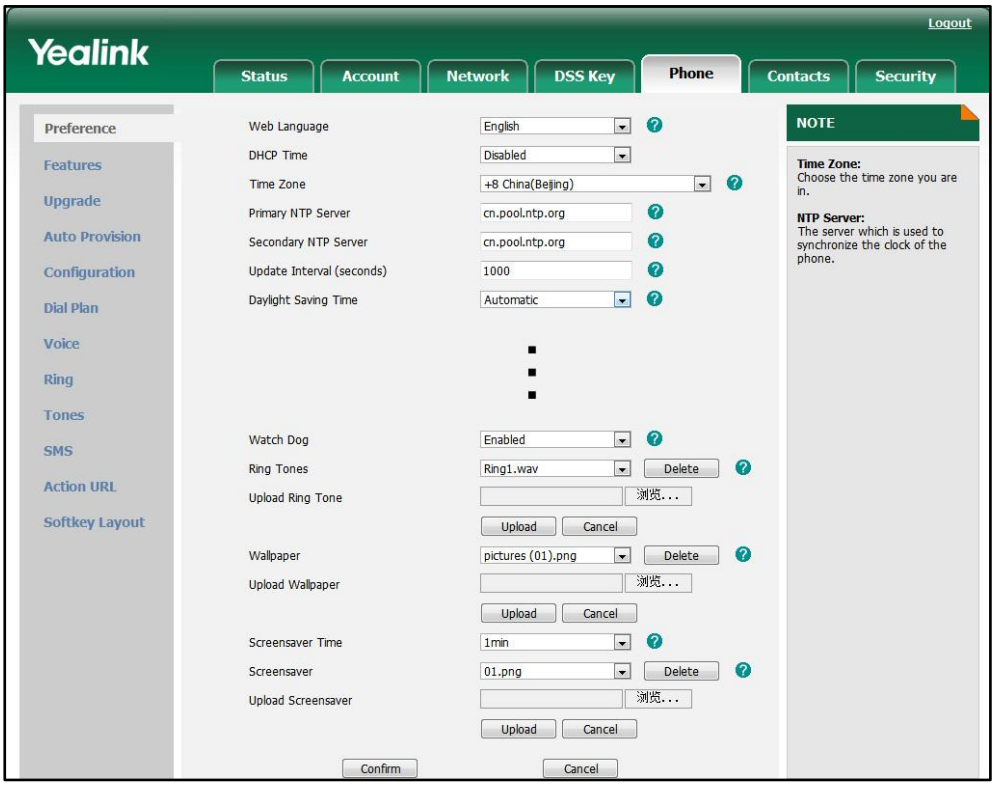
The IP phone provides a troubleshooting feature called "Watch Dog", which helps you monitor the IP phone status and provides the ability to get stack traces from the last time the IP phone failed. When the Watch Dog feature is enabled, the IP phone will automatically reboot when it detects a fatal failure. This feature can be configured using the configuration files or the web user interface.

You can use the "watch_dog.enable" parameter to configure the Watch Dog feature in the configuration files. For more information, refer to Watch Dog on page 317.

To configure the Watch Dog feature via web user interface:

1. Click on **Phone->Preference**.

2. Select the desired value from the pull-down list of **Watch Dog**.



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

Getting Information from Status Indicators

In some cases, the status indicators consist of power LED, message key indicator, line key indicator, headset key indicator and the on-screen icon/error messages, which are useful for you to figure out the cause of your phone's failure.

The following are two examples of getting the device information from status indicators:

- If a LINK failure of the IP phone is detected, a prompting message "Network Unavailable" and the icon indicate the current network LINK status.
- If the power LED is off, which indicates the IP phone is powered off.

For more information about the icons, refer to Reading Icons on page 14.

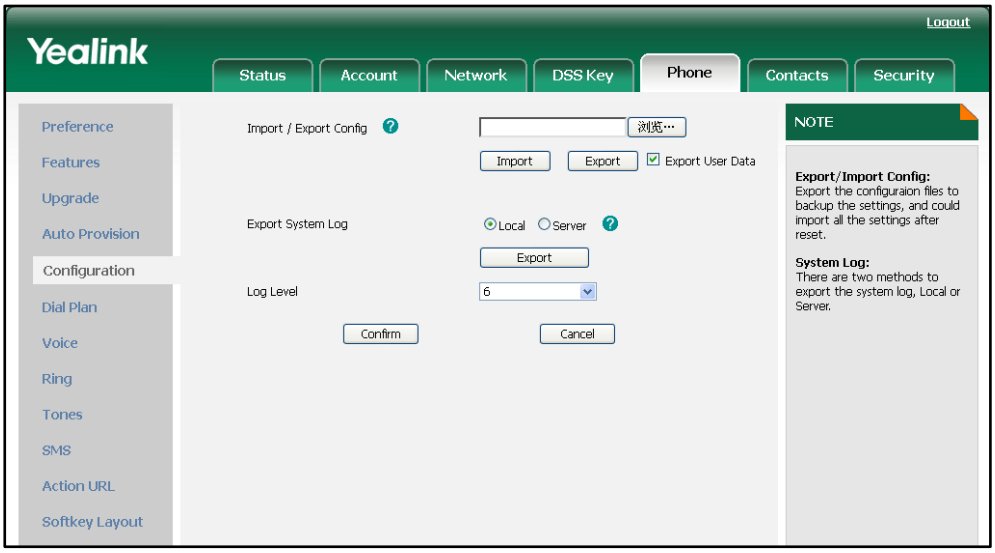
Analyzing Configuration Files

Sometimes, configuration errors may lead to your phone's failure. You can export configuration files to view the current configuration of the IP phone and troubleshoot as necessary.

To export configuration files via web user interface:

1. Click on **Phone**->**Configuration**.

2. In the **Import / Export Config** field, click **Export** to open the file download window, and then save the file to your local system.



Troubleshooting Solutions

This section describes solutions to some most common problems that may occur while using the IP phone. If you encounter a problem which is not listed in this section, contact your Yealink reseller for further support.

Why is the phone LCD screen blank?

Do one of the followings:

- Check that the power LED is on to ensure the IP phone is powered on.
- Ensure the IP phone is properly plugged into a functional AC outlet.
- Ensure that the IP phone isn't plugged into a plug controlled by a switch that is off.
- If the IP phone is plugged into a power strip, try plugging it directly into a wall outlet instead.
- If your phone is powered from PoE, ensure you use a PoE compliant switch or hub.

Why can the IP phone not obtain the IP address?

Do one of the followings:

- Ensure that the Ethernet cable is plugged into the Internet port on the IP phone and the Ethernet cable is not loose.
- Ensure the Ethernet cable is not damaged.
- Ensure the IP address and other network parameters are set correctly.

- Ensure that the switch or hub in your network is operational.

Why does the IP phone display "No Service"?

The phone LCD screen prompts "No Service" message when there is no any available SIP account on the IP phone.

Do one of the followings:

- Confirm if any account is actively registered on the IP phone at the path **Menu->Status->More->Accounts**.
- Check if the SIP parameters of the account have been set up correctly.

How can I know the basic information of the IP phone?

Press the OK key when the IP phone is idle to check the basic information of the IP phone, such as IP address and firmware version.

Why can the IP phone not upgrade successfully?

Do one of the followings:

- Ensure that the target firmware is not the same as the current used 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 the web browser is not closed and refreshed when upgrading the firmware using the web user interface.

Why does the IP phone not display time and date correctly?

Check if you have configured your phone to obtain the time and date from the NTP server automatically. If your phone is unable to access the NTP server, configure the time and date manually.

Why do I get poor audio during a call?

During a call, you may experience poor audio, including intermittent voice, low volume, echo or other noise. The root cause of audio anomalies can be difficult to diagnose.

- Problems may occur simply because the users are seated too far out of recommended microphone range and sound faint, or are seated too close to

sensitive microphones and cause feedback.

- Intermittent voice is mainly caused by packet loss and jitter. Packet loss may due to network congestion. Jitter is mainly due to message recombination of transmission or receiving equipment, such as timeout handling, retransmission mechanism or buffer under run.
- Noisy equipment, such as a computer or a fan, may make it difficult for hear the voice from the other party clearly. Turn off any other noisy equipment in the room such as fans.
- A line issue may also cause this problem. Disconnect the old line and redial the call to see if another line provides better connection.

What is the difference between a remote phonebook and a local phonebook?

A remote phonebook is placed on a server, while a local phonebook is placed on the IP phone flash. A remote phonebook can be used by everyone that can access the server, while a local phonebook can only be used by a specific phone itself. A remote phonebook is always used as a central phonebook for a company. That is, every stuff in the company can load this phonebook and each time they are trying to open a remote phonebook, the data is passed real-time from the certain server.

What is the difference of 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 phone LCD screen. Some servers also define the display name so this parameter set on the IP phone may not take effect.

Is there a SIP message that can make the IP phone reboot?

Yes. The IP phone will reboot only if the header in a SIP NOTIFY message contains an additional string "reboot=true". The 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
```

Why do IP phones use DOB format logo file instead of popular BMP, JPG and so on?

The picture the IP phone can recognize has some special requirements. It is not easy for the IP phone to resolve a popular picture format such as BMP and JPG. To make it easy, we enable only DOB file. There is a tool for you to convert a BMP file to DOB. For more information, refer to the document "Yealink Auto Provisioning User Guide".

What can I do if I forget the administrator password?

A factory reset can restore the original password. Please try to long press the OK key when the IP phone is idle, which should lead you to make a factory reset.

How to increase the volume on Speaker & on Headset?

The volumes in different cases are separated. Anytime you want to increase or reduce the voice you are hearing, just use the volume button under the navigation keys. When in idle, it tunes the ringer volume. In talking, it tunes the receiving volume. In dialing mode, it tunes the volume for dial tone. When you are using speaker, it tunes for speaker and when you are using headset, it tunes for headset.

What will happen if I connect both PoE cable and power adapter? Which has the higher priority?

The ones manufactured before last third of January 2010 will use the power adapter preferentially, while the after use PoE preferentially.

What is auto provisioning?

It is a term referring to the update of the IP phones, including updates on most of the configuration parameters, local phonebook, firmware and so on. You can make auto provisioning on a single phone, while it makes more sense in mass updates.

What is PnP?

Plug and Play (PnP) is a method for IP phones to get the provisioning server address. If

the IP phone is PnP enabled, it broadcast the PNP subscribe message to obtain a provisioning server address during booting up, any SIP server recognizing the message will respond with the preconfigured provisioning server address, so the IP phone will be able to download the CFG files from that server address. It depends on support from a SIP server.

Why does the IP phone not apply the configuration?

Do one of the followings:

- Ensure the configuration is set correctly.
- Reboot the IP phone, some configurations need reboot to take effect.
- Ensure the configuration is applicable to the IP phone model.
- The configuration may depend on support from the server.

What is "BLF List URI" used for?

This parameter is for BroadSoft platform. On BroadSoft, you can set up a BLF group containing several extension numbers. A name should be specified to this group that is the so-called BLF List URI. Normally when it comes to BLF, you should set them up in DSS keys and the IP phone will subscribe to the server for each extension, while with BLF List URI, the subscription will be simplified. The IP phone will only send subscription of the BLF List URI to the server and the server will know to subscribe all the extension numbers in that group.

For example, if you have 10 extensions, normally you will have to subscribe with the server for 10 times from the first extension number to the last. However, if you specify a BLF List URI including these 10 extensions and name it "Sales", you will only need to subscribe "Sales" with the server, which happens only for once.

What do "on code" and "off code" mean?

They are the codes that a phone will send to the server when there's a certain action. On code is related to the action of activating a feature, while off code of deactivating a feature.

Take the on code for Always forward for example, if you set the on code to be *78 (this code is supposed to be a feature code to activate Always forward on the server), and the target as 201. When you enable Always forward, the Forward feature on the IP phone-side is for sure activated, at the same time the code *78201 will be sent to the server, hence the server-side will also know that this phone is set to Always forward its calls to 201. So, the server-side will be able to get the right status of the extension.

How to solve the IP conflict problem?

Do one of the followings:

- Try to set another available IP address for the IP phone.
- Check the configuration of the network via phone user interface at the path **Menu**->**Settings**->**Advanced Settings**->**Network**->**WAN Port**. If Static IP Client is selected, select DHCP IP Client instead.

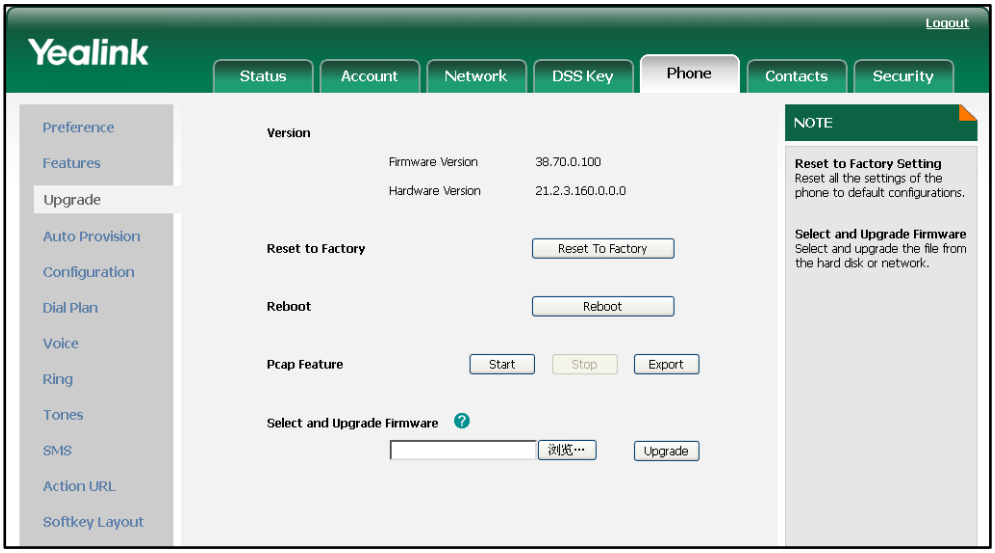
How to reset your phone to factory configurations?

Reset your phone to factory configurations after you have tried almost all troubleshooting suggestions but do not correct the problem. You need to note that all customized settings will be overwritten after resetting.

To reset your phone via web user interface:

1. Click on **Phone**->**Upgrade**.
2. Click **Reset to Factory** in the **Reset to Factory** field.

The web user interface prompts the message "Do you want to reset to factory?".



3. Click **OK** to confirm the resetting.

Appendix

Appendix A: Glossary

802.1x--an IEEE Standard for port-based Network Access Control (PNAC). It is 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.

ACD (Automatic Call Distribution)--used to distribute calls from large volumes of incoming calls to the registered IP phone users.

ACS (Auto Configuration server)--responsible for auto-configuration of the Central Processing Element (CPE).

Cryptographic Key--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)--built on a client-server model, where designated DHCP server hosts allocate network addresses and deliver configuration parameters to dynamically configured hosts.

DHCP Option--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)--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)--only provides authentication of the EAP peer to the EAP server but not mutual authentication.

FAC (Feature Access Code)--special patterns of characters that are dialed from a phone keypad to invoke particular features.

HTTP (Hypertext Transfer Protocol)--used to request and transmit data on the World Wide Web.

HTTPS (Hypertext Transfer Protocol over Secure Socket Layer)--a widely-used communications protocol for secure communication over a network.

IEEE (Institute of Electrical and Electronics Engineers)--a non-profit professional association headquartered in New York City that is dedicated to advancing technological innovation and excellence.

LAN (Local Area Network)--used to interconnects network devices in a limited area such as a home, school, computer laboratory, or office building.

PNP (Plug and Play)--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)--a class of storage medium used in computers and other electronic devices.

RTP (Real-time Transport Protocol)--provides end-to-end service for real-time data.

TCP (Transmission Control Protocol)--a transport layer protocol used by applications that require guaranteed delivery.

UDP (User Datagram Protocol)--a protocol offers non-guaranteed datagram delivery.

URI (Uniform Resource Identifier)--a compact sequence of characters that identifies an abstract or physical resource.

URL (Uniform Resource Locator)--specifies the address of an Internet resource.

VLAN (Virtual LAN)-- 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)--a family of technologies used for the delivery of voice communications and multimedia sessions over IP networks.

WLAN (Wireless Local Area Network)--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)--which uses XML to encode its calls and HTTP as a transport mechanism.

Appendix B: Time Zones

Time Zone	Time Zone Name
−11:00	Samoa
−10:00	United States-Hawaii-Aleutian
−10:00	United States-Alaska-Aleutian
−09:00	United States-Alaska Time
−08:00	Canada(Vancouver, Whitehorse)
−08:00	Mexico(Tijuana, Mexicali)
−08:00	United States-Pacific Time
−07:00	Canada(Edmonton, Calgary)
−07:00	Mexico(Mazatlan, Chihuahua)
−07:00	United States-Mountain Time
−07:00	United States-MST no DST
−06:00	Canada-Manitoba(Winnipeg)
−06:00	Chile(Easter Islands)
−06:00	Mexico(Mexico City, Acapulco)
−06:00	United States-Central Time
−05:00	Bahamas(Nassau)
−05:00	Canada(Montreal, Ottawa, Quebec)
−05:00	Cuba(Havana)
−05:00	United States-Eastern Time
−04:30	Venezuela(Caracas)
−04:00	Canada(Halifax, Saint John)
−04:00	Chile(Santiago)
−04:00	Paraguay(Asuncion)
−04:00	United Kingdom-Bermuda(Bermuda)
−04:00	United Kingdom(Falkland Islands)
−04:00	Trinidad&Tobago
−03:30	Canada- New Foundland(St.Johns)
−03:00	Denmark-Greenland(Nuuk)
−03:00	Argentina(Buenos Aires)
−03:00	Brazil(no DST)
−03:00	Brazil(DST)
−02:00	Brazil(no DST)
−01:00	Portugal(Azores)
0	GMT
0	Greenland
0	Denmark-Faroe Islands(Torshavn)
0	Ireland(Dublin)
0	Portugal(Lisboa, Porto, Funchal)
0	Spain-Canary Islands(Las Palmas)

Time Zone	Time Zone Name
0	United Kingdom(London)
0	Morocco
+01:00	Albania(Tirane)
+01:00	Austria(Vienna)
+01:00	Belgium(Brussels)
+01:00	Caicos
+01:00	Chad
+01:00	Croatia(Zagreb)
+01:00	Czech Republic(Prague)
+01:00	Denmark(Kopenhagen)
+01:00	France(Paris)
+01:00	Germany(Berlin)
+01:00	Hungary(Budapest)
+01:00	Italy(Rome)
+01:00	Luxembourg(Luxembourg)
+01:00	Macedonia(Skopje)
+01:00	Netherlands(Amsterdam)
+01:00	Namibia(Windhoek)
+02:00	Estonia(Tallinn)
+02:00	Finland(Helsinki)
+02:00	Gaza Strip(Gaza)
+02:00	Greece(Athens)
+02:00	Israel(Tel Aviv)
+02:00	Jordan(Amman)
+02:00	Latvia(Riga)
+02:00	Lebanon(Beirut)
+02:00	Moldova(Kishinev)
+02:00	Russia(Kaliningrad)
+02:00	Romania(Bucharest)
+02:00	Syria(Damascus)
+02:00	Turkey(Ankara)
+02:00	Ukraine(Kyiv, Odessa)
+02:00	Syria(Damascus)
+03:00	East Africa Time
+03:00	Iraq(Baghdad)
+03:00	Russia(Moscow)
+03:30	Iran(Teheran)
+04:00	Armenia(Yerevan)
+04:00	Azerbaijan(Baku)
+04:00	Georgia(Tbilisi)
+04:00	Kazakhstan(Aktau)
+04:00	Russia(Samara)

Time Zone	Time Zone Name
+05:00	Kazakhstan(Aqtobe)
+05:00	Kyrgyzstan(Bishkek)
+05:00	Pakistan(Islamabad)
+05:00	Russia(Chelyabinsk)
+05:30	India(Calcutta)
+06:00	Kazakhstan(Astana, Almaty)
+06:00	Russia(Novosibirsk, Omsk)
+07:00	Russia(Krasnoyarsk)
+07:00	Thailand(Bangkok)
+08:00	China(Beijing)
+08:00	Singapore(Singapore)
+08:00	Australia(Perth)
+09:00	Korea(Seoul)
+09:00	Japan(Tokyo)
+09:30	Australia(Adelaide)
+09:30	Australia(Darwin)
+10:00	Australia(Sydney, Melbourne, Canberra)
+10:00	Australia(Brisbane)
+10:00	Australia(Hobart)
+10:00	Russia(Vladivostok)
+10:30	Australia(Lord Howe Islands)
+11:00	New Caledonia(Noumea)
+12:00	New Zealand(Wellington, Auckland)
+12:45	New Zealand(Chatham Islands)
+13:00	Tonga(Nukualofa)

Appendix C: Configuration Parameters

This appendix describes the parameters you can set in the configuration files for the IP phone. The configuration files are <y0000000000xx>.cfg and <MAC>.cfg.

Setting Parameters in Configuration Files

You can set specific parameters in the configuration files for configuring the IP phones. The <y0000000000xx>.cfg and <MAC>.cfg files are stored on the provisioning server. The IP phone checks for configuration files and looks for resource files when restarting the IP phone. The <y0000000000xx>.cfg file stores configurations for all phones of the same model. The <MAC>.cfg file stores configurations specific to the IP phone with that MAC address.

Configuration changes made in the <MAC>.cfg file override the configuration settings in the <y0000000000xx>.cfg file.

Basic and Advanced Parameters

DHCP

Parameter- network.internet_port.type	**Configuration File** <y0000000000xx>.cfg
Description	Defines 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

Static Network Settings

Parameter- network.internet_port.type	Configuration File <y0000000000xx>.cfg
Description	Defines 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.internet_port.ip	Configuration File <y0000000000xx>.cfg
Description	Configures the IP address when the Internet port type is configured as Static IP Address. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.ip = 192.168.1.20

Parameter- network.internet_port.mask	Configuration File <y0000000000xx>.cfg
Description	Configures the subnet mask when the Internet port type is configured as Static IP Address. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.

Format	IP Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.mask = 255.255.255.0

Parameter- network.internet_port.gateway	**Configuration File** <y0000000000xx>.cfg
Description	Configures the default gateway when the Internet port type is configured as Static IP Address. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.gateway = 192.168.1.254

Parameter- network.primary_dns	**Configuration File** <y0000000000xx>.cfg
Description	Configures the primary DNS server when the Internet port type is configured as Static IP Address. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	202.101.103.55
Range	Not Applicable
Example	network.primary_dns = 202.101.103.5

Parameter- network.secondary_dns	Configuration File <y0000000000xx>.cfg
Description	Configures the secondary DNS server when the Internet port type is configured as Static IP Address. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	202.101.103.56
Range	Not Applicable
Example	network.secondary_dns = 202.101.103.6

PPPoE

Parameter- network.internet_port.type	Configuration File <y0000000000xx>.cfg
Description	Defines 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 <y0000000000xx>.cfg
Description	Configures the PPPoE username when the Internet port type is configured as PPPoE. **Note**: If you change this parameter, the IP phone will reboot to make the change take

	effect.
Format	String
Default Value	Blank
Range	Not Applicable
Example	network.pppoe.user = xmyealink

Parameter- network.pppoe.password	**Configuration File** <y0000000000xx>.cfg
Description	Configures the PPPoE password when the Internet port type is configured as PPPoE. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	String
Default Value	Blank
Range	Not Applicable
Example	network.pppoe.password = yealink123

PC Port Mode

Parameter- network.bridge_mode	**Configuration File** <y0000000000xx>.cfg
Description	Configures the PC port mode. **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**-Router **1**-Bridge
Example	network.bridge_mode = 1

Parameter- network.pc_port.ip	Configuration File <y0000000000xx>.cfg
Description	Configures the IP address for the PC port when the PC port is configured as Router. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	10.0.0.1
Range	Not Applicable
Example	network.pc_port.ip = 10.0.0.1

Parameter- network.pc_port.mask	Configuration File <y0000000000xx>.cfg
Description	Configures the subnet mask for the PC port when the PC port is configured as Router. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	255.255.255.0
Range	Not Applicable
Example	network.pc_port.mask = 255.255.255.0

Parameter- network.pc_port.dhcp_server	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the DHCP service for the PC attached to the PC port when the PC port is configured as Router. **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.pc_port.dhcp_server = 1

Parameter- network.dhcp.start_ip	**Configuration File** <y0000000000xx>.cfg
Description	Configure the start IP address that the IP phone assigns for the PC attached to the PC port when the PC port is configured as Router. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	10.0.0.10
Range	Not Applicable
Example	network.dhcp.start_ip = 10.0.0.10

Parameter- network.dhcp.end_ip	**Configuration File** <y0000000000xx>.cfg
Description	Configure the end IP address that the IP phone assigns for the PC attached to the PC port when the PC port is configured as Router. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	10.0.0.100
Range	Not Applicable
Example	network.dhcp.end_ip = 10.0.0.100

Dial Plan

Replace Rule

Parameter- dialplan.replace.prefix.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the numbers you want to replace.
Format	String
Default Value	Blank
Range	Not Applicable
Example	dialplan.replace.prefix.1 = 123

Parameter- dialplan.replace.replace.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the alternate string instead of what the user enters.
Format	String
Default Value	Blank
Range	Not Applicable
Example	dialplan.replace.replace.1 = 0592

Parameter- dialplan.replace.line_id.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply this replace rule. **Note:** Multiple line IDs are separated by comma.
Format	Integer
Default Value	Blank
Range	**Valid values are**: 1 to 6 (for T38G) 1 to 3 (for T32G)
Example	dialplan.replace.line_id.1 = 1,2,3

Dial-now

Parameter- dialplan.dialnow.rule.x	Configuration File <y0000000000xx>.cfg
Description	Specifies 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 20.
Format	String
Default Value	Blank
Range	Not Applicable
Example	dialplan.dialnow.rule.1 = 2216

Parameter- dialplan.dialnow.line_id.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply this dial-now rule. X ranges from 1 to 20. **Note**: Multiple line IDs are separated by comma.
Format	Integer
Default Value	Blank
Range	**Valid values are**: 1 to 6 (for T38G) 1 to 3 (for T32G)
Example	dialplan.dialnow.line_id.1 = 1,2,3

Parameter- phone_setting.dialnow_delay	Configuration File <y0000000000xx>.cfg
Description	Configures the delay time for the dial-now rule. When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the number after the delay time.

Format	Integer
Default Value	1
Range	Not Applicable
Example	phone_setting.dialnow_delay = 1

Area Code

Parameter- dialplan.area_code.code	**Configuration File** <y0000000000xx>.cfg
Description	Defines the area code to add before the entered numbers.
Format	Integer
Default Value	Blank
Range	Not Applicable
Example	dialplan.area_code.code = 010

Parameter- dialplan.area_code.min_len	**Configuration File** <y0000000000xx>.cfg
Description	Sets the minimum length of the entered numbers.
Format	Integer
Default Value	1
Range	1 to 15
Example	dialplan.area_code.min_len = 2

Parameter- dialplan.area_code.max_len	**Configuration File** <y0000000000xx>.cfg
Description	Sets 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 = 13

Parameter- dialplan.area_code.line_id	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply this area code rule. Note: Multiple line IDs are separated by comma.
Format	Integer
Default Value	Blank (for all lines)
Range	Valid values are: 1 to 6 (for T38G) 1 to 3 (for T32G)
Example	dialplan.area_code.line_id = 1,2

Block Out

Parameter- dialplan.block_out.number.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the block out numbers. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	Not Applicable
Example	dialplan.block_out.number.1 = 0000

Parameter- dialplan.block_out.line_id.x	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply this block out rule. X ranges from 1 to 10. Note: Multiple line IDs are separated by comma.
Format	Integer
Default Value	Blank (for all lines)

Range	**Valid values are**: 1 to 6 (for T38G) 1 to 3 (for T32G)
Example	dialplan.block_out.line_id.1 = 1,2,3

Backlight

Parameter- phone_setting.active_backlight_level	**Configuration File** <y0000000000xx>.cfg
Description	Configures the backlight level used to adjust the backlight intensity of the LCD screen Level 1 is the least bright and level 10 is the most bright.
Format	Integer
Default Value	8
Range	1 to 10
Example	phone_setting.active_backlight_level = 1

Parameter- phone_setting.inactive_backlight_level	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the phone to completely turn off the backlight after a period of inactivity.
Format	Boolean
Default Value	1
Range	**Valid values are:** **0**-Enable **1**-Disable
Example	phone_setting.inactive_backlight_level = 1

Parameter- phone_setting.backlight_time	Configuration File <y0000000000xx>.cfg
Description	Configures the backlight time (in seconds) used to specify the delay time to turn off the backlight when the IP phone is inactive. If set to 60 (60s), the LCD backlight is turned off when the IP phone is inactive for 60 seconds.
Format	Integer
Default Value	60
Range	Valid values are: 1-Always on 60-1min 120-2min 300-5min 600-10min 1800-30min
Example	phone_setting.backlight_time = 60

User Password

Parameter- security.user_password	Configuration File <y0000000000xx>.cfg
Description	Sets a new user password for the IP phone. The IP phone uses "user" as the default user password. Note: The IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.
Format	username:new password
Default Value	user
Range	ASCII characters 32-126(0x20-0x7E)
Example	security.user_password = user:password123

Administrator Password

Parameter- security.user_password	Configuration File <y0000000000xx>.cfg
Description	Sets a new administrator password for the IP phone. The IP phone uses "admin" as the default administrator password. **Note**: The IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.
Format	administrator username:new password
Default Value	admin
Range	ASCII characters 32-126(0x20-0x7E)
Example	security.user_password = admin:password000

Phone Lock

Parameter- phone_setting.lock	Configuration File <y0000000000xx>.cfg
Description	Specifies the type of phone lock. Menu Key: The Menu soft key is locked. Function Key: MESSAGE, RD, CONF, HOLD, MUTE, TRAN, OK, X, navigation keys, soft keys, line keys and memory keys are locked (For T32G, CONF, HOLD, MUTE and memory keys do not exist). All Keys: All keys are locked. Answer call only: All keys are locked. If set to 0 (Disabled), the IP phone lock feature is disabled.
Format	Integer
Default Value	0
Range	**Valid values are:** **0**-Disabled **1**-Menu Key **2**-Function Keys **3**-All Keys **4**-Answer call only

Example	phone_setting.lock = 2

Parameter- phone_setting.phone_lock.unlock_pin	Configuration File <y0000000000xx>.cfg
Description	Sets a new unlock password. Once the IP phone is locked, you can use "123" as the default password to unlock it. **Note**: The IP phones support numeric characters only in password.
Format	Numeric characters only
Default Value	123
Range	0 to 32768
Example	phone_setting.phone_lock.unlock_pin = 123456

Parameter- phone_setting.phone_lock.lock_time_out	Configuration File <y0000000000xx>.cfg
Description	Configure the IP phone to automatically lock the keypad after a delay time (in seconds). If set to 0 (0s), the keypad will not be locked automatically. In this case, you can long press the pound key to lock the keypad only. **Note**: This parameter works only if the IP phone lock type is preset.
Format	Integer
Default Value	10
Range	0 to 3600s
Example	phone_setting.phone_lock.lock_time_out = 8

Time and Date

NTP Server

Parameter- local_time.manual_time_enable	**Configuration File** <y0000000000xx>.cfg
Description	Configures the phone to obtain the time and date manually or dynamically from the NTP server.
Format	Boolean
Default Value	1
Range	**Valid values are:** **0**-Manual **1**-NTP Server
Example	local_time.manual_time_enable = 1

Parameter- local_time.ntp_server1	**Configuration File** <y0000000000xx>.cfg
Description	Sets the IP address or the domain name of the primary NTP server. **Note**: It works only if the parameter "local_time.manual_time_enable" is set to 1 (NTP Server).
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	Not Applicable
Example	local_time.ntp_server1 = 192.168.0.5

Parameter- local_time.ntp_server2	**Configuration File** <y0000000000xx>.cfg
Description	Sets 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. **Note**: It works only if the parameter

	"local_time.manual_time_enable" is set to 1 (NTP Server).
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	Not Applicable
Example	local_time.ntp_server2 = 192.168.0.5

Parameter- local_time.interval	**Configuration File** <y0000000000xx>.cfg
Description	Sets the IP phone to update time and date from the NTP server at regular intervals (in seconds). **Note**: It works only if the parameter "local_time.manual_time_enable" is set to 1 (NTP Server).
Format	Integer
Default Value	1000
Range	Not Applicable
Example	local_time.interval = 1200

Time Zone

Parameter- local_time.time_zone	**Configuration File** <y0000000000xx>.cfg
Description	Defines the time zone. For more available time zone list, refer to Appendix B: Time Zones on page 221.
Format	Not Applicable
Default Value	+8
Range	-11 to +13
Example	local_time.time_zone = +9

Parameter- local_time.time_zone_name	**Configuration File** <y0000000000xx>.cfg
Description	Defines the desired time zone name.

	For more available time zone name list, refer to Appendix B: Time Zones on page 221.
Format	String
Default Value	China(Beijing)
Range	Not Applicable
Example	local_time.time_zone_name = Korea(Seoul)

DST

Parameter- local_time.summer_time	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the use of Daylight Saving Time (DST).
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** <y0000000000xx>.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 = 1

Parameter- local_time.start_time	Configuration File <y0000000000xx>.cfg
Description	Specifies 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:** • MM/DD/HH (For By Date) • Month/Week of Month/Day of Week/Hour of Day (For By Week)
Default Value	1/1/0
Range	1to 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 = 5/20/12

Parameter- local_time.end_time	Configuration File <y0000000000xx>.cfg
Description	Specifies the time to end 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:** MM/DD/HH (For By Date) Month/Week of Month/Day of Week/Hour of Day (For By Week)
Default Value	12/31/23
Range	1to 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 = 10/25/22

Parameter- local_time.offset_time	**Configuration File** <y0000000000xx>.cfg
Description	Sets the offset time (in minutes) of DST. **Note**: It works only if the parameter "local_time.summer_time" is set to 1 (Enabled).
Format	Integer
Default Value	60
Range	Not Applicable
Example	local_time.offset_time = 120

Time Format

Parameter- local_time.time_format	**Configuration File** <y0000000000xx>.cfg
Description	Sets the time format. If set to 0 (12 Hour), the time display uses 12 hour format. If set to 1 (24 Hour), the time display uses 24

	hour format.
Format	Integer
Default Value	Integer
Range	**0**-12 Hour **1**-24 Hour
Example	local_time.time_format = 0

Date Format

Parameter- local_time.date_format	**Configuration File** <y0000000000xx>.cfg
Description	Sets the date format. The IP phones support various date formats. You can change the date to your desired format according to your requirement.
Format	Integer
Default Value	0
Range	**Valid values are:** **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
Example	local_time.date_format = 1

Language

Parameter- gui_lang.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the language pack. **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.

Format	URL
Default Value	Blank
Range	Not Applicable
Example	The following example uses HTTP to download the language pack "lang-Italian.txt"(Italian) from the provisioning server 192.168.10.25. gui_lang.url = http://192.168.10.25/lang-Italian.txt

Parameter- lang.wui	**Configuration File** <y0000000000xx>.cfg
Description	Specifies 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	Text
Default Value	Not Applicable
Range	**Valid values are:** English Chinese_S Deutsch French Italian Portuguese Spanish Turkish
Example	lang.wui = French

Parameter- lang.gui	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the language used on the phone

	user interface.
Format	Text
Default Value	English
Range	**Valid values are:** English Chinese_S Chinese_T German French Italian Portuguese Polish Spanish Turkish
Example	lang.gui = Italian

Key as Send

Parameter- features.pound_key.mode	**Configuration File** <y0000000000xx>.cfg
Description	Defines the "#" or "*" key as the send key. If set to 0 (Disabled), neither "#" nor "*" can be used as a send key. If set to 1(# key), the pound key is defined as the send key. If set to 2(* key), the asterisk key is defined as the send key.
Format	Integer
Default Value	1
Range	**Valid values are:** **0**-Disabled **1**-# key **2**-* key
Example	features.pound_key.mode = 0

Parameter- features.send_key_tone	**Configuration File** <y0000000000xx>.cfg

Description	Enables or disables the IP phone to play a tone when a user presses a send key. If set to 1 (Enabled), the IP phone plays a tone when a user presses a send key. **Note**: It works only if the key tone is enabled. So you should set the parameter "features.key_tone" to 1 (Enabled) in advance.
Format	Integer
Default Value	1
Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	features.send_key_tone = 0

Hotline

Parameter- features.hotline_number	**Configuration File** <y0000000000xx>.cfg
Description	Configures the hotline number. It specifies a number that the IP phone automatically dials out when lifting the handset, pressing the speakerphone key or pressing the line key. Leaving it blank disables the hotline feature.
Format	String
Default Value	Blank
Range	Not Applicable
Example	features.hotline_number = 3601

Parameter- features.hotline_delay	Configuration File <y0000000000xx>.cfg
Description	Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number. If set to 0 (0s), the IP phone immediately dials out the preconfigured hotline number when you lift the handset, press the speakerphone key or press the line key. If set to a value greater than 0, the IP phone waits the specified seconds before dialing out the dials out the predefined hotline number when you lift the handset, press the speakerphone key or press the line key.
Format	Integer
Default Value	2
Range	0 to 180
Example	features.hotline_delay = 30

Call Log

Parameter- features.save_call_history	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to save call log. If set to 0 (Disabled), the IP phone cannot log the dialed calls, received calls, missed calls and the forwarded calls in the call log lists.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	features.save_call_history = 0

Missed Call Log

Parameter- account.x.missed_calllog	Configuration File <MAC>.cfg
Description	Enables or disables the missed call log feature for account X. If 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. If 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. X ranges from 1 to 6.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	account.1.missed_calllog = 1

Live Dialpad

Parameter- phone_setting.predial_autodial	Configuration File <y0000000000xx>.cfg
Description	Configures live dialpad feature. If set to 1 (Enabled), the IP phone automatically dials out the entered phone number without having to press any key.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	phone_setting.predial_autodial = 1

Call Waiting

Parameter- call_waiting.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the call waiting feature. If set to 0 (Disabled), a new incoming call is automatically rejected by the IP phone with a busy message while during a call. If set to 1 (Enabled), the phone LCD screen presents a new incoming call while during a call.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	call_waiting.enable = 1

Parameter- call_waiting.tone	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the playing of a call waiting tone when the IP phone receives an incoming call during a call. If set to 1 (Enabled), the IP phone performs 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	**0**-Disabled **1**-Enabled
Example	call_waiting.tone = 1

Auto Redial

Parameter- auto_redial.enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to automatically redial the called number when it is busy. If set to 1 (Enabled), the IP phone dials the previous dialed out number automatically when the dialed number is busy.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	auto_redial.enable = 1

Parameter- auto_redial.interval	**Configuration File** <y0000000000xx>.cfg
Description	Sets the interval (in seconds) for the IP phone to wait before redial. 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 = 30

Parameter- auto_redial.times	**Configuration File** <y0000000000xx>.cfg
Description	Sets 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 = 8

Auto Answer

Parameter- account.x.auto_answer	**Configuration File** <MAC>.cfg
Description	Enables or disables the auto answer feature for account X. If set to 1 (Enabled), the IP phone can automatically answer an incoming call. X ranges from 1 to 6. **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	**0**-Disabled **1**-Enabled
Example	account.1.auto_answer = 1

Call Completion

Parameter- features.call_completion_enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the call completion feature. If a user places a call and the callee is temporarily not available to answer the call, the call completion feature allows notifying the user when the callee becomes available to receive a call. If set to 1 (Enabled), the caller failed is notified when the callee becomes available to receive a call.

Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.call_completion_enable = 1

Anonymous Call

Parameter- account.x.anonymous_call	Configuration File <MAC>.cfg
Description	Enables or disables the anonymous call feature for account X. If set to 1 (Enabled), the IP phone blocks 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. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.anonymous_call = 1

Parameter- account.x.anonymous_call_oncode	Configuration File <MAC>.cfg
Description	Sets the anonymous call on code to inform the server to enable the anonymous call feature for account X (optional). X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.anonymous_call_oncode = *72

Parameter- account.x.anonymous_call_offcode	Configuration File <MAC>.cfg
Description	Sets the anonymous call off code to inform the server to disable the anonymous call feature for account X (optional). X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.anonymous_call_offcode = *73

Anonymous Call Rejection

Parameter- account.x.reject_anonymous_call	Configuration File <MAC>.cfg
Description	Enables or disables the anonymous call rejection feature for account X. If set to 1 (Enabled), the IP phone automatically rejects incoming calls from users enabled the anonymous call feature. The anonymous user's phone LCD screen presents "Anonymity Disallowed". X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.reject_anonymous_call = 1

Parameter- account.x.anonymous_reject_oncode	Configuration File <MAC>.cfg
Description	Sets the anonymous call rejection on code to inform the server to enable the anonymous call rejection feature for account X

	(optional). X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.anonymous_reject_oncode = *74

Parameter- account.x.anonymous_reject_offcode	**Configuration File** <MAC>.cfg
Description	Sets the anonymous call rejection off code to inform the server to disable the anonymous call rejection feature for account X (optional). X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.anonymous_reject_offcode = *73

Do Not Disturb

Parameter- features.dnd.on_code	**Configuration File** <y0000000000xx>.cfg
Description	Sets the DND on code to inform the server to enable the DND feature (optional).
Format	String
Default Value	Blank
Range	Not Applicable
Example	features.dnd.on_code = *71

Parameter- features.dnd.off_code	Configuration File <y0000000000xx>.cfg
Description	Sets the DND off code to inform the server to disable the DND feature (optional).
Format	String
Default Value	Blank
Range	Not Applicable
Example	features.dnd.off_code = *72

Parameter- features.dnd.emergency_authorized_number	Configuration File <y0000000000xx>.cfg
Description	Specify the DND authorized numbers. If set to 1008, the IP phone can still receive the incoming call from 1008 even if DND is activated on the IP phone.
Format	String
Default Value	Blank
Range	0 to 200 characters
Example	features.dnd.emergency_authorized_number = 1008,1010

Return Message When DND

Parameter- features.dnd_refuse_code	Configuration File <y0000000000xx>.cfg
Description	Defines return codes and reason of the SIP response message when rejecting an incoming call for DND. A specific reason is displayed on the caller's phone LCD screen. If set to 486 (Busy here), the caller's phone LCD screen displays the reason "Busy here" when the callee enables the 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 = 486

Busy Tone Delay

Parameter- features.busy_tone_delay	**Configuration File** <y0000000000xx>.cfg
Description	Configure a period of time (in seconds) for which the busy tone is audible on the IP phone. When one party releases the call, a busy tone is audible to the other party indicating that the call connection breaks. If set to 3 (3s), a busy tone is audible for 3 seconds on the IP phone.
Format	Integer
Default Value	0
Range	**Valid values are:** **0**-0s **3**-3s **5**-5s
Example	features.busy_tone_delay = 3

Return Code When Refuse

Parameter- features.normal_refuse_code	**Configuration File** <y0000000000xx>.cfg
Description	Defines return codes and messages when rejecting an incoming call. A specific return message is displayed on the caller's phone LCD screen. If set to 486 (Busy here), the caller's phone LCD screen displays the message "Busy here" when the callee rejects the incoming call.

Format	Integer
Default Value	486
Range	**Valid values are:** **404**-No Found **480**-Temporarily not available **486**-Busy here
Example	features.normal_refuse_code = 480

180 Ring Workaround

Parameter- phone_setting.is_deal180	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message. If set to 1 (Enabled), the IP phone resumes and plays the local ringback tone upon a subsequent 180 message received.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	phone_setting.is_deal180 = 0

Use Outbound Proxy in Dialog

Parameter- sip.use_out_bound_in_dialog	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to send the SIP messages to the outbound proxy server. If set to 1 (Enabled), all the SIP request messages from the IP phone will be forced to send to the outbound proxy server.
Format	Boolean
Default Value	1

Range	**0**-Disabled **1**-Enabled
Example	sip.use_out_bound_in_dialog = 0

SIP Session Timer

Parameter- account.x.advanced.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 ranges from 1 to 6.
Format	Float
Default Value	0.5
Example	account.1.advanced.timer_t1 = 1

Parameter- account.x.advanced.timer_t2	**Configuration File** <MAC>.cfg
Description	Configures the session timer T2 (in seconds) for account X. T2 represents the maximum retransmitting time of any SIP request message. The re-transmitting and doubling of T1 continues until the retransmitting time reaches the T2 value. X ranges from 1 to 6.
Format	Float
Default Value	4
Example	account.1.advanced.timer_t2 = 5

Parameter- account.x.advanced.timer_t4	**Configuration File** <MAC>.cfg
Description	Configures the session timer of T4 (in

	seconds) for account X. T4 represents the time the network will take to clear messages between the SIP Client and SIP Server. X ranges from 1 to 6.
Format	Float
Default Value	5
Example	account.1.advanced.timer_t4 = 10

Session Timer

Parameter- account.x.session_timer.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the session timer for account X. If set to 1 (Enabled), IP phone sends periodic re-INVITE requests to refresh the session during a call. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**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 set to 180 (180s), the IP phone refreshes the session during a call before 180 seconds. X ranges from 1 to 6.
Format	Integer
Default Value	Blank

Range	1-999
Example	account.1.session_timer.expires = 300

Parameter- account.x.session_timer.refresher	**Configuration File** <MAC>.cfg
Description	Configures the session timer refresher for account X. If set to 0 (UAC), refreshing the session is performed by the IP phone. If set to 1 (UAS), refreshing the session is performed by a SIP server. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-UAC **1**-UAS
Example	account.1.session_timer.refresher = 1

Call Hold

Parameter- sip.rfc2543_hold	**Configuration File** <y0000000000xx>.cfg
Description	Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used. If set to 0 (Disabled), use SDP media direction attributes (such as a=sendonly) per RFC 3264 when putting a call on hold. If set to 0 (Enabled), use SDP media connection address c=0.0.0.0 per RFC 2543 when putting a call on hold.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	sip.rfc2543_hold = 1

Call Transfer

Parameter- transfer.blind_tran_on_hook_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to complete the blind transfer through on-hook.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	transfer.blind_tran_on_hook_enable = 1

Parameter- transfer.on_hook_trans_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to complete the semi-attended transfer or attended transfer through on-hook.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	transfer.on_hook_trans_enable = 1

Parameter- transfer.semi_attend_tran_enable	Configuration File <y0000000000xx>.cfg
Description	Specifies whether to display the missed call prompt on the destination party's phone.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	transfer.semi_attend_tran_enable = 1

Network Conference

Parameter- account.x.conf_type	Configuration File <MAC>.cfg
Description	Defines the conference type for account X. If set to 0 (Local), conferences are set up on the IP phone locally. If set to 2 (Network Conference), conferences are set up by the server. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	Valid values are: 0-Local 2-Network Conference
Example	account.1.conf_type = 2

Parameter- account.x.conf_uri	Configuration File <MAC>.cfg
Description	Defines the conference URI for account X. X ranges from 1 to 6. Note: It works only if the parameter "account.x.conf_type" is set to 2 (Network Conference).
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.conf_uri = conference@domain.com

Transfer on Conference Hang Up

Parameter- transfer.tran_others_after_conf_enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the Transfer on Conference Hang Up feature. If 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	**0**-Disabled **1**-Enabled
Example	transfer.tran_others_after_conf_enable = 1

Direct Pickup

Phone Basis

Parameter- features.pickup.direct_pickup_enable	**Configuration File** <y0000000000xx>.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	**0**-Disabled **1**-Enabled
Example	features.pickup.direct_pickup_enable = 1

Parameter- features.pickup.direct_pickup_code	Configuration File <y0000000000xx>.cfg
Description	Configures the direct pickup code on a phone basis. **Note**: The direct pickup code configured on a per-line basis takes precedence over that configured on a global basis.
Format	String
Default Value	Blank
Range	Not Applicable
Example	features.pickup.direct_pickup_code = *97

Per-account Basis

Parameter- account.x.direct_pickup_code	Configuration File <y0000000000xx>.cfg
Description	Configures the direct pickup code on a per-account basis. X ranges from 1 to 6. **Note**: The direct pickup code configured on a per-line basis takes precedence over that configured on a global basis.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.direct_pickup_code = *68

Group Pickup

Phone Basis

Parameter- features.pickup.group_pickup_enable	Configuration File <y0000000000xx>.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	**0**-Disabled **1**-Enabled
Example	features.pickup.group_pickup_enable = 1

Parameter- features.pickup.group_pickup_code	Configuration File <y0000000000xx>.cfg
Description	Configures the group pickup code on a phone basis. **Note**: The group pickup code configured on a per-line basis takes precedence over that configured on a global basis.
Format	String
Default Value	Blank
Range	Not Applicable
Example	features.pickup.group_pickup_code = *98

Per-account Basis

Parameter- account.x.group_pickup_code	Configuration File <y0000000000xx>.cfg
Description	Configures the group pickup code on a per-account basis. X ranges from 1 to 6. **Note**: The group pickup code configured on a per-line basis takes precedence over that configured on a global basis.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.group_pickup_code = *69

Dialog-Info Call Pickup

Parameter- account.x.dialoginfo_callpickup	Configuration File <MAC>.cfg
Description	Configures the Dialog-Info Call Pickup feature for account X. If set to 1 (Enabled), call pickup is implemented through SIP signals. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.dialoginfo_callpickup = 1

Web Server Type

Parameter- network.web_server_type	Configuration File <y0000000000xx>.cfg
Description	Specifies the access type of the web user interface of the IP phone. If set to 0 (Disabled), you are not allowed to access the web user interface of the IP phone. If set to 1 (HTTP & HTTPS), you can access the web user interface of the IP phone using HTTP protocol or HHTPS protocol. If set to 2 (HTTP Only), you can access the web user interface of the IP phone using HTTP protocol only. If set to 3 (HTTPS Only), you can access the web user interface of the IP phone using HTTPS protocol only. **Note**: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer

Default Value	1
Range	**Valid values are**: **0**-Disabled **1**-HTTP & HTTPS **2**-HTTP Only **3**-HTTPS Only
Example	network.web_server_type = 2

Parameter- network.port.http	**Configuration File** <y0000000000xx>.cfg
Description	Configures the HTTP port 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 = 90

Parameter- network.port.https	**Configuration File** <y0000000000xx>.cfg
Description	Configures the HTTPS port 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 = 100

Calling Line Identification Presentation

Parameter- account.x.cid_source	Configuration File <MAC>.cfg
Description	Configure the presentation of the caller identity for account X. **0**-FROM (Derives the name and number of the caller from the "From" header). **1**-PAI (Derives the name and number of the caller from the "PAI" header. If the server does not send the "PAI" header, displays "anonymity" on the callee's phone). **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). **3**-RPID-PAI-FROM **4**-PAI-RPID-FROM **5**-RPID-FROM X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	0 to 5
Example	account.1.cid_source = 2

Connected Line Identification Presentation

Parameter- account.x.cp_source	Configuration File <MAC>.cfg
Description	Configure the presentation of the callee identity for account X. **0**-RPID-FROM (Derives the name and number of the callee from the "RPID" header preferentially. If the server does not send the "RPID" header, derives from the "From" header). **1**-Dialed Digits (Preferentially displays the dialed digits on the caller's phone).

	2-RFC 4916 (Derives the name and number of the callee from "From" header in the Update message). 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. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	0 to 2
Example	account.1.cp_source = 2

DTMF

Parameter- account.x.dtmf.type	**Configuration File** <MAC>.cfg
Description	Specifies the DTMF type for account X. If set to 0 (INBAND), DTMF digits are transmitted in the voice band (G.711). If set to 1 (RFC 2833), DTMF digits are transmitted by RTP Events compliant to RFC 2833. If set to 2 (SIP INFO), DTMF digits are transmitted by the SIP INFO messages. If set to 3 (AUTO+SIP INFO), negotiates with the other end to use INBAND or RFC 2833, if there is no negotiation, using SIP INFO by default. X ranges from 1 to 6.
Format	Integer
Default Value	1
Range	**Valid values are:** **0**-INBAND **1**-RFC 2833

	2-SIP INFO **3**-AUTO+SIP INFO
Example	account.1.dtmf.type = 2

Parameter- account.x.dtmf.dtmf_payload	**Configuration File** <MAC>.cfg
Description	Configures the RFC 2833 payload type. X ranges from 1 to 6.
Format	Integer
Default Value	101
Range	96 to 126
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" or "AUTO+SIP INFO". X ranges from 1 to 6.
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 = 3

Parameter- features.dtmf.repetition	**Configuration File** <y0000000000xx>.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 = 2

Incoming Intercom calls

Parameter- features.intercom.allow	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to automatically answer an incoming intercom call. If set to 0 (Disabled), the IP phone rejects incoming intercom calls and sends a busy signal to the caller. If set to 1 (Enabled), the IP phone automatically answers an incoming intercom call.
Format	Boolean
Default Value	1
Range	0-Disabled 1-Enabled
Example	features.intercom.allow = 1

Parameter- features.intercom.mute	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to mute the microphone when answering an intercom call. If set to 0 (Disabled), the microphone is un-muted for incoming calls. If set to 1 (Enabled), the microphone is muted for intercom calls.
Format	Boolean
Default Value	0
Range	0-Disabled 1-Enabled
Example	features.intercom.mute = 1

Parameter- features.intercom.tone	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to play a warning tone when receiving an intercom call. If set to 0 (Disabled), the IP phone automatically answers the intercom call without a warning tone. If set to 1 (Enabled), the IP phone plays a warning tone to alert you before answering the intercom call.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	features.intercom.tone = 1

Parameter- features.intercom.barge	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to automatically answer an incoming intercom call while there is already an active call on the IP phone. If set to 0 (Disabled), the IP phone handles an incoming intercom call like a waiting call while there is already an active call on the IP phone. If set to 1 (Enabled), the IP phone automatically answers the intercom call while there is already an active call on the IP phone and put the active call on hold.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.intercom.barge = 1

Distinctive Ring Tones

Parameter- account.x.alert_info_url_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the distinctive ring tones feature for account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Enabled **1**-Disabled
Example	account.1.alert_info_url_enable = 1

Parameter- distinctive_ring_tones.alert_info.x.text	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the texts to map the keywords contained in the SIP header. X ranges from 1 to 10.
Format	Text
Default Value	Blank
Range	Not Applicable
Example	distinctive_ring_tones.alert_info.1.text = family

Parameter- distinctive_ring_tones.alert_info.x.ringer	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired ring tones for each text. The value ranges from 1 to 8, 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 **6**-Ring6.wav **7**-Ring7.wav **8**-Ring8.wav
Example	distinctive_ring_tones.alert_info.1.ringer = 2

Tones

Parameter- voice.tone.country	**Configuration File** <y0000000000xx>.cfg
Description	Configures the tone type for the IP phone.
Format	Text
Default Value	Custom
Range	**Valid values are:** • 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
Example	voice.tone.country = Austria

Parameter- voice.tone.dial voice.tone.ring voice.tone.busy voice.tone.congestion voice.tone.callwaiting voice.tone.dialrecall voice.tone.record voice.tone.info voice.tone.stutter voice.tone.message voice.tone.autoanswer	**Configuration File** <y0000000000xx>.cfg
Description	Customizes the tone for each condition. **F**: the frequency of the tone (ranges from 200 to 7000 Hz). If set to 0 (0Hz), it means a pause between tones. A tone can be composited at most four different frequencies (value format: F1+F2+F3+F4). **D**: the time duration (in milliseconds, ranges from 0 to 30000ms) of ringing the tone. You can configure at most eight different tones for one condition, each tone separated by comma (e.g. 250/200, 0/1000, 200+300/500, 600+700+800+1000/2000). **Note**: It works only if the parameter "voice.tone.country" is set to Custom.
Format	**F/D**
Default Value	Blank
Range	Not Applicable
Example	voice.tone.dial = 800+200/1000, 0/100, 500/1200, 500+600+950+1500/5000

Remote Phonebook

Parameter- features.remote_phonebook.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform a remote phonebook search when receiving an incoming call.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.remote_phonebook.enable = 1

Parameter- features.remote_phonebook.flash_time	Configuration File <y0000000000xx>.cfg
Description	Sets how often to refresh the local cache of the remote phonebook. If set to 3600 (3600s), the IP phone refreshes the local cache of the remote phonebook every 3600 seconds.
Format	Integer
Default Value	3600
Range	120 to 2592000
Example	features.remote_phonebook.flash_time = 1800

LDAP

Parameter- ldap.name_filter	Configuration File <y0000000000xx>.cfg
Description	Specifies the name attribute for LDAP searching. The "*" symbol in the filter stands for any character. The "%" symbol in the filter stands for the entering string used as the

	prefix of the filter condition.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.name_filter = (|(cn=%)(sn=%)) When the name prefix of the cn or sn of the contact record matches the search criteria, the record will be displayed on the phone LCD screen.

Parameter- ldap.number_filter	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the number attribute for LDAP searching. The "*" symbol in the filter stands for any character. The "%" symbol in the filter stands for the entering string used as the prefix of the filter condition.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.number_filter = (|(telephoneNumber=%)(Mobile=%)(ipPhone=%)) When the number prefix of the telephoneNumber, Mobile or ipPhone of the contact record matches the search criteria, the record will be displayed on the phone LCD screen.

Parameter- ldap.host	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the domain name or IP address of the LDAP server.
Format	IP Address or Domain Name
Default Value	0.0.0.0
Range	Not Applicable

Example	ldap.host = 192.168.1.20

Parameter- ldap.port	Configuration File <y0000000000xx>.cfg
Description	Specifies the LDAP server port.
Format	Integer
Default Value	389
Range	Not Applicable
Example	ldap.port = 390

Parameter- ldap.base	Configuration File <y0000000000xx>.cfg
Description	Specifies the LDAP search base which corresponds to the location in the LDAP phonebook from which the LDAP search request begins. The search base narrows the search scope and decreases directory search time.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.base = dc=yealink,dc=cn

Parameter- ldap.user	Configuration File <y0000000000xx>.cfg
Description	Specifies the user name to login the LDAP server. It can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the username to access the LDAP server.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.user =

	cn=manager,dc=yealink,dc=cn

Parameter- ldap.password	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the password to login the LDAP server. This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the password to access the LDAP server.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.password = secret

Parameter- ldap.max_hits	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the maximum number of search results to be returned by the LDAP server. If the value of the "Max.Hits" is blank, the LDAP server will return all searched results. Please note that a very large value of the "Max. Hits" will slow down the LDAP search speed, therefore it should be configured according to the available bandwidth.
Format	Integer
Default Value	50
Range	1 to 32000
Example	ldap.max_hits = 60

Parameter- ldap.name_attr	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the name attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple name attributes separated by space.

Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.name_attr = cn sn

Parameter- ldap.numb_attr	Configuration File <y0000000000xx>.cfg
Description	Specifies the number attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple number attributes separated by space.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.numb_attr = telephoneNumber

Parameter- ldap.display_name	Configuration File <y0000000000xx>.cfg
Description	Specifies the display name of the contact record displayed on the LCD screen. **Note**: It must start with "%" symbol.
Format	String
Default Value	Blank
Range	Not Applicable
Example	ldap.display_name = %cn The cn of the contact record is displayed on the LCD screen.

Parameter- ldap.version	Configuration File <y0000000000xx>.cfg
Description	Specifies the LDAP protocol version supported by the IP phone. Make sure the protocol value corresponds with the version assigned on the LDAP server.

Format	Integer
Default Value	3
Range	2 or 3
Example	ldap.version = 3

Parameter ldap.search_delay	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the delay time (in milliseconds) to display the search results on the LCD screen after searching.
Format	Integer
Default Value	2000
Range	0 to 2000
Example	ldap.search_delay = 20

Parameter ldap.call_in_lookup	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform an LDAP search when receiving an incoming call.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	ldap.call_in_lookup = 1

Parameter ldap.ldap_sort	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to sort the search results in alphabetical order or numerical order.
Format	Boolean
Default Value	0
Range	**0**-Disabled

	1-Enabled
Example	ldap.ldap_sort = 1

Parameter- ldap.dial_lookup	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform an LDAP search when pre-dialing or dialing a call.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	ldap.dial_lookup = 1

BLF

Visual and Audio Alert for BLF Pickup

Parameter- features.pickup.blf_visual_enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to display a visual prompt when the monitored user receives an incoming call.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.pickup.blf_visual_enable = 1

Parameter- features.pickup.blf_audio_enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the IP phone to play an alert tone when the monitored user receives an incoming call.
Format	Boolean

Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	features.pickup.blf_audio_enable = 1

BLF List

Parameter- account.x.blf.blf_list_uri	**Configuration File** <MAC>.cfg
Description	Specifies the URI used to access the BLF list configured on the SIP server for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.blf.blf_list_uri = blf_3601

Parameter- account.x.blf_list_code	**Configuration File** <MAC>.cfg
Description	Configures the feature access code used to pick up the ringing call of the monitored user for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.blf_list_code = *65

Shared Call Appearance

Use the following parameters to register the shared line on the IP phone.

Parameter- account.x.shared_line	**Configuration File** <MAC>.cfg
Description	Configures the line type for account X.

	X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	**Valid values are:** **0**-Disabled **1**-Broadsoft SCA **2**-BLA
Example	account.1.shared_line = 1

Parameter- account.x.enable	**Configuration File** <MAC>.cfg
Description	Enables or disables account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.enable = 1

Parameter- account.x.label	**Configuration File** <MAC>.cfg
Description	Configures the label of the account X to be displayed on the IP phone. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.label = sca

Parameter- account.x.display_name	**Configuration File** <MAC>.cfg
Description	Configures the display name of the account X. X ranges from 1 to 6.

Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.display_name = 2413333601

Parameter- account.x.auth_name	**Configuration File** <MAC>.cfg
Description	Configures the register name of the account X. X ranges from 1 to 6. **Note:** If configuring the secondary line on the IP phone, enter the register name of the primary line for this parameter.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.auth_name = 2413333601

Parameter- account.x.password	**Configuration File** <MAC>.cfg
Description	Configures the password of the account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.password = userpassword

Parameter- account.x.user_name	**Configuration File** <MAC>.cfg
Description	Configures the user name of the account X. X ranges from 1 to 6.
Format	String
Default Value	Blank

Range	Not Applicable
Example	account.1.user_name = 2413333601

Parameter- account.x.sip_server_host	**Configuration File** <MAC>.cfg
Description	Configures the SIP server address for account X. X ranges from 1 to 6.
Format	IP Address or Domain Name
Default Value	Blank
Range	Not Applicable
Example	account.1.sip_server_host = server@domain name.net

Parameter- account.x.sip_server_port	**Configuration File** <MAC>.cfg
Description	Configures the SIP server port for account X. X ranges from 1 to 6.
Format	Integer
Default Value	5060
Range	Not Applicable
Example	account.1.sip_server_port = 5060

Parameter- account.x.outbound_proxy_enable	**Configuration File** <MAC>.cfg
Description	Enables or disables the outbound proxy server for account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.outbound_proxy_enable = 1

Parameter- account.x.outbound_host	Configuration File <MAC>.cfg
Description	Configures the address of the outbound proxy server for account X. X ranges from 1 to 6.
Format	IP Address
Default Value	Blank
Range	Not Applicable
Example	account.1.outbound_host = 199.19.195.10

Parameter- account.x.outbound_port	Configuration File <MAC>.cfg
Description	Configures the outbound proxy server port for account X. X ranges from 1 to 6.
Format	Integer
Default Value	5060
Range	Not Applicable
Example	account.1.outbound_port = 5060

As-Feature-Event

Parameter- bw.feature_key_sync	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the as-feature-event feature. If set to 1 (Enabled), the IP phone and the server can synchronize the status of the following features with each other: • Do Not Disturb • Call Forwarding Always (CFA) • Call Forwarding Busy (CFB) • Call Forwarding No Answer (CFNA)

	• ACD
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	bw.feature_key_sync = 1

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 for account X. If set to 1 (Disabled), the IP phone sends a SUBSCRIBE message to the server for message-summary updates. X ranges from 1 to 6.
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	Configures MWI subscribe expiry time (in seconds) for account X. The IP phone is able to successfully refresh the SUBCRIBE for message-summary events before expiration of the SUBSCRIBE dialog. X ranges from 1 to 6. **Note**: It works only if the parameter "account.x.subscribe_mwi" is set to 1 (Enabled).

Format	Integer
Default Value	3600
Value	0 to 84600
Example	account.1.subscribe_mwi_expires = 3600

Action URL

Parameter-	**Configuration File**
action_url.setup_completed	<y0000000000xx>.cfg
action_url.log_on	
action_url.log_off	
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.forward_incoming_call action_url.reject_incoming_call action_url.answer_new_incoming_call action_url.transfer_finished action_url.transfer_failed	
Description	Specifies the URL for the predefined event. The value format is: http://IP address of server/help.xml? variable name=variable value **Valid variable values are:** • $mac • $ip • $model • $firmware • $active_url • $active_user • $active_host • $local • $remote • $display_local • $display_remote • $call_id
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	action_url.mute = http://192.168.0.20/help.xml?model=$model

Action URI

Parameter- features.action_uri_limit_ip	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the address(es) from which Action URI will be accepted. Multiple IP addresses are separated by comma. If left blank, the IP phone cannot receive or handle any HTTP GET request. If set to "any", the IP phone accepts and

	handles HTTP GET requests from any IP address.
Format	IP Address
Default Value	Blank
Range	IP address or any
Example	features.action_uri_limit_ip = any

Server Redundancy

Parameter- account.x.transport	**Configuration File** <MAC>.cfg
Description	Configures the transport type for account X. If set to 3 (DNS SRV), the IP phone is able to perform DNS SRV query, and fail over the request to the secondary server when there is no response from the primary server. X ranges from 1 to 6.
Format	Integer
Default Value	0 (UDP)
Range	**Valid values are:** **0**-UDP **1**-TCP **2**-TLS **3**-DNS SRV
Example	account.1.transport = 3

LLDP

Parameter- network.lldp.enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the 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	**0**-Disabled **1**-Enabled
Example	network.lldp.enable = 1

Parameter- network.lldp.packet_interval	Configuration File <y0000000000xx>.cfg
Description	Configures the amount of time (in seconds) between the transmission of LLDP packets. **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 = 150

VLAN

Internet Port

Parameter- network.vlan.internet_port_enable	Configuration File <y0000000000xx>.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	**0**-Disabled **1**-Enabled
Example	network.vlan.internet_port_enable = 1

Parameter- network.vlan.internet_port_vid	Configuration File <y0000000000xx>.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	0 to 4094
Example	network.vlan.internet_port_vid = 1

Parameter- network.vlan.internet_port_priority	Configuration File <y0000000000xx>.cfg
Description	Specifies 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.internet_port_priority = 1

PC Port

Parameter- network.vlan.pc_port_enable	Configuration File <y0000000000xx>.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	**0**-Disabled **1**-Enabled

Example	network.vlan.pc_port_enable = 1

Parameter- network.vlan.pc_port_vid	Configuration File <y0000000000xx>.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	0
Range	0 to 4094
Example	network.vlan.pc_port_vid = 1

Parameter- network.vlan.pc_port_priority	Configuration File <y0000000000xx>.cfg
Description	Specifies 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 = 1

VPN

Parameter- network.vpn_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the VPN 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	0
Range	**0**-Disabled **1**-Enabled
Example	network.vpn_enable = 1

Parameter- openvpn.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the OpenVPN tar package.
Format	String
Default Value	Blank
Range	Not Applicable
Example	openvpn.url = http://192.168.10.25/OpenVPN.tar

QOS

Parameter- network.qos.rtptos	**Configuration File** <y0000000000xx>.cfg
Description	Configure 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.rtptos = 50

Parameter- network.qos.signaltos	**Configuration File** <y0000000000xx>.cfg
Description	Configure 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.signaltos = 30

Network Address Translation

Parameter- account.x.nat.nat_traversal	**Configuration File** <MAC>.cfg
Description	Enables or disables the NAT traversal for account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	account.1.nat.nat_traversal = 1

Parameter- account.x.nat.stun_server	**Configuration File** <MAC>.cfg
Description	Specifies the IP address or the domain name of the STUN server for account X. X ranges from 1 to 6.
Format	IP Address or Domain Name
Default Value	Blank
Range	Not Applicable
Example	account.1.nat.stun_server = 192.168.1.20

Parameter- account.x.nat.stun_port	Configuration File <MAC>.cfg
Description	Specifies the port of the STUN server. X ranges from 1 to 6.
Format	Integer
Default Value	3478
Range	Not Applicable
Example	account.1.nat.stun_port = 3479

802.1X

Parameter- network.802_1x.mode	Configuration File <y0000000000xx>.cfg
Description	Specifies 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	**0**-Disabled **1**-EAP-MD5
Example	network.802_1x.mode = 1

Parameter- network.802_1x.identity	Configuration File <y0000000000xx>.cfg
Description	Enters 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	Not Applicable
Example	network.802_1x.identity = admin

Parameter- network.802_1x.md5_password	Configuration File <y0000000000xx>.cfg
Description	Enters 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.
Format	String
Default Value	Blank
Range	Not Applicable
Example	network.802_1x.md5_password = admin123

Audio Features Parameters

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 ranges from 1 to 6. Y ranges from 1 to 13.
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 0; When Y=4, the default value is 0; When Y=5, the default value is 1; When Y=6, the default value is 1; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0; When Y=10, the default value is 0;

	When Y=11, the default value is 0; When Y=12, the default value is 0; When Y=13, the default value is 0.
Range	**0**-Disabled **1**-Enabled
Example	account.1.codec.1.enable = 1

Parameter- account.X.codec.Y.payload_type	**Configuration File** <MAC>.cfg
Description	Specifies the codec for account X to use. X ranges from 1 to 6. Y ranges from 1 to 13.
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 G723_53; When Y=4, the default value is G723_63; When Y=5, the default value is G729; When Y=6, the default value is G722; When Y=8, the default value is G726_16; When Y=9, the default value is G726_24; When Y=10, the default value is G726_32; When Y=11, the default value is G726_40; When Y=12, the default value is iLBC_13_3; When Y=13, the default value is iLBC_15_2.
Range	**Valid values are:** • PCMU • PCMA • G729 • G722 • G723_53 • G723_63 • G726_16 • G726_24 • G726_32 • G726_40 • iLBC_13_3 • iLBC_15_2

Example	account.1.codec.1.payload_type = G723_53

Parameter- account.X.codec.Y.priority	**Configuration File** <MAC>.cfg
Description	Specifies the priority for the codec. X ranges from 1 to 6. Y ranges from 1 to 13.
Format	Integer
Default Value	When Y=1, the default value is 1; When Y=2, the default value is 2; When Y=3, the default value is 4; When Y=4, the default value is 0; When Y=5, the default value is 3; When Y=6, the default value is 4; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0; When Y=10, the default value is 0; When Y=11, the default value is 0; When Y=12, the default value is 0; When Y=13, the default value is 0.
Range	Not Applicable
Example	account.1.codec.1.priority = 1

Parameter- account.X.codec.Y.rtpmap	**Configuration File** <MAC>.cfg
Description	Configure the rtpmap. X ranges from 1 to 6. Y ranges from 1 to 13.
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 4;

	When Y=4, the default value is 4; When Y=5, the default value is 18; When Y=6, the default value is 9; When Y=7, the default value is 102; When Y=8, the default value is 112; When Y=9, the default value is 102; When Y=10, the default value is 2; When Y=11, the default value is 104; When Y=12, the default value is 97; When Y=13, the default value is 97.
Range	0 to 127
Example	account.1.codec.1.rtpmap = 120

Ptime

Parameter- account.x.ptime	**Configuration File** <MAC>.cfg
Description	Configure the ptime (in milliseconds) for the codec. X ranges from 1 to 6.
Format	Integer
Default Value	20
Range	**Valid values are:** 0 (Disabled) 10, 20, 30, 40, 50, 60
Example	account.1.ptime = 30

Acoustic Echo Cancellation

Parameter- voice.echo_cancellation	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the AEC feature on the IP phone.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled

Example	voice.echo_cancellation = 1

Voice Activity Detection

Parameter- voice.vad	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the VAD feature on the IP phone.
Format	Boolean
Default Value	0
Range	0-Disabled 1-Enabled
Example	voice.vad = 1

Comfort Noise Generation

Parameter- voice.cng	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the CNG feature on the IP phone.
Format	Boolean
Default Value	1
Range	0-Disabled 1-Enabled
Example	voice.cng = 1

Jitter Buffer

Parameter- voice.jib.adaptive	Configuration File <y0000000000xx>.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** <y0000000000xx>.cfg
Description	Configures the minimum delay time (in milliseconds) for jitter buffer. **Note**: It works only if the parameter "voice.jib.adaptive" is set to 1 (Adaptive).
Format	Integer
Default Value	0
Range	Not Applicable
Example	voice.jib.min = 1

Parameter- voice.jib.max	**Configuration File** <y0000000000xx>.cfg
Description	Configures the maximum delay time (in milliseconds) for jitter buffer. **Note**: It works only if the parameter "voice.jib.adaptive" is set to 1 (Adaptive).
Format	Integer
Default Value	300
Range	Not Applicable
Example	voice.jib.max = 200

Parameter- voice.jib.normal	**Configuration File** <y0000000000xx>.cfg
Description	Configures the fixed delay time (in milliseconds) for jitter buffer. **Note**: It works only if the parameter "voice.jib.adaptive" is set to 0 (Fixed).
Format	Integer
Default Value	120
Range	Not Applicable

Example	voice.jib.mormal = 100

Security Feature Parameters

TLS

Parameter- account.x.transport	Configuration File <MAC>.cfg
Description	Configures the transport type for account X. If set to 2 (TLS), the SIP message of this account will be encrypted after the successful TLS negotiation. X ranges from 1 to 6.
Format	Integer
Default Value	0 (UDP)
Range	Valid values are: 0-UDP 1-TCP 2-TLS 3-DNS SRV
Example	account.1.transport = 2

Parameter- security.trust_certificates	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to authenticate the connected server using the certificate in the trusted certificate list.
Format	Boolean
Default Value	1
Range	0-Disabled 1-Enabled
Example	security.trust_certificates = 1

Parameter- account.x.enable_signal_encode	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to encode SIP signal using RC4 encryption algorithm. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	account.1.srtp_encryption = 0

Parameter- account.x.signal_encode_key	Configuration File <MAC>.cfg
Description	Configures the key for the IP phone to encode the SIP signal with RC4. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	Not Applicable
Example	account.1.signal_encode_key = 123abc

Uploading Certificates

Parameter- trusted_certificates.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the trusted certificate. Note: The trusted certificate you want to add must have a .crt or .cer extension.
Format	String
Default Value	Blank
Range	Not Applicable
Example	trusted_certificates.url =

	http://192.168.1.20/tc.crt

Parameter- server_certificates.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the server certificate. **Note**: The server certificate you want to add must have a .pem extension.
Format	String
Default Value	Blank
Range	Not Applicable
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 the set to 1 (Forced), the IP phone is forced to using SRTP during a call. If set to 2 (Negotiated), the IP phone will negotiate with the other IP phone what type of encryption to utilize for the session. X ranges from 1 to 6.
Format	Integer
Default Value	0
Value	**Valid values are:** **0**-Disabled **1**-Forced **2**-Negotiated
Example	account.1.srtp_encryption = 0

Configuring AES Keys

Parameter- auto_provision.aes_key_16.com	Configuration File <y0000000000xx>.cfg
Description	Configures the AES key which is used to encrypt or decrypt the <y0000000000xx>.cfg file.
Format	String () >< | "& cannot be included.
Default Value	Blank
Range	16 characters
Example	auto_provision.aes_key_16.com = 0123456789abcdef

Parameter- auto_provision.aes_key_16.mac	Configuration File <y0000000000xx>.cfg
Description	Configures the AES key which is used to encrypt or decrypt the <MAC>.cfg file.
Format	String () >< | "& cannot be included.
Default Value	Blank
Range	16 characters
Example	auto_provision.aes_key_16.mac = 0123456789abmins

Upgrading the Firmware

Parameter- auto_provision.mode	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to check for new configuration files during booting up.
Format	Boolean
Default Value	1

Range	**Valid values are:** **0**-Disabled **1**-Enabled
Example	auto_provision.mode = 1

Parameter- auto_provision.repeat.enable	**Configuration File** <y000000000068>.cfg
Description	Enable or disable the IP phone to check the new configuration repeatedly.
Format	Boolean
Default Value	0
Range	**0**-Disabled **1**-Enabled
Example	auto_provision.repeat.enable = 1

Parameter- auto_provision.repeat.minutes	**Configuration File** <y000000000068>.cfg
Description	Configure the interval (in minutes) the phone repeatedly checks the new configuration files. **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.enable = 1

Parameter- auto_provision.weekly.enable	**Configuration File** <y000000000068>.cfg
Description	Enable or disable the phone to check the new configuration files weekly.
Format	Boolean
Default Value	0

Range	**0**-Disabled **1**-Enabled
Example	auto_provision.weekly.enable = 1

Parameter- auto_provision.weekly.mask	**Configuration File** <y000000000068>.cfg
Description	Defines the desired day(s) of a week for the phone to check new configuration. **Note**: It works only if the parameter "auto_provision.weekly.enable" is set to 1 (Enabled).
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.mask = 123

Parameter- auto_provision.weekly.begin_time	**Configuration File** <y000000000068>.cfg
Description	Sets the start time of day in 24-hour period for the phone to check new configuration files. **Note**: It works only if the parameter "auto_provision.weekly.enable" is set to 1 (Enabled).
Format	00:00
Default Value	00:00
Range	00:00 to 23:59
Example	auto_provision.weekly.begin_time = 01:30

Parameter- auto_provision.weekly.end_time	Configuration File <y000000000068>.cfg
Description	Sets the end time of day in 24-hour period for the phone to check new configuration files. Note: It works only if the parameter "auto_provision.weekly.enable" is set to 1 (Enabled).
Format	00:00
Default Value	00:00
Range	00:00 to 23:59
Example	auto_provision.weekly.end_time = 02:00

Parameter- firmware.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the firmware.
Format	String
Default Value	Blank
Range	Not Applicable
Example	firmware.url = http://192.168.1.20/2.70.0.50.rom

Resource Files

Access URL of Replace Rule Template

Parameter- dialplan_replace_rule.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the replace rule template.
Format	URL
Default Value	Blank
Range	Not Applicable
Example	dialplan_replace_rule.url =

	http://192.168.10.25/dialplan.xml

Access URL of Dial-now Template

Parameter- dialplan_dialnow.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the dial-now template.
Format	URL
Default Value	Blank
Range	Not Applicable
Example	dialplan_dialnow.url = http://192.168.10.25/dialnow.xml

Access URL of Wallpaper Image

Parameter- wallpaper_upload.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the wallpaper image.
Format	URL
Default Value	Blank
Range	Not Applicable
Example	wallpaper_upload.url = http://192.168.10.25/wallpaper.jpg

Access URL of Screensaver Image

Parameter- screen_saver.pic.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the screensaver image.
Format	URL
Default Value	Blank

Range	Not Applicable
Example	screen_saver.pic.url = http://192.168.10.25/screensaver.jpg

Access URL of Softkey Layout Template

Parameter custom_softkey_call_failed.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the CallFailed state.
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_call_failed.url = http://10.2.8.16:8080/XMLfiles/CallFailed.xml

Parameter custom_softkey_call_in.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the CallIn state.
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_call_in.url = http://10.2.8.16:8080/XMLfiles/CallIn.xml

Parameter- custom_softkey_connecting.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the Connecting state.
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_connecting.url = http://10.2.8.16:8080/XMLfiles/Connecting.xml

Parameter- custom_softkey_dialing.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the Dialing state.
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_dialing.url = http://10.2.8.16:8080/XMLfiles/Dialing.xml

Parameter- custom_softkey_ring_back.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the RingBack state.
Format	URL

Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_ring_back.url = http://10.2.8.16:8080/XMLfiles/RingBack.xml

Parameter- custom_softkey_talking.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the customized file for the soft key presented on the phone LCD screen when in the Talking state.
Format	URL
Default Value	Not Applicable
Range	Not Applicable
Example	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. custom_softkey_talking.url = http://10.2.8.16:8080/XMLfiles/Talking.xml

Access URL of Local Contact File

Parameter- local_contact.data.url	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the access URL of the local contact file.
Format	URL
Default Value	Blank
Range	Not Applicable
Example	local_contact.data.url = http://192.168.10.25/contactData1.xml

Access URL of Remote XML Phonebook

Parameter- remote_phonebook.data.x.url	Configuration File <y0000000000xx>.cfg
Description	Specifies the access URL of the remote XML phonebook. X ranges from 1 to 5.
Format	URL
Default Value	Blank
Range	Not Applicable
Example	remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml

Troubleshooting

Log Settings

Parameter- syslog.server	Configuration File <y0000000000xx>.cfg
Description	Specifies the IP address of the syslog server where to export the log files. **Note:** If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address
Default Value	Blank
Range	Not Applicable
Example	syslog.server = 192.168.1.50

Parameter- syslog.log_level	Configuration File <y0000000000xx>.cfg
Description	Specifies 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 = 2

Watch Dog

Parameter- watch_dog.enable	**Configuration File** <y0000000000xx>.cfg
Description	Enables or disables the Watch Dog feature.
Format	Boolean
Default Value	1
Range	**0**-Disabled **1**-Enabled
Example	watch_dog.enable = 1

Configuring DSS Key

This section provides the DSS key parameters you can configure on the IP phone. The DSS key is consist of the memory key and line key. The following table lists the number of DSS keys you can configure for each phone model:

Phone Model	Line Key	Memory Key
T38G	6	10
T32G	3	/

Various key features can be assigned to the DSS key. The SIP-T38G IP phone supports to be assigned features to the memory keys and line keys. The SIP-T32G IP phone only supports to be assigned features to the line keys. The configurations of the line key are basically the same as the memory key. The parameters of the DSS key are detailed in the following (take the memory key as an example):

Parameter memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the key feature. X ranges from 1 to 10. The value 0 stands for **Auto** (the first available line). **0 stands for Line1 when assigning the following features:** • BLF • Shared Line • BLF List • Call Park • Direct Pickup • ACD • Voice Mail **When assigning the following features, you do not need to configure this parameter:** • DTMF • Prefix • Local Group • XML Group • XML Browser • LDAP • BroadSoft Group • Conference • Forward • Hold • DND • Redial • Call Return • SMS • Record • URL Record • Group Listening • Public Hold • Private Hold

	• Hot Desking • Zero Touch • URL • Keypad Lock
Format	Integer
Default Value	0 (Auto)
Range	**Valid values are**: 0 to 6 (for T38G) 0 to 3 (for T32G)
Example	memorykey.1.line = 2

Parameter- memorykey.x.value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the value for some key features. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	Not Applicable
Example	When assigning the Speed Dial to the memory key, this parameter is used to specify the number you want to dial out. memorykey.1.value = 1001

Parameter- memorykey.x.pickup_value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the pickup code for the BLF feature. This parameter only applies to the BLF feature. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	Not Applicable
Example	memorykey.1.pickup_value = *88

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Specifies the key feature for the memory key. X ranges from 1 to 10. **Valid types are:** • N/A (default for memory key) • Conference • Forward • Transfer • Hold • DND • Redial • Call Return • SMS • Call Pickup • Call Park • DTMF • Voicemail • Speed Dial • Intercom • Line (default for line key) • BLF • URL • Group Listening • Public Hold • Shared Line • Private Hold • XML Group • Group Pickup • Multicast Paging • Record • Hot Desking • XML Browser • URL Record • LDAP • BLF List • Prefix • Zero Touch • ACD • Local Group • BroadSoft Group • Keypad Lock

Format	Integer
Default Value	0 (N/A)
Range	Valid values are: 0-N/A(default for memory key) 1-Conference 2-Forward 3-Transfer 4-Hold 5-DND 6-Redial 7-Call Return 8-SMS 9-Call Pickup 10-Call Park 11-DTMF 12-Voicemail 13-SpeedDial 14-Intercom 15-Line(default for line key) 16-BLF 17-URL 18-Group Listening 19-Public Hold 20-Private Hold 21- Shared Line 22-XML Group 23-Group Pickup 25-Record 27-XML Browser 34-Hot Desking 35-URL Record 38-LDAP 39-BLF List 40-Prefix 41-Zero Touch 42-ACD

	45-Local Group **46**-BroadSoft Group **50**-Keypad Lock
Example	memorykey.1.type = 8

Parameter- memorykey.x.xml_phonebook	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired phonebook when multiple phonebooks are configured on the IP phone. This parameter only applies to the Local Group/XML Group/BroadSoft Group features. X ranges from 1 to 10.
Format	Integer
Default Value	0
Range	Not Applicable
Example	Specify the second phonebook when there are three BroadSoft groups are configured on the IP phone. memorykey.1.xml_phonebook = 2

Keypad Lock Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be **Keypad Lock** key on the IP phone. The digit 50 stands for the key type **Keypad Lock**. X ranges from 1 to 10.
Format	Integer
Value	50
Example	memorykey.1.type = 50

DND Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be DND key on the IP phone. The digit **5** stands for the key type **DND**. X ranges from 1 to 10.
Format	Integer
Value	5
Example	memorykey.1.type = 5

Direct Pickup Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be direct pickup key on the IP phone. The digit **9** stands for the key type **Call Pickup**. X ranges from 1 to 10.
Format	Integer
Value	9
Example	memorykey.1.type = 9

Parameter- memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the direct pickup key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.1.line = 1

Parameter- memorykey.x.value	Configuration File <y0000000000xx>.cfg
Description	Specifies the direct pickup feature code followed by the number of monitored extension. X ranges from 1 to 10.
Format	String
Range	Not Applicable
Example	memorykey.1.value = *971001

Group Pickup Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a line key to be group pickup key on the IP phone. The digit **23** stands for the key type **Group Pickup**. X ranges from 1 to 10.
Format	Integer
Value	23
Example	memorykey.1.type = 23

Parameter- memorykey.x.line	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply the group pickup key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 6 (for T38G) 0 to 3 (for T32G)
Example	memorykey.1.line = 1

Parameter- memorykey.x.value	Configuration File <y0000000000xx>.cfg
Description	Specifies the group pickup feature code. X ranges from 1 to 10.
Format	String
Range	Not Applicable
Example	memorykey.1.value = *98

Call Return Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a memory key to be call return key on the IP phone. The digit **7** stands for the key type **Call Return**. X ranges from 1 to 10.
Format	Integer
Value	7
Example	memorykey.2.type = 7

Call Park Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a memory key to be call park key on the IP phone. The digit **10** stands for the key type **Call Park**. X ranges from 1 to 10.
Format	Integer
Value	10
Example	memorykey.2.type = 10

Parameter- memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the call park key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.2.line = 0

Parameter- memorykey.x.value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the call park feature code. X ranges from 1 to 10.
Format	String
Range	Not Applicable
Example	memorykey.2.value = *99

Intercom Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be the intercom key. The digit **14** stands for the key type **Intercom**. X ranges from 1 to 10.
Format	Integer
Value	14
Example	memorykey.2.type = 14

Parameter- memorykey.x.line	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply the intercom key. X ranges from 1 to 10.
Format	Integer
Range	Valid values are: 0 to 6 (for T38G) 0 to 3 (for T32G)
Example	memorykey.2.line = 1

Parameter- memorykey.x.value	Configuration File <y0000000000xx>.cfg
Description	Specifies the intercom number. X ranges from 1 to 10.
Format	String
Range	Not Applicable
Example	memorykey.2.value = 1008

LDAP Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a memory key to be LDAP key on the IP phone. The digit **38** stands for the key type **LDAP**. X ranges from 1 to 10.
Format	Integer
Value	38
Example	memorykey.2.type = 38

BLF Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be BLF key on the IP phone. The digit **16** stands for the key type **BLF**. X ranges from 1 to 10.
Format	Integer
Value	16
Example	memorykey.3.type = 16

Parameter- memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the BLF key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.3.line = 2

Parameter- memorykey.x.value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the number of the monitored user. X ranges from 1 to 10.
Format	String
Range	Not Applicable
Example	memorykey.3.value = 1008

Parameter- memorykey.x.pickup_value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the pickup code for the BLF

	feature. This parameter only applies to the BLF feature. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	Not Applicable
Example	memorykey.3.pickup_value = *88

BLF List Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be BLF list key on the IP phone. The digit **39** stands for the key type **BLF list**. X ranges from 1 to 10.
Format	Integer
Value	39
Example	memorykey.3.type = 39

Parameter- memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the BLF list key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.2.line = 1

Shared Line Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be a shared line key on the IP phone. The digit **21** stands for the key type **Shared Line**. X ranges from 1 to 10.
Format	Integer
Value	21
Example	memorykey.2.type = 21

Parameter- memorykey.x.value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the primary account.
Format	String
Value	Not Applicable
Example	memorykey.x.value = 2413333612

Parameter- memorykey.x.line	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the desired line to apply the shared line key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.2.line = 1

ACD Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a memory key to be an ACD key on the IP phone. The digit **42** stands for the key type **ACD**. X ranges from 1 to 10.
Format	Integer
Value	42
Example	memorykey.2.type = 42

Parameter- memorykey.x.line	Configuration File <y0000000000xx>.cfg
Description	Specifies the desired line to apply the ACD key. X ranges from 1 to 10.
Format	Integer
Range	**Valid values are**: 0 to 5 (for T38G) 0 to 2 (for T32G)
Example	memorykey.2.line = 1

Record Key

Parameter- memorykey.x.type	Configuration File <y0000000000xx>.cfg
Description	Configures a memory key to be a record key on the IP phone. The digit **25** stands for the key type **Record**. X ranges from 1 to 10.
Format	Integer
Value	25
Example	memorykey.2.type = 25

URL Record Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be a URL record key on the IP phone. The digit **35** stands for the key type **URL Record**. X ranges from 1 to 10.
Format	Integer
Value	35
Example	memorykey.2.type = 35

Parameter- memorykey.x.value	**Configuration File** <y0000000000xx>.cfg
Description	Specifies the URL to record a call. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	Not Applicable
Example	memorykey.1.value = http://10.1.2.224/phonerecording.cgi

Hot Desking Key

Parameter- memorykey.x.type	**Configuration File** <y0000000000xx>.cfg
Description	Configures a memory key to be a hot desking key on the IP phone. The digit **34** stands for the key type **hot desking**. X ranges from 1 to 10.
Format	Integer
Value	34
Example	memorykey.2.type = 34

Appendix D: SIP (Session Initiation Protocol)

This section describes how the Yealink SIP-T3xG IP phones comply with the IETF definition of SIP as described in RFC 3261.

This section contains compliance information in the following:

- RFC and Internet Draft Support
- SIP Request
- SIP Header
- SIP Responses
- SIP Session Description Protocol (SDP) Usage

RFC and Internet Draft Support

The following RFC's and Internet drafts are supported:

- RFC 1321—The MD5 Message-Digest Algorithm
- RFC 2327—SDP: Session Description Protocol
- RFC 2387—The MIME Multipart / Related Content-type
- RFC 2976—The SIP INFO Method
- RFC 3261—SIP: Session Initiation Protocol (replacement for RFC 2543)
- RFC 3262—Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 3263—Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264—An Offer/Answer Model with the Session Description Protocol (SDP)
- RFC 3265—Session Initiation Protocol (SIP) - Specific Event Notification
- RFC 3311—The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3325—SIP Asserted Identity
- RFC 3515—The Session Initiation Protocol (SIP) Refer Method
- RFC 3555—MIME Type of RTP Payload Formats
- RFC 3611—RTP Control Protocol Extended reports (RTCP XR)
- RFC 3665—Session Initiation Protocol (SIP) Basic Call Flow Examples
- draft-ietf-sip-cc-transfer-05.txt—SIP Call Control - Transfer
- RFC 3725—Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)
- RFC 3842—A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 3856—A Presence Event Package for Session Initiation Protocol (SIP)

- RFC 3891—The Session Initiation Protocol (SIP) "Replaces" Header
- RFC 3892—The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3968—The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session Initiation Protocol (SIP)
- RFC 3969—The Internet Assigned Number Authority (IANA) Uniform Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP)
- RFC 4028—Session Timers in the Session Initiation Protocol (SIP)
- RFC 4235—An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4662—Session Initiation Protocol (SIP) Event Notification Extension for Resource Lists
- draft-levy-sip-diversion-04.txt—Diversion Indication in SIP
- draft-anil-sipping-bla-02.txt—Implementing Bridged Line Appearances (BLA) Using Session Initiation Protocol (SIP)
- draft-ietf-sip-privacy-04.txt—SIP Extensions for Network-Asserted Caller Identity and Privacy within Trusted Networks
- draft-levy-sip-diversion-06.txt—Diversion Indication in SIP
- draft-ietf-sipping-cc-conferencing-03.txt—SIP Call Control - Conferencing for User Agents
- draft-ietf-sipping-rtcp-summary-02.txt —Session Initiation Protocol Package for Voice Quality Reporting Event
- draft-ietf-sip-connect-reuse-04.txt—Connection Reuse in the Session Initiation Protocol (SIP)

To find the applicable Request for Comments (RFC) document, go to http://www.ietf.org/rfc.html and enter the RFC number.

SIP Request

The following SIP request messages are supported:

Method	Supported	Notes
REGISTER	Yes	
INVITE	Yes	The Yealink SIP-T3xG IP phones support mid-call changes such as putting a call on hold as signaled by a new INVITE that contains an existing Call-ID.

Method	Supported	Notes
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	
NOTIFY	Yes	
REFER	Yes	
PRACK	Yes	
INFO	Yes	
MESSAGE	Yes	
UPDATE	Yes	
PUBLISH	Yes	

SIP Header

The following SIP request headers are supported:

Method	Supported	Notes
Accept	Yes	
Alert-Info	Yes	
Allow	Yes	
Allow-Events	Yes	
Authorization	Yes	
Call-ID	Yes	
Call-Info	Yes	
Contact	Yes	
Content-Length	Yes	
Content-Type	Yes	
CSeq	Yes	
Diversion	Yes	
Event	Yes	

Method	Supported	Notes
Expires	Yes	
From	Yes	
Max-Forwards	Yes	
Min-SE	Yes	
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	
RAck	Yes	
Record-Route	Yes	
Refer-To	Yes	
Referred-By	Yes	
Remote-Party-ID	Yes	
Replaces	Yes	
Require	Yes	
Route	Yes	
RSeq	Yes	
Session-Expires	Yes	
Subscription-State	Yes	
Supported	Yes	
To	Yes	
User-Agent	Yes	
Via	Yes	

SIP Responses

The following SIP responses are supported:

1xx Response—Information Responses

1xx Response	Supported	Notes
100 Trying	Yes	

1xx Response	Supported	Notes
180 Ringing	Yes	
181 Call Is Being Forwarded	Yes	
183 Session Progress	Yes	

2xx Response—Successful Responses

2xx Response	Supported	Notes
200 OK	Yes	
202 Accepted	Yes	In REFER transfer.

3xx Response—Redirection Responses

3xx Response	Supported	Notes
300 Multiple Choices	Yes	
301 Moved Permanently	Yes	
302 Moved Temporarily	Yes	

4xx Response—Request Failure Responses

4xx Response	Supported	Notes
400 Bad Request	Yes	
401 Unauthorized	Yes	
402 Payment Required	Yes	
403 Forbidden	Yes	
404 Not Found	Yes	
405 Method Not Allowed	Yes	
406 Not Acceptable	No	
407 Proxy Authentication Required	Yes	
408 Request Timeout	Yes	
409 Conflict	No	
410 Gone	No	
411 Length Required	No	

4xx Response	Supported	Notes
413 Request Entity Too Large	No	
414 Request-URI Too Long	Yes	
415 Unsupported Media Type	Yes	
416 Unsupported URI Scheme	No	
420 Bad Extension	No	
421 Extension Required	No	
423 Interval Too Brief	Yes	
480 Temporarily Unavailable	Yes	
481 Call/Transaction Does Not Exist	Yes	
482 Loop Detected	Yes	
483 Too Many Hops	No	
484 Address Incomplete	Yes	
485 Ambiguous	No	
486 Busy Here	Yes	
487 Request Terminated	Yes	
488 Not Acceptable Here	Yes	
491 Request Pending	No	
493 Undecipherable	No	

5xx Response—Server Failure Responses

5xx Response	Supported	Notes
500 Internal Server Error	Yes	
501 Not Implemented	Yes	
502 Bad Gateway	No	
503 Service Unavailable	No	
504 Gateway Timeout	No	
505 Version Not Supported	No	

6xx Response—Global Responses

6xx Response	Supported	Notes
600 Busy Everywhere	Yes	
603 Decline	Yes	
604 Does Not Exist Anywhere	No	
606 Not Acceptable	No	

SIP Session Description Protocol (SDP) Usage

SDP Headers	Supported
v—Protocol version	Yes
o—Owner/creator and session identifier	Yes
a—Media attribute	Yes
c—Connection information	Yes
m—Media name and transport address	Yes
s—Session name	Yes
t—Active time	Yes

Appendix E: SIP Call Flows

SIP uses six request methods:

- INVITE—Indicates a user is being invited to participate in a call session.
- ACK—Confirms that the client has received a final response to an INVITE request.
- BYE—Terminates a call and can be sent by either the caller or the callee.
- CANCEL—Cancels any pending searches but does not terminate a call that has already been accepted.
- OPTIONS—Queries the capabilities of servers.
- REGISTER—Registers the address listed in the To header field with a SIP server.

The following types of responses are used by SIP and generated by the IP phone or the SIP server:

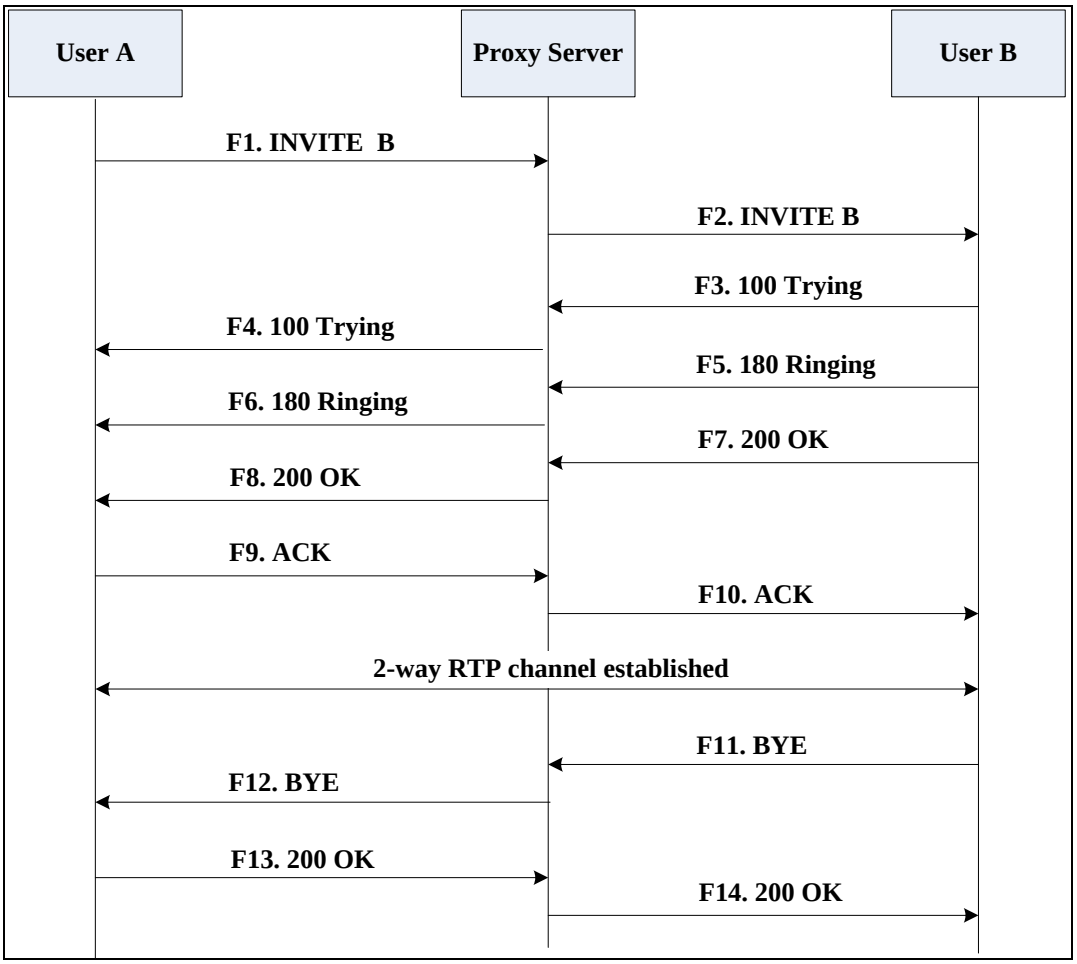
- SIP 1xx—Informational Responses
- SIP 2xx—Successful Responses
- SIP 3xx—Redirection Responses
- SIP 4xx—Client Failure Responses
- SIP 5xx—Server Failure Responses
- SIP 6xx—Global Failure Responses

Successful Call Setup and Disconnect

The following figure illustrates the scenario of a successful call. In this scenario, the two end users are User A and User B. User A and User B are located at the Yealink SIP IP phones.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B hangs up.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends a SIP INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	100 Trying—User B to Proxy Server	User B sends a SIP 100 Trying response to the proxy server. The 100 Trying response indicates that the INVITE request has been received by User B.
F4	100 Trying—Proxy Server to User A	The proxy server forwards the SIP 100 Trying to User A to indicate that the INVITE request has been received by User B.
F5	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the User B is being alerted.
F6	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.

Step	Action	Description
F7	200 OK— User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F8	200OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.
F9	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F10	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F11	BYE—User B to Proxy Server	User B terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User B wants to release the call.
F12	BYE—Proxy Server to User A	The proxy server forwards the SIP BYE request to User A to notify that User B wants to release the call.
F13	200 OK—User A to Proxy Server	User A sends a SIP 200 OK response to the proxy server. The 200 OK response indicates that User A has received the BYE request. The call session is now terminated.
F14	200 OK—Proxy Server to User B	The proxy server forwards the SIP 200 OK response to User B to indicate that User A has received the BYE request. The call session is now terminated.

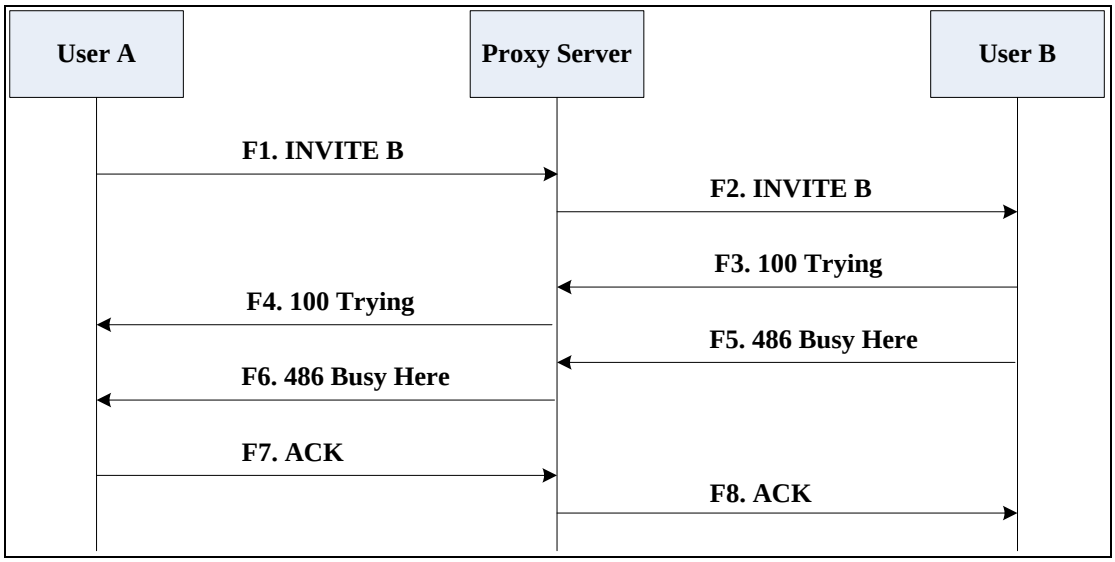
Unsuccessful Call Setup—Called User is Busy

The following figure illustrates the scenario of an unsuccessful call due to the reason of the called user being busy. In this scenario, the two end users are User A and User B. User A and User B are located at the Yealink SIP IP phones.

The call flow scenario is as follows:

1. User A calls User B.
2. User B is busy on the IP phone and unable or unwilling to take another call.

   The call cannot be set up successfully.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	100 Trying—User B to Proxy Server	User B sends a SIP 100 Trying response to the proxy server. The 100 Trying response indicates that the INVITE request has been received by User B.
F4	100 Trying—Proxy Server to User A	The proxy server forwards the SIP 100 Trying to User A to indicate that the INVITE request has already been received.
F5	486 Busy Here—User B to Proxy Server	User B sends a SIP 486 Busy Here response to the proxy server. The 486 Busy Here response is a client error response indicating that User B is successfully connected but User B is busy on the IP phone and unable or unwilling to take the call.

Step	Action	Description
F6	486 Busy Here—Proxy Server to User A	The proxy server forwards the 486 Busy Here response to notify User A that User B is busy.
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The SIP ACK message indicates that User A has received the 486 Busy Here message.
F8	ACK—Proxy Server to User B	The proxy server forwards the SIP ACK to User B to indicate that the 486 Busy Here message has already been received.

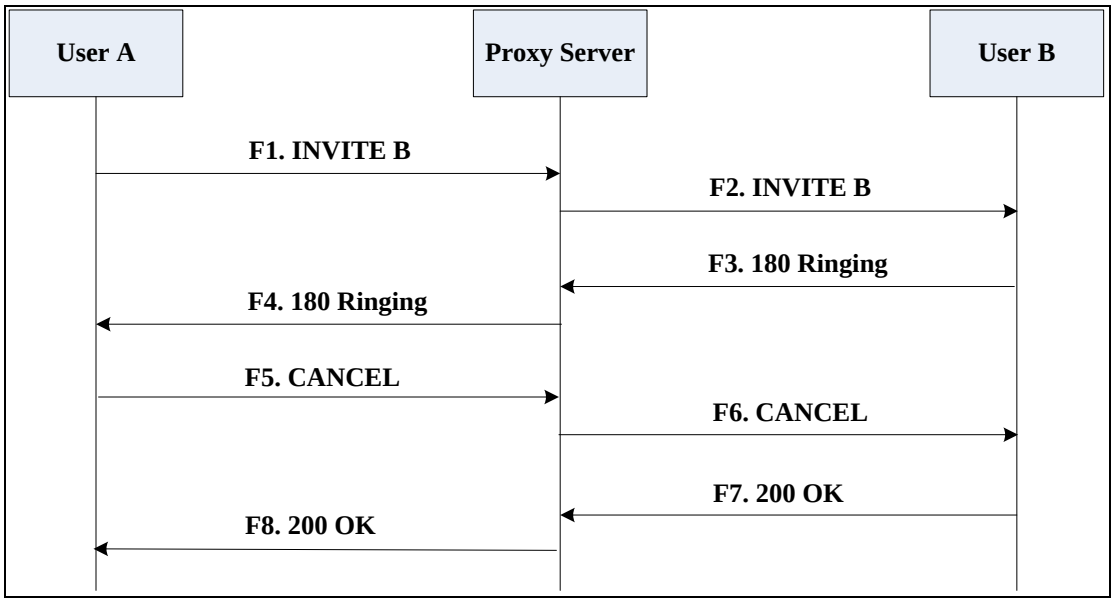
Unsuccessful Call Setup—Called User Does Not Answer

The following figure illustrates the scenario of an unsuccessful call due to the reason of the called user not answering the call. In this scenario, the two end users are User A and User B. User A and User B are located at the Yealink SIP IP phones.

The call flow scenario is as follows:

1. User A calls User B.
2. User B does not answer the call.
3. User A hangs up.

   The call cannot be set up successfully.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	CANCEL—User A to Proxy Server	User A sends a SIP CANCEL request to the proxy server after not receiving an appropriate response within the time allocated in the INVITE request. The SIP CANCEL request indicates that User A wants to disconnect the call.
F6	CANCEL—Proxy Server to	The proxy server forwards the SIP CANCEL request to notify User B that

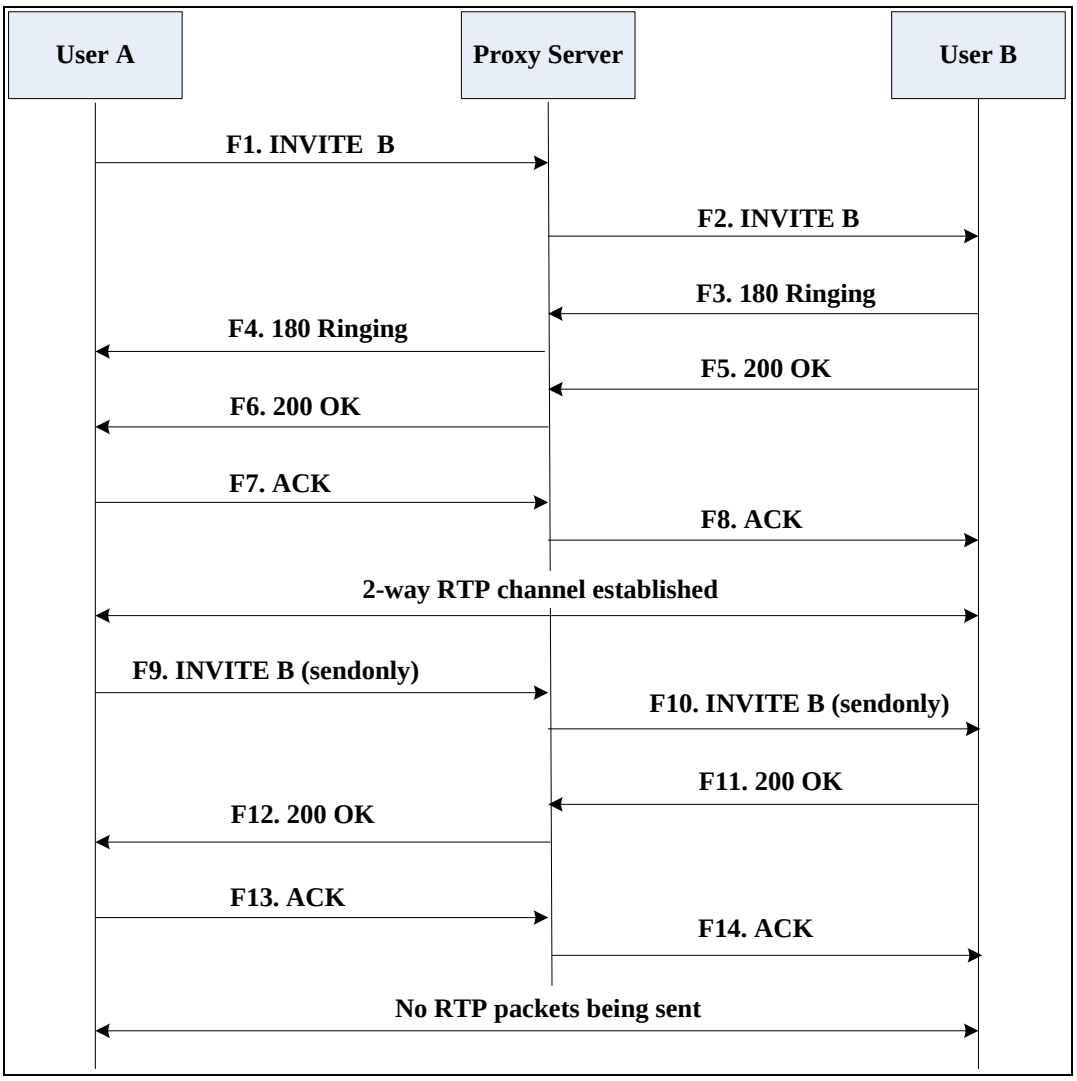
Step	Action	Description
	User B	User A wants to disconnect the call.
F7	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The SIP 200 OK response indicates that User B has received the CANCEL request.
F8	200 OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to notify User A that the CANCEL request has been processed successfully.

Successful Call Setup and Call Hold

The following figure illustrates a successful call setup and call hold. In this scenario, the two end users are User A and User B. User A and User B are located at the Yealink SIP IP phones.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User A puts User B on hold.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies the proxy server that the connection has been made.
F6	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE is successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.

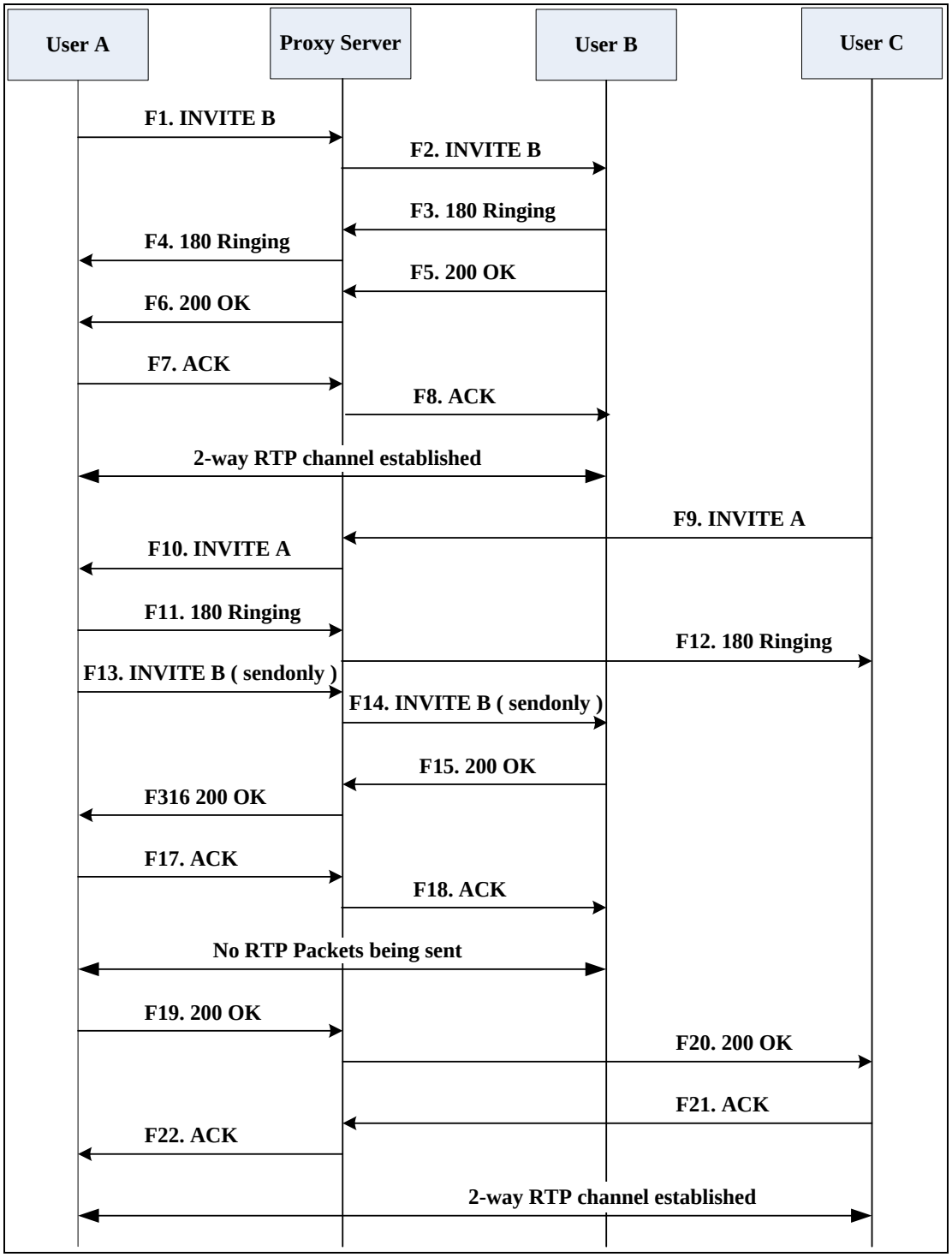
Successful Call Setup and Call Waiting

The following figure illustrates a successful call between Yealink SIP IP phones in which parties are in a call, one of the participants receives a call from a third party, then answers the incoming call. In this call flow scenario, the end users are User A, User B,

and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User C calls User B.
4. User B accepts the call from User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies proxy server that the connection has been made.
F6	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server, The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE—User C to Proxy Server	User C sends a SIP INVITE message to the proxy server. The INVITE request is an invitation to User A to participate in a call session. In the INVITE request: • The IP address of User A is inserted in the Request-URI field. • User C is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User C is ready to receive is specified. • The port on which User A is prepared to receive the RTP data is specified.
F10	INVITE—Proxy Server to User A	The proxy server maps the SIP URI in the To field to User A. The proxy server sends the INVITE message to User A.
F11	180 Ringing—User A to Proxy Server	User A sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing—Proxy Server to User C	The proxy server forwards the 180 Ringing response to User C. User C hears the ring-back tone indicating that User A is being alerted.

Step	Action	Description
F13	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F14	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F15	200 OK—User B to Proxy Server	User B sends a 200 OK to the proxy server. The 200 OK response indicates that the INVITE was successfully processed.
F16	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F17	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F18	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F19	200 OK—User A to Proxy Server	User A sends a 200 OK response to the proxy server. The 200 OK response notifies that the connection has been made.
F20	200 OK—Proxy Server User C	The proxy server forwards the 200 OK message to User C.
F21	ACK—User C to Proxy Server	User C sends a SIP ACK to the proxy server. The ACK confirms that User C has received the 200 OK response. The call session is now active.
F22	ACK—Proxy Server to User A	The proxy server forwards the SIP ACK to User A to confirm that User C has received the 200 OK response.

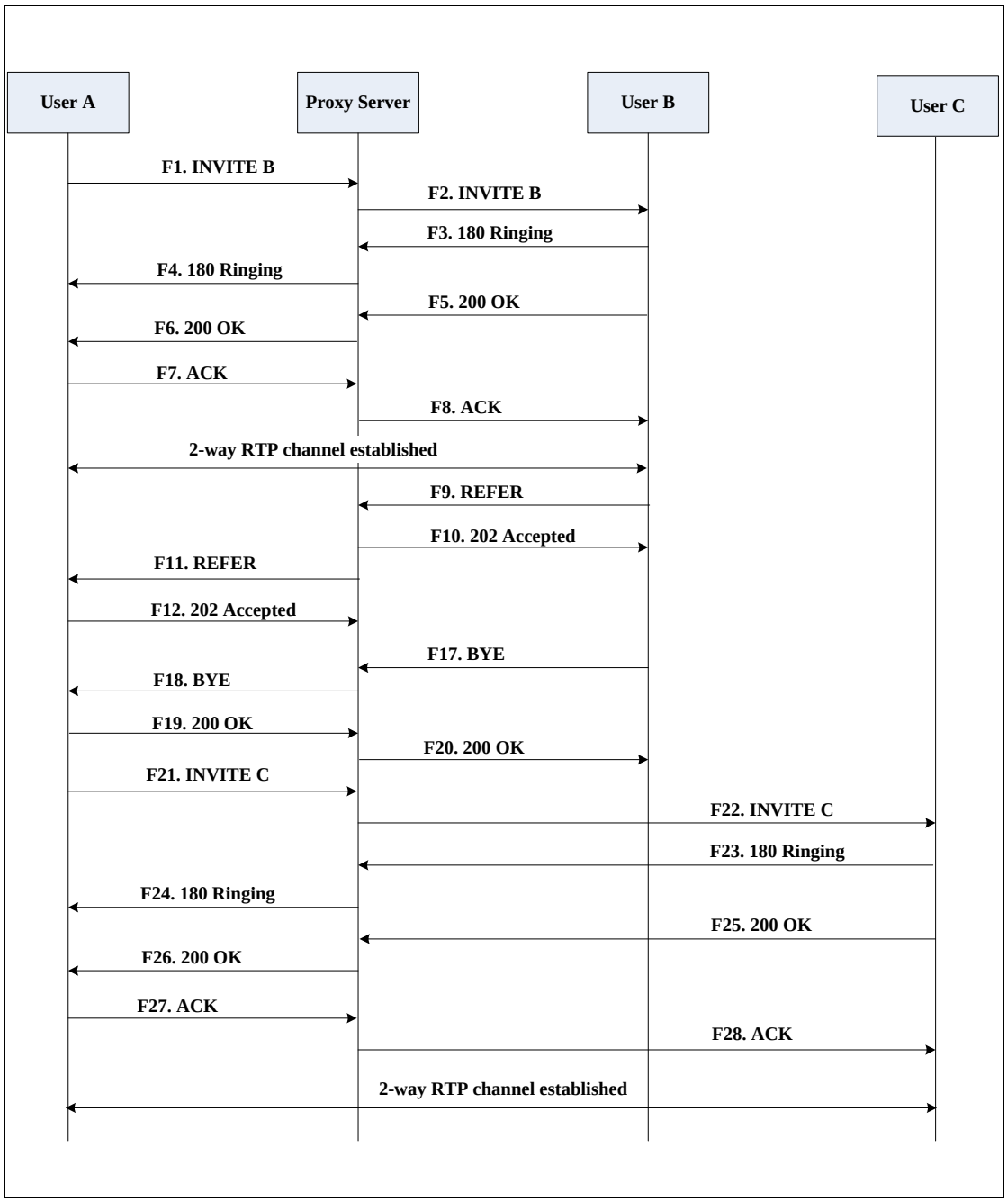
Call Transfer without Consultation

The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to a third party without consulting the third party. This is called a blind transfer. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B transfers the call to User C.
4. User C answers the call.

   Call is established between User A and User C.


User A
Proxy Server
User B
User C
F1. INVITE B
F2. INVITE B
F3. 180 Ringing
F4. 180 Ringing
F5. 200 OK
F6. 200 OK
F7. ACK
F8. ACK
2-way RTP channel established
F9. REFER
F10. 202 Accepted
F11. REFER
F12. 202 Accepted
F17. BYE
F18. BYE
F19. 200 OK
F20. 200 OK
F21. INVITE C
F22. INVITE C
F23. 180 Ringing
F24. 180 Ringing
F25. 200 OK
F26. 200 OK
F27. ACK
F28. ACK
2-way RTP channel established

Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to the proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server, The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	REFER—User B to Proxy Server	User B sends a REFER message to the proxy server. User B performs a blind transfer of User A to User C.
F10	202 Accepted—Proxy Server to User B	The proxy server sends a SIP 202 Accept response to User B. The 202 Accepted response notifies User B that the proxy server has received the REFER message.
F11	REFER—Proxy Server to User A	The proxy server forwards the REFER message to User A.
F12	202 Accepted—User A to Proxy Server	User A sends a SIP 202 Accept response to the proxy server. The 202 Accepted response indicates that User A accepts the transfer.
F13	BYE—User B to Proxy Server	User B terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User B wants to release the call.
F14	BYE—Proxy Server to User A	The proxy server forwards the BYE request to User A.
F15	200OK—User A to Proxy Server	User A sends a SIP 200 OK response to the proxy server. The 200 OK response confirms that User A has received the BYE request.
F16	200OK—Proxy Server to User B	The proxy server forwards the SIP 200 OK response to User B.
F17	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A

Step	Action	Description
		requests the call.
F18	INVITE—Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C.
F19	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F20	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted
F21	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies the proxy server that the connection has been made.
F22	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.
F23	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F24	ACK—Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that User A has received the 200 OK response. The call session is now active.

Call Transfer with Consultation

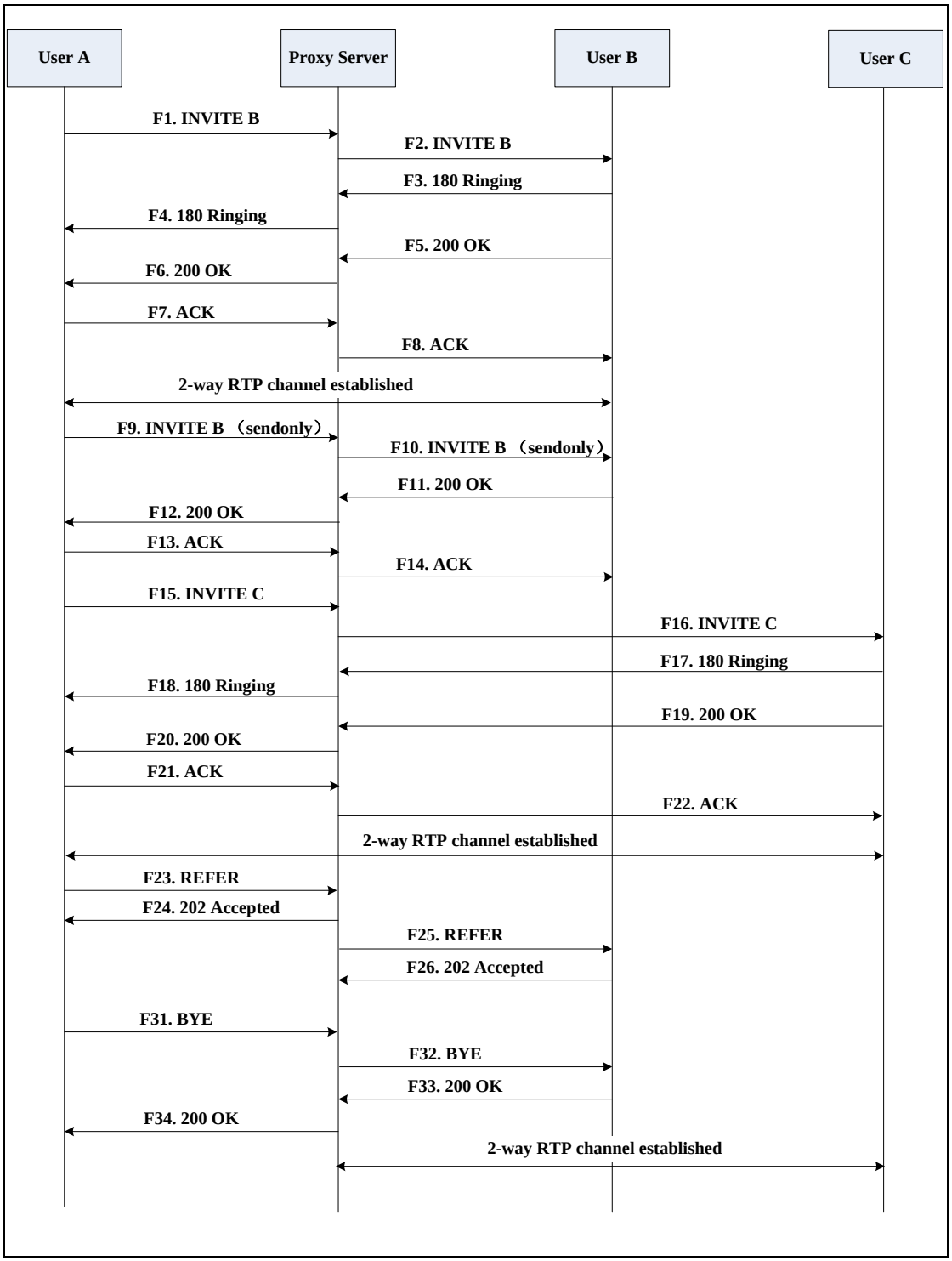
The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to the third party with consultation. This is called attended transfer. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User A calls User C.
4. User C answers the call.

5. User A transfers the call to User C.

   Call is established between User B and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server, The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE was successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F16	INVITE—Proxy Server to User	The proxy server maps the SIP URI to in the To field to User C. The proxy server

Step	Action	Description
	C	sends the INVITE request to User C.
F17	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F18	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F19	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F20	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F21	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F22	ACK—Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F23	REFER—User A to Proxy Server	User A sends a REFER message to the proxy server. User A performs a transfer of User B to User C.
F24	202 Accepted—Proxy Server to User A	The proxy server sends a SIP 202 Accepted response to User A. The 202 Accepted response notifies User A that the proxy server has received the REFER message.
F25	REFER—Proxy Server to User B	The proxy server forwards the REFER message to User B.
F26	202 Accepted—User B to Proxy Server	User B sends a SIP 202 Accept response to the proxy server. The 202 Accepted

Step	Action	Description
		response indicates that User B accepts the transfer.
F27	BYE—User A to Proxy Server	User A terminates the call session by sending a SIP BYE request to the proxy server. The BYE request indicates that User A wants to release the call.
F28	BYE—Proxy Server to User B	The proxy server forwards the BYE request to User B.
F29	200OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that User B has received the BYE request.
F30	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.

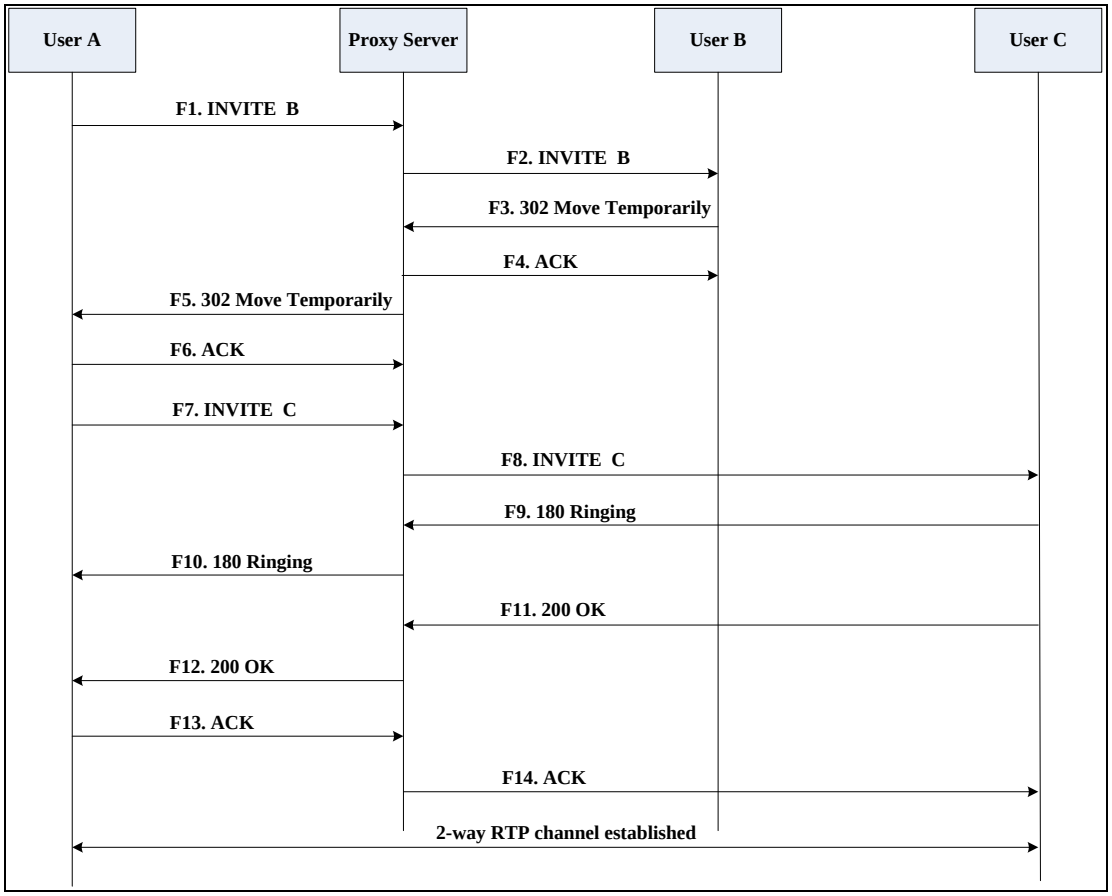
Always Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled always call forward. The incoming call is immediately forwarded to User C when User A calls User B. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User B enables always call forward, and the destination number is User C.
2. User A calls User B.
3. User B forwards the incoming call to User C.
4. User C answers the call.

   Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of the User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	302 Move Temporarily—User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F4	ACK—Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the 302 Move Temporarily message.
F5	302 Move Temporarily—Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F6	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the 302 Move Temporarily message.

Step	Action	Description
F7	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requested the call.
F8	INVITE—Proxy Server to User C	The proxy server maps the SIP URI in the To field to User C. The proxy server sends the SIP INVITE request to User C.
F9	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F10	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F11	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F12	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F13	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F14	ACK—Proxy Server to User C	The proxy server forwards the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.

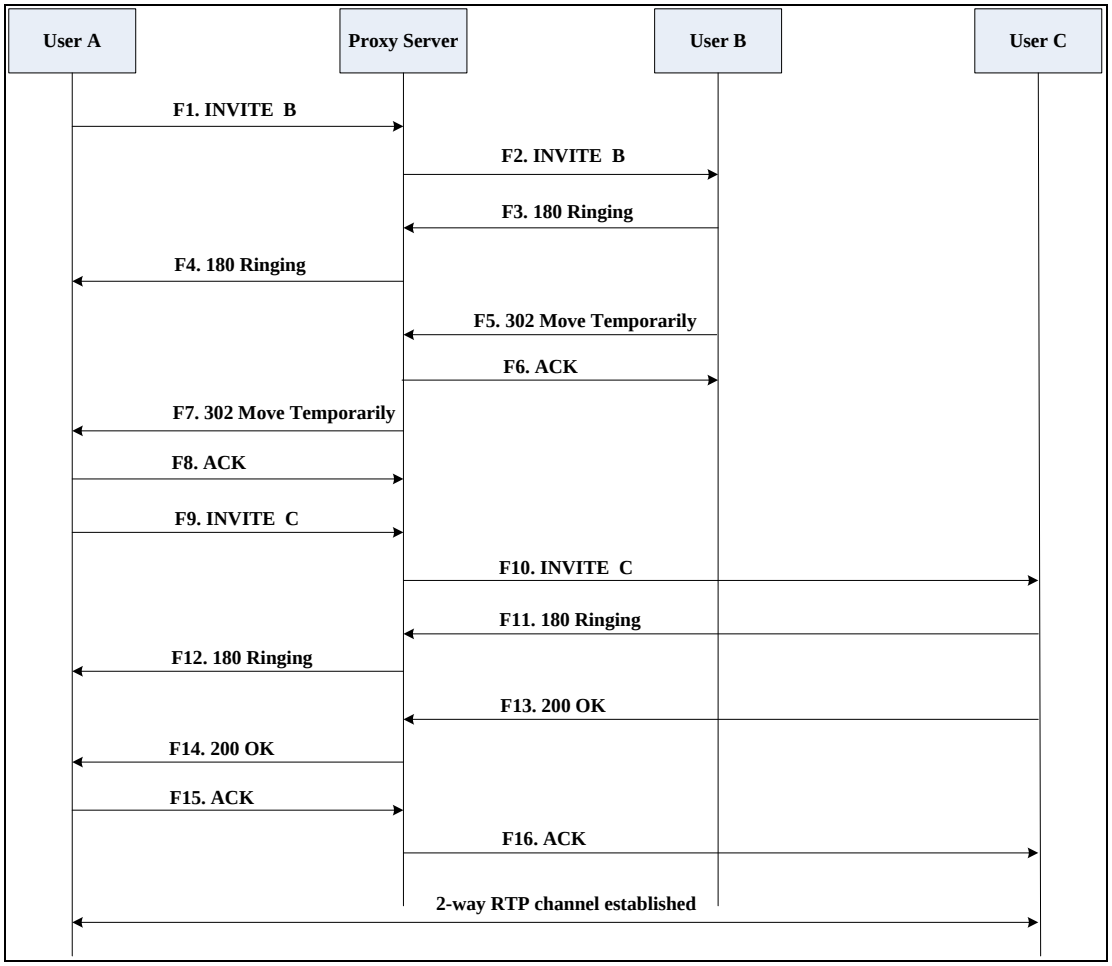
Busy Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled busy call forward. The incoming call is forwarded to User C when User B is busy. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User B enables busy call forward, and the destination number is User C.
2. User A calls User B.
3. User B is busy.
4. User B forwards the incoming call to User C.
5. User C answers the call.

   Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	302 Move Temporarily—User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F6	ACK—Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the

Step	Action	Description
		ACK message.
F7	302 Move Temporarily—Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F8	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the ACK message.
F9	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F10	INVITE—Proxy Server to User C	The proxy server forwards the SIP INVITE request to User C.
F11	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F13	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F14	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A.
F15	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F16	ACK—Proxy Server to User C	The proxy server sends the ACK message to User C.

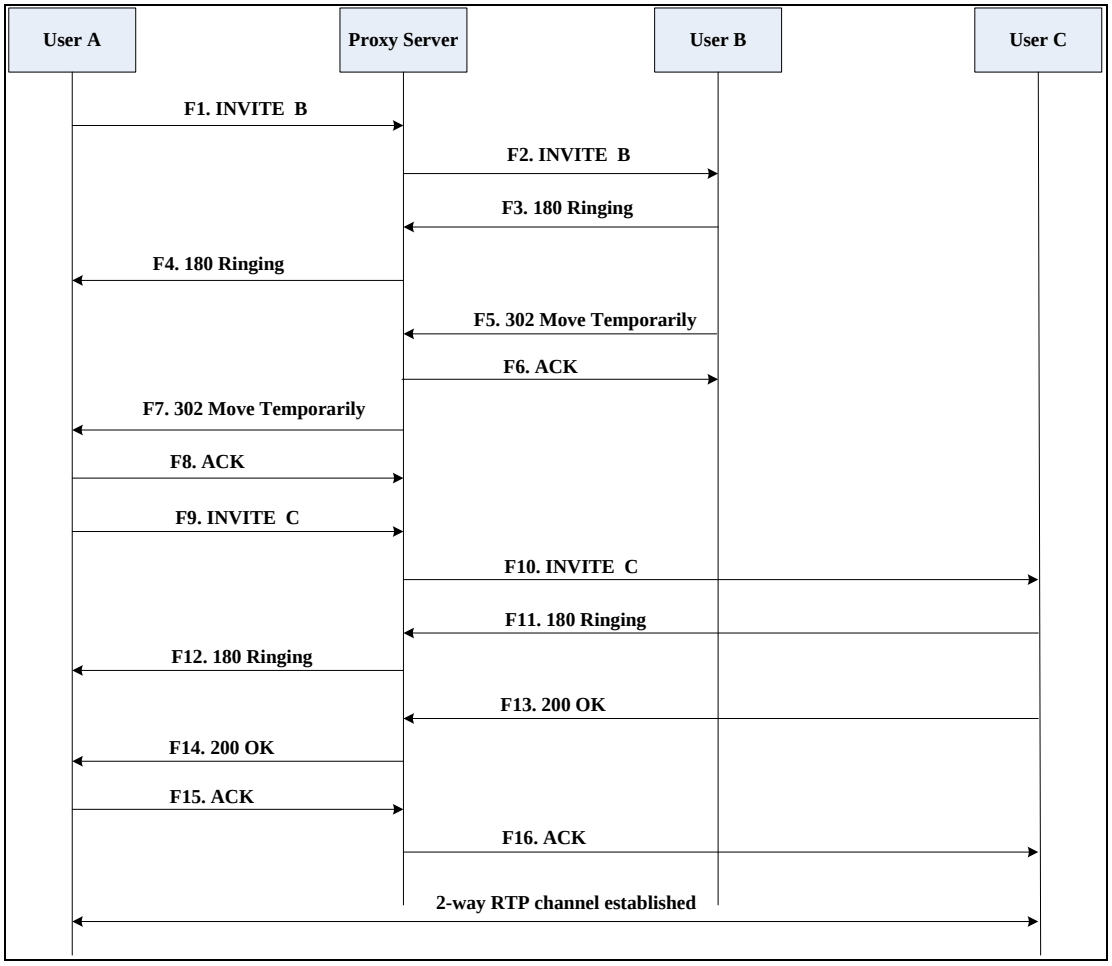
No Answer Call Forward

The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled no answer call forward. The incoming call is forwarded to User C when User B does not answer the incoming call after a period of time. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User B enables no answer call forward, and the destination number is User C.
2. User A calls User B.
3. User B does not answer the incoming call.
4. User B forwards the incoming call to User C.
5. User C answers the call.

   Call is established between User A and User C.



Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. The proxy server sends the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	302 Move Temporarily—User B to Proxy Server	User B sends a SIP 302 Moved Temporarily message to the proxy server. The message indicates that User B is not available at SIP phone B. User B rewrites the contact-URI.
F6	ACK—Proxy Server to User B	The proxy server sends a SIP ACK to User B, the ACK message notifies User B that the proxy server has received the

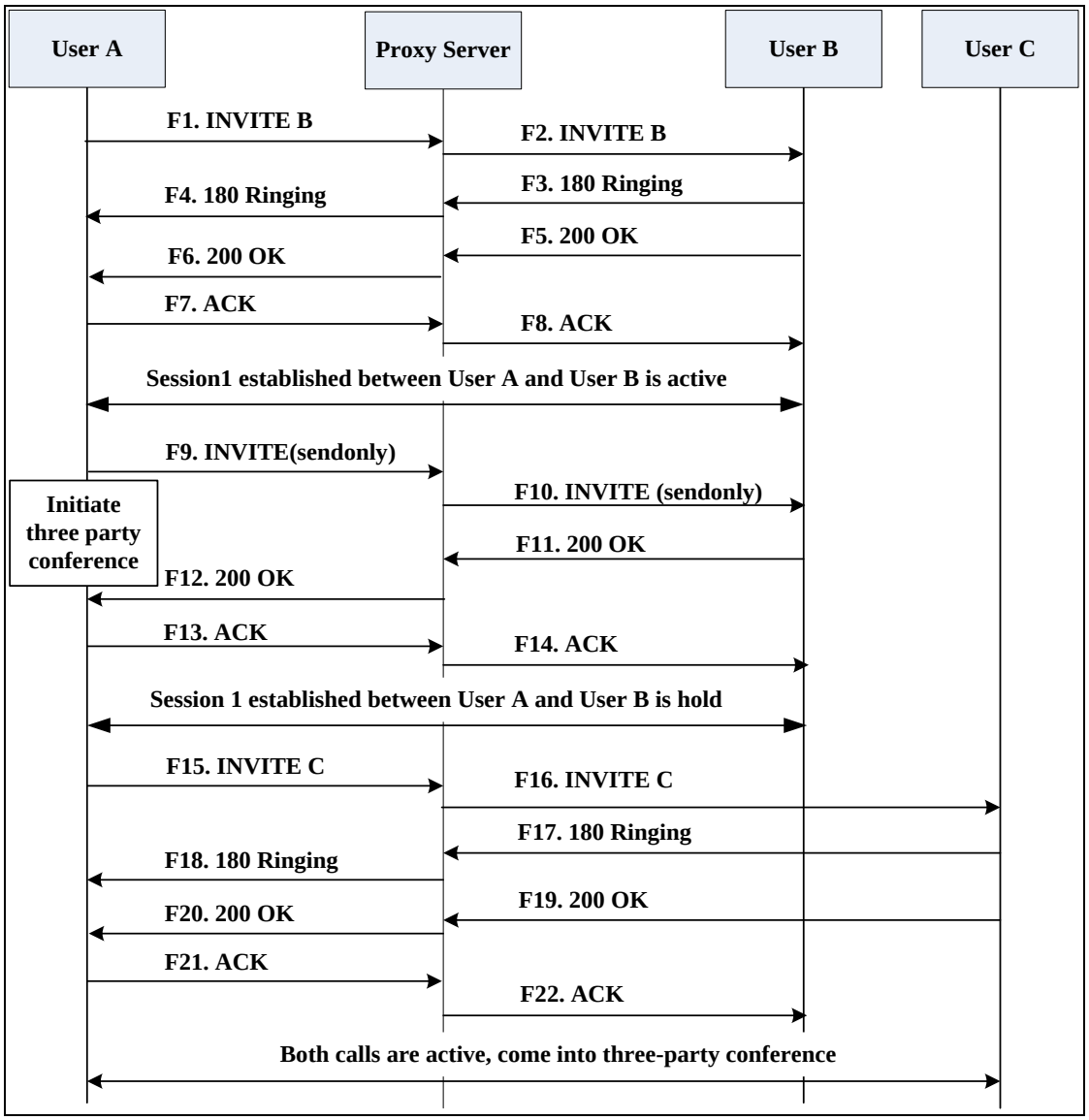
Step	Action	Description
		ACK message.
F7	302 Move Temporarily—Proxy Server to User A	The proxy server forwards the 302 Moved Temporarily message to User A.
F8	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK message notifies the proxy server that User A has received the ACK message.
F9	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F10	INVITE—Proxy Server to User C	The proxy server forwards the SIP INVITE request to User C.
F11	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F13	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F14	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F15	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F16	ACK—Proxy Server to User C	The proxy server sends the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response.

Call Conference

The following figure illustrates successful 3-way calling between Yealink SIP-T3xG IP phones in which User A mixes two RTP channels and therefore establishes a conference between User B and User C. In this call flow scenario, the end users are User A, User B, and User C. They are all using Yealink SIP IP phones, which are connected via an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User A put User B on hold.
4. User A calls User C.
5. User C answers the call.
6. User A mixes the RTP channels and establishes a conference between User B and User C.


User A
Proxy Server
User B
User C
F1. INVITE B
F2. INVITE B
F3. 180 Ringing
F4. 180 Ringing
F5. 200 OK
F6. 200 OK
F7. ACK
F8. ACK
Session1 established between User A and User B is active
F9. INVITE(sendonly)
Initiate three party conference
F10. INVITE (sendonly)
F11. 200 OK
F12. 200 OK
F13. ACK
F14. ACK
Session 1 established between User A and User B is hold
F15. INVITE C
F16. INVITE C
F17. 180 Ringing
F18. 180 Ringing
F19. 200 OK
F20. 200 OK
F21. ACK
F22. ACK
Both calls are active, come into three-party conference

Step	Action	Description
F1	INVITE—User A to Proxy Server	User A sends the INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: • The IP address of User B is inserted in the Request-URI field. • User A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which User B is prepared to receive the RTP data is specified.
F2	INVITE—Proxy Server to User B	The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.
F3	180 Ringing—User B to Proxy Server	User B sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F4	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User B is being alerted.
F5	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F6	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK message to User A. The 200 OK response notifies User A that the connection has been made.

Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE is successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User A that User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends the ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F16	INVITE—Proxy Server to User	The proxy server maps the SIP URI in the To field to User C. The proxy server

Step	Action	Description
	C	sends the SIP INVITE request to User C.
F17	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F18	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F19	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F20	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F21	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F22	ACK—Proxy Server to User C	The proxy server sends the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response.

Appendix F: Sample Configuration File

This section provides the sample configuration file necessary to configure the IP phone. Any line starts with a pound sign (#) is considered to be a comment, unless the # is contained within double quotes. For Boolean fields, 0 = disabled, 1 = enabled.

This file contains sample configurations for the <y0000000000xx>.cfg or <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 you required. The settings in the <y0000000000xx>.cfg file will be overridden by settings which also appear in the <MAC>.cfg file.

T3xG Sample Configuration File

```
#!version:1.0.0.1
#Note: This file header cannot be edited or deleted.

#Network Settings

network.internet_port.type =

#Configure the WAN port type; 0-DHCP, 1-PPPoE, 2-Static IP Address.
#If the WAN port type is configured as DHCP, you do not need to set the
#following network parameters.
#If the WAN port type is configured as Static IP Address, configure the
#following parameters.

network.internet_port.ip =

network.internet_port.mask =

network.internet_port.gateway =

network.primary_dns=

network.secondary_dns =

#If the WAN port type is configured as PPPoE, configure the following
#parameters.
network.pppoe.user =
network.pppoe.password =

#Dial Plan Settings

dialplan.area_code.code =
dialplan.area_code.min_len =
dialplan.area_code.max_len =
dialplan.area_code.line_id =
dialplan.block_out.number.1 =
dialplan.block_out.line_id.1 =
dialplan.dialnow.rule.1 =
dialplan.dialnow.line_id.1 =
```

```
dialplan.replace.prefix.1 =
dialplan.replace.replace.1 =
dialplan.replace.line_id.1 =
```

#Time Settings

```
local_time.time_zone =
local_time.time_zone_name =
local_time.ntp_server1 =
local_time.ntp_server2 =
local_time.interval =

#Use the following parameters to set the time and date manually.
local_time.manual_time_enable =
local_time.date_format =
local_time.time_format =
```

#Auto DST Settings

```
local_time.summer_time =
local_time.dst_time_type =
local_time.start_time =
local_time.end_time =
local_time.offset_time =
```

#Phone Lock

```
phone_setting.lock =
phone_setting.phone_lock.unlock_pin =
phone_setting.phone_lock.lock_time_out =
```

#Language

```
lang.wui =
lang.gui =
```

#Call Waiting

```
call_waiting.enable =
call_waiting.tone =
```

#Auto Redial

```
auto_redial.enable =
auto_redial.interval =
auto_redial.times =
```

#Call Hold

```
sip.rfc2543_hold =
```

#Hotline

```
features.hotline_number =
features.hotline_delay =
```

#Web Server Type

```
network.web_server_type =
network.port.http =
network.port.https =
```

#Call Transfer

```
transfer.semi_attend_tran_enable =
transfer.blind_tran_on_hook_enable =
transfer.on_hook_trans_enable =
transfer.tran_others_after_conf_enable =
```

#Call Conference

```
account.1.conf_type =
account.1.conf_uri =
```

#DTMF

```
account.1.dtmf.type =
account.1.dtmf.dtmf_payload =
account.1.dtmf.info_type =
```

#Distinctive Ring Tones

```
account.1.alert_info_url_enable =
distinctive_ring_tones.alert_info.1.text =
distinctive_ring_tones.alert_info.1.ringer =
```

#Tones

```
voice.tone.dial =
voice.tone.ring =
voice.tone.busy =
voice.tone.congestion =
voice.tone.callwaiting =
voice.tone.dialrecall =
voice.tone.record=
voice.tone.info =
voice.tone.stutter =
voice.tone.message =
voice.tone.autoanswer =
```

#Remote Phonebook

```
features.remote_phonebook.enable =
```

```
features.remote_phonebook.flash_time =
```

#LDAP

```
ldap.name_filter =
ldap.number_filter =
ldap.host = 0.0.0.0
ldap.port = 389
ldap.base =
ldap.user =
ldap.password =
ldap.max_hits =
ldap.name_attr =
ldap.numb_attr =
ldap.display_name =
ldap.version =
ldap.search_delay =
ldap.call_in_lookup =
ldap.ldap_sort =
ldap.dial_lookup =
```

#BLF List

```
account.5.blf.blf_list_uri =
account.5.blf_list_code =
```

#Shared Call Appearance

```
account.1.shared_line =
account.1.enable =
account.1.label =
account.1.display_name =
account.1.auth_name =
account.1.password =
account.1.user_name =
account.1.sip_server_host =
account.1.sip_server_port =
account.1.outbound_proxy_enable =
account.1.outbound_host =
account.1.outbound_port =
```

#Action URL

```
action_url.setup_completed =
action_url.log_on =
action_url.log_off =
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.forward_incoming_call =
action_url.reject_incoming_call =
action_url.answer_new_incoming_call =
action_url.transfer_finished =
action_url.transfer_failed =
```

#Access URL of Resource Files

```
dialplan_dialnow.url =
dialplan_replace_rule.url =
local_contact.data.url =
remote_phonebook.data.1.url =
```

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