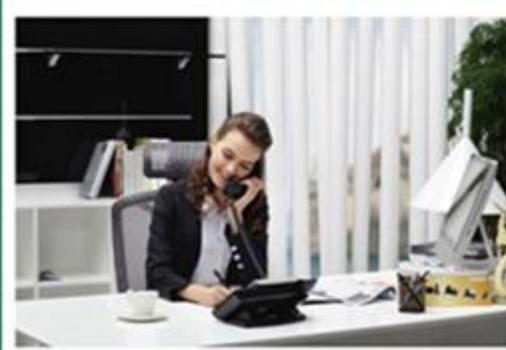


Yealink



SIP-T2xP and SIP-T19P IP Phones Administrator Guide

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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 experienced radio/TV technician for help.

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About This Guide

This guide is intended for administrators who need to properly configure, customize, manage, and troubleshoot the IP phone system rather than end-users. It provides details on the functionality and configuration of IP phones.

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

Documentations

This guide covers SIP-T28P, SIP-T26P, SIP-T22P, SIP-T21P, SIP-T20P and SIP-T19P IP phones. The following related documents are available:

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

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

In This Guide

The information detailed in this guide is applicable to firmware version 72 or higher. The firmware format is like x.x.x.x.rom. The second x from left must be greater than or equal to 72 (e.g., the firmware version of SIP-T28P IP phone: 2.72.0.1.rom). 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 IP phones and the configuration methods.
- Chapter 3, "[Configuring Basic Features](#)" describes how to configure the basic features on IP phones.

- Chapter 4, “[Configuring Advanced Features](#)” describes how to configure the advanced features on IP phones.
- Chapter 5, “[Configuring Audio Features](#)” describes how to configure the audio features on IP phones.
- Chapter 6, “[Configuring Security Features](#)” describes how to configure the security features on IP phones.
- Chapter 7, “[Upgrading Firmware](#)” describes how to upgrade firmware of IP phones.
- Chapter 8, “[Resource Files](#)” describes the resource files that can be downloaded by IP phones.
- Chapter 9, “[Troubleshooting](#)” describes how to troubleshoot IP phones and provides some common troubleshooting solutions.
- Chapter 10, “[Appendix](#)” provides the glossary, reference information about IP phones compliant with RFC 3261, SIP call flows and the sample configuration files.

Summary of Changes

This section describes the changes to this guide for each release and guide version.

Changes for Release 72, Guide Version 72.1

The following section is new for this version:

- [Power Indicator LED](#) on page 40

Major updates have occurred to the following sections:

- [DHCP](#) on page 21
- [Replace Rule](#) on page 33
- [Dial-now](#) on page 34
- [Contrast](#) on page 42
- [Backlight](#) on page 43
- [Time and Date](#) on page 50
- [Key as Send](#) on page 64
- [Anonymous Call](#) on page 81
- [LDAP](#) on page 139
- [Busy Lamp Field](#) on page 142
- [Action URL](#) on page 163
- [IPv6 Support](#) on page 196
- [Transport Layer Security](#) on page 211

- [Upgrading Firmware](#) on page 225
- [Resource Files](#) on page 229

Changes for Release 71, Guide Version 71.165

Documentations of the newly released SIP-T19P and SIP-T21P IP phones have also been added.

Changes for Release 71, Guide Version 71.141

Major updates have occurred to the following sections:

- [Action URL](#) on page 163
- [Action URI](#) on page 166

Changes for Release 71, Guide Version 71.140

Major updates have occurred to the following sections:

- [Logo Customization](#) on page 59
- [Anonymous Call](#) on page 81
- [Distinctive Ring Tones](#) on page 131
- [Server Redundancy](#) on page 169
- [Transport Layer Security](#) on page 211
- [Secure Real-Time Transport Protocol](#) on page 217
- [Encrypting Configuration Files](#) on page 219
- [Local Contact File](#) on page 233
- [Viewing Log Files](#) on page 241
- [Capturing Packets](#) on page 244

Changes for Release 71, Guide Version 71.125

Major updates have occurred to the following section:

- [Appendix B: Time Zones](#) on page 257

Changes for Release 71, Guide Version 71.120

Major updates have occurred to the following section:

- [Configuring DSS Key](#) on page 399

Changes for Release 71, Guide Version 71.110

The following sections are new for this version:

- [Hot Desking](#) on page 162
- [TR-069 Device Management](#) on page 194
- [IPv6 Support](#) on page 196

Major updates have occurred to the following sections:

- [Configuring Network Parameters Manually](#) on page 24
- [Softkey Layout](#) on page 61
- [Directed Call Pickup](#) on page 107
- [Distinctive Ring Tones](#) on page 131
- [Automatic Call Distribution](#) on page 149
- [Action URL](#) on page 166
- [Server Redundancy](#) on page 169
- [VLAN](#) on page 179
- [Transport Layer Security](#) on page 211
- [Local Contact File](#) on page 233

Changes for Release 70, Guide Version 70

The following sections are new for this version:

- [Configuring Network Parameters Manually](#) on page 24
- [Contrast](#) on page 42
- [Backlight](#) on page 43
- [Logo Customization](#) on page 59
- [Softkey Layout](#) on page 61
- [Key as Send](#) on page 64
- [Call Log](#) on page 68
- [Live Dialpad](#) on page 73
- [Auto Answer](#) on page 77
- [Call Completion](#) on page 79
- [Anonymous Call](#) on page 81
- [Anonymous Call Rejection](#) on page 82
- [Busy Tone Delay](#) on page 88

- [Return Code When Refuse](#) on page 89
- [Early Media](#) on page 90
- [180 Ring Workaround](#) on page 90
- [Use Outbound Proxy in Dialog](#) on page 92
- [SIP Session Timer](#) on page 93
- [Session Timer](#) on page 94
- [Call Return](#) on page 115
- [Transfer via DTMF](#) on page 125
- [Intercom](#) on page 126
- [Music on Hold](#) on page 148
- [Automatic Call Distribution](#) on page 149
- [Message Waiting Indicator](#) on page 151
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- [Configuring Audio Features](#) on page 199
- [Secure Real-Time Transport Protocol](#) on page 217
- [Appendix B: Time Zones](#) on page 257
- Phone user interface for each feature

Major updates have occurred to the following sections:

- [Creating Dial Plan](#) on page 32
- [Transport Layer Security](#) on page 211
- [Encrypting Configuration Files](#) on page 219
- [Troubleshooting](#) on page 241
- Web user interface for each feature

Changes for Release 70, Guide Version 2.0

The following sections are new for this version:

- [Dialog Info Call Pickup](#) on page 113
- [Web Server Type](#) on page 117
- [Tones](#) on page 135

- [Hot Desking](#) on page 162
- [Action URL](#) on page 166
- [Action URI](#) on page 166
- [Resource Files](#) on page 229
- [Appendix C: Configuration Parameters](#) on page 260
- [Appendix F: Sample Configuration File](#) on page 470

Major updates have occurred to the following sections:

- [Creating Dial Plan](#) on page 32
- [Phone Lock](#) on page 48
- [Time and Date](#) on page 50
- [Busy Lamp Field](#) on page 142

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Product Overview

This chapter contains the following information about 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 implementation.

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 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 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 attributes of an end-to-end call.

SIP provides capabilities to:

- Determine the location of the target endpoint -- SIP supports address resolution, name mapping, and call redirection.
- Determine media capabilities of the target endpoint -- Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between endpoints. Conferences are established using only 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 does not answer in the allotted number of rings. It then returns a message indicating why the target endpoint is unavailable.
- Establish a session between the origin and target endpoint -- The call can be completed, SIP establishes a session between endpoints. SIP also supports mid-call changes, such as the addition of another endpoint to the conference or the change 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 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 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. It may be

preferential to use this method when not using an application layer firewall. Application layer firewalls like to know what applications are flowing through which ports and it is possible to use content types of other applications other than the one you are trying to let through what has been denied.

User agent server (UAS)

UAS is a server that hosts the application responsible for receiving the SIP requests from a UAC, and on reception it 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 SIP IP phone models. 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. 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.

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 log and directory access. IP phones also support advanced functionalities, including LDAP, Busy Lamp Field, Sever Redundancy and Network Conference.

The following IP phone models are described:

- SIP-T28P
- SIP-T26P
- SIP-T22P
- SIP-T21P
- SIP-T20P
- SIP-T19P

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

In order to operate as SIP endpoints in your network successfully, 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 IP phones is available.
- A call server is active and configured to receive and send SIP messages.

Physical Features of IP Phones

This section lists the available physical features of IP phones.

SIPT28P



Physical Features:

- TI TITAN chipset and TI voice engine
- 320x160 graphic LCD with 4-level greyscales
- 6 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 48 keys including 16 DSS keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- 1XRJ12 (6P6C) expansion module port
- 19 LEDs: 1xpower, 6xline, 1xmessage, 1xheadset, 10xmemory
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)

SIPT26P



Physical Features:

- TI TITAN chipset and TI voice engine
- 132x64 graphic LCD
- 3 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 45 keys including 13 DSS keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- 1XRJ12 (6P6C) expansion module port
- 16 LEDs: 1xpower, 3xline, 1xmessage, 1xheadset, 10xmemory
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)

SIP-T22P



Physical Features:

- TI TITAN chipset and TI voice engine
- 132x64 graphic LCD
- 3 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 32 keys including 4 soft keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- 5 LEDs: 1xpower, 3xline, 1xmessage
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)
- Wall Mount

SIPT21P



Physical Features:

- 132x64 graphic LCD
- 2 VoIP accounts
- 31 keys including 4 soft keys
- 4 LEDs: 1xpower, 2xline, 1xmessage
- HD Voice: HD Codec, HD Handset, HD Speaker
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- Power adapter: AC 100~240V input and DC 5V/600mA output
- Power over Ethernet (IEEE 802.3af)
- Wall Mount

SIP-T20P



Physical Features:

- TI TITAN chipset and TI voice engine
- 3-line LCD consists of an icon line and two 15-character lines
- 2 VoIP accounts, BroadSoft/Avaya/Asterisk validated
- HD Voice: HD Codec, HD Handset, HD Speaker
- 31 keys including 9 function keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- 4 LEDs: 1xpower, 2xline, 1xmessage
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)
- Wall Mount

SIPT19P



Physical Features:

- 132x64 graphic LCD
- Single VoIP account
- 29 keys including 4 soft keys
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100Mbps Ethernet ports
- 1 LED: 1xpower
- Power adapter: AC 100~240V input and DC 5V/600mA output
- Power over Ethernet (IEEE 802.3af)
- Wall Mount

Key Features of IP Phones

In addition to physical features introduced above, 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, conference.
 - **Basic Features:** DND, phone lock, auto redial, live dialpad, dial plan, hotline, caller identity, auto answer.
 - **Advanced Features:** BLF, server redundancy, distinctive ring tones, remote phone book (not applicable to SIP-T20P IP phones), LDAP (not applicable to SIP-T19P and SIP-T20P IP phones), 802.1X authentication.

- **Codecs and Voice Features**
 - Wideband codec: G.722
 - Narrowband codec: G.711, G.723, G.726, G.729AB, iLBC
 - 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
 - VLAN assignment: LLDP/Static/DHCP
 - Bridge/Router mode for PC port (Router mode is not applicable to SIP-T19P and SIP-T21P IP phones)
 - TFTP/DHCP/PPPoE client
 - HTTP/HTTPS server
 - DNS client
 - NAT/DHCP server
 - IPv6 support

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

- Dial URL via SIP server
- TR-069
- **Security**
 - HTTPS (server/client)
 - SRTP (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 provides basic information and installation instructions of IP phones.

This chapter provides the following sections:

- [Connecting the IP Phones](#)
- [Initialization Process Overview](#)
- [Verifying Startup](#)
- [Configuration Methods](#)
- [Reading Icons](#)
- [Configuring Basic Network Parameters](#)
- [Creating Dial Plan](#)

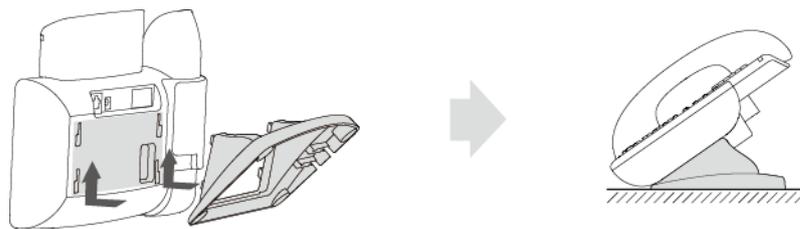
Connecting the IP Phones

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

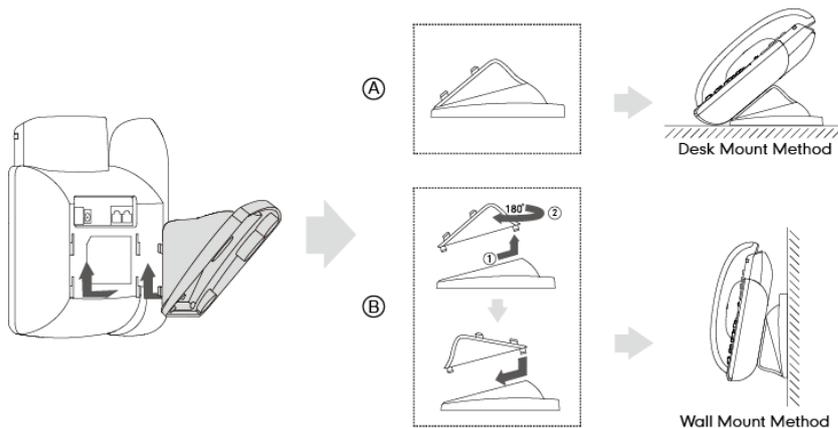
1. Attach the stand
2. Connect the handset and optional headset
3. Connect the network and power

Note A headset is not included in packaging contents.

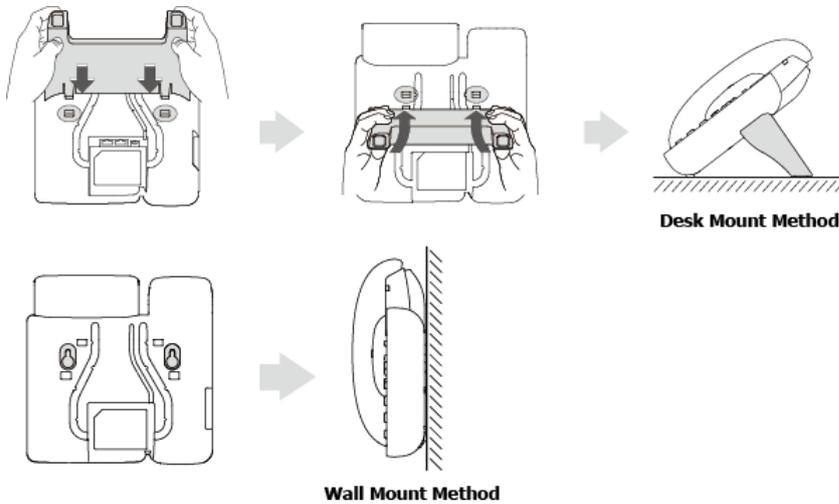
1) Attach the stand:



SIP-T28P/T26P

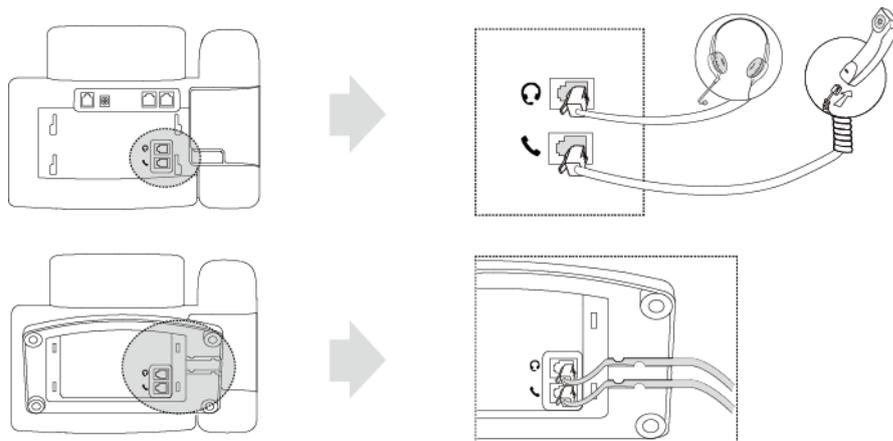


SIP-T22P/T21P/T20P

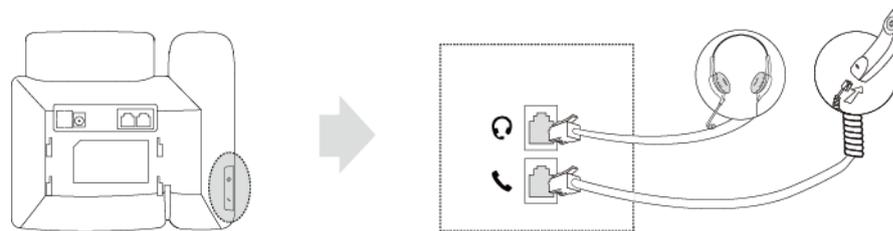


SIP-T19P

2) Connect the handset and optional headset:



SIPT28P/T26P



SIPT22P/T21P/T20P/T19P

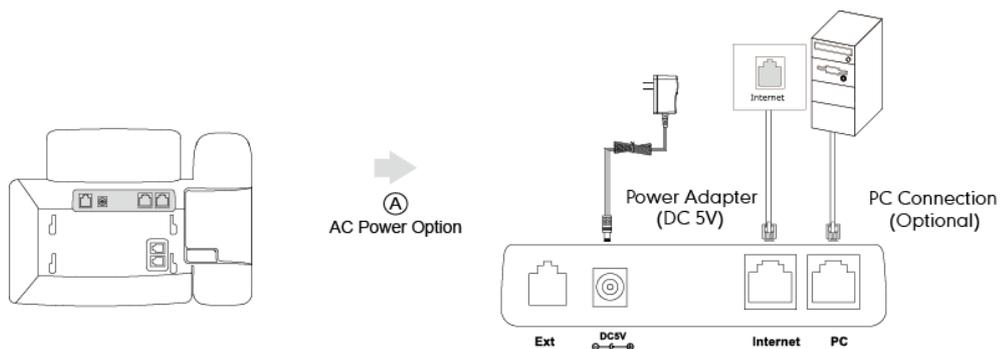
3) Connect the network and power:

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

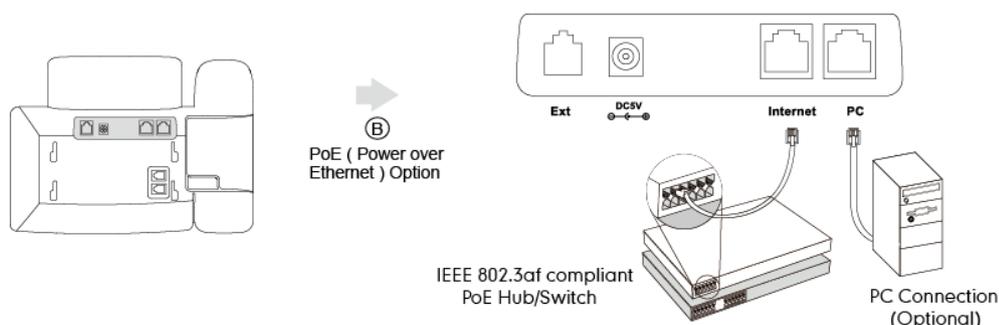


Power over Ethernet

With the included or a regular Ethernet cable, IP phones can be powered from a PoE-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 switch/hub is provided, you don't need to connect the phone to the power adapter. Make sure the switch/hub is PoE-compliant.

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

Important! Do not unplug or remove the 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).

Querying the DHCP (Dynamic Host Configuration Protocol) Server

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 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 24.

Contacting the provisioning server

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

Updating firmware

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

Downloading the resource files

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

The followings show examples of resource files:

- Language packs
- Ring tones
- Contact files

Verifying Startup

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

1. The power indicator LED illuminates.

2. The message "Initializing, Please Wait" appears on the LCD screen when the IP phone starts up.
3. The main LCD screen displays the following:
 - Time and date
 - Soft key labels (not applicable to SIP-T20P IP phones)
4. Press the OK key to check the IP phone status, the LCD screen displays the valid IP address, MAC address, firmware version, etc.

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

Configuration Methods

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

- [Phone User Interface](#)
- [Web User Interface](#)
- [Configuration Files](#)

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

Phone User Interface

An administrator or a user can configure and use IP phones via phone user interface. Access to specific features is restricted to the administrator. The default password is "admin"(case-sensitive). Not all features are available on phone user interface.

Web User Interface

An administrator or a user can configure IP phones via web user interface. The default user name and password for the administrator to log into the web user interface are both "admin" (case-sensitive). Almost all features are available on web user interface. IP phones support both HTTP and HTTPS protocols for accessing the web user interface. For more information, refer to [Web Server Type](#) on page 117.

Configuration Files

You can deploy IP phones using 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 a SIP-T22P IP phone is 001565113af8, names of these two configuration files must be: y000000000005.cfg and 001565113af8.cfg.

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

- SIP-T28P: y000000000000.cfg
- SIP-T26P: y000000000004.cfg
- SIP-T22P: y000000000005.cfg
- SIP-T21P: y000000000034.cfg
- SIP-T20P: y000000000007.cfg
- SIP-T19P: y000000000031.cfg

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

IP phones can obtain the address of the provisioning server during startup through one of the following processes: Zero Touch, PnP, DHCP Options and Phone Flash. Then IP phones download configuration files from the provisioning server, resolve and update the configurations written in configuration files. This entire process is called auto provisioning. For more information on auto provisioning, refer to *Yealink_SIP-T2 Series_T19P_T4_Series_IP_Phones_Auto_Provisioning_Guide*.

When modifying parameters, learn the following:

- Parameters in configuration files override those stored in the IP phone's flash memory.
- The .cfg extension of 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.

IP phones can accept two sources of configuration data:

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

The latest values configured on the IP phone take effect finally.

Reading Icons

Icons associated with different features may appear on the LCD screen. The following table provides a description for each icon on IP phones.

SIP-T28P	SIP-T26P	SIP-T22P	SIP-T21P	SIP-T20P	SIP-T19P	Description
						Network is unavailable
				/		Registered successfully
				/		Registration failed
				/		Registering
						Hands-free speakerphone mode
						Handset mode
						Headset mode
						Voice Mail
				/		Text Message
				AA		Auto Answer
				DND		Do Not Disturb
						Call Forward/Forwarded Calls
				/		Call Hold
						Call Mute

SIP-T28P	SIP-T26P	SIPT22P	SIP-T21P	SIP-T20P	SIPT19P	Description
				/		Ringer volume is 0
						Phone Lock
						Received Calls
						Placed Calls
						Missed Calls
				/	/	Recording box is full
				/	/	A call cannot be recorded
				/	/	Recording starts successfully
				/	/	Recording cannot be started
				/	/	Recording cannot be stopped

Configuring Basic Network Parameters

This section describes how to configure basic network parameters for the IP phone.

Note This section mainly introduces IPv4 network parameters. IP phones also support IPv6. For more information on IPv6, refer to [IPv6 Support](#) on page 196.

DHCP

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

be configured and used when DHCP is enabled.

DHCP Option

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

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

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

Parameter	DHCP Option	Description
Subnet Mask	1	Specify the client's subnet mask.
Time Offset	2	Specify the offset of the client's subnet in seconds from Coordinated Universal Time (UTC).
Router	3	Specify a list of IP addresses for routers on the client's subnet.
Time Server	4	Specify a list of time servers available to the client.
Domain Name Server	6	Specify a list of domain name servers available to the client.
Log Server	7	Specify a list of MIT-LCS UDP servers available to the client.
Host Name	12	Specify the name of the client.
Domain Server	15	Specify the domain name that client should use when resolving 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 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	66	Identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP

Parameter	DHCP Option	Description
Name		options.
Boot file Name	67	Identify a boot file when the 'file' field in the DHCP header has been used for DHCP options.

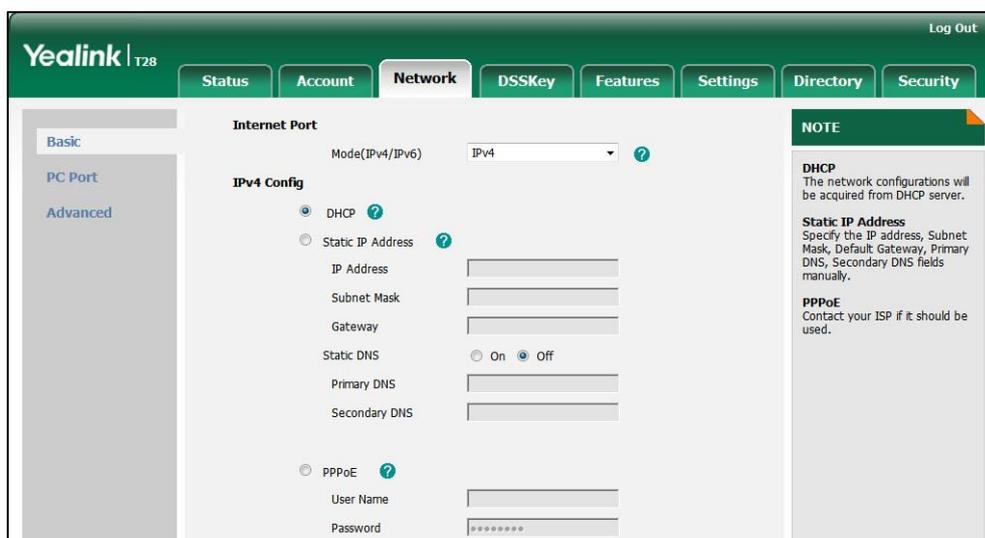
Procedure

DHCP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure DHCP on the IP phone. Configure static DNS address when DHCP is used. For more information, refer to DHCP on page 260.
Local	Web User Interface	Configure DHCP on the IP phone. Configure static DNS address when DHCP is used. Navigate to: http://<phoneIPAddress>/servlet?p=network&q=load
	Phone User Interface	Configure DHCP on the IP phone.

To configure DHCP via web user interface:

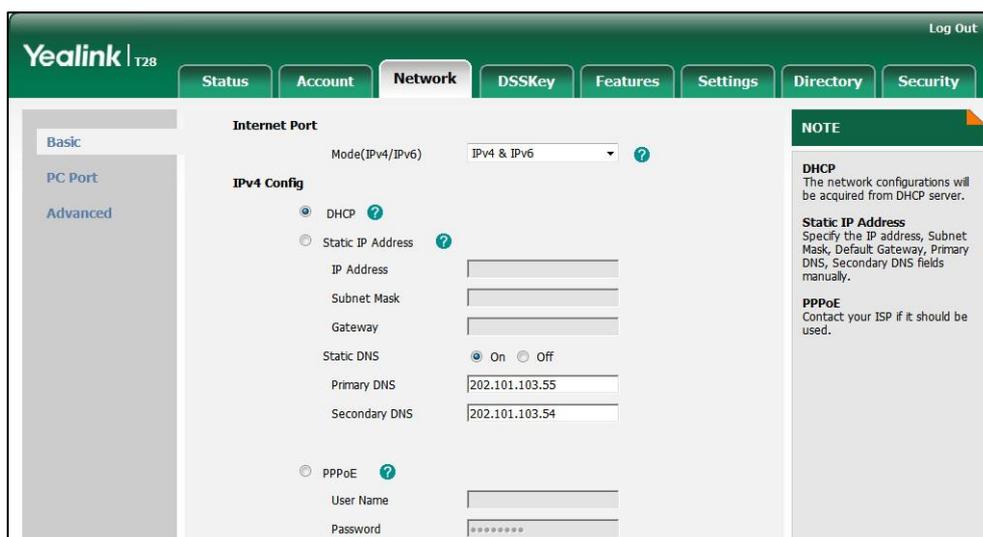
1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **DHCP** radio box.



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

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

1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **DHCP** radio box.
3. Mark the **Static DNS** radio box.
4. Enter the desired values in the **Primary DNS** and **Secondary DNS** fields.



5. Click **Confirm** to accept the change.
A dialog box pops up to prompt that settings will take effect after a reboot.
6. 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->IPv4**.
2. Press **▲** or **▼** to highlight the **DHCP IP Client** field.
3. Press the **Save** soft key to accept the change.

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

Configuring Network Parameters Manually

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

- IP Address
- Subnet Mask

- Default Gateway
- Primary DNS
- Secondary DNS

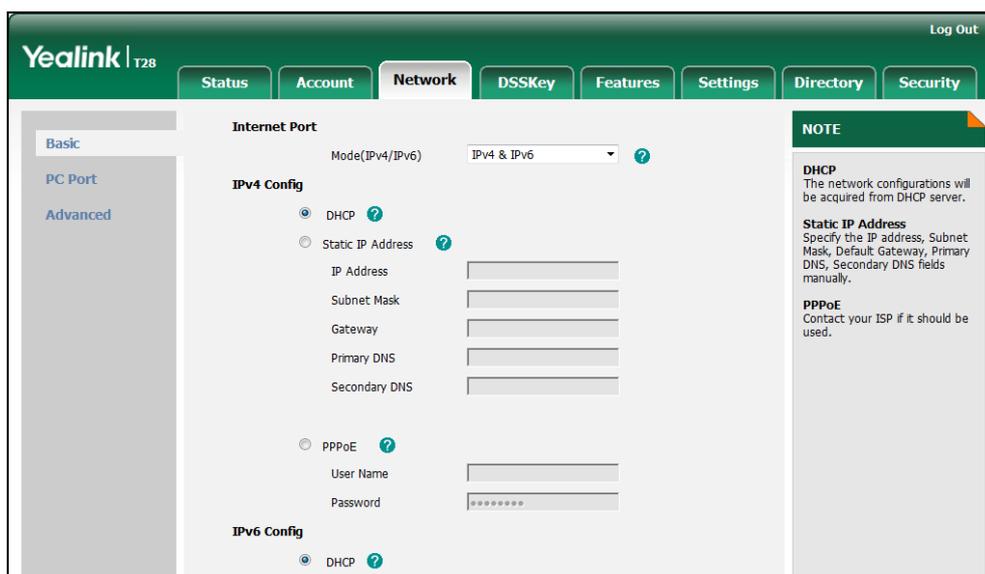
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure network parameters of the IP phone manually. For more information, refer to Static Network Settings on page 261.
Local	Web User Interface	Configure network parameters of the IP phone manually. Navigate to: http://<phoneIPAddress>/servlet?p=network&q=load
	Phone User Interface	Configure network parameters of the IP phone manually.

To configure the IP address mode via web user interface:

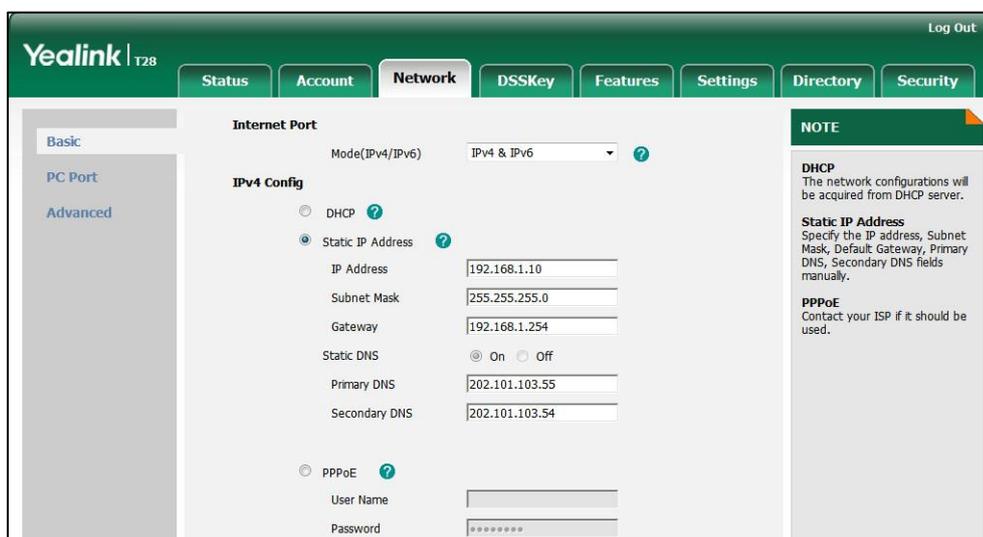
1. Click on **Network->Basic**.
2. Select desired value from the pull-down list of **Mode (IPv4/IPv6)**.



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

To configure a static IPv4 address via web user interface:

1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **Static IP Address** radio box.
3. Enter the desired values in the **IP Address, Subnet Mask, Gateway, Primary DNS** and **Secondary DNS** fields.



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

To configure the IP address mode via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) ->**Network->WAN Port**.
2. Press **←** or **→** to select **IPv4, IPv6** or **IPv4&IPv6** from the **IP Mode** field.
3. Press the **Save** soft key to accept the change.
The IP phone reboots automatically to make settings effective after a period of time.

To configure a static IPv4 address via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) ->**Network->WAN Port->IPv4->Static IP Client**.
2. Enter the desired values in the **IPv4, Subnet Mask, Default Gateway, Pri DNS** and **Sec DNS** fields.
3. Press the **Save** soft key to accept the change.
The IP phone reboots automatically to make 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 on these parameters, contact your network administrator.

PPPoE

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

Procedure

PPPoE can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure PPPoE on the IP phone. For more information, refer to PPPoE on page 264.
Local	Web User Interface	Configure PPPoE on the IP phone. Navigate to: http://<phoneIPAddress>/servlet ?p=network&q=load
	Phone User Interface	Configure PPPoE on the IP phone.

To configure PPPoE via web user interface:

1. Click on **Network->Basic**.
2. In the **IPv4 Config** block, mark the **PPPoE** radio box.
3. Enter the user name and password in corresponding fields.

The screenshot shows the Yealink T28 web interface. The 'Network' tab is active, and the 'Basic' sub-tab is selected. Under 'Internet Port', the 'Mode(IPv4/IPv6)' is set to 'IPv4 & IPv6'. In the 'IPv4 Config' section, the 'PPPoE' radio button is selected. The 'User Name' field is filled with 'Xmy0592123' and the 'Password' field is masked with dots. The 'IPv6 Config' section has 'DHCP' selected. A 'NOTE' box on the right contains the following text:

NOTE

DHCP
The network configurations will be acquired from DHCP server.

Static IP Address
Specify the IP address, Subnet Mask, Default Gateway, Primary DNS, Secondary DNS fields manually.

PPPoE
Contact your ISP if it should be used.

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

A dialog box pops up to prompt that settings will take effect after a 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->IPv4->PPPoE IP Client**.
2. Enter the user name and password in corresponding fields.
3. Press the **Save** soft key to accept the change.

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

Configuring Transmission Methods of the Internet Port and PC

Port

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

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

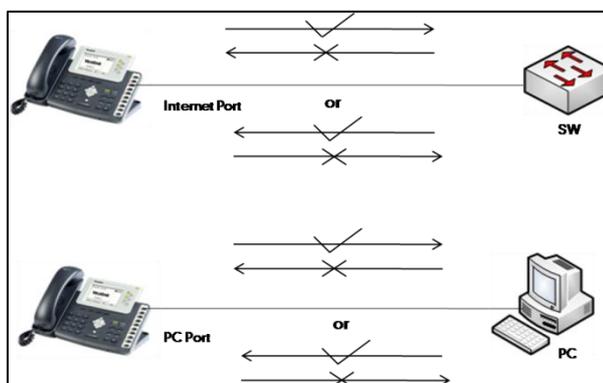
Auto-negotiation is configured for both Internet and PC ports on the IP phone by default.

Auto-negotiation

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

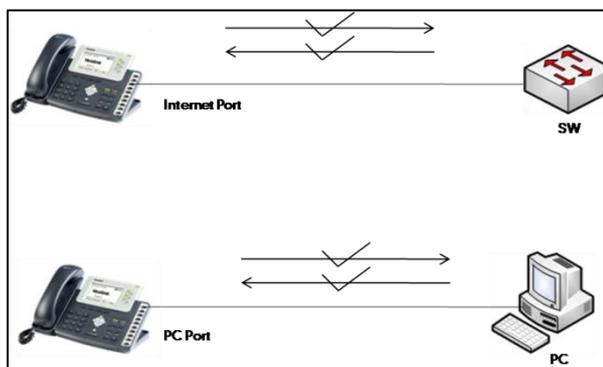
Half-duplex

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



Full-duplex

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



Procedure

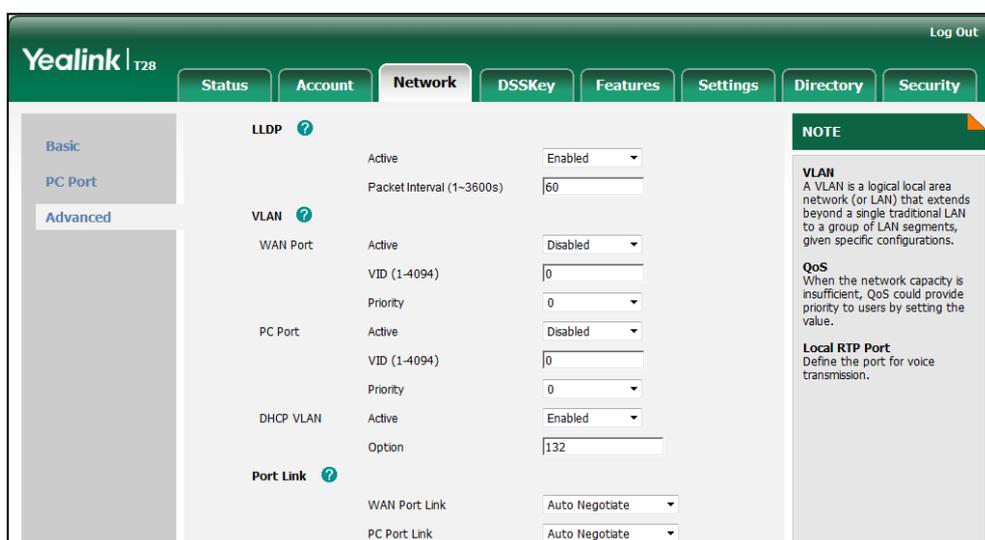
The transmission methods of Ethernet ports can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the transmission methods of Ethernet ports. For more information, refer to Internet and PC Ports Transmission Methods on page 265.
---------------------------	---------------------	---

Local	Web User Interface	<p>Configure the transmission methods of Ethernet ports.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=network-adv&q=load</p>
-------	--------------------	---

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

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **WAN Port Link**.
3. Select the desired value from the pull-down list of **PC Port Link**.



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

Configuring PC Port Mode

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

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

Note

The router mode is not applicable to SIP-T19P and SIP-T21P IP phones.

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 266.
Local	Web User Interface	Configure the PC port mode. Navigate to: http://<phoneIPAddress>/servlet?p=network-pcport&q=load
	Phone User Interface	Configure the PC port mode.

To configure the PC port mode via web user interface:

1. Click on **Network->PC Port**.
2. Select the desired value from the pull-down list of **PC Port Active**.
3. Mark the desired radio box.

If you mark the **As Router** radio box, 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 Server**.
- 4) Enter the start IP address in the **Start IP Address** field.
- 5) Enter the end IP address in the **End IP Address** field.

4. Click **Confirm** to accept the change.
A dialog box pops up to prompt that settings will take effect after a reboot.
5. 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 **IPv4** 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.
 - 4) Select the desired value from the **Server Status** field.
 - 5) Enter the start IP address in the **Start IP** field.
 - 6) 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 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 IP phone dial plan. Dial plan is a string of characters that governs the way for IP phones to process the inputs received from the IP phone’s keypads. IP phones support the following dial plan features:

- [Replace Rule](#)
- [Dial-now](#)
- [Area Code](#)
- [Block Out](#)

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. IP phones support up to 100 replace rules, which 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 229.

Procedure

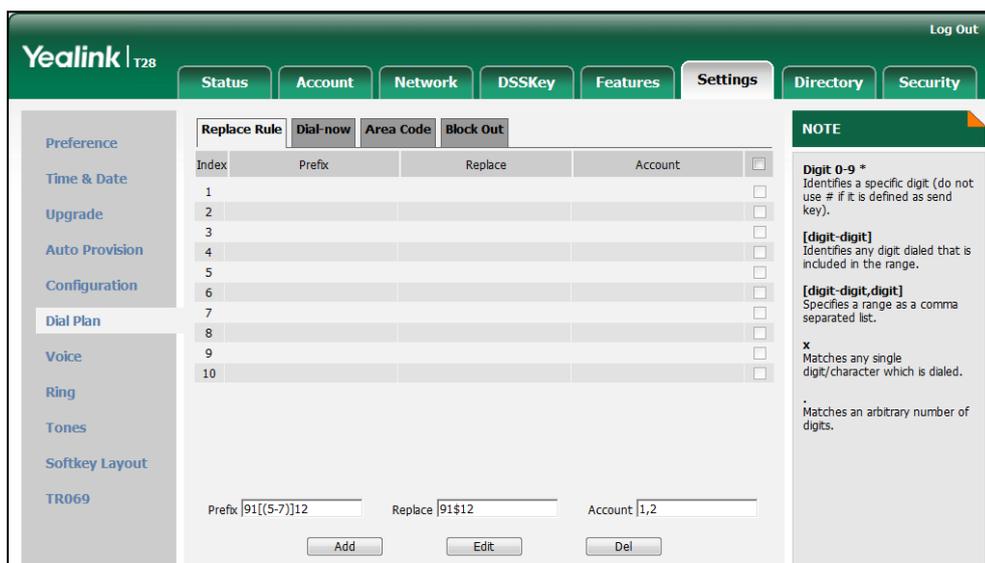
Replace rule can be created using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Create the replace rule for the IP phone. For more information, refer to Dial Plan on page 269.
Local	Web User Interface	Create the replace rule for the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=settings-dialplan&q=load

To create a replace rule via web user interface:

1. Click on **Settings->Dial Plan->Replace Rule**.
2. Enter the string in the **Prefix** 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 this field blank or enter 0, the replace rule will apply to all accounts on the IP phone.



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

Dial-now

Dial-now is a string used to match 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. IP phones support up to 100 dial-now rules, which can be created either one by one or in batch using a dial-now rule template. For more information on the dial-now template, refer to [Dial-now Template](#) on page 230.

Delay Time for Dial-now Rule

The IP phone will automatically dial out the entered number, which matches the dial-now rule, after a specified period of time.

Procedure

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

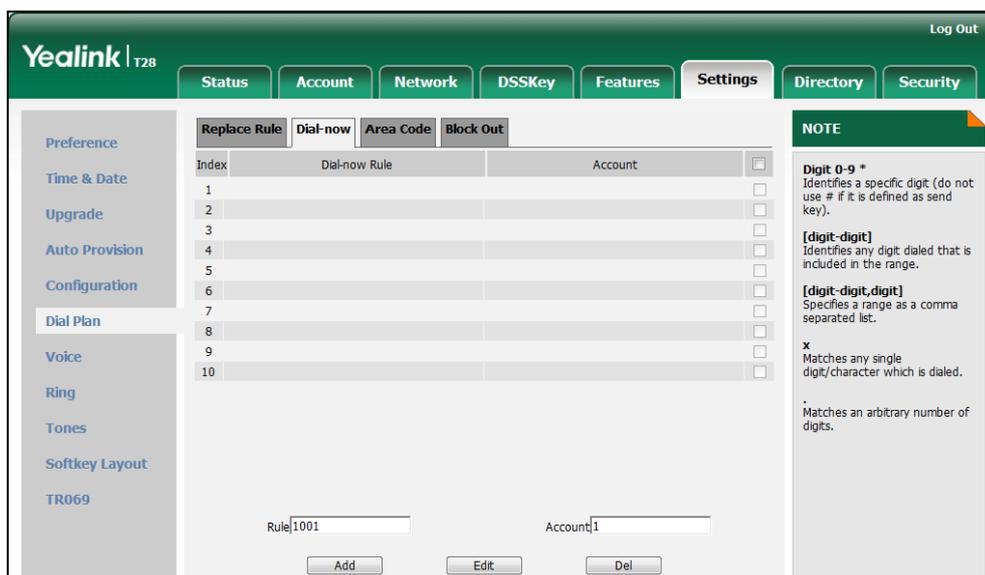
Configuration File	<y0000000000xx>.cfg	Create the dial-now rule for the IP phone. Configure the delay time for the
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		dial-now rule. For more information, refer to Dial Plan on page 269.
Local	Web User Interface	Create the dial-now rule for the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=settings-dialnow&q=load Configure the delay time for the dial-now rule. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

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

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

If you leave this field blank or enter 0, the dial-now rule will apply to all accounts on the IP phone.

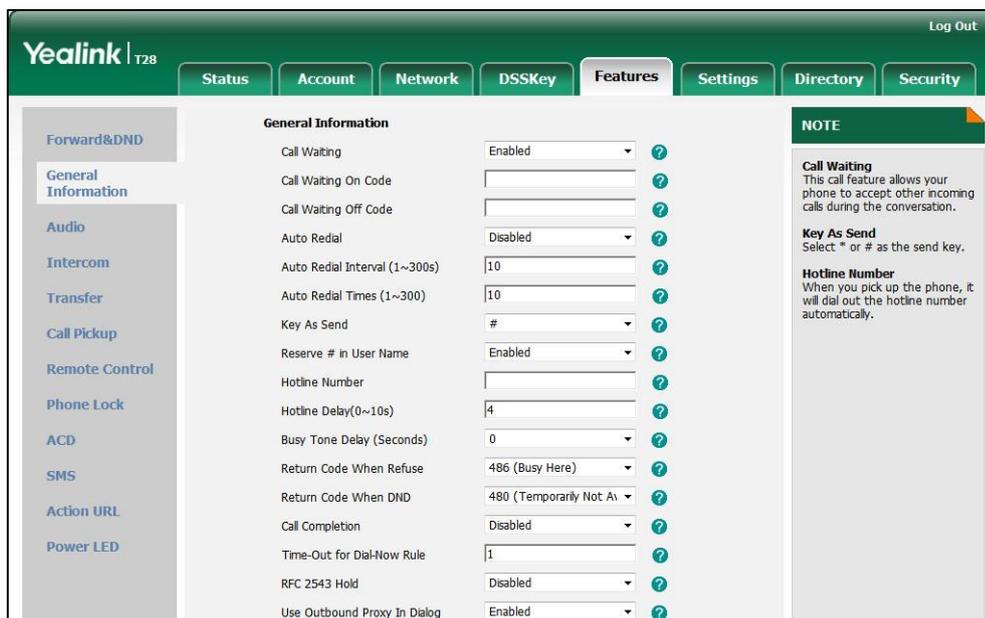


4. Click **Add** to add the dial-now rule.

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

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

- Enter the desired time within 1-14 (in seconds) in the **Time-Out for Dial-Now Rule** field.



- 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 before the numbers when dialing out them. 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 entered numbers. For more information, refer to Dial Plan on page 269.
Local	Web User Interface	Create the area code rule and specify the maximum and minimum lengths of entered numbers. Navigate to: http://<phoneIPAddress>/servlet

		?p=settings-areacode&q=load
--	--	-----------------------------

To configure an area code rule via web user interface:

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

If you leave this field blank or enter 0, the area code rule will apply to all accounts on the IP phone.

The screenshot shows the Yealink T28 web interface. The top navigation bar includes Status, Account, Network, DSSKey, Features, Settings, Directory, and Security. The left sidebar lists various settings categories like Preference, Time & Date, Upgrade, Auto Provision, Configuration, Dial Plan, Voice, Ring, Tones, Softkey Layout, and TR069. The main content area is titled 'Area Code' and contains the following fields:

- Code: 0592
- Min Length (1-15): 1
- Max Length (1-15): 15
- Account: 1

Buttons for 'Confirm' and 'Cancel' are visible below the fields. A 'NOTE' section on the right provides definitions for symbols used in dial plans:

- Digit 0-9 ***: Identifies a specific digit (do not use # if it is defined as send key).
- [digit-digit]**: Identifies any digit dialed that is included in the range.
- [digit-digit,digit]**: Specifies a range as a comma separated list.
- x**: Matches any single digit/character which is dialed.
- .**: Matches an arbitrary number of digits.

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

Block Out

Block out rule prevents users from dialing out specific numbers. When entered numbers match the predefined block out rule, the LCD screen prompts "Forbidden Number". IP phones support up to 10 block out rules.

Procedure

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

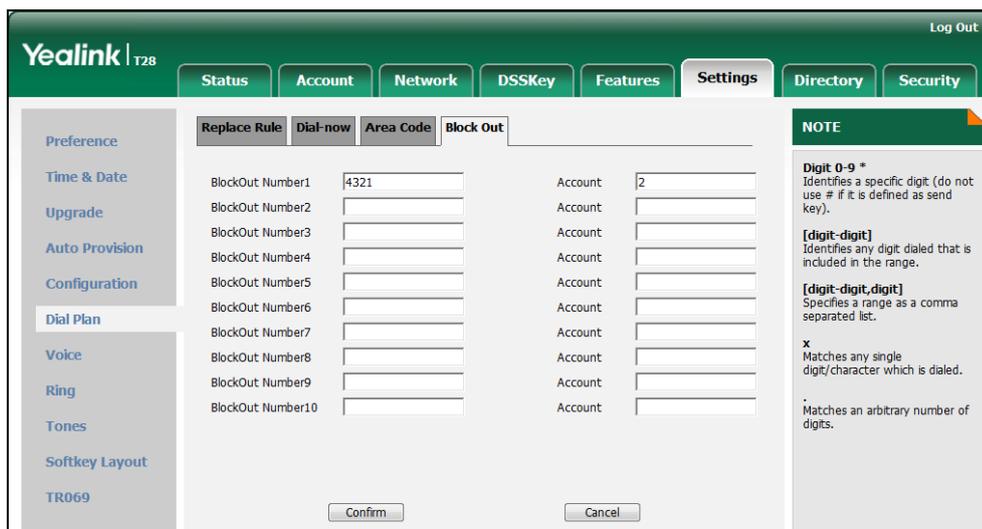
Configuration File	<y0000000000xx>.cfg	Create the block out rule for the IP phone. For more information, refer to Dial Plan on page 269.
Local	Web User Interface	Create the block out rule for the desired line. Navigate to: http://<phoneIPAddress>/servlet

		?p=settings-blackout&q=load
--	--	-----------------------------

To create a block out rule via web user interface:

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

If you leave this field blank or enter 0, the block out rule will apply to all accounts on the IP phone.



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:

- [Power Indicator LED](#)
- [Contrast](#)
- [Backlight](#)
- [User Password](#)
- [Administrator Password](#)
- [Phone Lock](#)
- [Time and Date](#)
- [Language](#)
- [Logo Customization](#)
- [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 Delay](#)
- [Return Code When Refuse](#)
- [Early Media](#)
- [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](#)
- [Directed Call Pickup](#)
- [Group Call Pickup](#)
- [Dialog Info Call Pickup](#)
- [Call Return](#)
- [Call Park](#)
- [Web Server Type](#)
- [Calling Line Identification Presentation](#)
- [Connected Line Identification Presentation](#)
- [DTMF](#)
- [Suppress DTMF Display](#)
- [Transfer via DTMF](#)
- [Intercom](#)

Power Indicator LED

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

Common Power Light On

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

Ring Power Light Flash

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

Voice/Text Mail Power Light Flash

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

Mute Power Light Flash

Mute Power Light Flash allows the power indicator LED to flash when a call is mute. If

this option is disabled, the status of the power indicator LED is determined by the option “Common Power Light On”.

Hold/Held Power Light Flash

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

Talk/Dial Power Light On

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

Procedure

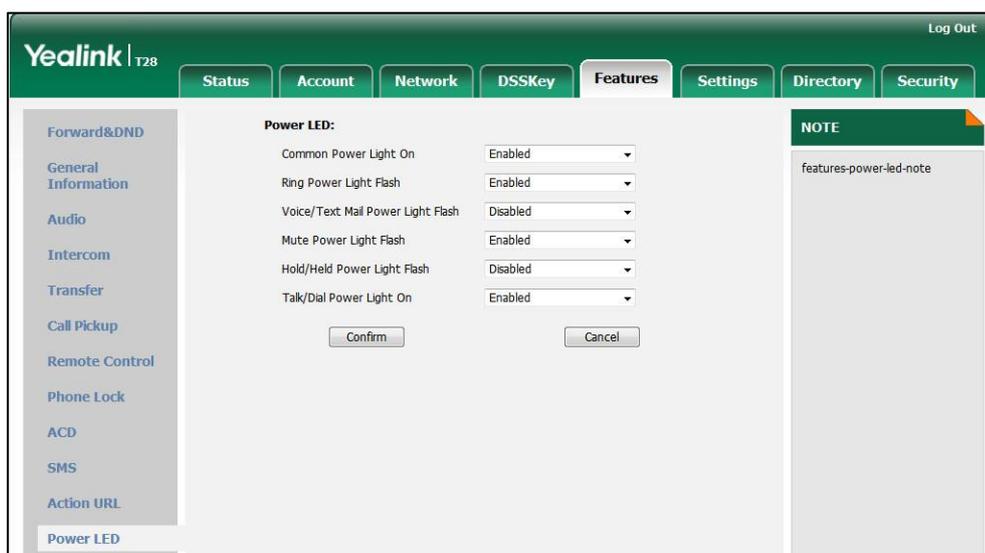
Power indicator LED can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the power indicator LED. For more information, refer to Power Indicator LED on page 273.
Local	Web User Interface	Configure the power indicator LED. Navigate to: http://<phoneIPAddress>/servlet?p=features-poweredled&q=load

To configure the power Indicator LED via web user interface:

1. Click on **Features->Power LED**.
2. Select the desired value from the pull-down list of **Common Power Light On**.
3. Select the desired value from the pull-down list of **Ring Power Light Flash**
4. Select the desired value from the pull-down list of **Voice/Text Mail Power Light Flash**.
5. Select the desired value from the pull-down list of **Mute Power Light Flash**.
6. Select the desired value from the pull-down list of **Hold/Held Power Light Flash**.

7. Select the desired value from the pull-down list of **Talk/Dial Power Light On**.



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

Contrast

Contrast determines the readability of the texts displayed on the LCD screen. Adjusting the contrast to a comfortable level can optimize the screen viewing experience. When configured properly, contrast allows users to read the LCD's display with minimal eyestrain. The contrast of the LCD screen is only applicable to SIP-T19P, SIP-T21P and SIP-T28P IP phones, and EXP39 connected to SIP-T26P and SIP-T28P IP phones.

Procedure

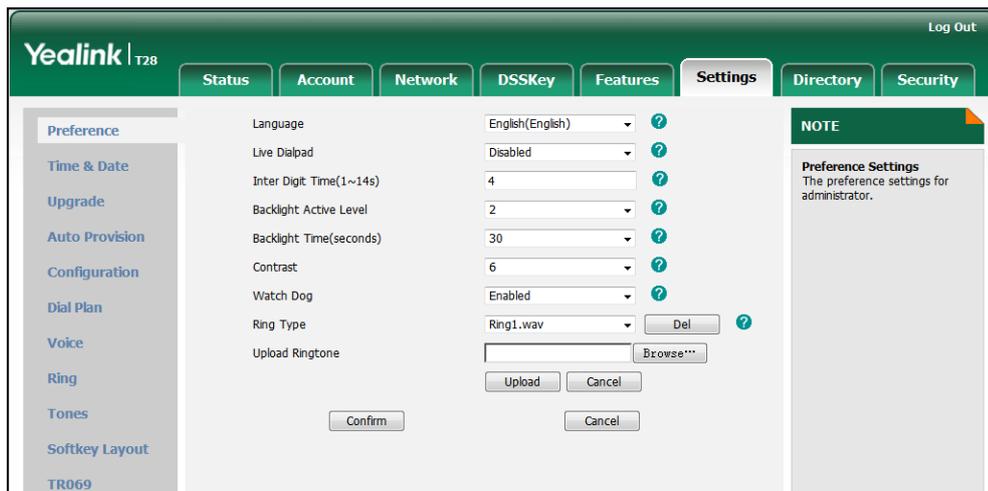
Contrast can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the contrast of the LCD screen. For more information, refer to Contrast on page 276.
Local	Web User Interface	Configure the contrast of the LCD screen. Navigate to: http://<phoneIPAddress>/servlet?p=settings-preference&q=load
	Phone User Interface	Configure the contrast of the LCD screen.

To configure contrast via web user interface:

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

2. Select the desired value from the pull-down list of **Contrast**.



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

To configure contrast via phone user interface (applicable to SIP-T28P IP phones and EXP39 connected to SIP-T26P and SIP-T28P IP phones):

1. Press **Menu->Settings->Basic Settings->Display->Contrast**.
2. Press **◀** or **▶**, or the **Switch** soft key to increase or decrease the intensity of contrast.
The default contrast level is 6.
3. Press the **Save** soft key to accept the change.

Note

Before you adjust the LCD's contrast of the expansion module, make sure the expansion module has been connected to the IP phone.

To configure contrast via phone user interface (applicable to SIP-T19P and SIP-T21P IP phones):

1. Press **Menu->Settings->Basic Setting->Contrast**.
2. Press **◀** or **▶**, or the **Switch** soft key to increase or decrease the intensity of contrast.
The default contrast level is 6.
3. Press the **Save** soft key to accept the change.

Backlight

Backlight determines the brightness of the LCD screen display, allowing users to read easily in dark environments. Backlight time specifies the delay time to turn off the backlight when the IP phone is inactive. Backlight time is applicable to SIP-T22P, SIP-T26P and SIP-T28P IP phones, and EXP39 connected to SIP-T26P and SIP-T28P IP phones. Backlight turns off quickly if a short backlight time is configured, this may not give users

enough time to read messages. Backlight active level is used to adjust the backlight intensity of the LCD screen. Backlight active level is only applicable to SIP-T28P IP phones and the connected EXP39.

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

- **Always Off:** Backlight is turned off permanently.
- **Always On:** Backlight is turned on permanently.
- **15, 30, 60, 120, 300, 600 or 1800:** Backlight is turned off when the IP phone is inactive after a preset period of time (in seconds), but it is automatically turned on if the status of the IP phone changes or any key is pressed.

The following table lists available methods and configuration options to configure the backlight of each phone model.

Phone Model	Configuration Methods	Configuration Options
SIP-T28P	Configuration Files Web User Interface Phone User Interface	Backlight Active Level Backlight Time
SIP-T26P	Configuration Files Web User Interface Phone User Interface (only applicable to the connected EXP39)	Backlight Active Level (only applicable to the connected EXP39) Backlight Time
SIP-T22P	Configuration Files Web User Interface	Backlight Time

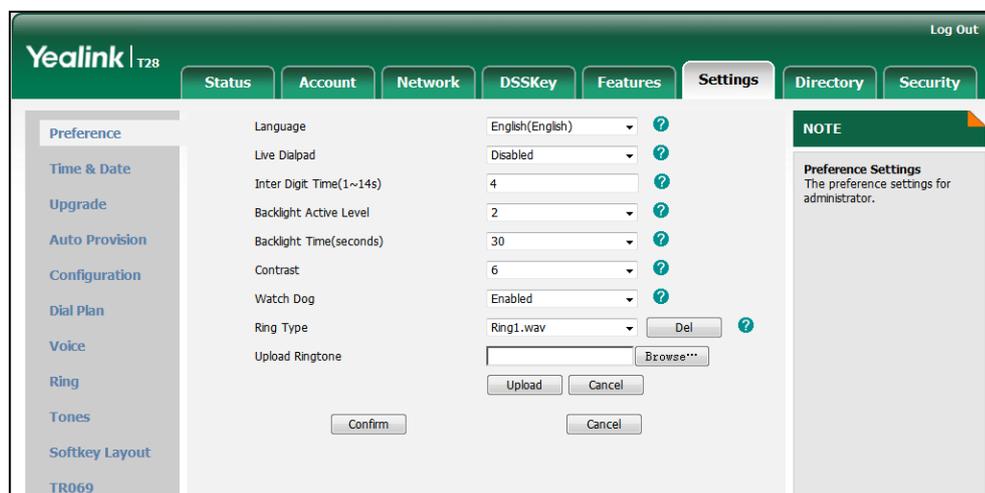
Procedure

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 277.
Local	Web User Interface	Configure the backlight of the LCD screen. Navigate to: http://<phoneIPAddress>/servlet?p=settings-preference&q=load
	Phone User Interface	Configure the backlight of the LCD screen.

To configure backlight via web user interface:

1. Click on **Settings->Preference**.
2. Select the desired value from the pull-down list of **Backlight Active Level** (only applicable to SIP-T28P IP phones and the connected EXP39).
3. Select the desired value from the pull-down list of **Backlight Time (seconds)**.



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

To configure backlight via phone user interface (only applicable to SIP-T28P IP phones and EXP39 connected to SIP-T26P and SIP-T28P IP phones):

1. Press **Menu->Settings->Basic Settings->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 type from the **Backlight Time** field.
4. Press the **Save** soft key to accept the change.

Note

Before you adjust the LCD's backlight of expansion module, make sure the expansion module has been connected to the IP phone.

User Password

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

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

Procedure

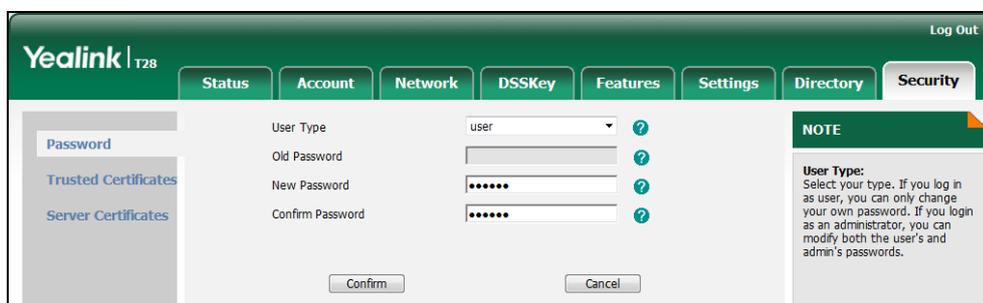
User password can be changed using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Change the user password of the IP phone. For more information, refer to User Password on page 278.
Local	Web User Interface	Change the user password of the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=security&q=load

To change the user password via web user interface:

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

The new password should be complex and contains at least 6 characters, where at least one character is numeric, and one character is alphabetic. Valid characters contain A-Z, a-z, 0-9,#,!,@,-,.,*,+ and \$.



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

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

Administrator Password

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

Procedure

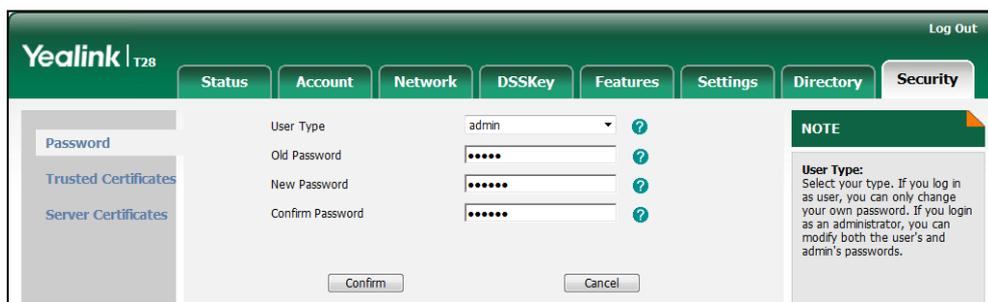
Administrator password can be changed using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Change the administrator password. For more information, refer to Administrator Password on page 278.
Local	Web User Interface	Change the administrator password. Navigate to: http://<phoneIPAddress>/servlet?p=security&q=load
	Phone User Interface	Change the administrator password.

To change the administrator password via web user interface:

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

The new password should be complex and contains at least 6 characters, where at least one character is numeric, and one character is alphabetic. Valid characters contain A-Z, a-z, 0-9,#,!,@,-,.,*,+ and \$.



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

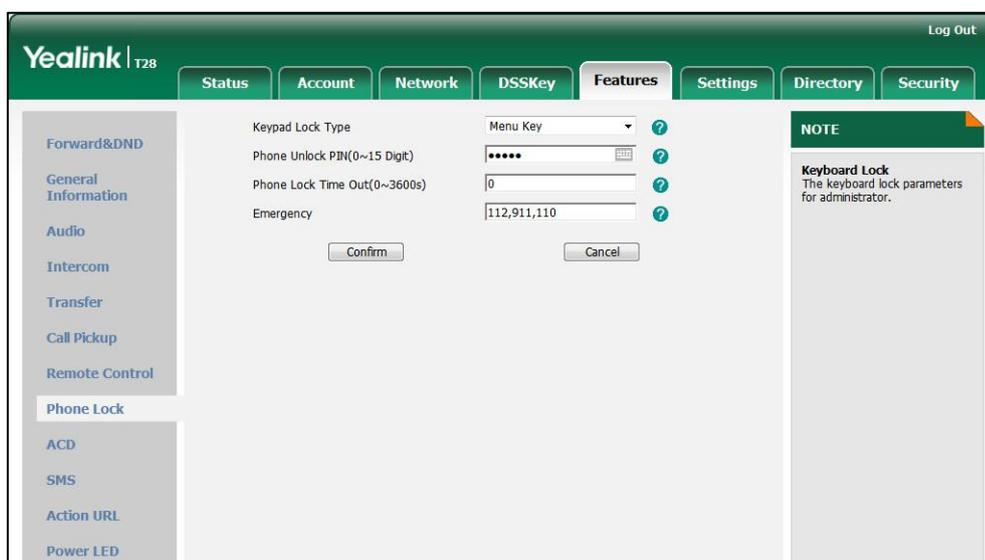
To change the administrator password via phone user interface:

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

		0
	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 **Features->Phone Lock**.
2. Select the desired type from the pull-down list of **Keypad Lock Type**.
3. Enter the unlock PIN in the **Phone Unlock PIN (0~15 Digit)** field.
4. Enter the desired time in the **Phone Lock Time Out (0~3600s)** field.



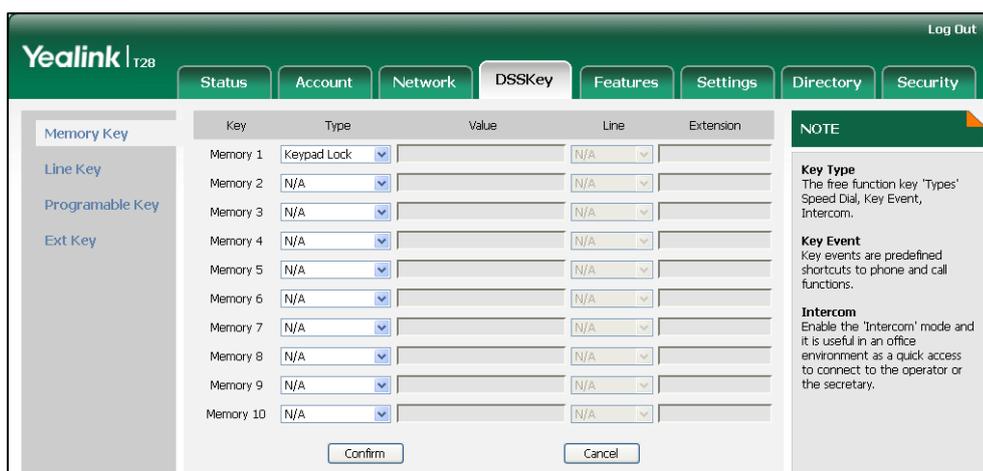
5. Click **Confirm** to accept the change.

To configure a keypad lock key via web user interface:

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

SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.

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



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

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

- Press **Menu->Settings->Advanced Settings** (password: admin) -> **Phone Settings->Keypad Lock**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired type from the **Keypad Lock** field.
- Press the **Save** soft key to accept the change.

To change the unlock PIN via phone user interface:

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

To configure a keypad lock key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Keypad Lock** from the **Type** field.
- Press the **Save** soft key to accept the change.

Time and Date

IP phones maintain a local clock and calendar. Time and date are displayed on the idle screen of IP phones. Time and date are synced automatically from the NTP server by default. The NTP server can be obtained by DHCP or configured manually. If IP phones cannot obtain the time and date from the NTP server, you need to manually configure

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

Time Zone

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

Daylight Saving Time

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

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

Option	Configuration Methods
Time Zone	Configuration Files Web User Interface Phone User Interface
Time	Web User Interface Phone User Interface
Time Format	Configuration Files Web User Interface Phone User Interface
Date	Web User Interface Phone User Interface
Date Format	Configuration Files Web User Interface Phone User Interface
Daylight Saving Time	Configuration Files Web User Interface

Procedure

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

Configuration File	<MAC>.cfg	Configure NTP by DHCP priority feature.
---------------------------	-----------	---

		<p>Configure the NTP server, time zone and DST.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p> <p>For more information, refer to Time and Date on page 281.</p>
Local	Web User Interface	<p>Configure NTP by DHCP priority feature.</p> <p>Configure the NTP server, time zone and DST.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-datetime&q=load">http://<phoneIPAddress>/servlet?p=settings-datetime&q=load</p>
	Phone User Interface	<p>Configure the NTP server and time zone.</p> <p>Configure the time and date manually.</p> <p>Configure the time and date formats.</p>

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

1. Click on **Settings->Time & Date**.

- Select the desired value from the pull-down list of **NTP By DHCP Priority**.

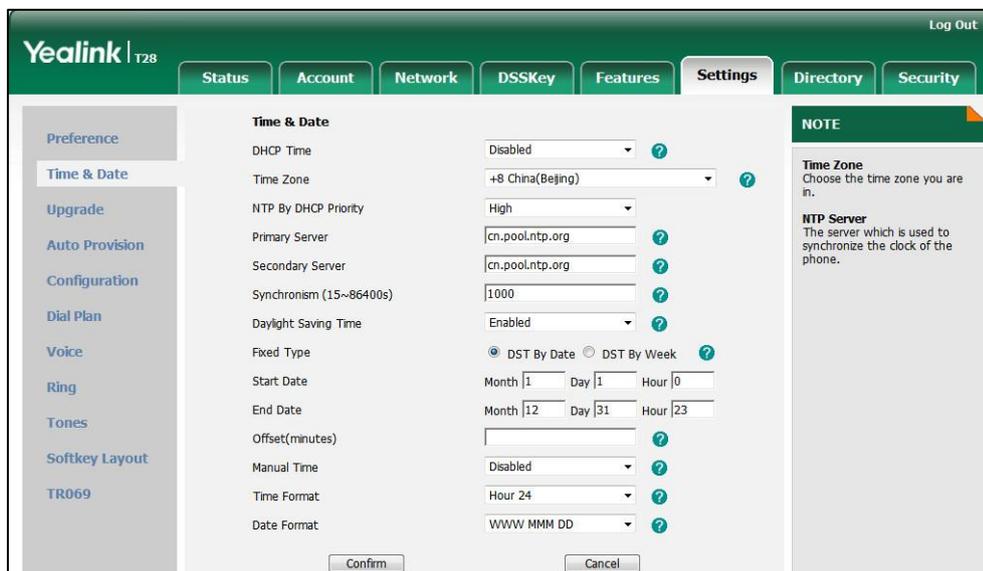
- Click **Confirm** to accept the change.

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

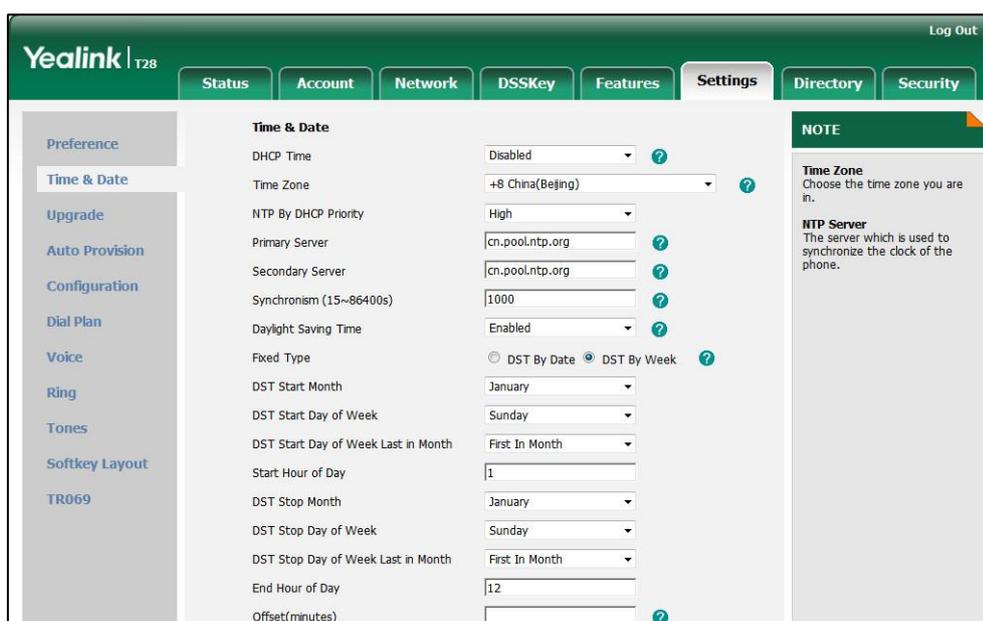
- Click on **Settings->Time & Date**.
- Select **Disabled** from the pull-down list of **Manual Time**.
- Select the desired time zone from the pull-down list of **Time Zone**.
- Enter the domain names or IP addresses in the **Primary Server** and **Secondary Server** fields respectively.
- Enter the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

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

- Mark the **DST By Date** radio box in the **Fixed Type** field.
Enter the start time in the **Start Date** field.
Enter the end time in the **End Date** field.



- Mark the **DST By Week** radio box in the **Fixed Type** field.
Select the desired values from the pull-down lists of **DST Start Month**, **DST Start Day of Week**, **DST Start Day of Week Last in Month**, **DST Stop Month**, **DST Stop Day of Week** and **DST Stop Day of Week Last in Month**.
Enter the desired time in the **Start Hour of Day** field.
Enter the desired time in the **End Hour of Day** field.

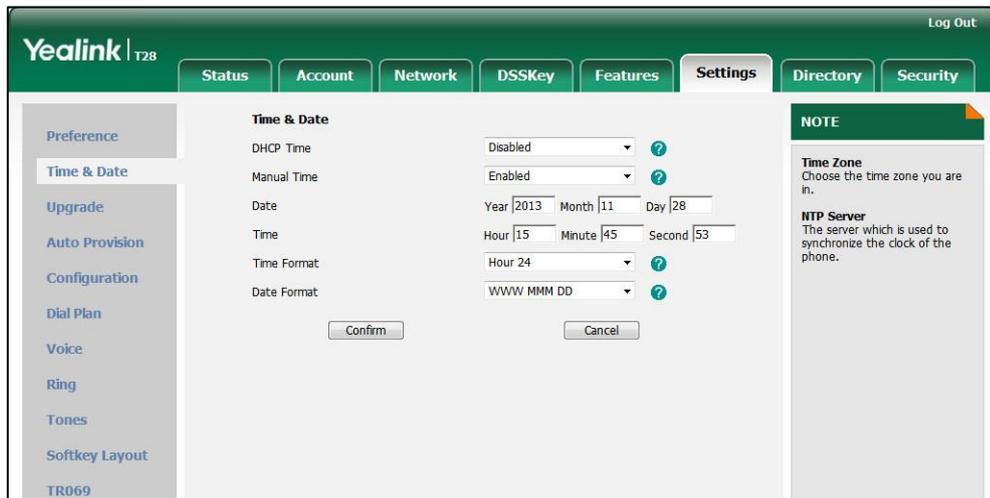


7. Enter the desired offset time in the **Offset (minutes)** field.

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

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

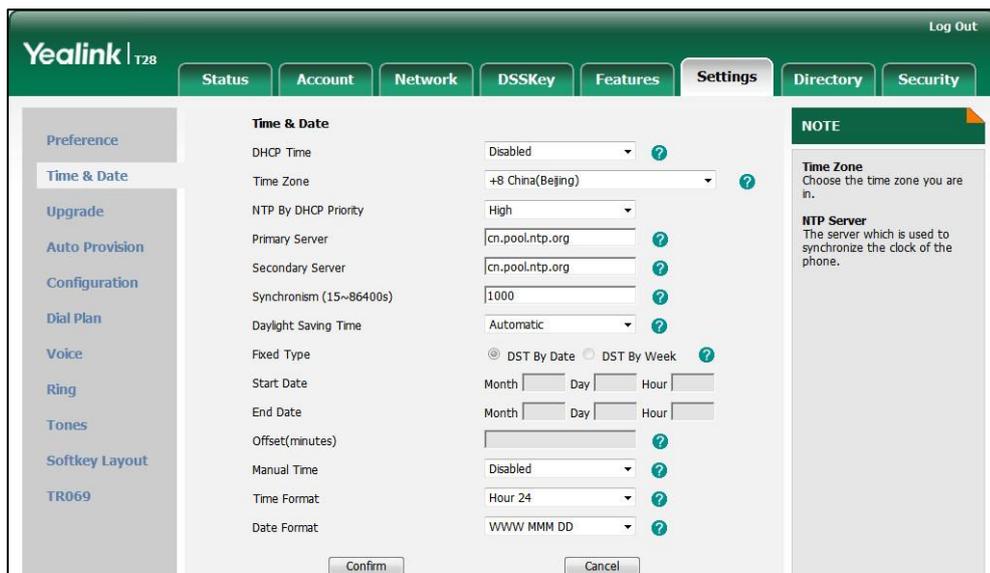
- Click on **Settings->Time & Date**.
- Select **Enabled** from the pull-down list of **Manual Time**.
- Enter the time and date in the corresponding fields.



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

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

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



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

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

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

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

1. Press **Menu->Settings->Basic Settings->Time & Date->Time & Date Format**.
2. Press  or  , or the **Switch** soft key to select the desired time format 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

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

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

Phone User Interface	Web User Interface
English	English
German	Chinese_S (only applicable to SIP-T19P and SIP-T21P IP phones)
Chinese_S (only applicable to SIP-T19P and SIP-T21P IP phones)	German
Chinese_T (only applicable to SIP-T19P and SIP-T21P IP phones)	French (not applicable to SIP-T19P and SIP-T21P IP phones)
French	Italian
Italian	Portuguese (not applicable to

Phone User Interface	Web User Interface
Portuguese	SIP-T19P and SIP-T21P IP phones)
Polish	Spanish (not applicable to
Spanish	SIP-T19P and SIP-T21P IP phones)
Turkish	Turkish

Loading Language Packs

Not all of supported languages are available for selection. Languages available for selection depend on language packs currently loaded to the IP phone. You can make languages available for use on the phone user interface by loading language packs to the IP phone. Language packs can only be loaded using configuration files.

The following table lists available languages and associated language packs.

Available Language	Associated Language Pack
English	lang+English.txt
Chinese_S (only applicable to SIP-T19P and SIP-T21P IP phones)	lang-Chinese_S.txt
Chinese_T (only applicable to SIP-T19P and SIP-T21P IP phones)	lang-Chinese_T.txt
German	lang-German.txt
French	lang-French.txt
Italian	lang-Italian.txt
Portuguese	lang-Portuguese.txt
Polish	lang-Polish.txt
Spanish	lang-Spanish.txt
Turkish	lang-Turkish.txt

Procedure

Loading language pack can only be 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 287.
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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.

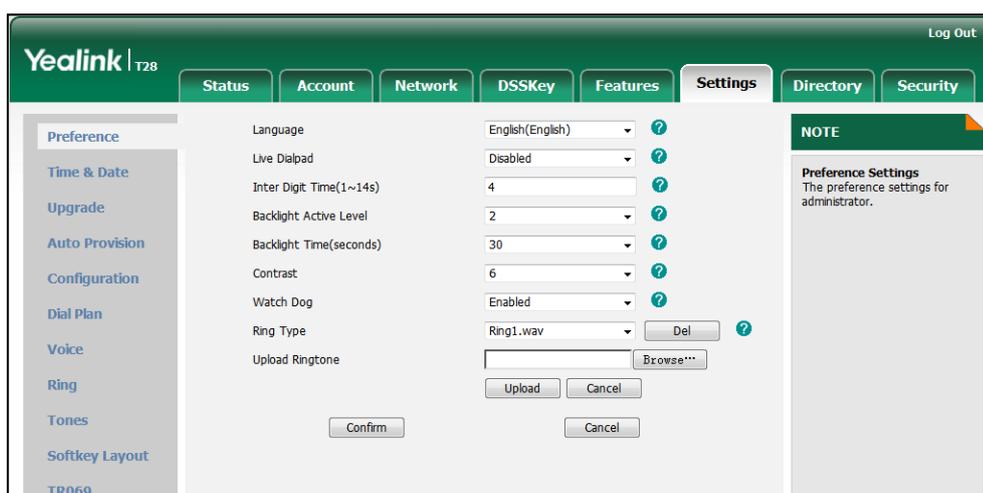
Procedure

Specify the language for the phone user interface or the web 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 287.
Local	Web User Interface	Specify the language for the web user interface. Navigate to: http://<phoneIPAddress>/servlet?&p=settings-preference&q=load
	Phone User Interface	Specify the language for the phone user interface.

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

1. Click on **Settings->Preference**.
2. Select the desired language from the pull-down list of **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.

Logo Customization

Logo customization allows unifying the IP phone appearance or displaying a custom image on the idle screen such as a company logo, instead of the default system logo. SIP-T20P IP phones only support a text logo.

The following table lists the logo file format and resolution for each phone model.

Phone Model	Logo File Format	Resolution
SIP-T28P	.dob	<=236*82 2 gray scale
SIP-T26P	.dob	<=132*64 2 gray scale
SIP-T22P/T21P/T19P	.dob	<=132*64 2 gray scale

Note

The format of the logo file must be *.dob. Before uploading your custom logo to IP phones, ensure your logo file is correctly formatted. For more information on customizing a logo file, refer to *Yealink SIP-T2 Series_T19P_T4_Series_IP_Phones_Auto_Provisioning_Guide*.

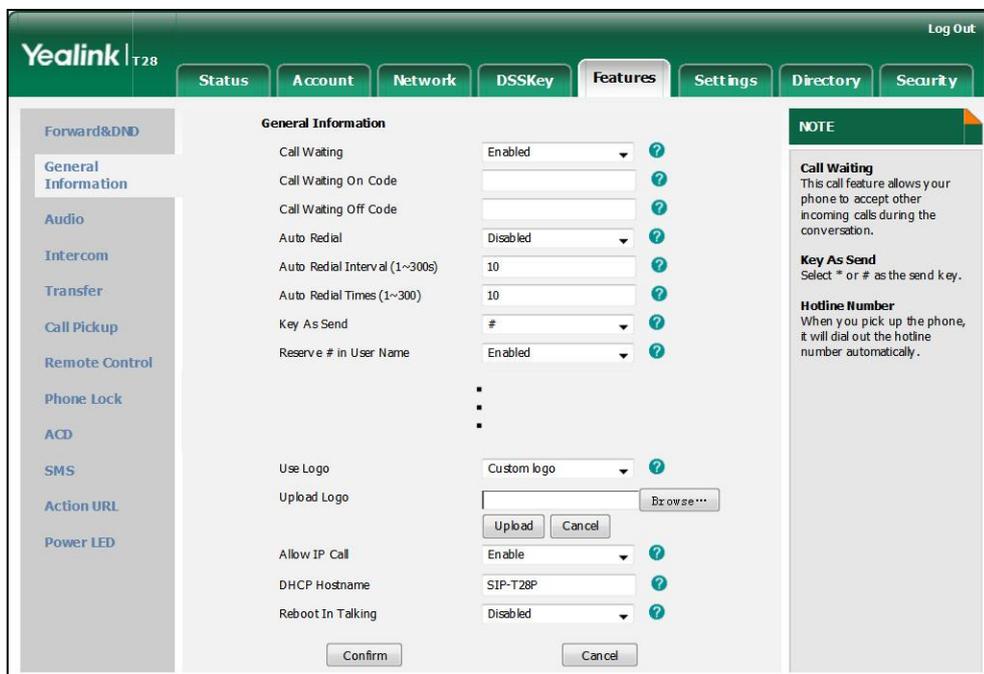
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the logo shown on the idle screen. For more information, refer to Logo Customization on page 289.
Local	Web User Interface	Configure the logo shown on the idle screen. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

To configure an image logo via web user interface (not applicable to SIP-T20P IP phones):

1. Click on **Features->General Information**.
2. Select **Custom logo** from the pull-down list of **Use Logo**.



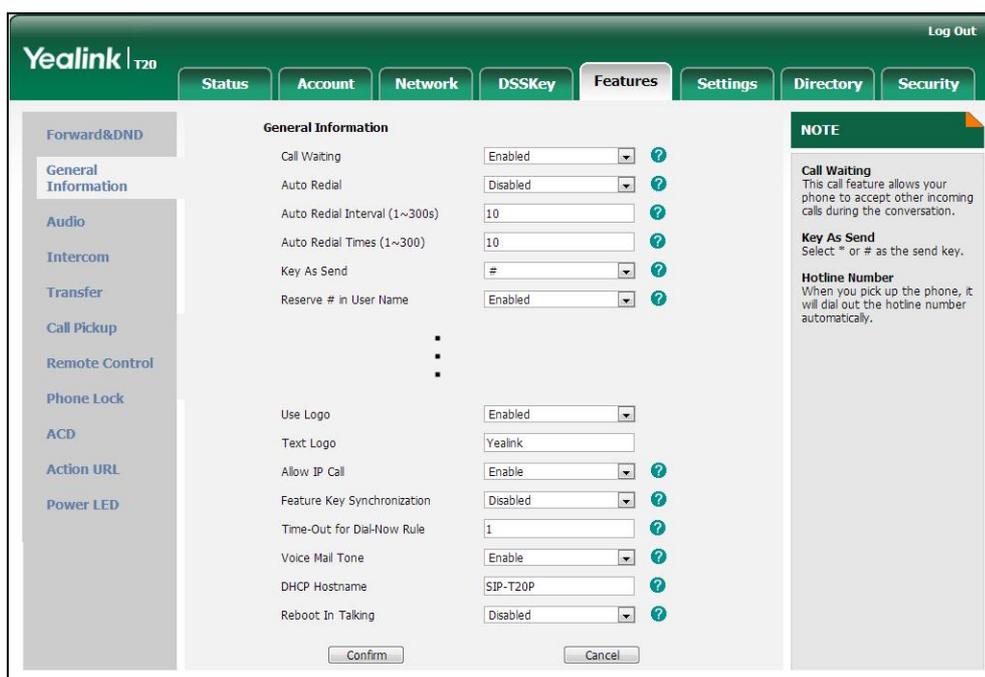
3. Click **Browse** to select the logo file from your local system.
4. Click **Upload** to upload the file.
5. Click **Confirm** to accept the change.

For SIP-T28P IP phones, the image logo is displayed on the idle screen. For SIP-T26P/T22P IP phones, the image logo screen and the idle screen are displayed alternately.

To configure a text logo via web user interface (only applicable to SIP-T20P IP phones):

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

- Enter the desired text (0~15 characters) in the **Text Logo** field.



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

The registered account and the configured text logo are displayed alternately.

Softkey Layout

Softkey layout is used to customize the soft keys at the bottom of the LCD screen to best meet users' requirements. It can be configured based on call states. In addition to specifying which soft keys to display, you can determine their display order. Softkey layout is not applicable to SIP-T20P IP phones. You can create softkey layout templates for different call states. For more information on the softkey layout template, refer to [Softkey Layout Template](#) on page 231.

The following table lists soft keys available for IP phones in different call states.

Call State	Default Soft Keys	Optional Soft Keys
CallFailed	NewCall Empty Empty Empty	Empty Switch Cancel
CallIn	Answer Forward Silence Reject	Empty Switch

Call State		Default Soft Keys	Optional Soft Keys
Connecting	Connecting	Empty Empty Empty Cancel	Empty Switch
	SemiAttendTrans	Transfer Empty Empty Cancel	Empty Switch
Dialing		Send IME Delete Cancel	Empty History Switch Line Favorite GPickup DPickup
RingBack	RingBack	Empty Empty Empty Cancel	Empty Switch CC
	SemiAttendTransBack	Transfer Empty Empty Cancel	Empty Switch CC
Talking	Talk	Transfer Hold Conference Cancel	Empty Mute SWAP NewCall Switch Answer Reject
	Hold	Transfer Resume NewCall Cancel	Empty Switch Answer Reject

Call State		Default Soft Keys	Optional Soft Keys
	Held	Empty Empty Empty Cancel	Empty Switch Answer Reject NewCall
	PreTrans	Transfer IME Delete Cancel	Empty Directory Switch Send
	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 on page 393.
Local	Web User Interface	Configure the softkey layout. Navigate to: http://<phoneIPAddress>/servlet?p=settings-softkey&q=load

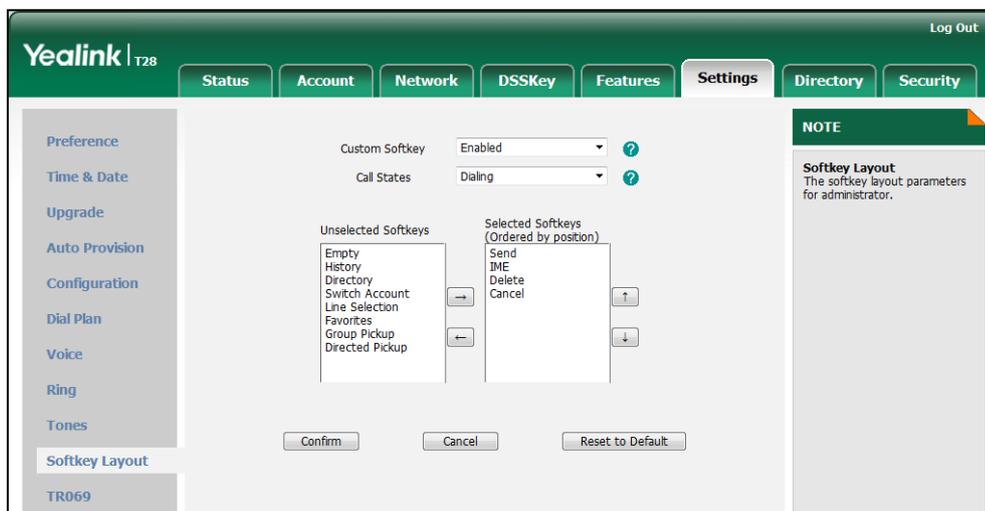
To configure softkey layout via web user interface:

1. Click on **Settings->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. Select the desired soft key from the **Unselected Softkeys** column and then click  .
The selected soft key appears in the **Selected Softkeys** column.
5. Repeat the step 4 to add more soft keys to the **Selected Softkeys** column.
6. To remove the soft key from the **Selected Softkeys** column, select the desired soft

key and then click  .

- To adjust the display order of soft keys, select the desired soft key and then click  or  .

The LCD screen displays the soft keys in the adjusted order.



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

Key as Send

Key as send allows assigning the pound key or star key as a send key. Send sound allows the IP phone to play a key tone when a user presses the send key. Key tone allows the IP phone to play a key tone when a user presses any key. Send sound works only if Key tone is enabled.

Procedure

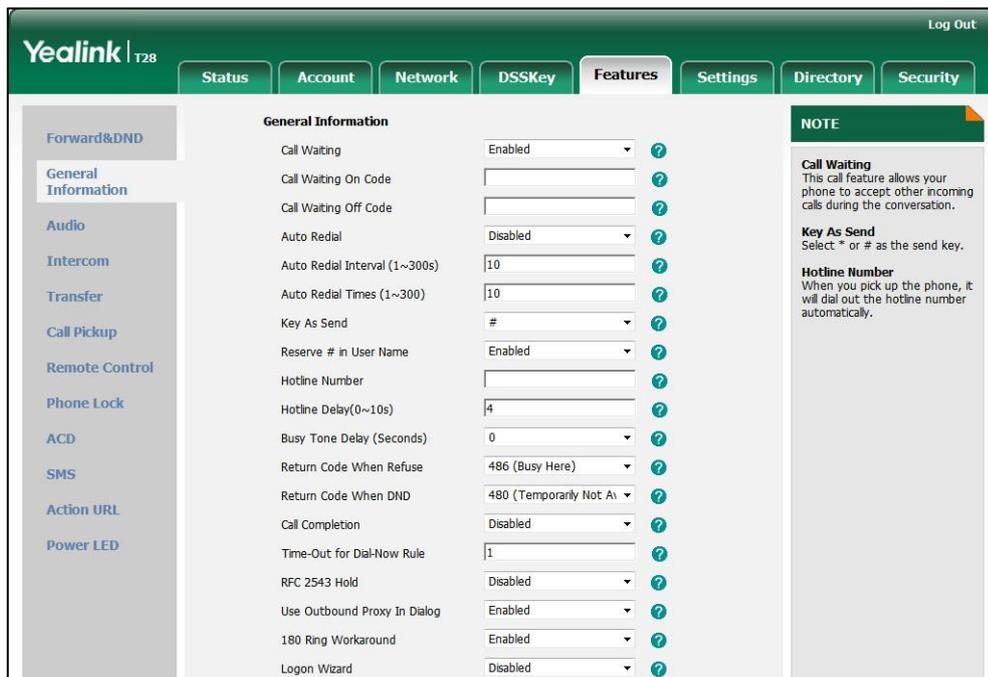
Key as send can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure a send key. Configure a send sound. Configure a key tone. For more information, refer to Key as Send on page 290.
Local	Web User Interface	Configure a send key. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load Configure a send sound and key tone. Navigate to:

		http://<phoneIPAddress>/servlet ?p=features-audio&q=load
	Phone User Interface	Configure the send key.

To configure a send key via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Key As Send**.

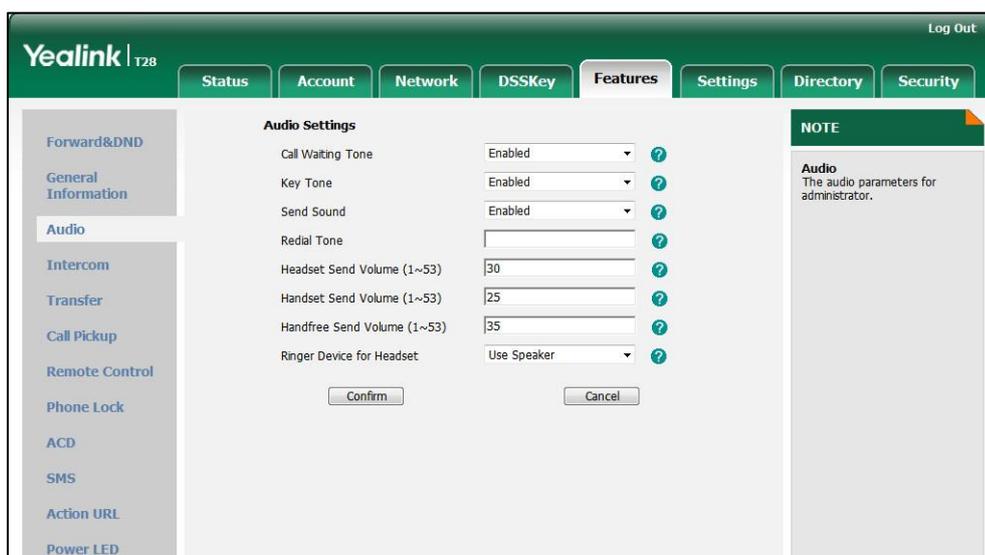


3. Click **Confirm** to accept the change.

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

1. Click on **Features->Audio**.
2. Select the desired value from the pull-down list of **Key Sound**.

3. Select the desired value from the pull-down list of **Send Sound**.



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

To configure 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.

Hotline

Hotline is a point-to-point communication 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 specified time interval when off-hook. IP phones only support one hotline number.

Procedure

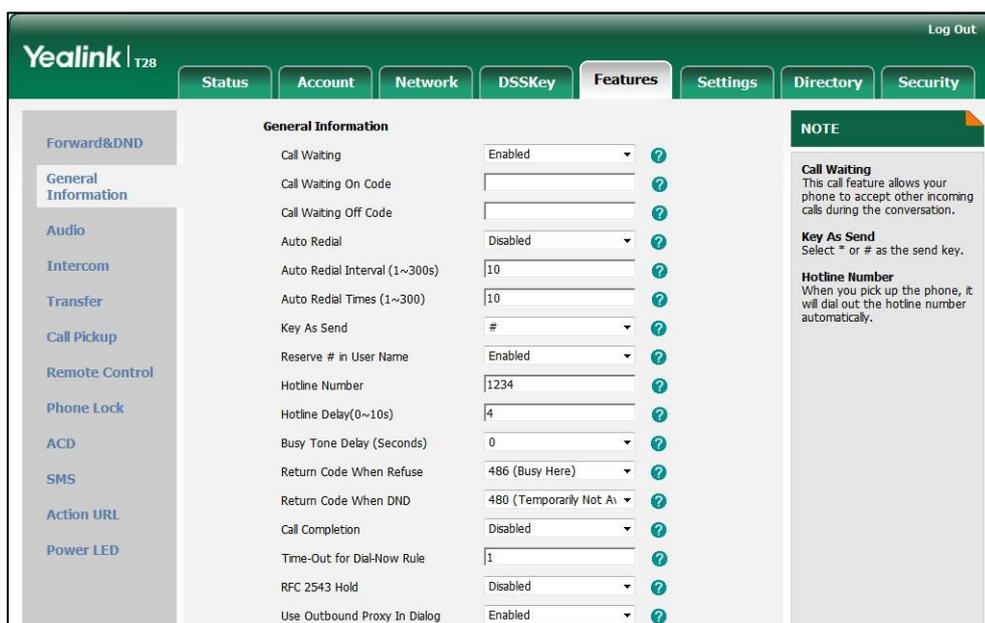
Hotline can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the hotline number. Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number. For more information, refer to Hotline on page 292.
Local	Web User Interface	Configure the hotline number. Specify the time (in seconds) the

		<p>IP phone waits before automatically dial out the hotline number.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	Phone User Interface	<p>Configure the hotline number.</p> <p>Specify the time (in seconds) the IP phone waits before automatically dialing out the hotline number.</p>

To configure hotline via web user interface:

1. Click on **Features->General Information**.
2. Enter the hotline number in the **Hotline Number** field.
3. Enter the delay time in the **Hotline Delay (0~10s)** 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 **Hot Number** field.
3. Enter the waiting time (in seconds) in the **HotLine Delay** field.
4. Press the **Save** soft key to accept the change.

Call Log

Call log contains call information such as remote party identification, time and date, and call duration. IP phones maintain a local call log. Call log consists of four lists: Placed Calls, Received Calls, Missed Calls and Forwarded Calls. Call log lists support 100 entries in all. To store call information, you must enable save call log feature in advance.

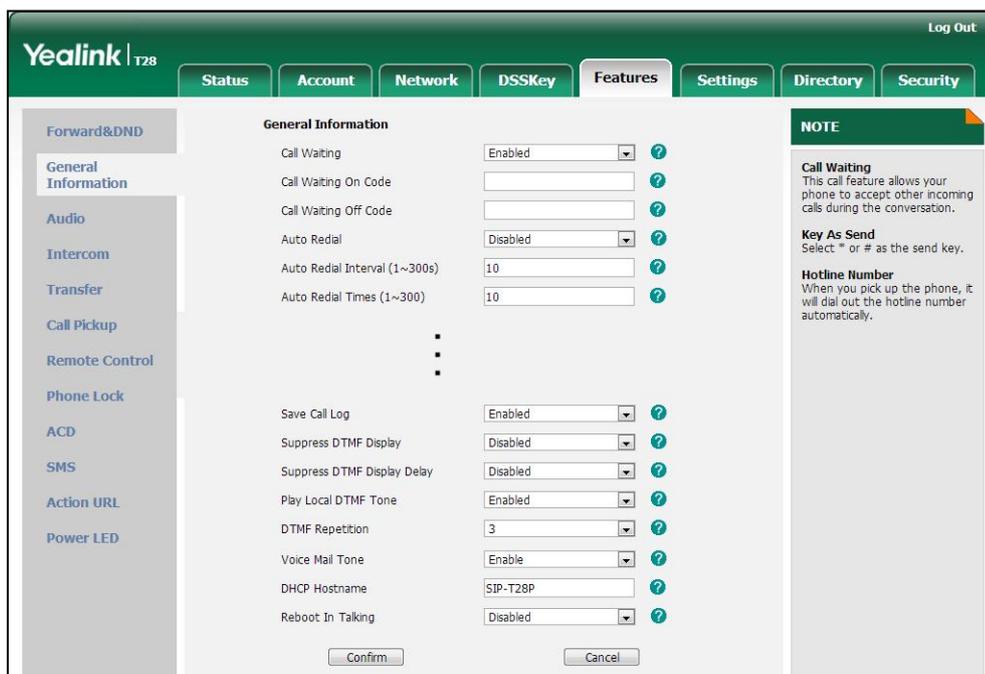
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure call log feature. For more information, refer to Call Log on page 293.
Local	Web User Interface	Configure call log feature. Navigate to: http://<phoneIPAddress>/servlet? p=features-general&q=load
	Phone User Interface	Configure the call log.

To configure call log feature via web user interface:

1. Click on **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 call log feature 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

Missed call log allows the IP phone to display the number of missed calls with an indicator icon on the idle screen, and to log missed calls in the Missed Calls list when the IP phone misses calls. It is configurable on a per-line basis. Once the user accesses the Missed Calls list, the prompt message and indicator icon on the idle screen disappear.

Procedure

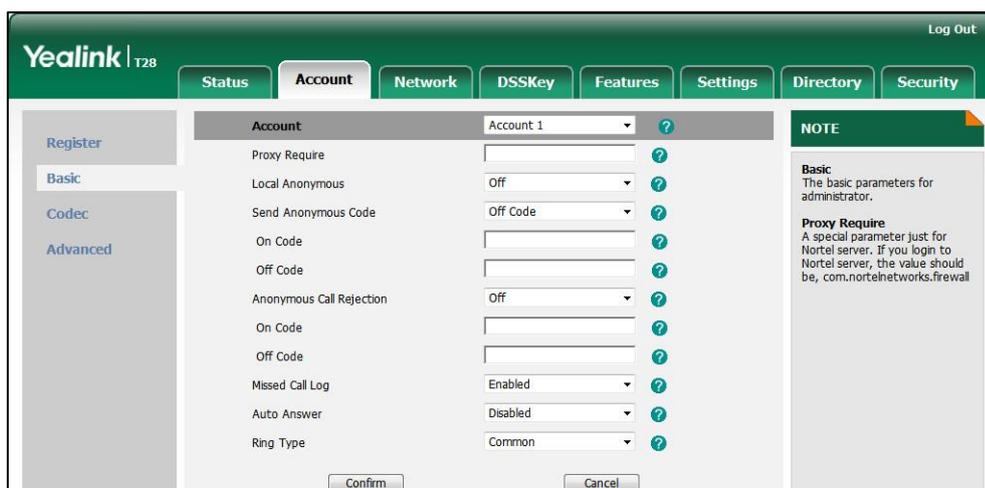
Missed call log can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure missed call log feature. For more information, refer to Missed Call Log on page 293.
Local	Web User Interface	Configure missed call log feature. Navigate to: http://<phoneIPAddress>/servlet ?p=account-basic&q=load&acc =0

To configure missed call log via web user interface:

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

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



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

Local Directory

IP phones maintain a local directory. The local directory can store up to 1000 contacts and 5 groups. When adding a contact to the local directory, in addition to name and phone numbers, you can also specify the account, ring tone and group for the contact. Contacts and groups can be added either one by one or in batch using a local contact file. For more information on the contact file, refer to [Local Contact File](#) on page 233.

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 396.
Local	Web User Interface	Add a group and a contact to the local directory. Navigate to: http://<phoneIPAddress>/servlet?p=contactsbasic&q=load&num=1&group=
	Phone User Interface	Add a group and a contact to the local directory.

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

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

The screenshot shows the Yealink T28 web user interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Directory' tab is active, showing the 'Local Directory' section. A table lists contacts with columns for Index, Name, Office Number, Mobile Number, and Other Number. Below the table are navigation buttons like 'Page 1', 'Prev', 'Next', 'Hang Up', 'Delete All', 'Delete', 'Move To', and 'All Contact'. The 'Group Setting' form is visible, with fields for Name, Office Number, Mobile Number, Other Number, Ring Tone (set to 'Auto'), and Group (set to 'All Contacts'). The 'Add' button is highlighted. A 'NOTE' section on the right provides instructions for adding, deleting, and moving contacts.

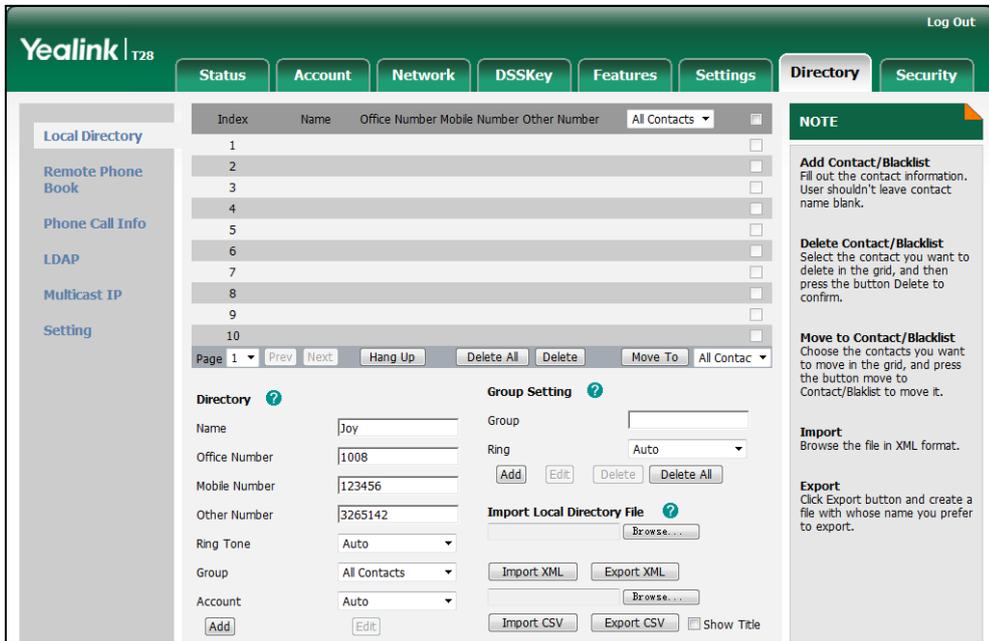
4. Click **Add** to add the group.

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

1. Click on **Directory->Local Directory**.
2. In the **Directory** block, enter the name and the office, mobile or other numbers in the corresponding fields.
3. Select the desired ring tone from the pull-down list of **Ring Tone**.
4. Select the desired group from the pull-down list of **Group**.

5. Select the desired account from the pull-down list of **Account**.

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



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

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

1. Press **Menu->Directory->Local Directory**.
2. Press the **AddGrp** soft key.
3. Enter the desired group name in the **Name** field.
4. Press **←** or **→**, or the **Switch** soft key to select the desired group ring tone from the **Ring Tones** field.
5. Press the **Add** soft key to accept the change.

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

1. Press **Menu->Directory->Local Directory**.
2. Select the desired contact group.
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 Tones** field.
7. Press the **Save** soft key to accept the change.

Live Dialpad

Live dialpad allows IP phones to automatically dial out the entered phone number after a specified period of time.

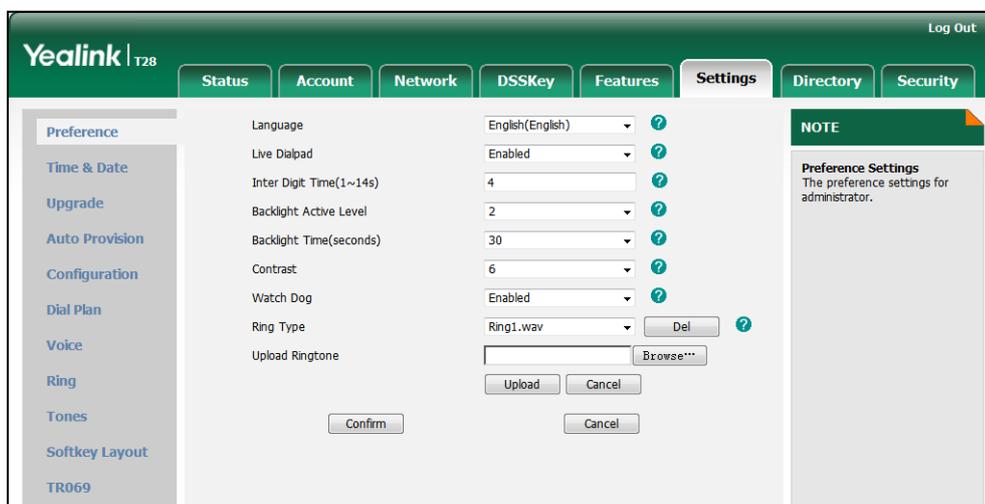
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure live dialpad. For more information, refer to Live Dialpad on page 294.
Local	Web User Interface	Configure live dialpad. Navigate to: http://<phoneIPAddress>/servlet ?p=settings-preference&q=load

To configure live dialpad via web user interface:

1. Click on **Settings->Preference**.
2. Select the desired value from the pull-down list of **Live Dialpad**.
3. Enter the desired delay time in the **Inter Digit Time (1~14s)** field.



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

Call Waiting

Call waiting allows IP phones to receive a new call when there is already an active call. The new incoming call is presented to the user visually on the LCD screen. Call waiting tone allows the phone to play a short tone, to remind the user audibly of a new incoming call during conversation. Call waiting tone works only if call waiting is

enabled.

Procedure

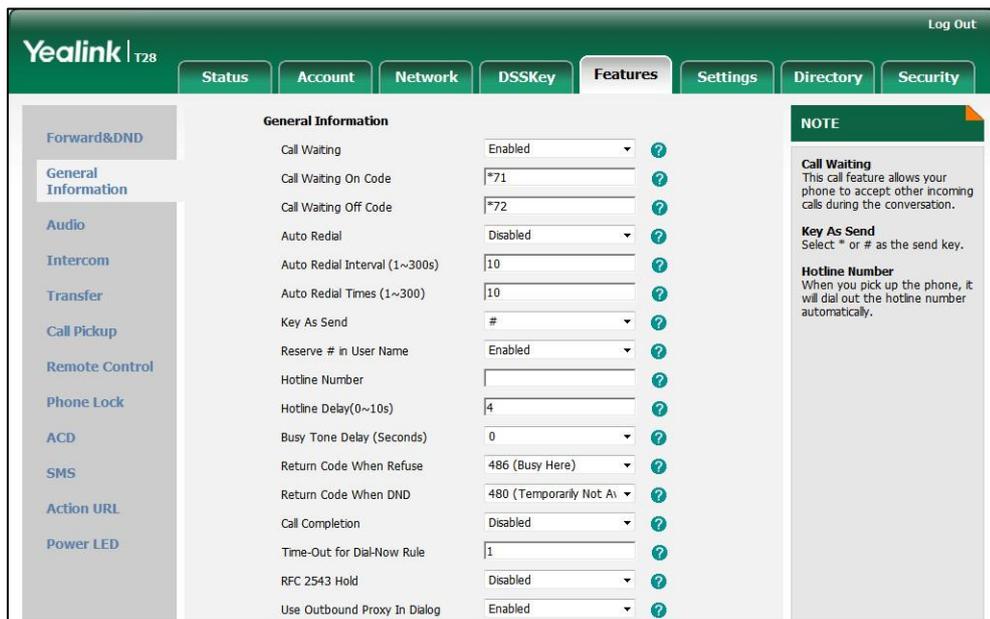
Call waiting and call waiting tone can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure call waiting and call waiting tone. For more information, refer to Call Waiting on page 295.
Local	Web User Interface	Configure call waiting. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load Configure call waiting tone. Navigate to: http://<phoneIPAddress>/servlet?p=features-audio&q=load
	Phone User Interface	Configure call waiting and call waiting tone.

To configure call waiting via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Call Waiting**.
3. (Optional.) Enter the call waiting on code in the **Call Waiting On Code** field.

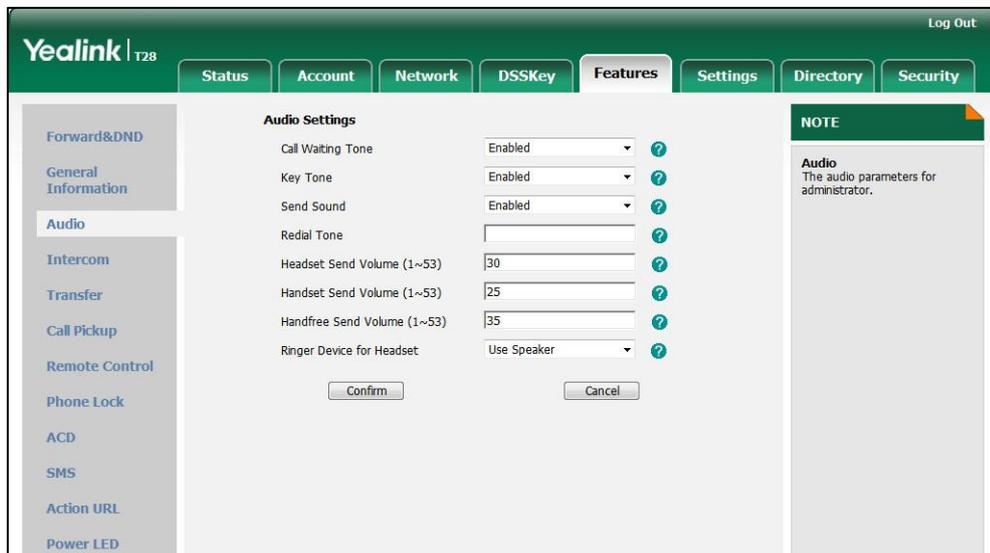
- (Optional.) Enter the call waiting off code in the **Call Waiting Off Code** field.



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

To configure call waiting tone via web user interface:

- Click on **Features->Audio**.
- Select the desired value from the pull-down list of **Call Waiting Tone**.



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

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

- Press **Menu->Features->Call Waiting**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Call Waiting** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Play**

Tone field.

4. (Optional.) Enter the call waiting on code in the **CW On Code** field.
5. (Optional.) Enter the call waiting off code in the **CW Off Code** field.
6. Press the **Save** soft key to accept the change.

Auto Redial

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

Procedure

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

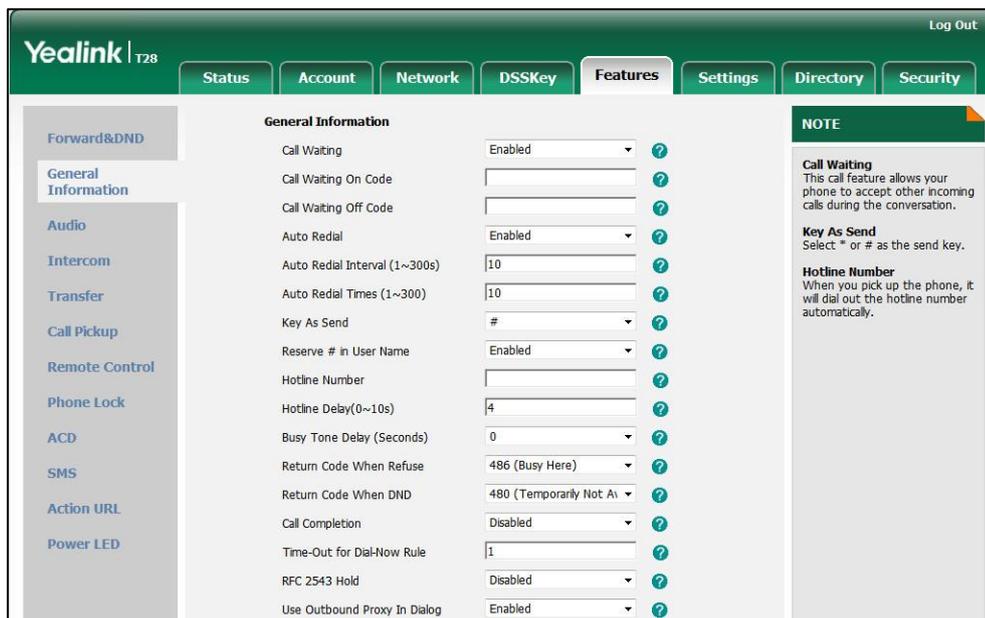
Configuration File	<y0000000000xx>.cfg	Configure auto redial feature. For more information, refer to Auto Redial on page 296.
Local	Web User Interface	Configure auto redial feature. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure auto redial feature.

To configure auto redial via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Auto Redial**.
3. Enter the waiting time in the **Auto Redial Interval (1~300s)** field.
The default waiting time is 10s.

4. Enter the desired times in the **Auto Redial Times (1~300)** field.

The default value is 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 waiting time (in seconds) in the **Redial Interval** field.
4. Enter the desired times in the **Redial Times** field.
5. Press the **Save** soft key to accept the change.

Auto Answer

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

Procedure

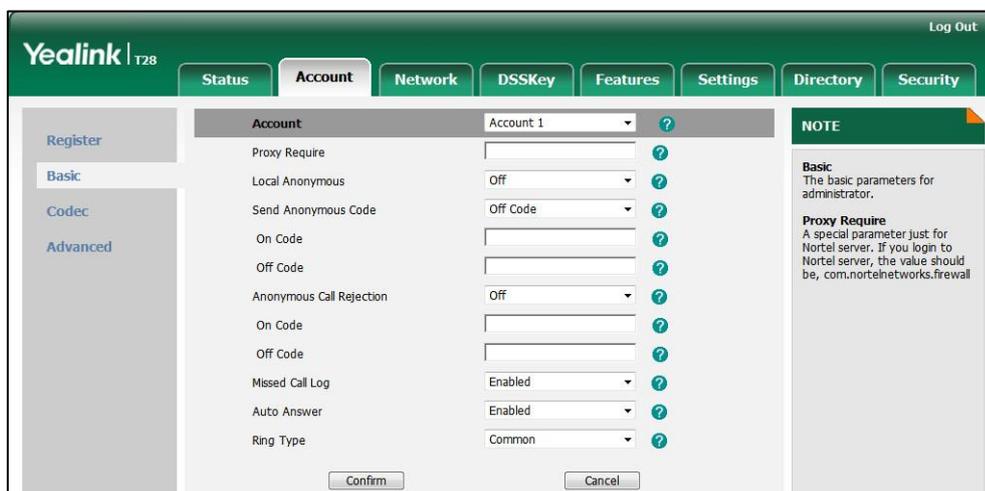
Auto answer can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure auto answer. For more information, refer to Auto Answer on page 297.
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	<y0000000000xx>.cfg	Specify a period of delay time for auto answer. For more information, refer to Auto Answer on page 297.
Local	Web User Interface	Configure auto answer. Navigate to: http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0 Specify a period of delay time for auto answer. Navigate to: http://<phoneIPAddress>servlet?p=features-general&q=load
	Phone User Interface	Configure auto answer.

To configure auto answer via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Basic**.
4. Select the desired value from the pull-down list of **Auto Answer**.



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

To configure a period of delay time for auto answer via web user interface:

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

- Enter the desired time in the **Auto-Answer Delay (1~4s)** field.

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

To configure auto answer via phone user interface:

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

Call Completion

Call completion allows users to monitor the busy party and establish a call when the busy party becomes available to receive a call. Two factors commonly prevent a call from connecting successfully:

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

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

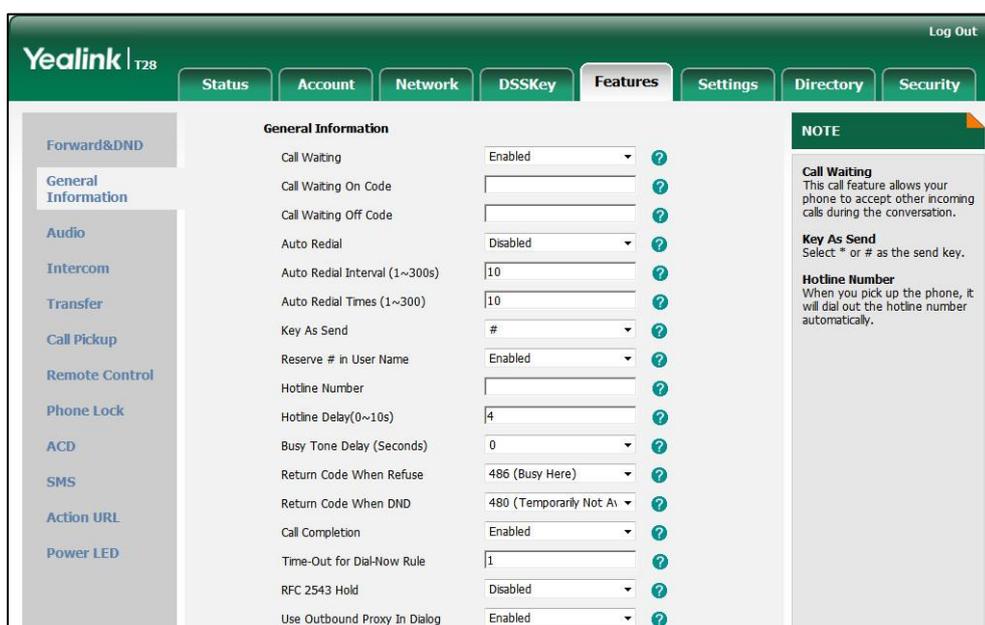
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure call completion. For more information, refer to Call Completion on page 298.
Local	Web User Interface	Configure call completion. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure call completion.

To configure call completion via web user interface:

1. Click on **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

Anonymous call allows the caller to conceal the identity information displayed on the callee's screen. The callee's phone LCD screen prompts an incoming call from anonymity. Anonymous call is configurable on a per-line basis.

Example of anonymous SIP header:

```
Via: SIP/2.0/UDP 10.2.8.183:5063;branch=z9hG4bK1535948896
From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag=128043702
To: <sip:1011@10.2.1.199>
Call-ID: 1773251036@10.2.8.183
CSeq: 1 INVITE
Contact: <sip:1012@10.2.8.183:5063>
Content-Type: application/sdp
Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER,
PUBLISH, UPDATE, MESSAGE
Max-Forwards: 70
User-Agent: Yealink SIP-T28P 2.72.0.1
Privacy: id
Supported: replaces
Allow-Events: talk,hold,conference,refer,check-sync
P-Preferred-Identity: <sip:1012@10.2.1.199>
Content-Length: 302
```

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

Procedure

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

Configuration File	<MAC>.cfg	Configure anonymous call. For more information, refer to Anonymous Call on page 299.
Local	Web User Interface	Configure anonymous call. Navigate to: http://<phoneIPAddress>/servlet ?p=account-basic&q=load&acc=0
	Phone User Interface	Configure anonymous call.

To configure anonymous call via web user interface:

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

The screenshot shows the Yealink T28 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is selected, and the 'Basic' sub-tab is active. The 'Account' dropdown is set to 'Account 1'. The 'Local Anonymous' field is set to 'On'. The 'Send Anonymous Code' field is set to 'On Code'. The 'On Code' field contains '*71' and the 'Off Code' field contains '*72'. The 'Anonymous Call Rejection' field is set to 'Off'. The 'Missed Call Log' field is set to 'Enabled', the 'Auto Answer' field is set to 'Disabled', and the 'Ring Type' field is set to 'Common'. A 'NOTE' section on the right explains the 'Basic' parameters and the 'Proxy Require' field.

8. 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 **Send Anon** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Anon Code** field.
5. (Optional.) Enter the anonymous call on code in the **Call On Code** field.
6. (Optional.) Enter the anonymous call off code in the **Call Off Code** field.
7. Press the **Save** soft key to accept the change.

Anonymous Call Rejection

Anonymous call rejection allows IP phones to automatically reject incoming calls from callers whose identity has been deliberately concealed. The anonymous caller's phone LCD screen presents "Anonymity Disallowed". Anonymous call rejection is configurable on a per-line basis.

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

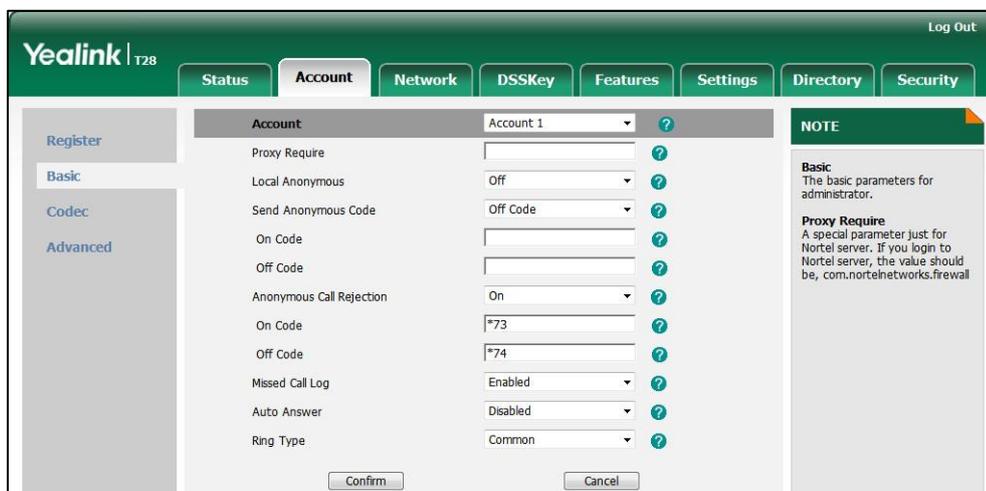
Procedure

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

Configuration File	<MAC>.cfg	Configure anonymous call rejection. For more information, refer to Anonymous Call Rejection on page 301.
Local	Web User Interface	Configure anonymous call rejection. Navigate to: http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0
	Phone User Interface	Configure anonymous call rejection.

To configure anonymous call rejection via web user interface:

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



7. 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 **Anon 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 IP phones to ignore incoming calls. DND feature can be configured on a phone or a per-line basis depending on the DND mode. Two DND modes:

- **Phone** (default): DND feature is effective for the IP phone.
- **Custom**: DND feature can be configured for each or all accounts.

A user can activate or deactivate DND using the DND key or DND soft key (not applicable to SIP-T20P IP phones). DND activated on the IP phone disables the local call forward settings. The DND configurations on IP phones may be overridden by the server settings.

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

Return Message When DND

This feature defines the return code and the reason of the SIP response message for the rejected incoming call when DND is enabled on the IP phone. The caller's phone LCD screen displays the received return code.

Procedure

DND can be configured using the configuration files or locally.

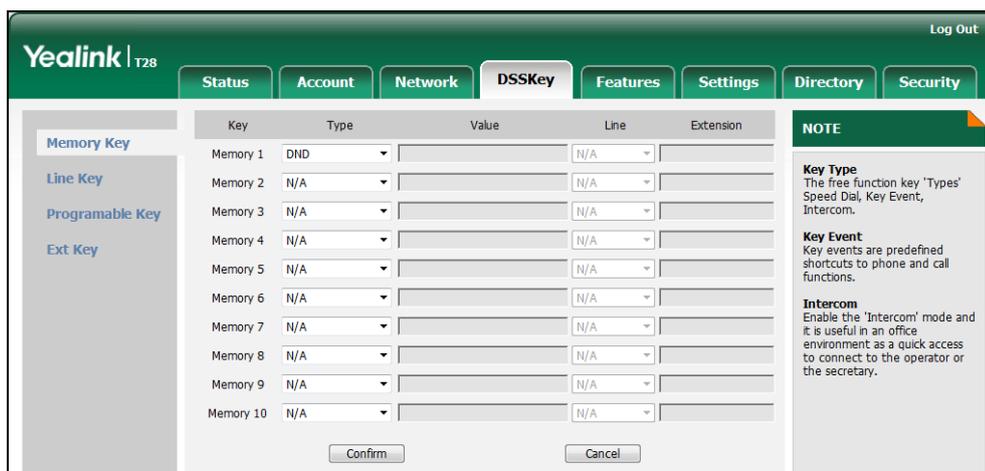
Configuration File	<MAC>.cfg	Configure DND in the custom mode. For more information, refer to Do Not Disturb on page 302.
	<y0000000000xx>.cfg	Assign a DND key. For more information, refer to DND Key on page 408.

		<p>Configure the DND mode.</p> <p>Configure DND in the phone mode.</p> <p>Specify the return code and the reason of the SIP response message when DND is enabled.</p> <p>For more information, refer to Do Not Disturb on page 302.</p>
Local	Web User Interface	<p>Assign a DND key.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</p> <p>Configure DND.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-forward&q=load</p> <p>Specify the return code and the reason of the SIP response message when DND is enabled.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	Phone User Interface	<p>Assign a DND key.</p> <p>Configure DND.</p>

To configure a DND key via web user interface:

1. Click on **DSSKey->Memory Key (Line Key or Programable Key)**.
 SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.

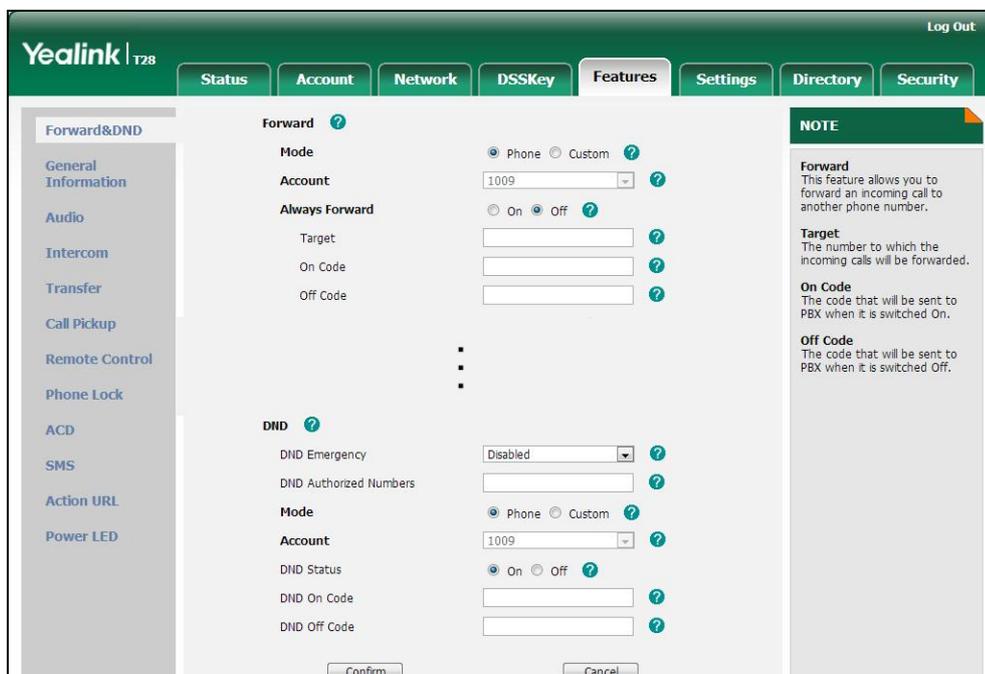
- In the desired DSS key field, select **DND** from the pull-down list of **Type**.



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

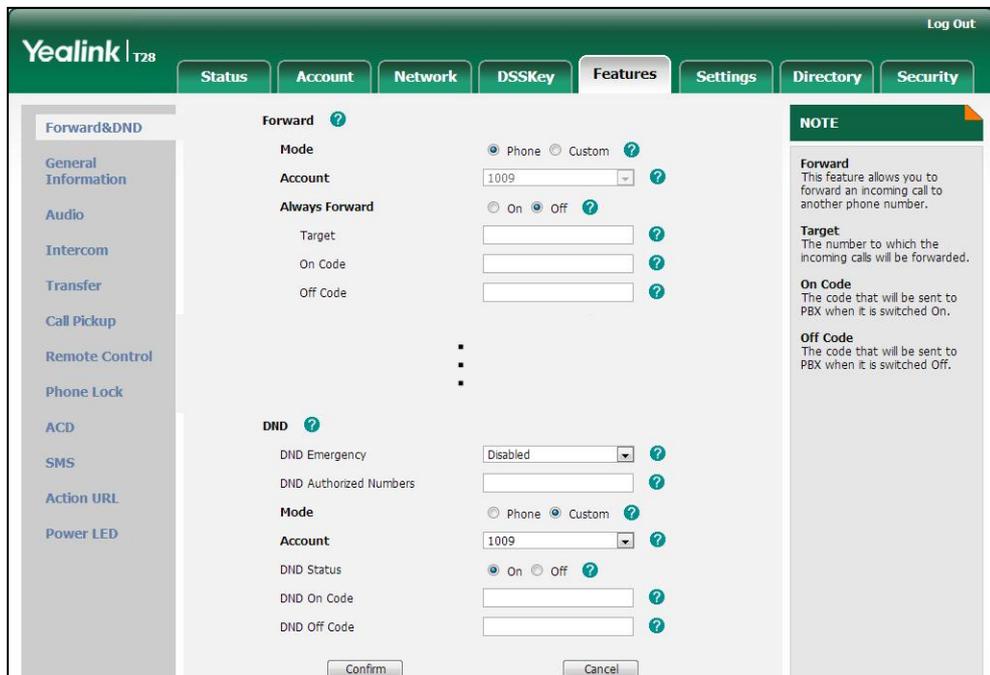
To configure DND feature via web user interface:

- Click on **Features->Forward & DND**.
- In the **DND** block, mark the desired radio box in the **Mode** field.
 - If you mark the **Phone** radio box:
 - Mark the desired radio box in the **DND Status** field.
 - (Optional.) Enter the DND on code in the **DND On Code** field.
 - (Optional.) Enter the DND off code in the **DND Off Code** field.



- If you mark the **Custom** radio box:
 - Select the desired account from the pull-down list of **Account**.

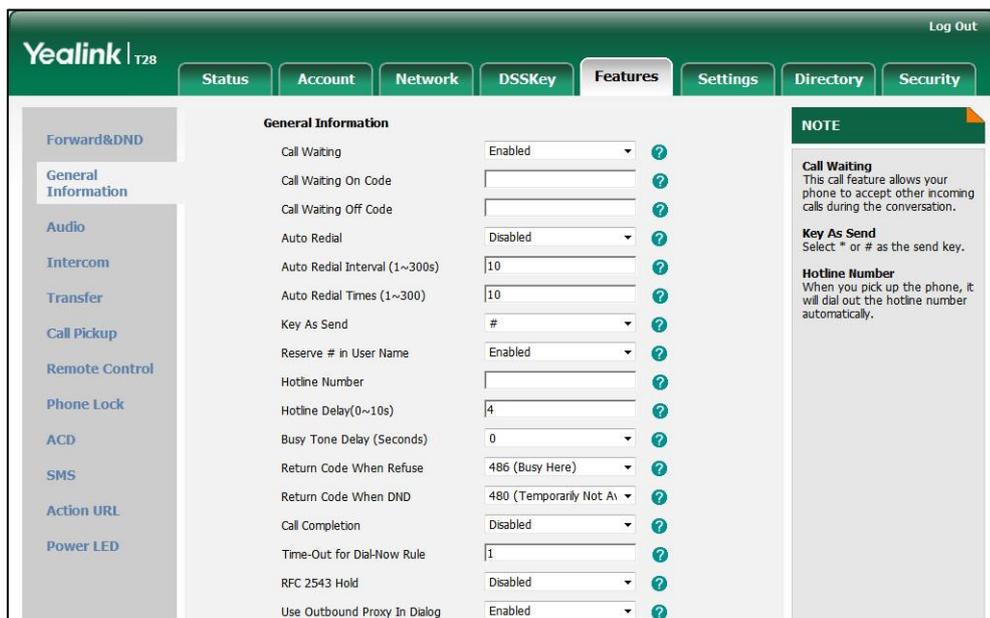
- 2) Mark the desired radio box in the **DND Status** field.
- 3) (Optional.) Enter the DND on code in the **DND On Code** field.
- 4) (Optional.) Enter the DND off code in the **DND Off Code** field.



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

To specify the return code and the reason when DND is enabled via web user interface:

1. Click on **Features->General Information**.
2. Select the desired type 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.

To configure DND in the phone mode via phone user interface:

1. Press the **DND** soft key or the DND key when the IP phone is idle.

To configure DND in the custom mode for a specific account via phone user interface:

1. Press the **DND** soft key or the DND key when the IP phone is idle.
The LCD screen displays a list of accounts registered on the IP phone.
2. Press  or  to select the desired account.
3. Press  or  to select **On** to activate DND.
You can configure DND in the custom mode for all accounts by pressing the **All On** soft key.
4. Press the **Save** soft key to accept the change.

Busy Tone Delay

Busy tone is audible to the other party, indicating that the call connection has been broken when one party releases a call. Busy tone delay can define a period of time during which the busy tone is audible.

Procedure

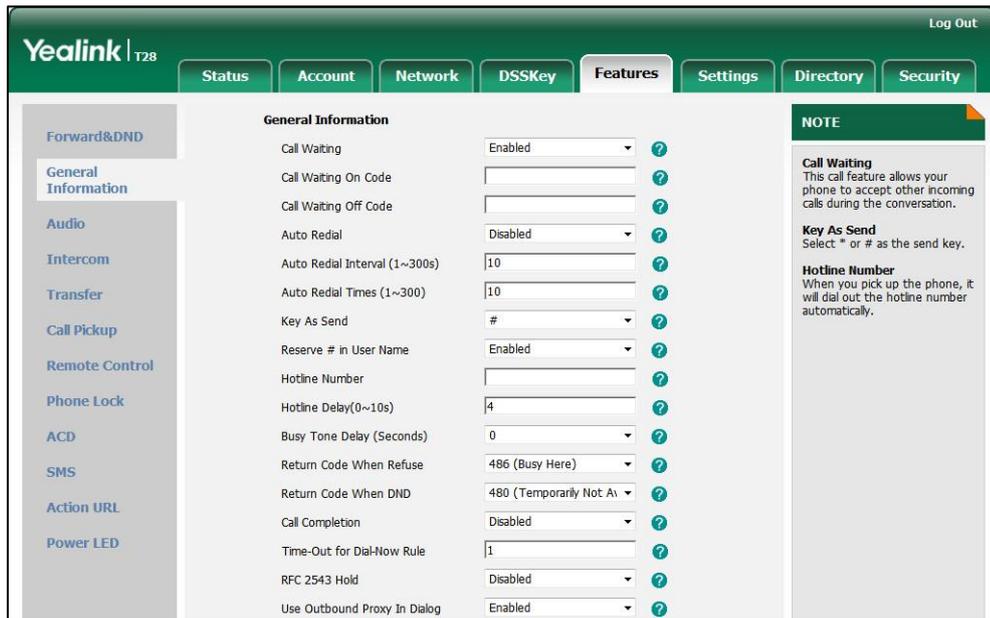
Busy tone delay can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure busy tone delay. For more information, refer to Busy Tone Delay on page 305.
Local	Web User Interface	Configure busy tone delay. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

To configure busy tone delay via web user interface:

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

- Select the desired value from the pull-down list of **Busy Tone Delay (Seconds)**.



- 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 for the refused call. The caller’s phone LCD screen displays the reason according to the received return code. Available return codes and reasons are:

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

Procedure

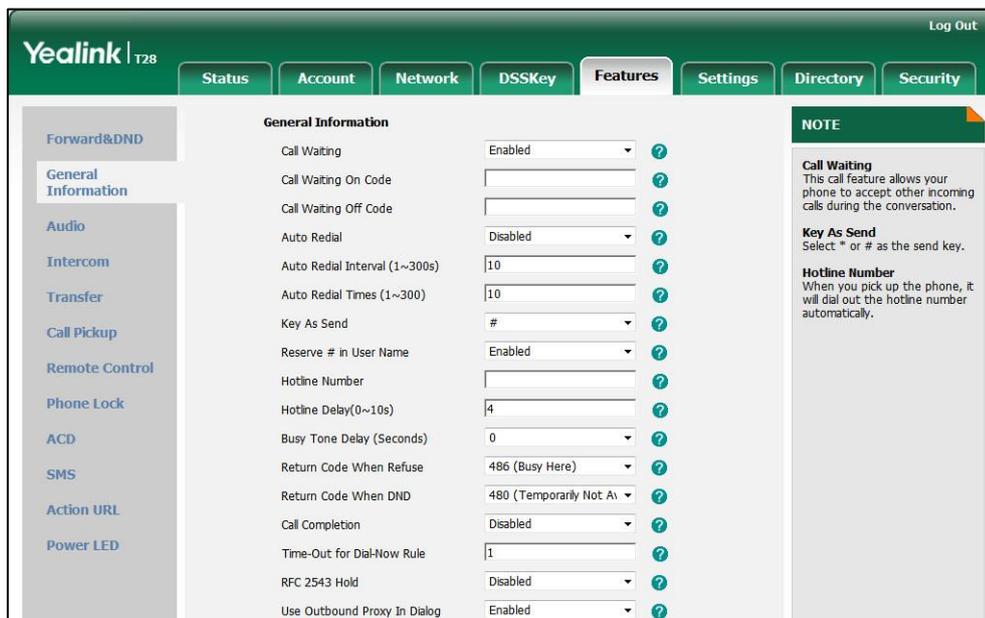
Return code for refused call can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify the return code and the reason of the SIP response message when refusing a call. For more information, refer to Return Code When Refuse on page 305.
Local	Web User Interface	Specify the return code and the reason of the SIP response message when refusing a call. Navigate to:

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

To specify the return code and the reason when refusing a call via web user interface:

1. Click on **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.

Early Media

Early media refers to media (e.g., audio and video) played to the caller before a SIP call is actually established. Current implementation supports early media through the 183 message. When the caller receives a 183 message with SDP before the call is established, a media channel is established. This channel is used to provide the early media stream for the caller.

180 Ring Workaround

180 ring workaround defines whether to deal with the 180 message received after the 183 message. When the caller receives a 183 message, it suppresses any local ringback tone and begins to play the media received. 180 ring workaround allows IP phones to resume and play the local ringback tone upon a subsequent 180 message received.

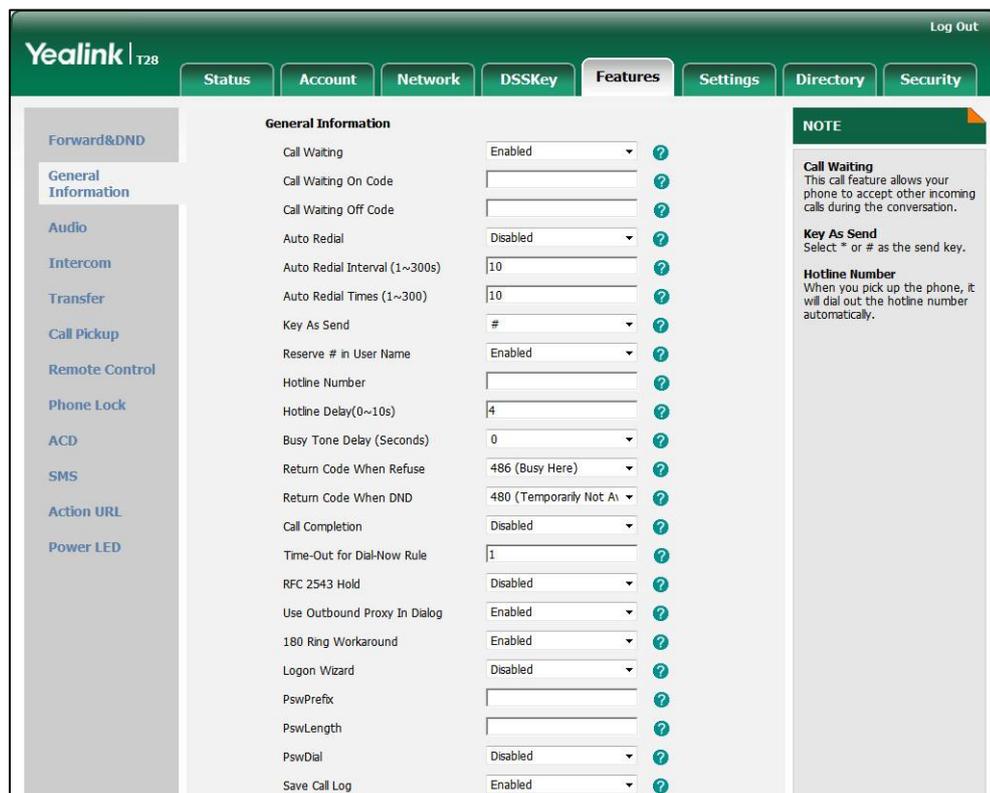
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure 180 ring workaround. For more information, refer to 180 Ring Workaround on page 306.
Local	Web User Interface	Configur 180 ring workaround. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

To configure 180 ring workaround via web user interface:

1. Click on **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 SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.

Note To use this feature, make sure the outbound server has been correctly configured on the IP phone.

Procedure

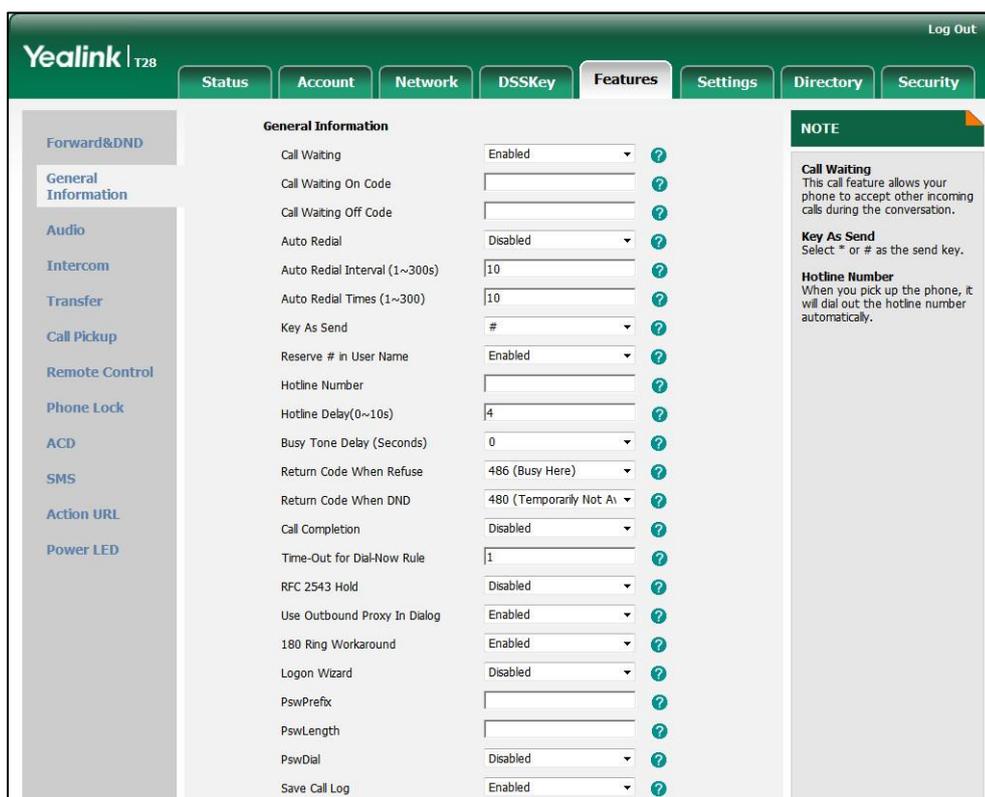
Use outbound proxy in dialog can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Specify whether to use outbound proxy in a dialog. For more information, refer to Use Outbound Proxy in Dialog on page 306.
Local	Web User Interface	Specify whether to use outbound proxy in a dialog. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

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

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

- Select the desired value from the pull-down list of **Use Outbound Proxy In Dialog**.



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

SIP Session Timer

SIP session timers T1, T2 and T4 are SIP transaction layer timers defined in RFC 3261. Timer T1 is an estimate of the Round Trip Time (RTT) of transactions between a SIP client and SIP server. Timer T2 represents the maximum retransmit interval for non-INVITE requests and INVITE responses. Timer T4 represents the maximum duration a message will remain in the network. These session timers are configurable on IP phones.

Procedure

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

Configuration File	<MAC>.cfg	Configure SIP session timer. For more information, refer to SIP Session Timer on page 307.
Local	Web User Interface	Configure SIP session timer. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc=0

To configure session timer via web user interface:

1. Click on **Account**.
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 T1 (0.5~10s)** field.
The default value is 0.5s.
5. Enter the desired value in the **SIP Session Timer T2 (2~40s)** field.
The default value is 4s.
6. Enter the desired value in the **SIP Session Timer T4 (2.5~60s)** field.
The default value is 5s.

Parameter	Value
Account	Account 1
Keep Alive Type	Default
Keep Alive Interval (Seconds)	30
Local SIP Port	5060
RPort	Disabled
SIP Session Timer T1 (0.5~10s)	0.5
SIP Session Timer T2 (2~40s)	4
SIP Session Timer T4 (2.5~60s)	5
Subscribe Period (Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~255)	101
Retransmission	Disabled
Subscribe for MWI	Disabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Disabled
Voice Mail	
Caller ID Source	FROM
Session Timer	Disabled
Session Expires(30~7200s)	1800
Session Refresher	UAC
Send user=phone	Disabled

NOTE
Advanced
The Advanced parameters for administrator.

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

Session Timer

Session timer allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active. Session timer is specified in RFC 4028. IP phones support two refresher modes: UAC and UAS. The UAC mode means refreshing the session from the client, while the UAS mode means refreshing the session from the server. The session expiration and session refresher are negotiated via the Session-Expires header in the INVITE message. The negotiated refresher will send a re-INVITE/UPDATE request at or before the negotiated session expiration.

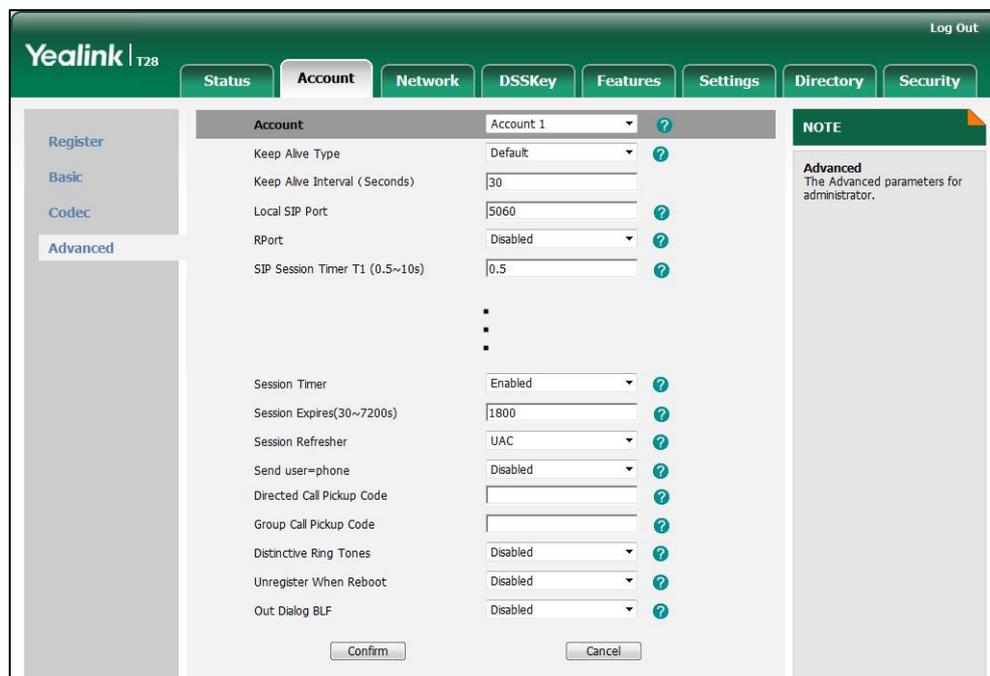
Procedure

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

Configuration File	<MAC>.cfg	Configure session timer. For more information, refer to Session Timer on page 308.
Local	Web User Interface	Configure session timer. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc= 0

To configure session timer via web user interface:

1. Click on **Account**.
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 **Session Timer**.
5. Enter the desired time interval in the **Session Expires (30~7200s)** field.
6. Select the desired refresher from the pull-down list of **Session Refresher**.



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

Call Hold

Call hold provides a service of placing 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. IP phones support two call hold methods, one is RFC 3264, which sets the "a" (media attribute) in the SDP to sendonly, recvonly or inactive (e.g., a=sendonly). The other is RFC 2543, which sets the "c" (connection addresses for the media streams) in the SDP to zero (e.g., c=0.0.0.0). Call hold tone allows IP phones to play a hold tone at regular intervals when there is a call on hold.

Procedure

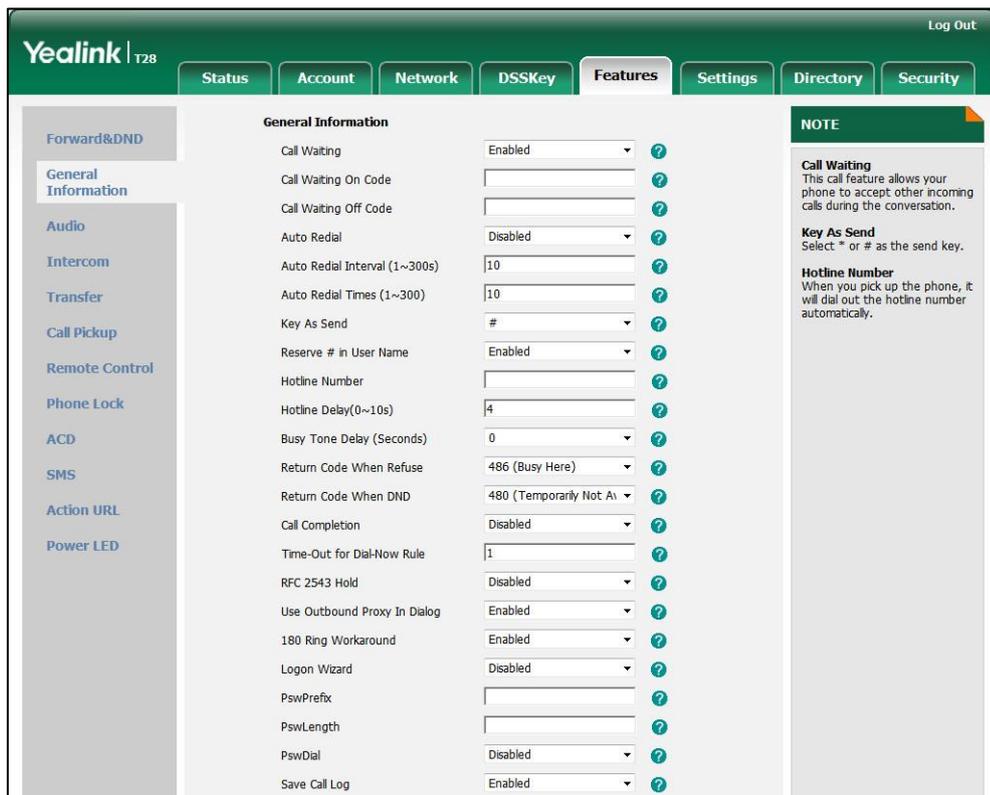
Call hold can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the call hold tone and call hold tone delay.</p> <p>Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.</p> <p>For more information, refer to Call Hold on page 309.</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the call hold tone and call hold tone delay.</p> <p>Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>

To configure call hold method via web user interface:

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

- Select the desired value from the pull-down list of **RFC 2543 Hold**.

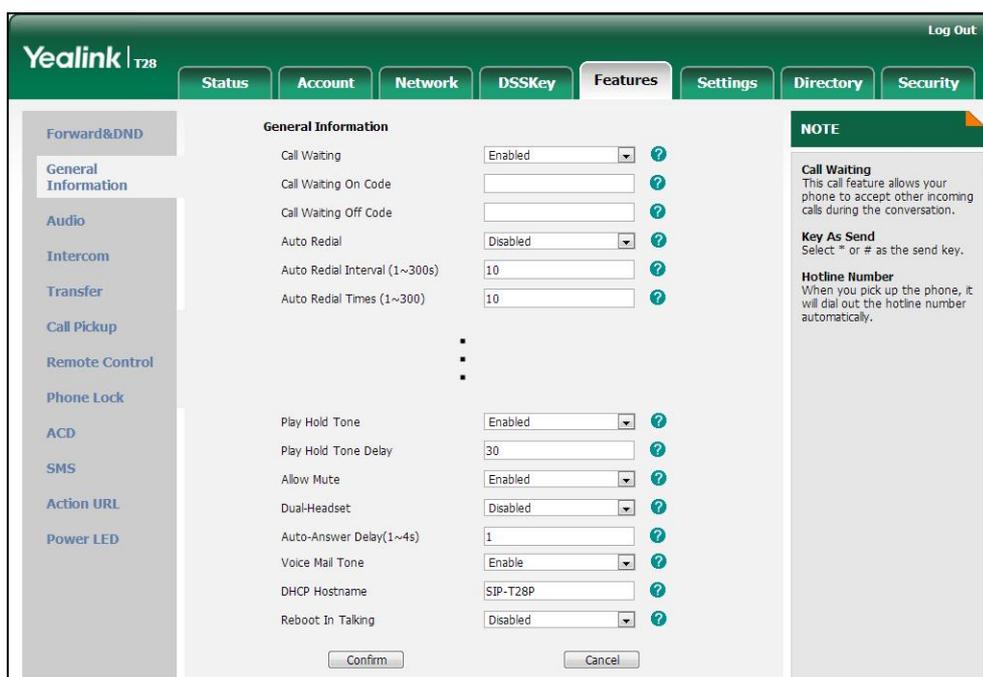


- Click **Confirm** to accept the change.

To configure call hold tone and call hold tone delay via web user interface:

- Click on **Features->General Information**.
- Select the desired value from the pull-down list of **Play Hold Tone**.

3. Enter the desired time in the **Play Hold Tone Delay** field.



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

Call Forward

Call forward allows users to redirect an incoming call to a third party. IP phones redirect an incoming INVITE message by responding with a 302 Moved Temporarily message, which contains a Contact header with a new URI that should be tried. Three types of call forward:

- **Always Forward** -- Forward the incoming call 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.

Call forward can be configured on a phone or a per-line basis depending on the call forward mode. The following describes the call forward modes:

- **Phone** (default): Call forward feature is effective for the IP phone.
- **Custom**: Call forward feature can be configured for each or all accounts.

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

Forward International

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

Procedure

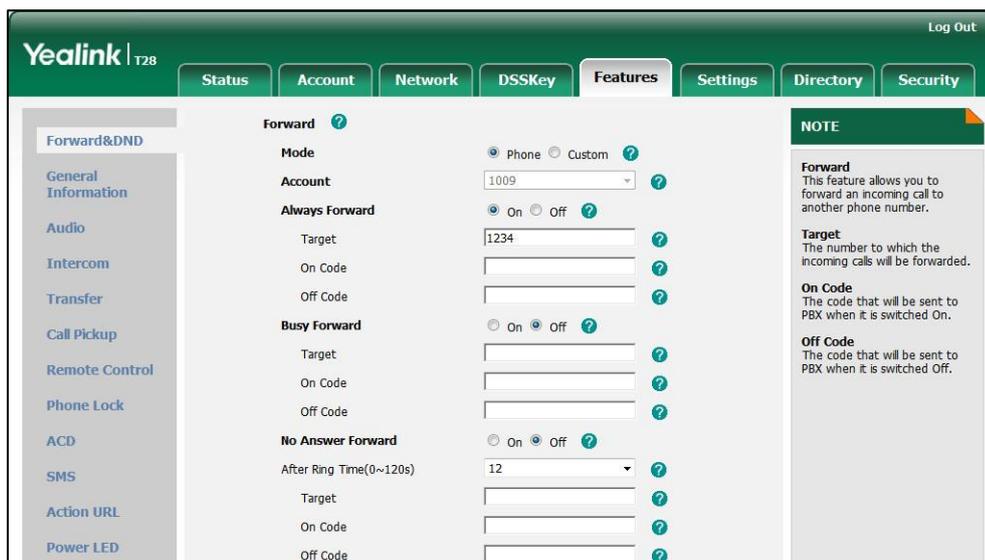
Call forward can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure call forward in custom mode. For more information, refer to Call Forward on page 311.
	<y0000000000xx>.cfg	Configure the call forward mode. Configure call forward in phone mode. Configure forward international. For more information, refer to Call Forward on page 311.
Local	Web User Interface	Configure call forward. Navigate to: http://<phoneIPAddress>/servlet?p=features-forward&q=load Configure forward international. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load
	Phone User Interface	Configure call forward.

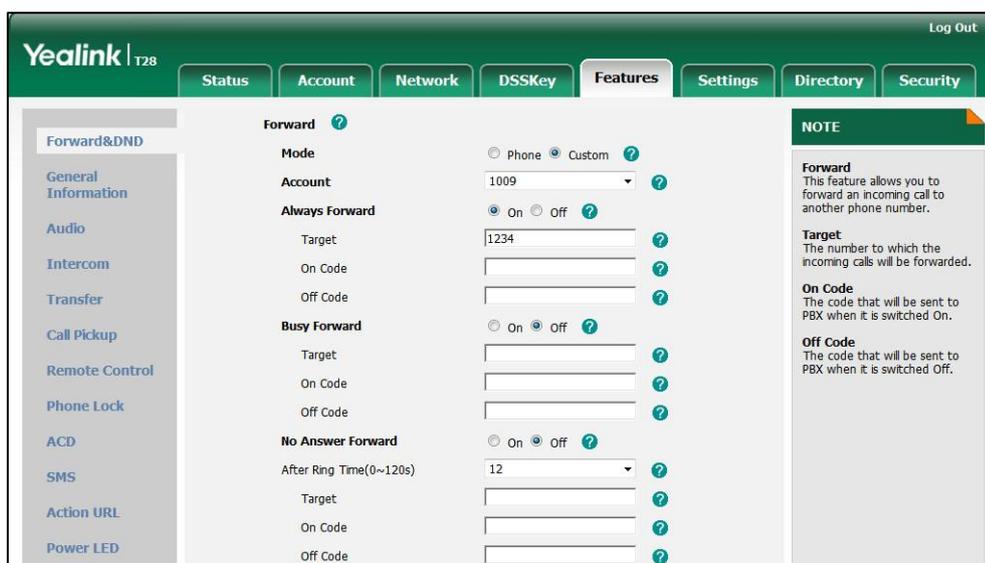
To configure call forward via web user interface:

1. Click on **Features->Forward & DND**.
2. In the **Forward** block, mark the desired radio box in the **Mode** field.
 - a) If you mark the **Phone** radio box:
 - 1) Mark the desired radio box in the **Always/Busy/No Answer Forward** field.
 - 2) Enter the destination number you want to forward in the **Target** field.
 - 3) (Optional.) Enter the on code and off code in the **On Code** and **Off Code** fields.

- 4) Select the ring time to wait before forwarding from the pull-down list of **After Ring Time (0~120s)** (only for the no answer forward).



- b) If you mark the **Custom** radio box:
 - 1) Select the desired account from the pull-down list of **Account**.
 - 2) Mark the desired radio box in the **Always/Busy/No Answer Forward** field.
 - 3) Enter the destination number you want to forward in the **Target** field.
 - 4) Enter the on code and off code in the **On Code** and **Off Code** fields.
 - 5) Select the ring time to wait before forwarding from the pull-down list of **After Ring Time (0~120s)** (only for the no answer forward).



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

To configure forward international via web user interface:

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

- Select the desired value from the pull-down list of **Fwd International**.

The screenshot shows the Yealink T28 web interface with the 'Features' tab selected. The 'General Information' section is active, displaying various settings. The 'Fwd International' setting is set to 'Enabled'. Other settings include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) set to 10, Auto Redial Times (1~300) set to 10, Diversion/History-Info (Enabled), Allow Trans Exist Call (Enabled), BLF LED Mode (0), Auto-Logout Time(1~1000min) set to 5, Voice Mail Tone (Enable), DHCP Hostname (SIP-T28P), and Reboot In Talking (Disabled). A 'NOTE' section on the right provides details for Call Waiting, Key As Send, and Hotline Number. 'Confirm' and 'Cancel' buttons are at the bottom.

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

To configure call forward in phone mode via phone user interface:

- Press **Menu->Features->Call Forward**.
- Press \uparrow or \downarrow to select the desired forwarding type, and then press the **Enter** soft key.
- Depending on your selection:
 - If you select **Always Forward**:
 - Press \leftarrow or \rightarrow , or the **Switch** soft key to select the desired value from the **Always** field.
 - Enter the destination number you want to forward all incoming calls to in the **Forward To** field.
 - (Optional.) Enter the always forward on code and off code respectively in the **On Code** and **Off Code** fields.
 - If you select **Busy Forward**:
 - Press \leftarrow or \rightarrow , or the **Switch** soft key to select the desired value from the **Busy** field.
 - Enter the destination number you want to forward all incoming calls to when the IP phone is busy in the **Forward To** field.
 - (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **Off Code** fields.
 - If you select **No Answer Forward**:
 - Press \leftarrow or \rightarrow , or the **Switch** soft key to select the desired value 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 Time** field.

The default ring time is 12 seconds.

- 4) (Optional.) Enter the no answer forward on code and off code respectively in the **On Code** and **Off Code** fields.

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

To configure call forward in custom mode via phone user interface:

1. Press **Menu->Features->Call Forward**.
2. Press  or  to select the desired account, and then press the **Enter** soft key.
3. Press  or  to select the desired forwarding type, and then press the **Enter** soft key.
4. Depending on your selection:

a) If you select **Always Forward**, you can configure it for a specific account.

- 1) Press  or  , or the **Switch** soft key to select the desired value 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 and off code respectively in the **On Code** and **Off Code** fields.

You can also configure the always forward for all accounts. After the always forward was configured for a specific account, do the following:

- 1) Press  or  to highlight the **Always** field.
- 2) Press the **All Lines** soft key.

The LCD screen prompts "Copy to All Lines?".

- 3) Press the **OK** soft key to accept the change.

b) If you select **Busy Forward**, you can configure it for a specific account.

- 1) Press  or  , or the **Switch** soft key to select the desired value from the **Busy** field.
- 2) Enter the destination number you want to forward all incoming calls to when the IP phone is busy in the **Forward To** field.
- 3) (Optional.) Enter the busy forward on code and off code respectively in the **On Code** and **Off Code** fields.

You can also configure the busy forward for all accounts. After the busy forward was configured for a specific account, do the following:

- 1) Press  or  to highlight the **Busy** field.
 - 2) Press the **All Lines** soft key.
The LCD screen prompts "Copy to All Lines?".
 - 3) Press the **OK** soft key to accept the change.
- c) If you select **No Answer Forward**, you can configure it for a specific account.
- 1) Press  or , or the **Switch** soft key to select the desired value 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 Time** field
The default ring time is 12 seconds.
 - 4) (Optional.) Enter the no answer forward on code and off code respectively in the **On Code** and **Off Code** fields.

You can also configure the no answer forward for all accounts. After the no answer forward was configured for a specific account, do the following:

- 1) Press  or  to highlight the **No Answer** field.
 - 2) Press the **All Lines** soft key.
The LCD screen prompts "Copy to All Lines?".
 - 3) Press the **OK** soft key to accept the change.
5. Press the **Save** soft key to accept the change.

Call Transfer

Call transfer enables IP phones to transfer an existing call to another party. IP phones support call transfer using the REFER method specified in RFC 3515 and offer three types of transfer:

- **Blind Transfer** -- Transfer a call directly to another party without consulting. Blind transfer is implemented by a simple REFER method without Replaces in the Refer-To header.
- **Semi-attended Transfer** -- Transfer a call after hearing the ringback tone. Semi-attended transfer is implemented by a REFER method with Replaces in the Refer-To header.
- **Attended Transfer** -- Transfer a call with prior consulting. Attended transfer is implemented by a REFER method with Replaces in the Refer-To header.

Normally, call transfer is completed by pressing the transfer key. Blind transfer on hook and semi-attended transfer on hook features allow the IP phone to complete the

transfer through on-hook.

When a user performs a semi-attended transfer, semi-attended transfer feature determines whether to display the prompt "**n New Missed Call(s)**" ("n" indicates the number of the missed calls) on the destination party's phone LCD screen.

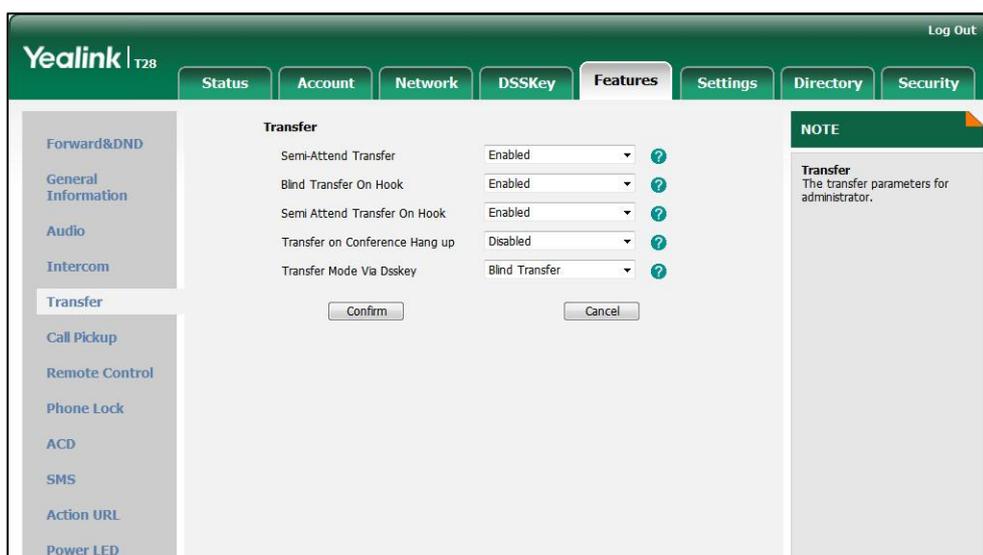
Procedure

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

Configuration File	<y0000000000xx>.cfg	Specify whether to complete the transfer through on-hook. Configure semi-attended transfer feature. For more information, refer to Call Transfer on page 320.
Local	Web User Interface	Specify whether to complete the transfer through on-hook. Configure semi-attended transfer feature. Navigate to: http://<phoneIPAddress>/servlet?p=features-transfer&q=load

To configure call transfer via web user interface:

1. Click on **Features->Transfer**.
2. Select the desired values from the pull-down lists of **Semi-Attend Transfer**, **Blind Transfer On Hook** and **Semi Attend 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). IP phones implement network conference using the REFER method specified in RFC 4579. This feature depends on support from a SIP server.

Procedure

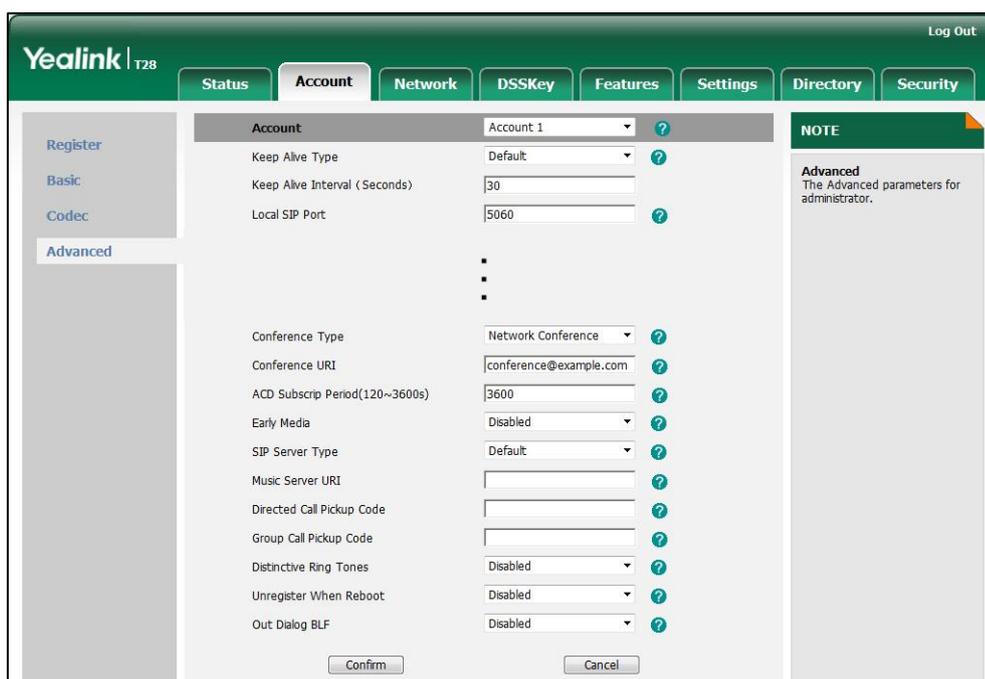
Network conference can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure network conference. For more information, refer to Network Conference on page 321.
Local	Web User Interface	Configure network conference. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc= 0

To configure the network conference via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select **Network Conference** 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 drop the call when the conference initiator drops the conference call. Transfer on conference hang up allows the other two parties to remain connected when the conference initiator drops the conference call.

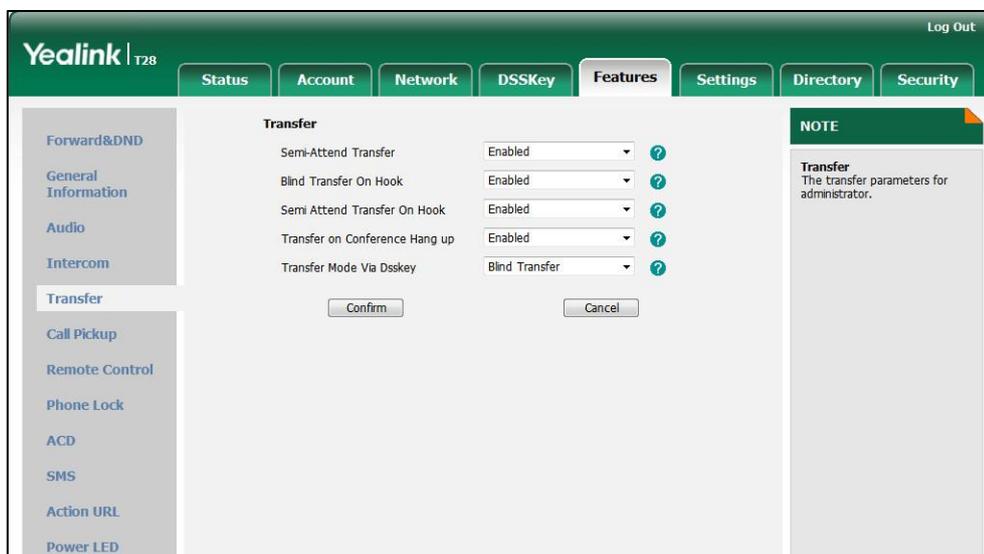
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the transfer on conference hang up. For more information, refer to Transfer on Conference Hang Up on page 322.
Local	Web User Interface	Configure the transfer on conference hang up. Navigate to: http://<phoneIPAddress>/servlet?p=features-transfer&q=load

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

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



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

Directed Call Pickup

Directed call pickup is used for picking up an incoming call on a specific extension. A user can pick up the incoming call using a directed pickup key or the DPickup soft key (not applicable to SIP-T20P IP phones). This feature depends on support from a SIP server. For many SIP servers, directed call pickup requires a directed pickup code, which can be configured on a phone or a per-line basis.

Note It is recommended not to configure the directed call pickup key and the DPickup soft key simultaneously. If you do, the directed call pickup key will not be used correctly.

Procedure

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

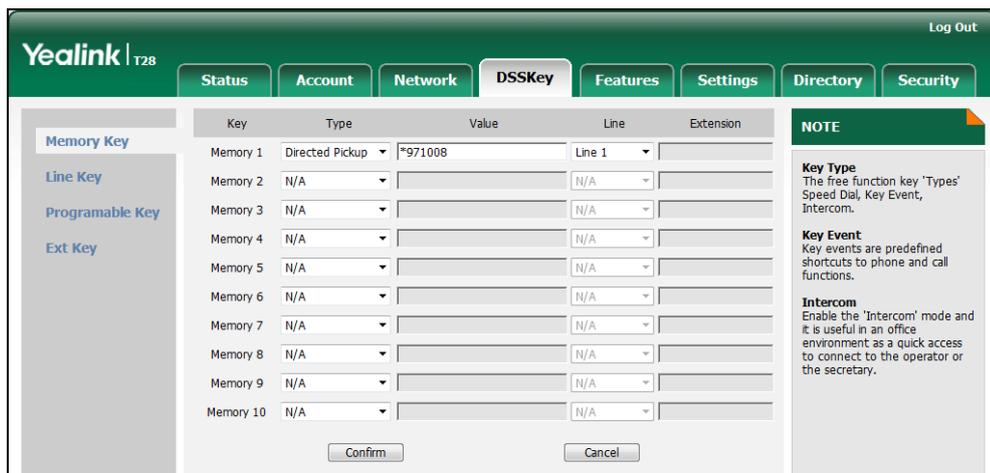
<p>Configuration File</p>	<p><MAC>.cfg</p>	<p>Configure the directed call pickup code on a per-line basis.</p> <p>Configure directed call pickup feature on a phone basis.</p> <p>For more information, refer to Directed Call Pickup on page 323.</p>
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	<y0000000000xx>.cfg	<p>Assign a directed call pickup key.</p> <p>For more information, refer to Directed Call Pickup Key on page 409.</p>
Local	Web User Interface	<p>Assign a directed call pickup key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0">http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</p> <p>Configure directed call pickup feature on a phone basis.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-callpickup&q=load">http://<phoneIPAddress>/servlet?p=features-callpickup&q=load</p> <p>Configure directed call pickup code on a per-line basis.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>
	Phone User Interface	<p>Assign a directed call pickup key.</p>

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

1. Click on **DSSKey->Memory Key (Line Key or Programmable Key)**.
 SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.
2. In the desired DSS key field, select **Directed Pickup** from the pull-down list of **Type**.
3. Enter the directed call pickup code followed by the specific extension in the **Value** field.

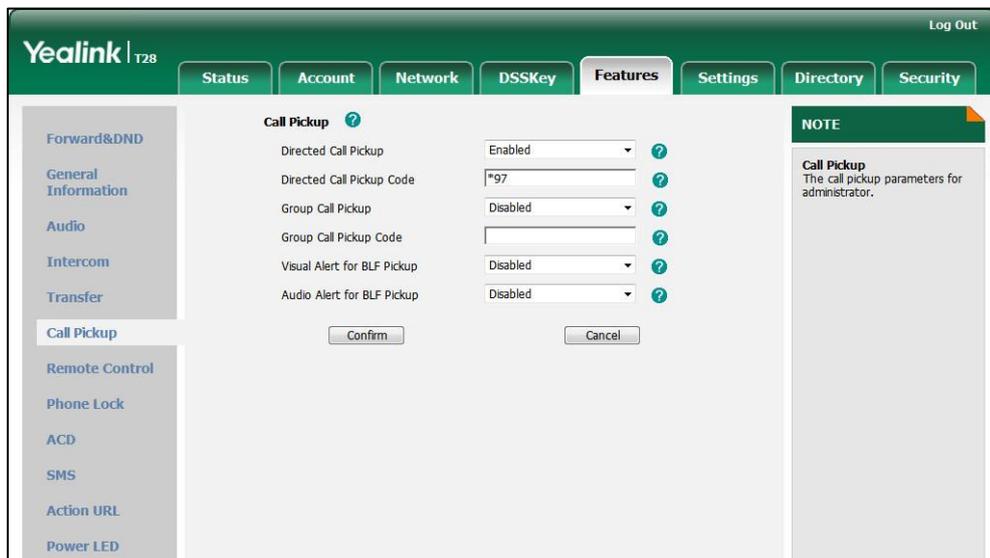
4. Select the desired line from the pull-down list of **Line**.



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

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

1. Click on **Features->Call Pickup**.
2. Select the desired value from the pull-down list of **Directed Call Pickup** (not applicable to SIP-T20P IP phones).
3. Enter the directed call pickup code in the **Directed Call Pickup Code** field.

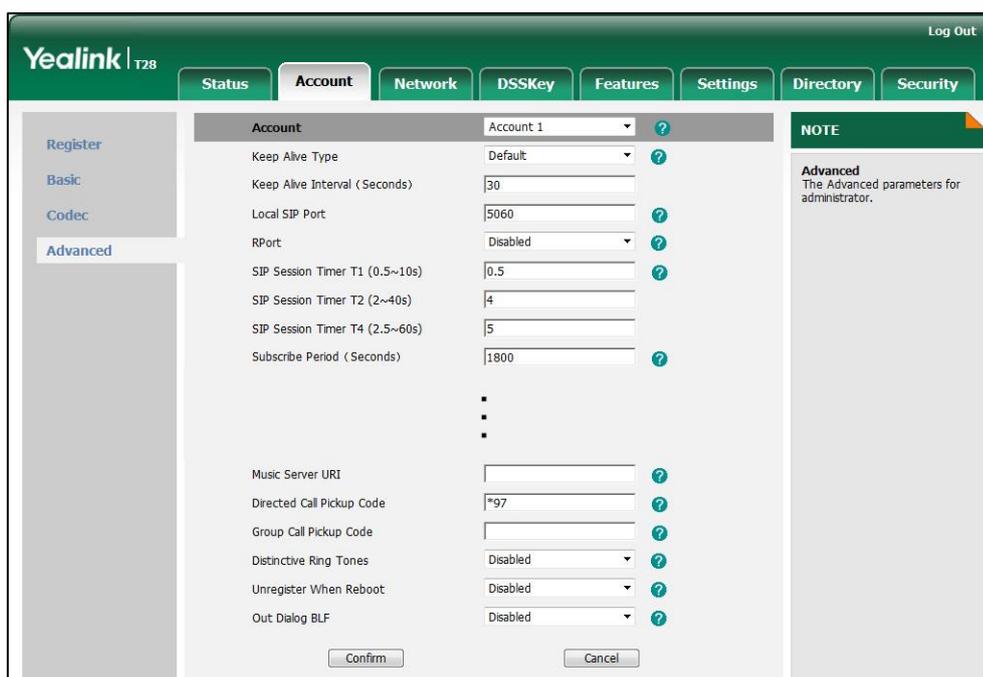


4. Click **Confirm** to accept the change.

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

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

4. Enter the directed call pickup code in the **Directed Call Pickup Code** field.



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

To configure a directed 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 **Directed 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 directed call pickup code followed by the specific extension in the **Value** field.
7. Press the **Save** soft key to accept the change.

Group Call Pickup

Group call pickup is used for picking up incoming calls within a pre-defined group. If the group receives many incoming calls at once, the user will pick up the first incoming call, using a group pickup key or the GPickup soft key (not applicable to SIP-T20P IP phones). This feature depends on support from a SIP server. For many SIP servers, group call pickup requires a group pickup code, which can be configured on a phone or a per-line basis.

Procedure

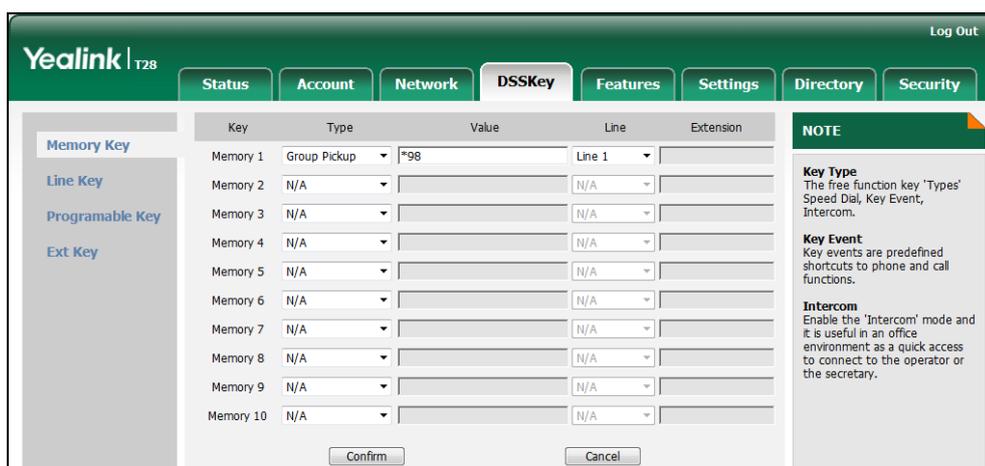
Group call pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the group call pickup code on a per-line basis. Configure group call pickup feature on a phone basis. For more information, refer to Group Call Pickup on page 324.
	<y000000000xx>.cfg	Assign a group call pickup key. For more information, refer to Group Call Pickup Key on page 411.
Local	Web User Interface	Assign a group call pickup key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Configure group call pickup feature on a phone basis. Navigate to: http://<phoneIPAddress>/servlet?p=features-callpickup&q=load Configure the group call pickup code on a per-line basis. Navigate to: http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0
	Phone User Interface	Assign a group call pickup key.

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

- Click on **DSSKey->Memory Key (Line Key or Programable Key)**.
SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.
- In the desired DSS key field, select **Group Pickup** from the pull-down list of **Type**.
- Enter the group call pickup code in the **Value** field.

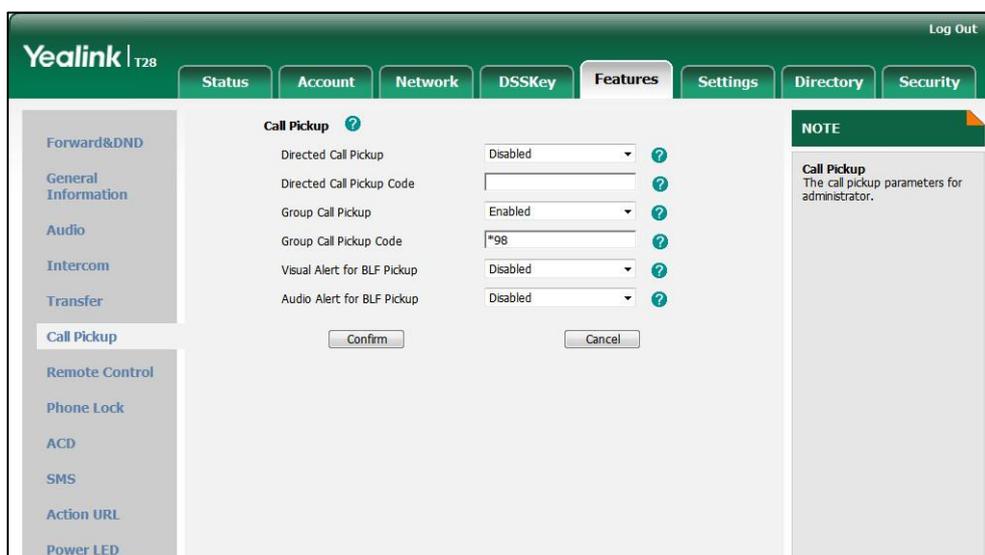
4. Select the desired line from the pull-down list of **Line**.



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

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

1. Click on **Features->Call Pickup**.
2. Select the desired value from the pull-down list of **Group Call Pickup** (not applicable to SIP-T20P IP phones).
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**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.

- Enter the group call pickup code in the **Group Call Pickup Code** field.

The screenshot shows the Yealink T28 web interface with the 'Account' tab selected. The 'Account' configuration page is displayed for 'Account 1'. The 'Group Call Pickup Code' field is set to '*98'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5060), 'RPort' (Disabled), 'SIP Session Timer T1 (0.5~10s)' (0.5), 'SIP Session Timer T2 (2~40s)' (4), 'SIP Session Timer T4 (2.5~60s)' (5), and 'Subscribe Period (Seconds)' (1800). There are also fields for 'Music Server URI', 'Directed Call Pickup Code', 'Distinctive Ring Tones' (Disabled), 'Unregister When Reboot' (Disabled), and 'Out Dialog BLF' (Disabled). A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator.' The interface includes a 'Confirm' button and a 'Cancel' button at the bottom.

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

To configure a group pickup key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys (or Line Keys)**.
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **Group Pickup** from the **Key Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired line from the **Account ID** field.
- Enter the group call pickup code in the **Value** field.
- Press the **Save** soft key to accept the change.

Dialog Info Call Pickup

Call pickup is implemented through SIP signals on some specific servers. 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 the DSS key used to monitor a specific extension (such as the BLF key). This feature is not applicable to SIP-T19P IP phones.

Example of the dialog-info message carried in NOTIFY message:

```
<?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

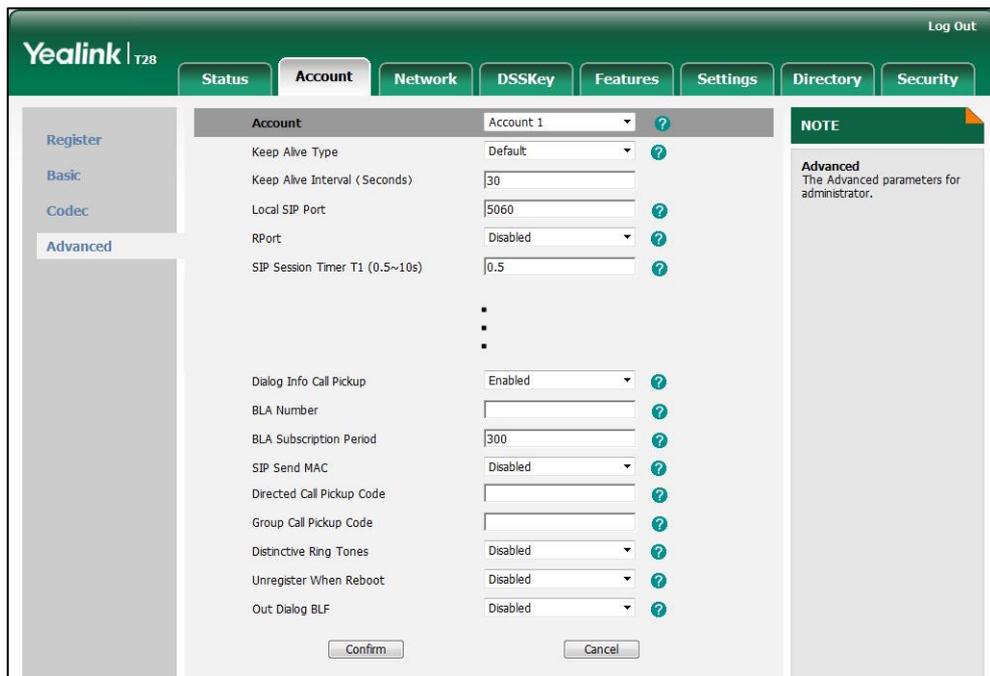
Dialog info call pickup can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure dialog info call pickup. For more information, refer to Dialog Info Call Pickup on page 325.
Local	Web User Interface	Configure dialog info call pickup. Navigate to: http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

To configure dialog info call pickup via web user interface:

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

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



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

Call Return

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

Procedure

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

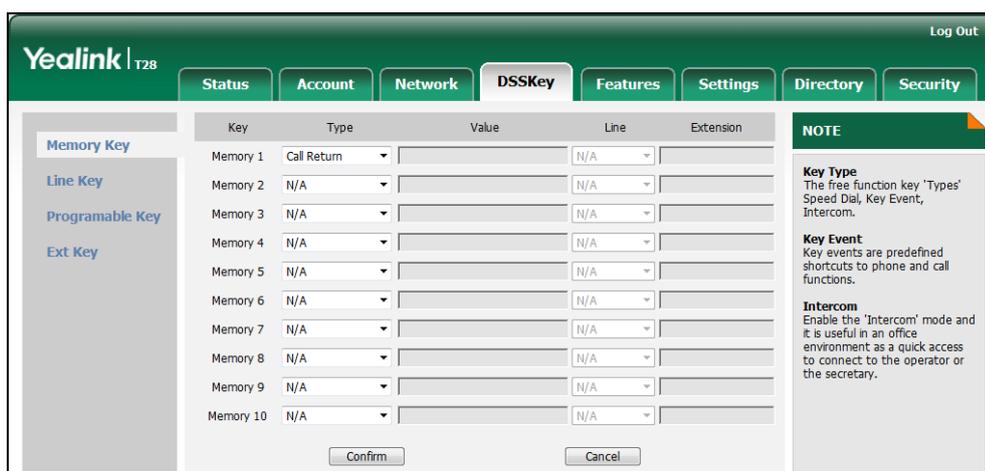
Configuration File	<y0000000000xx>.cfg	Assign a call return key. For more information, refer to Call Return Key on page 413.
Local	Web User Interface	Assign a call return key. Navigate to: http://<phoneIPAddress>/servlet ?p=dsskey&q=load&model=0
	Phone User Interface	Assign a call return key.

To configure a call return key via web user interface:

- Click on **DSSKey->Memory Key (Line Key or Programable Key)**.

SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.

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



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

To configure a call return key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **Call Return** from the **Key Type** field.
- Press the **Save** soft key to accept the change.

Call Park

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

Note

SIP-T19P IP phones support call park feature for BroadSoft server only. For more information, refer to *Yealink IP Phones Deployment Guide for BroadSoft UC-One Environments*.

Procedure

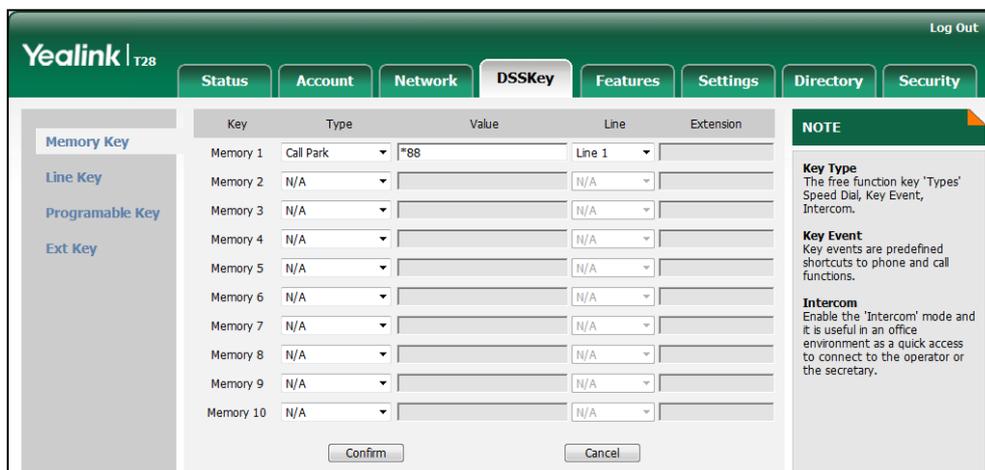
Call park key can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Assign a call park key. For more information, refer to Call Park Key on page 414.
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Local	Web User Interface	Assign a call park key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0
	Phone User Interface	Assign a call park key.

To configure a call park key via web user interface:

1. Click on **DSSKey->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 **Line**.



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

Web server type determines access protocol of the IP phone’s web user interface. IP phones support both HTTP and HTTPS protocols for accessing the web user interface.

HTTP is an application protocol that runs on top of the TCP/IP suite of protocols. HTTPS is a web protocol that encrypts and decrypts user page requests as well as pages returned by the web server. Both HTTP and HTTPS port numbers are configurable.

Procedure

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

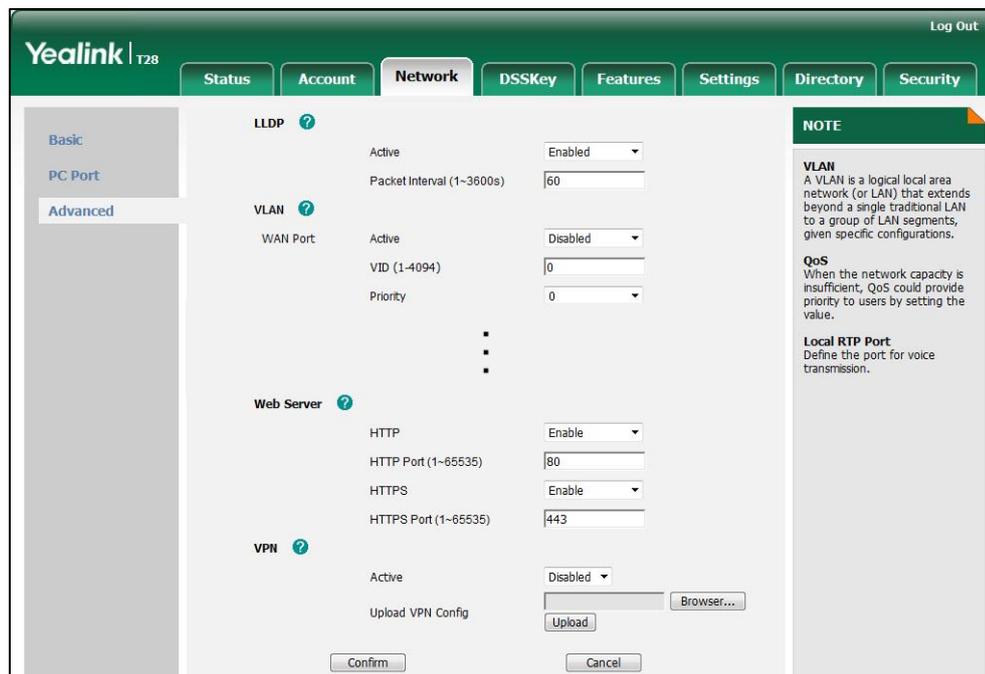
Configuration File	<y0000000000xx>.cfg	Configure the web access type, HTTP port and HTTPS port. For more information, refer to Web Server Type on page 326.
Local	Web User Interface	Configure the web access type, HTTP port and HTTPS port. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure the web access type, HTTP port and HTTPS port.

To configure web server type via web user interface:

1. Click on **Network->Advanced**.
2. Select the desired value from the pull-down list of **HTTP**.
3. Enter the HTTP port number in the **HTTP Port (1~65535)** field.
The default HTTP port number is 80.
4. Select the desired value from the pull-down list of **HTTPS**.

- Enter the HTTPS port number in the **HTTPS Port (1~65535)** field.

The default HTTPS port number is 443.



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

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

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

To configure web server type via phone user interface:

- Press **Menu->Settings->Advanced Settings** (password: admin)
->**Network->Webserver Type**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **HTTP Status** field.
- Enter the HTTP port number in the **HTTP Port** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **HTTPS Status** field.
- Enter the HTTPS port number in the **HTTPS Port** field.
- Press the **Save** soft key to accept the change.

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

Calling Line Identification Presentation

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

identity in the relevant SIP header.

If the caller has existed in the local directory, the local name assigned to the caller should be preferentially displayed.

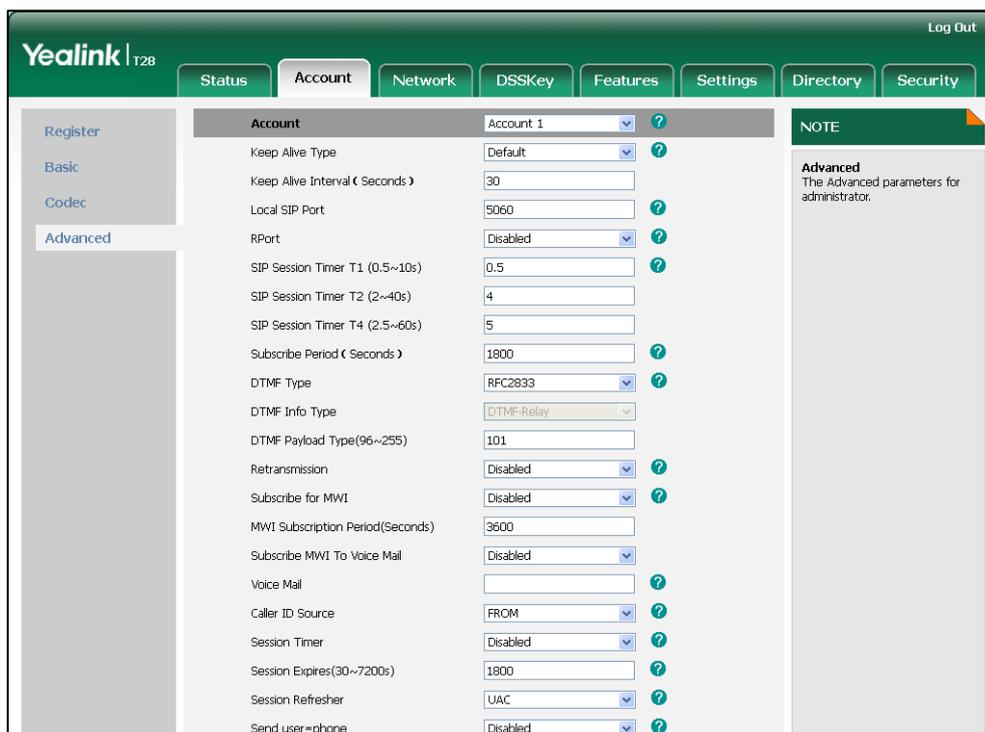
Procedure

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 327.
Local	Web User Interface	Configure the presentation of the caller identity. Navigate to: http://<phoneIPAddress>/servlet?account-adv&q=load&acc=0

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

1. Click on **Account**.
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 Source**.



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

Connected Line Identification Presentation

Connected line identification presentation (COLP) allows IP phones to display the identity of the callee specified for outgoing calls. IP phones can display the Dialed Digits, or the identity in a SIP header (Remote-Party-ID or P-Asserted-Identity) received, or the identity in the From header carried in the UPDATE message sent by the callee as described in RFC 4916.

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's identity. For more information, refer to Connected Line Identification Presentation on page 328.
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DTMF

DTMF (Dual Tone Multi-frequency), better known as touch-tone, is used for telecommunication signaling over analog telephone lines in the voice-frequency band. DTMF is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call. Each key pressed on the IP phone generates one sinusoidal tone of two frequencies. One is generated from a high frequency group and the other from a low frequency group.

The DTMF keypad is laid out in a 4×4 matrix, with each row representing a low frequency, and each column representing a high frequency. Pressing a digit key (such as '1') will generate a sinusoidal tone for each of two frequencies (697 and 1209 hertz (Hz)).

DTMF Keypad Frequencies:

	1209 Hz	1336 Hz	1447 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Three methods of transmitting DTMF digits on SIP calls:

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

The method of transmitting DTMF digits is configurable on a per-line 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 RTP Event packets is configurable. IP phones default to 101 for the payload type, which use the definition to negotiate with the other end during call establishment.

The RTP Event packet contains 4 bytes. The 4 bytes are distributed over several fields denoted as Event, End bit, R-bit, Volume and Duration. If the End bit is set to 1, the packet contains the end of the DTMF event. You can configure the sending times of the end RTP Event packet.

INBAND

DTMF digits are transmitted within the audio of the IP phone conversation. It uses the same codec as your voice and is audible to conversation partners.

SIP INFO

DTMF digits are transmitted by the SIP INFO messages when the voice stream is established after a successful SIP 200 OK-ACK message sequence. The SIP INFO message is sent along the signaling path of the call. The SIP INFO message can support transmitting DTMF digits in three ways: DTMF, DTMF-Relay and Telephone-Event.

Procedure

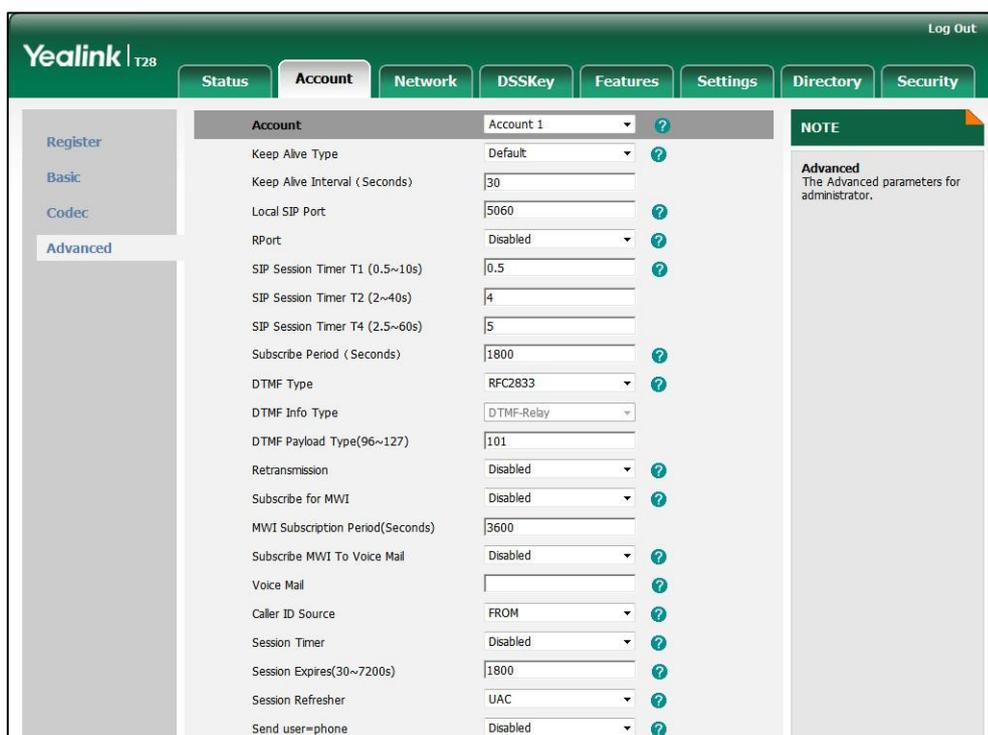
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the method of transmitting DTMF digit and the payload type. For more information, refer to DTMF on page 329 .
	<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 329 .

<p>Local</p>	<p>Web User Interface</p>	<p>Configure the method of transmitting DTMF digits and the payload type.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p> <p>Configure the number of times for the IP phone to send the end RTP Event packet.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
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To configure the method of transmitting DTMF digits via web user interface:

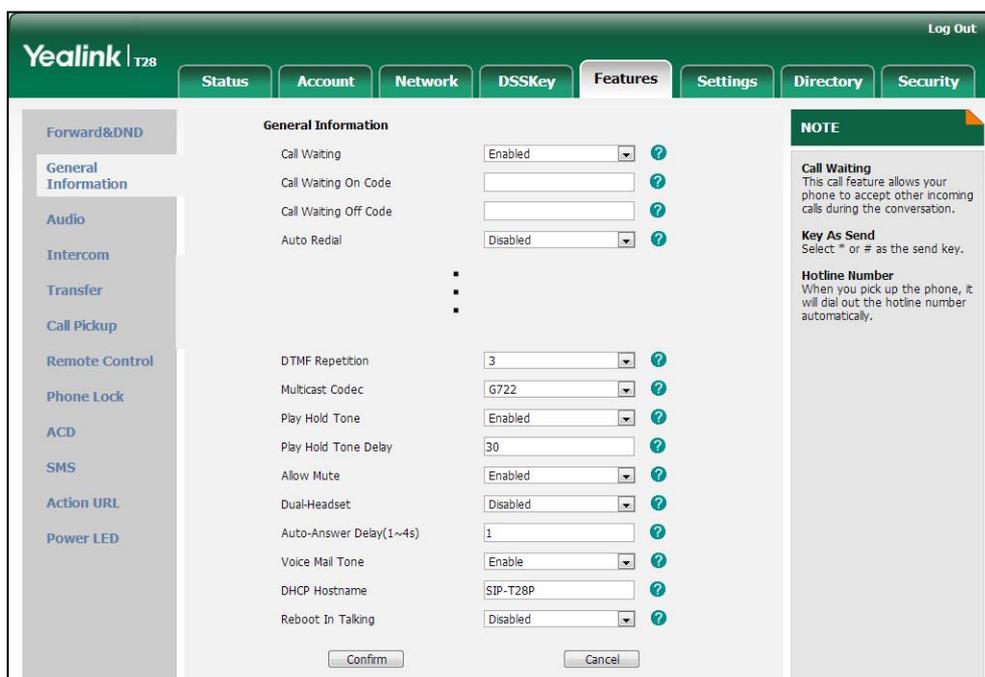
1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **DTMF Type**.
5. If **SIP INFO** or **AUTO** or **SIP INFO** is selected, select the desired value from the pull-down list of **DTMF Info Type**.
6. Enter the desired value in the **DTMF Payload Type (96~127)** field.



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

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

- Click on **Features->General Information**.
- Select the desired value (1-3) from the pull-down list of **DTMF Repetition**.



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

Suppress DTMF Display

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

Procedure

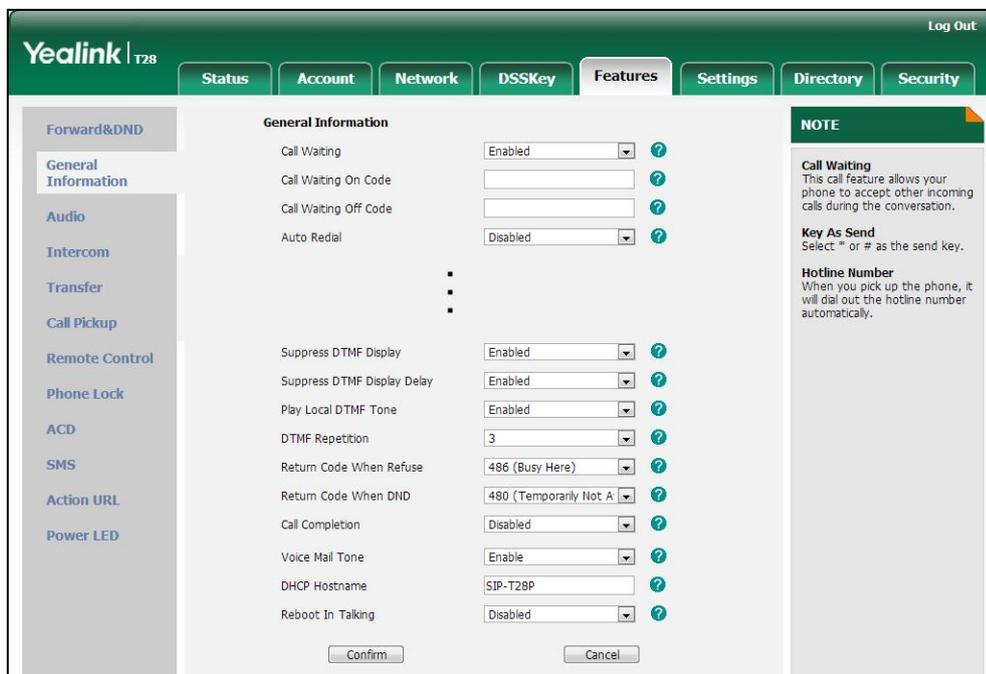
Configuration changes can be performed using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure suppress DTMF display and suppress DTMF display delay. For more information, refer to Suppress DTMF Display on page 331 .
Local	Web User Interface	Configure suppress DTMF display and suppress DTMF

		display delay. Navigate to: http://<phoneIPAddress>/servlet?<phoneIPAddress>/servlet?p=features-general&q=load
--	--	---

To configure suppress DTMF display and suppress DTMF display delay via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **Suppress DTMF Display**.
3. Select the desired value from the pull-down list of **Suppress DTMF Display Delay** (not applicable to SIP-T20P IP phones).



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

Transfer via DTMF

Call transfer is implemented via DTMF on some traditional servers. The IP phone sends specified DTMF digits to the server for transferring calls to third parties.

Procedure

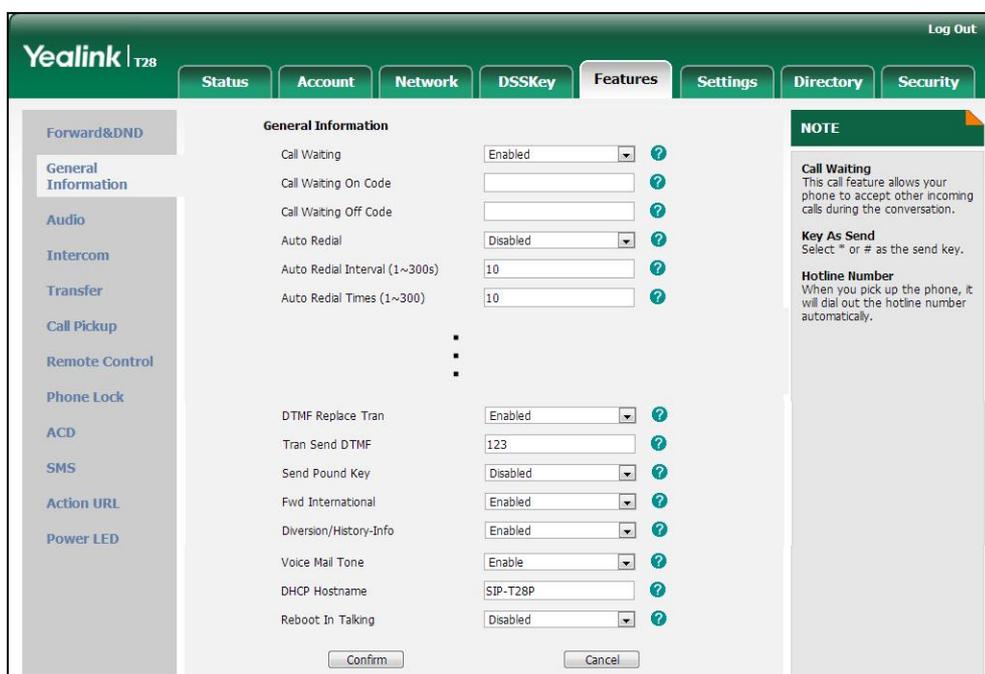
Configuration changes can be performed using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure transfer via DTMF. For more information, refer to Transfer via DTMF on page 331.
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Local	Web User Interface	Configure transfer via DTMF. Navigate to: http://<phoneIPAddress>/servlet? p=features-general&q=load
-------	--------------------	---

To configure transfer via DTMF via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **DTMF Replace Tran**.
3. Enter the specified DTMF digits in the **Tran Send DTMF** field.



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

Intercom

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

Outgoing Intercom Calls

Intercom is a useful feature in office environments to quickly connect with an operator or secretary. Users can press an intercom key to automatically initiate an outgoing intercom call with a remote extension. This feature is not applicable to SIPT19P IP phones.

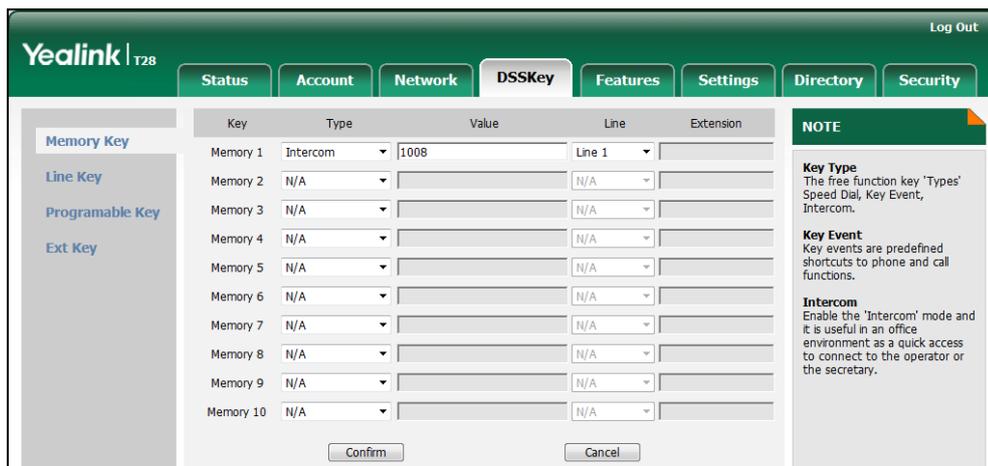
Procedure

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 415.
Local	Web User Interface	Assign an intercom key. Navigate to: http://<phoneIPAddress>/servlet ?p=dsskey&q=load&model=0
	Phone User Interface	Assign an intercom key.

To configure an intercom key via web user interface:

1. Click on **DSSKey->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 **Line**.



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 IP phone can process incoming calls differently depending on settings. There are four configuration options 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.

Intercom Tone

Intercom 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 an active call is in progress. The active call will be placed on hold.

Procedure

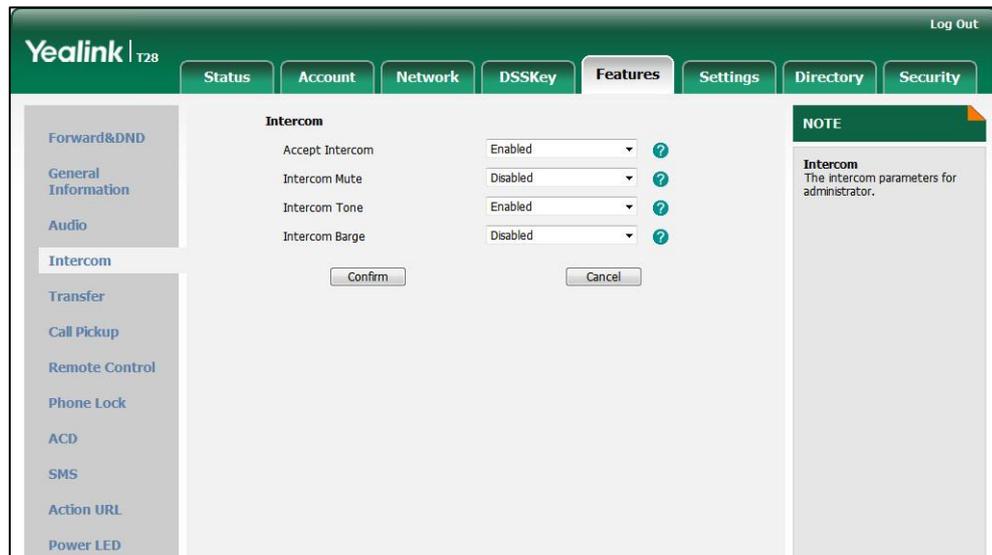
Incoming intercom calls can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure incoming intercom call feature. For more information, refer to Incoming Intercom calls on page 332.
Local	Web User Interface	Configure incoming intercom call feature. Navigate to: http://<phoneIPAddress>/servlet?p=features-intercom&q=load
	Phone User Interface	Configure incoming intercom call feature.

To configure intercom via web user interface:

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

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



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

To configure intercom via phone user interface:

- Press **Menu->Features->Intercom**.
- Press **◀** or **▶** , or the **Switch** soft key to select the desired values from the **Accept Intercom**, **Intercom Mute**, **Intercom Tone** and **Intercom Barge** fields.
- 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 Phone Book](#)
- [LDAP](#)
- [Busy Lamp Field](#)
- [Music on Hold](#)
- [Automatic Call Distribution](#)
- [Message Waiting Indicator](#)
- [Multicast Paging](#)
- [Call Recording](#)
- [Hot Desking](#)
- [Action URL](#)
- [Action URI](#)
- [Server Redundancy](#)
- [LLDP](#)
- [VLAN](#)
- [VPN](#)
- [Quality of Service](#)
- [Network Address Translation](#)
- [802.1X Authentication](#)
- [TR-069 Device Management](#)
- [IPv6 Support](#)

Distinctive Ring Tones

Distinctive ring tones allows certain incoming calls to trigger IP phones to play distinctive ring tones. The IP phone inspects the INVITE request for an "Alert-Info" header when receiving an incoming call. If the INVITE request contains an "Alert-Info" header, the IP phone strips out the URL and keyword parameter and maps them to the appropriate ring tone.

Alert-Info headers in the following two formats:

Alert-Info: http://localIP/Bellcore-drN

Alert-Info: <URL>;info=info text;x-line-id=0

- If the Alter-Info header contains the keyword "Bellcore-drN", the IP phone will play the Bellcore-drN ring tone (N=1, 2, 3, 4 or 5) (if the parameter "features.alert_info_tone" is set to 1).

Example:

Alert-Info: http://127.0.0.1/Bellcore-dr1

The following table identifies the different Bellcore ring tone patterns and cadences (These ring tones are designed for the BroadWorks server).

Bellcore Tone	Pattern ID	Pattern	Cadence	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
Bellcore-dr1 (standard)	1	Ringing	2s On	1800	2000	2200
		Silent	4s Off	3600	4000	4400
Bellcore-dr2	2	Ringing	Long	630	800	1025
		Silent		315	400	525
		Ringing	Long	630	800	1025
		Silent		3475	4000	4400
Bellcore-dr3	3	Ringing	Short	315	400	525
		Silent		145	200	525
		Ringing	Short	315	400	525
		Silent		145	200	525
		Ringing	Long	630	800	1025
		Silent		2975	4000	4400
Bellcore-dr4	4	Ringing	Short	200	300	525
		Silent		145	200	525
		Ringing	Long	800	1000	1100
		Silent		145	200	525
		Ringing	Short	200	300	525
		Silent		2975	4000	4400
Bellcore-dr5	5	Ringing		450	500	550

Note

"Bellcore-dr5" is a ring splash tone that reminds the user that the DND or Always Call Forward feature is enabled on the server side.

- If the Alert-Info header contains a remote URL, the IP phone will try to download the WAV ring tone file from the URL and then play the remote ring tone (if the parameter "account.X.alert_info_url_enable" is set to 1). If it fails to download the file, the IP phone will play the local ring tone associated with **info text**. If there is no text matched, the IP phone will play the preconfigured local ring tone in about ten seconds.

Example:

Alert-Info: http://192.168.0.12:8080/ring.wav/info=family;x-line-id=0

Procedure

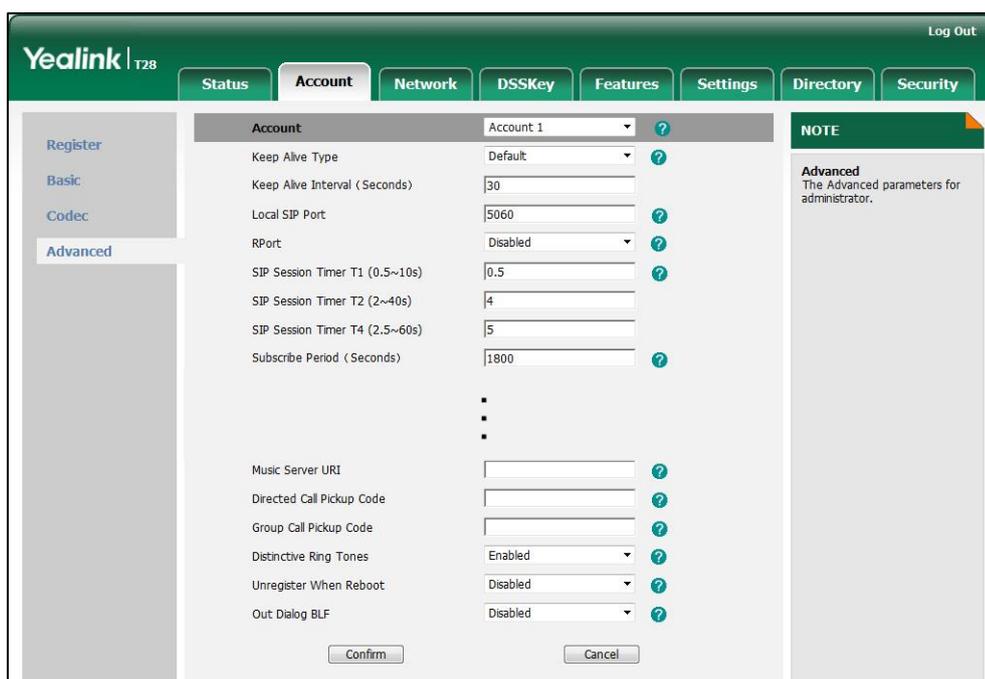
Distinctive ring tones can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure distinctive ring tones. For more information, refer to Distinctive Ring Tones on page 334.
	<y0000000000xx>.cfg	Configure the internal ringer text and internal ringer file. For more information, refer to Distinctive Ring Tones on page 334.
Local	Web User Interface	Configure distinctive ring tones. Navigate to: http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0 Configure the internal ringer text and internal ringer file. Navigate to: http://<phoneIPAddress>/servlet?p=settings-ring&q=load

To configure distinctive ring tones via web user interface:

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

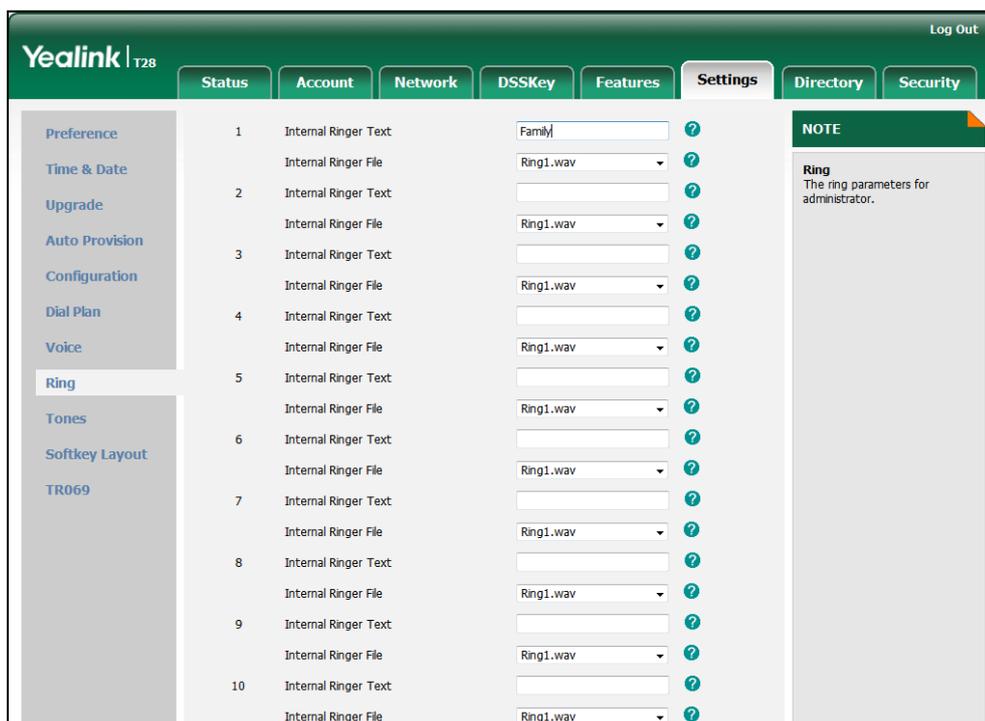
4. Select the desired value from the pull-down list of **Distinctive Ring Tones**.



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

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

1. Click on **Settings->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, the IP phone will play a warning tone. You can customize tones or select specialized tone sets (vary from country to country) to indicate different conditions of the IP phone. The default tones used on IP phones are the US tone sets.

Available tone sets for IP phones:

- Australia
- Austria
- Brazil
- Belgium
- China
- Czech
- Denmark
- Finland
- France
- Germany
- Great Britain
- Greece
- Hungary
- Lithuania
- India
- Italy
- Japan
- Mexico
- New Zealand
- Netherlands
- Norway
- Portugal
- Spain
- Switzerland
- Sweden
- Russia
- United States

- Chile
- Czech ETSI

Configured tones can be heard on IP phones for the following conditions.

Condition	Description
Dial	When in the pre-dialing interface
Ring Back	Ring-back tone
Busy	When the callee is busy
Congestion	When the network is congested
Call Waiting	Call waiting tone
Dial Recall	When receiving a call back
Info	When receiving a special message
Stutter	When receiving a voice mail
Message	When receiving a text message Note: It is not applicable to SIP-T20P IP phones.
Auto Answer	When automatically answering a call

Procedure

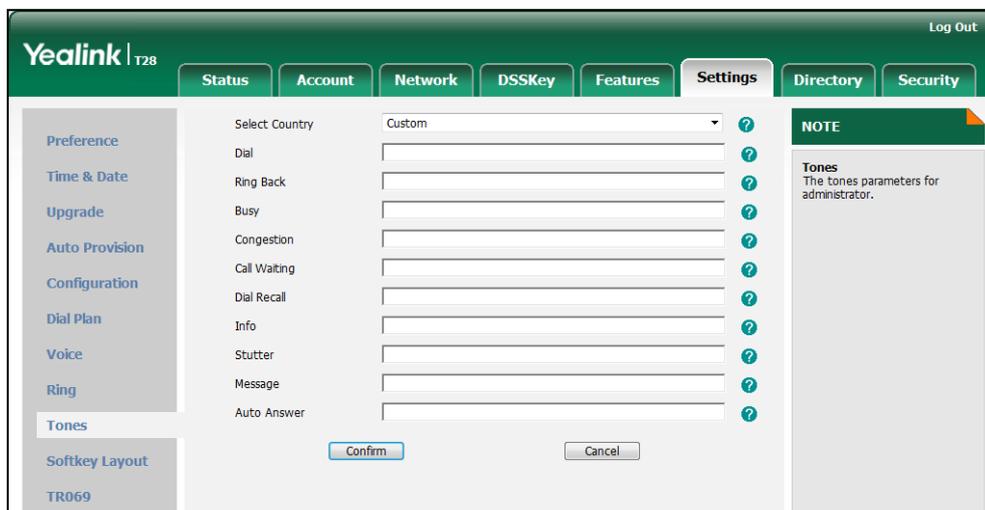
Tones can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the tones for the IP phone. For more information, refer to Tones on page 336.
Local	Web User Interface	Configure the tones for the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=settings-tones&q=load

To configure tones via web user interface:

1. Click on **Settings->Tones**.

- Select the desired type from the pull-down list of **Select Country**.
If you select **Custom**, you can customize a tone for each condition of the IP phone.



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

Remote Phone Book

Remote phone book is a centrally maintained phone book, stored on the remote server. Users only need the access URL of the remote phone book. The IP phone can establish a connection with the remote server and download the phone book, and then display the remote phone book entries on the phone user interface. IP phones support up to 5 remote phone books. SIP-T28/T26P/T22P IP phones support up to 2500 remote phone book entries. SIP-T21P/T19P IP phones support up to 2000 remote phone book entries. Remote phone book is customizable. For more information, refer to [Remote XML Phone Book](#) on page 234.

Search Remote Phonebook Name allows IP phones to search the entry names from the remote phone book when receiving incoming calls. Search Flash Time defines how often IP phones refresh the local cache of the remote phone book.

Note Remote phone book is not applicable to SIP-T20P IP phones.

Procedure

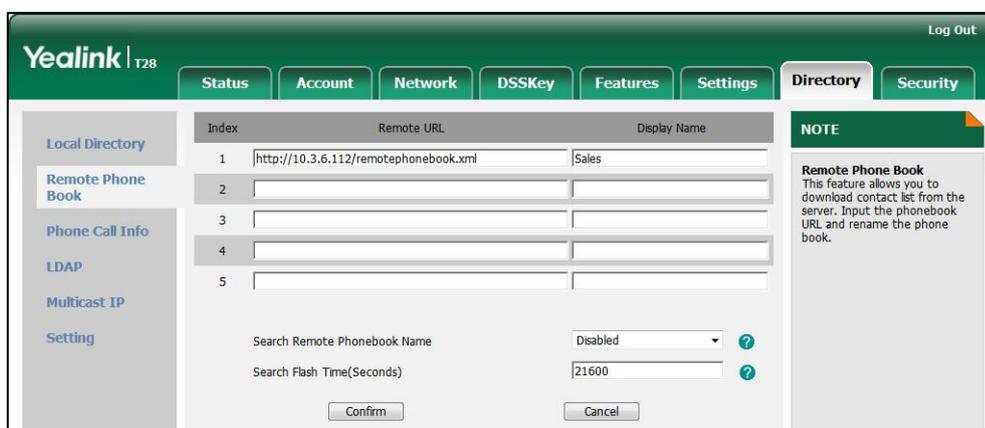
Remote phone book can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Specify the access URL of the remote phone book.</p> <p>Specify whether to query the entry name from the remote phone book when the IP phone</p>
----------------------------------	----------------------------------	---

		<p>receives an incoming call.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p>For more information, refer to Remote Phone Book on page 338.</p>
Local	Web User Interface	<p>Specify the access URL of the remote phone book.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=contacts-remote&q=load">http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p> <p>Specify whether to query the entry name from the remote phone book when the IP phone receives an incoming call.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=contacts-remote&q=load">http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p>

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

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



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

To configure Search Remote Phonebook Name and Search Flash Time via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Select the desired value from the pull-down list of **Search Remote Phonebook Name**.
3. Enter the desired time in the **Search Flash Time (Seconds)** field.

The screenshot shows the Yealink T28 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Directory' tab is active. On the left sidebar, 'Remote Phone Book' is selected. The main content area displays a table with columns 'Index', 'Remote URL', and 'Display Name'. The first row shows '1', 'http://10.3.6.112/remotephonebook.xml', and 'Sales'. Below the table, there are two configuration fields: 'Search Remote Phonebook Name' with a dropdown menu set to 'Enabled', and 'Search Flash Time(Seconds)' with an input field containing '21600'. A 'Confirm' button is located at the bottom center. A 'NOTE' box on the right states: 'Remote Phone Book: This feature allows you to download contact list from the server. Input the phonebook URL and rename the phone book.'

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

LDAP

LDAP (Lightweight Directory Access Protocol) is an application protocol for accessing and maintaining information services for the distributed directory over an IP network. IP phones 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 the corporation using IP phones, therefore they do not have to maintain the directory locally. Users can search and dial out from the LDAP directory, and save LDAP entries to the local directory. LDAP entries displayed on the IP phone are read only, which cannot be added, edited or deleted by users. When an LDAP server is properly configured, the IP phone can look up entries from the LDAP server in a wide variety of ways. The LDAP server indexes all the data in its entries, and "filters" can be used to select the desired entry or group, and return the desired information.

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

Note

LDAP feature is not applicable to SIP-T19P and SIP-T20P IP phones.

You can set a DSS key to be an LDAP key, and then press the LDAP key to enter the LDAP search screen when the IP phone is idle.

LDAP Attributes

The following table lists the most common attributes used to configure the LDAP lookup on IP phones.

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

Procedure

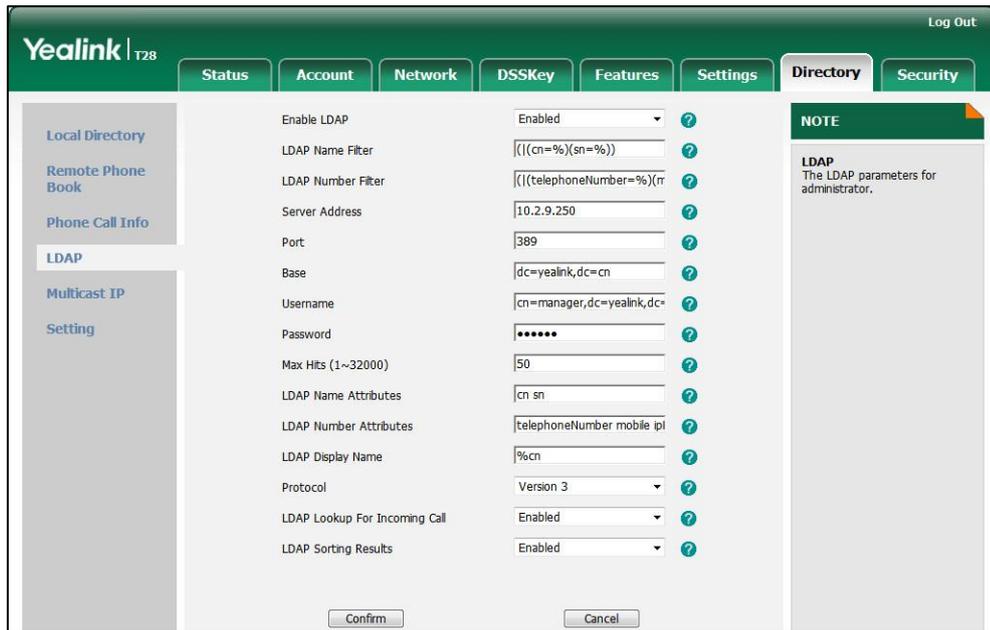
LDAP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	<p>Configure LDAP.</p> <p>For more information, refer to LDAP on page 340.</p> <p>Assign an LDAP key.</p> <p>For more information, refer to LDAP Key on page 416.</p>
Local	Web User Interface	<p>Configure LDAP.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=contacts-LDAP&q=load</p> <p>Assign an LDAP key.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0</p>
	Phone User Interface	Assign an LDAP key.

To configure LDAP via web user interface:

1. Click on **Directory->LDAP**.

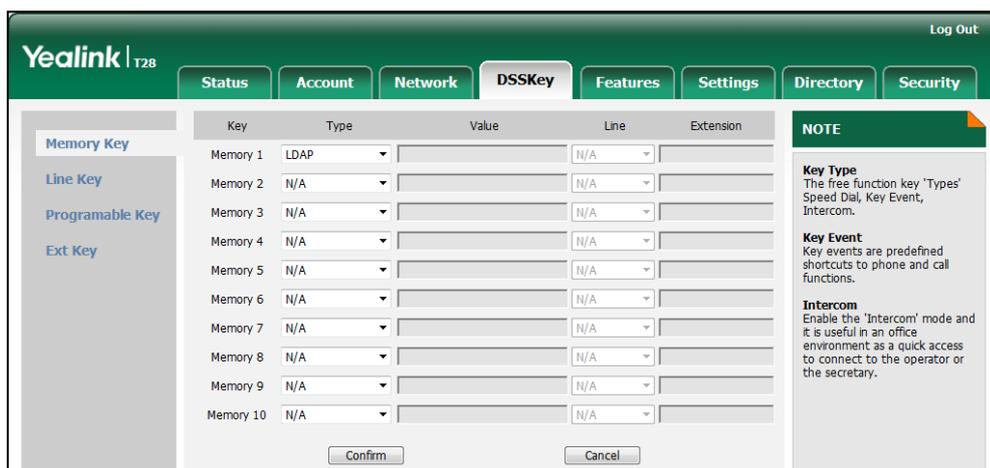
2. Enter the values in the corresponding fields.
3. Select the desired values from the corresponding pull-down lists.



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

To configure an LDAP key via web user interface:

1. Click on **DSSKey->Memory Key (Line Keys or Programable Key)**.
SIP-T19P IP phones only support programable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programable keys.
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 IP phones. For example, you can configure a BLF key on a supervisor's phone to monitor the phone user status (busy or idle). When the monitored user places a call, a busy indicator on the supervisor's phone indicates that the user's phone is in use.

When the monitored user is idle, the supervisor presses the BLF key to dial out the phone number. When the monitored user receives an incoming call, the supervisor presses the BLF key to pick up the call directly. When the monitored user is in a call, the supervisor presses the BLF key to interrupt and set up a conference call.

Note BLF feature is not applicable to SIPT19P IP phones.

Visual Alert and Audio Alert for BLF Pickup

Visual and audio alert for BLF pickup allow the supervisor's phone to play an alert tone and display a visual prompt (e.g., "6001 <-6002", 6001 is the monitored extension which receives an incoming call from 6002) when the monitored user receives an incoming call. In addition to the BLF key, visual alert for BLF pickup feature enables the supervisor to pick up the monitored user's incoming call by pressing the Pickup soft key. The directed call pickup code must be configured in advance. For more information on how to configure the directed call pickup code for the Pickup soft key, refer to [Directed Call Pickup](#) on page 107.

Note Visual alert for BLF pickup feature is not applicable to SIP-T20P IP phones.

BLF LED Mode

BLF LED Mode provides four kinds of definition for the BLF key LED status. The following table lists the LED statuses of the BLF key when BLF LED Mode is set to 0, 1, 2 or 3 respectively. The default value of BLF LED mode is 0.

Line key LED (configured as a BLF key and BLF LED Mode is set to 0)

LED Status	Description
Solid green	The monitored user is idle.
Fast flashing green (200ms)	The monitored user receives an incoming call.
Slow flashing green (500ms)	The monitored user is dialing. The monitored user is talking.

LED Status	Description
	The monitored user's conversation is placed on hold.
Slow flashing green (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user does not exist.

Memory key LED (configured as a BLF key and BLF LED Mode is set to 0)

LED Status	Description
Solid green	The monitored user is idle.
Fast flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking.
Slow flashing red (1s)	The call is parked against the monitored user's phone number. The monitored user's conversation is placed on hold.
Off	The monitored user does not exist.

Line key LED (configured as a BLF key and BLF LED Mode is set to 1)

LED Status	Description
Fast flashing green (200ms)	The monitored user receives an incoming call.
Solid green	The monitored user is dialing. The monitored user is talking.
Slow flashing green (500ms)	The monitored user's conversation is placed on hold.
Slow flashing green (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

Memory key LED (configured as a BLF key and BLF LED Mode is set to 1)

LED Status	Description
Fast flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking.
Slow flashing red (1s)	The call is parked against the monitored user's phone number. The monitored user's conversation is placed on hold.

LED Status	Description
Off	The monitored user is idle. The monitored user does not exist.

Line key LED (configured as a BLF key and BLF LED Mode is set to 2)

LED Status	Description
Fast flashing green (200ms)	The monitored user receives an incoming call.
Slow flashing green (500ms)	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold.
Slow flashing green (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

Memory key LED (configured as a BLF key and BLF LED Mode is set to 2)

LED Status	Description
Fast flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking.
Slow flashing red (1s)	The call is parked against the monitored user's phone number. The monitored user's conversation is placed on hold.
Off	The monitored user is idle. The monitored user does not exist.

Line key LED (configured as a BLF key and BLF LED Mode is set to 3)

LED Status	Description
Fast flashing green (200ms)	The monitored user receives an incoming call.
Solid green	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold.
Slow flashing green (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

Memory key LED (configured as a BLF key and BLF LED Mode is set to 3)

LED Status	Description
Fast flashing red (200ms)	The monitored user receives an incoming call.
Solid red	The monitored user is dialing. The monitored user is talking. The monitored user's conversation is placed on hold.
Slow flashing red (1s)	The call is parked against the monitored user's phone number.
Off	The monitored user is idle. The monitored user does not exist.

Procedure

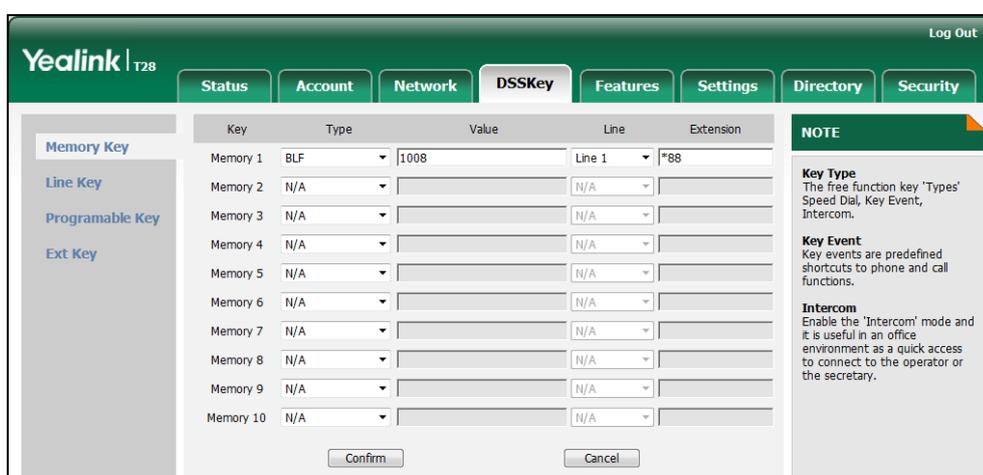
BLF can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Specify whether to use visual alert and audio alert for BLF pickup. For more information, refer to BLF on page 346 .
	y000000000xx.cfg	Assign a BLF key. For more information, refer to BLF Key on page 417 . Configure BLF LED mode. For more information, refer to BLF on page 346 .
Local	Web User Interface	Assign a BLF key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Specify whether to use visual alert and audio alert for BLF pickup. Navigate to: http://<phoneIPAddress>/servlet?p=features-callpickup&q=load Configure BLF LED mode. Navigate to:

		http://<phoneIPAddress>/servlet?op=features-general&q=load
	Phone User Interface	Assign a BLF key.

To configure a BLF key via web user interface:

1. Click on **DSSKey->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 **Line**.
5. (Optional.) Enter the directed call pickup code in the **Extension** field.

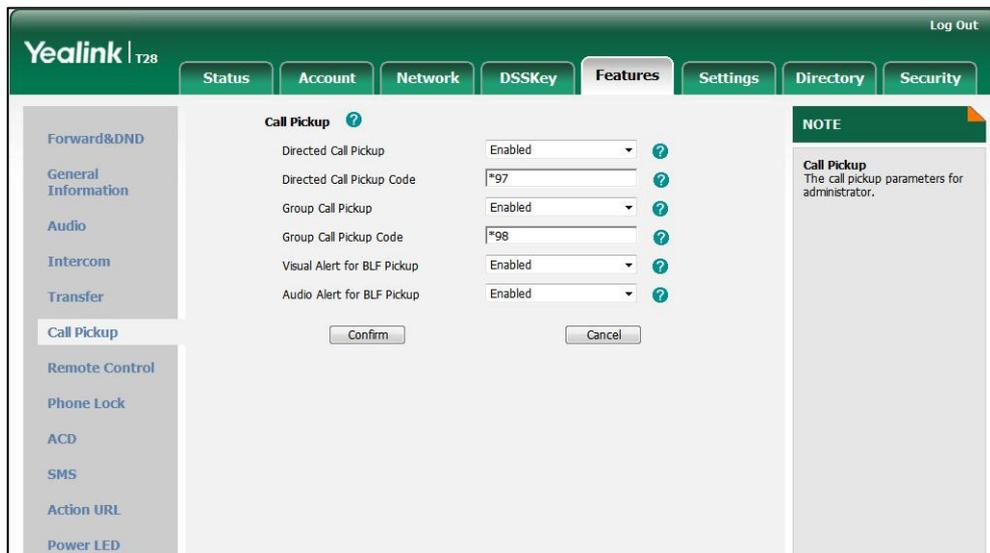


6. Click **Confirm** to accept the change.

To configure visual alert and audio alert for BLF pickup via web user interface:

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

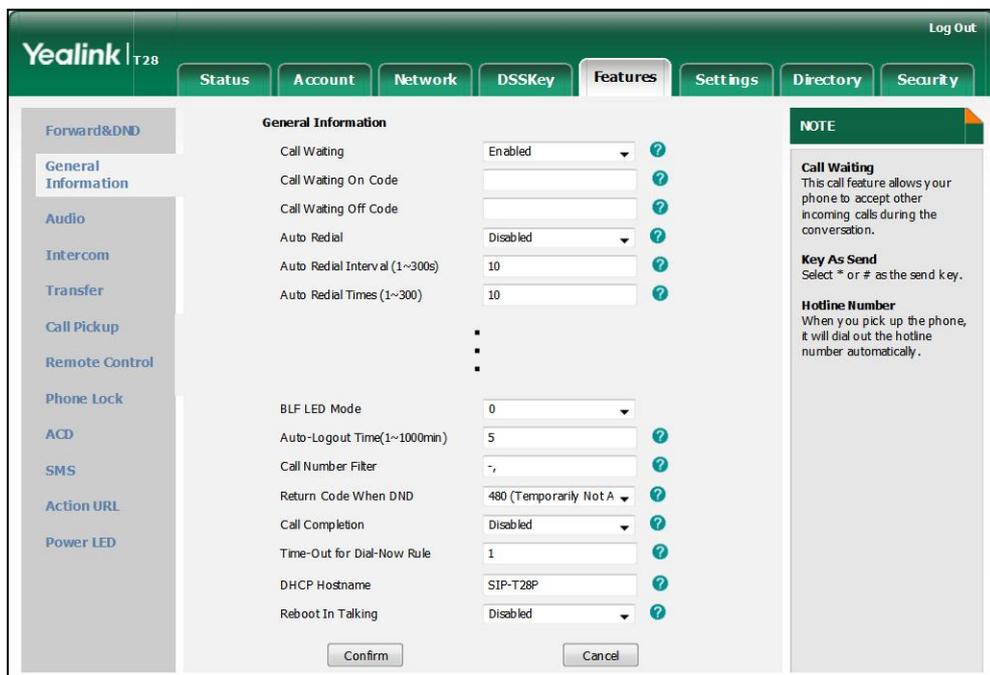
3. Select the desired value from the pull-down list of **Audio Alert for BLF Pickup**.



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

To configure BLF LED mode via web user interface:

1. Click on **Features->General Information**.
2. Select the desired value from the pull-down list of **BLF LED Mode**.



3. 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 directed call pickup code in the **Extension** field.
7. Press the **Save** soft key to accept the change.

Music on Hold

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

Procedure

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

Configuration File	<MAC>.cfg	Configure MoH on a per-line basis. For more information, refer to Music on Hold on page 348.
Local	Web User Interface	Configure MoH on a per-line basis. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc= 0

To configure MoH via web user interface:

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

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

The screenshot shows the Yealink T28 web interface with the 'Account' tab selected. The 'Advanced' sub-tab is active. The 'Music Server URI' field is populated with 'sip:moh@sip.com'. Other configuration options include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5060), 'RPort' (Disabled), 'SIP Session Timer T1 (0.5~10s)' (0.5), 'SIP Session Timer T2 (2~40s)' (4), 'SIP Session Timer T4 (2.5~60s)' (5), and 'Subscribe Period (Seconds)' (1800). A 'NOTE' box on the right indicates that advanced parameters are for administrators. The 'Confirm' button is visible at the bottom.

- 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 the use of IP phones in a call-center role by automatically distributing incoming calls to available users, or agents. ACD depends on support from a SIP server. ACD is disabled on the phone by default. You need to enable it on a per-line basis before logging into the ACD system.

After the IP phone user logs into the ACD system, the server monitors the phone status and then decides whether to assign an incoming call to the user's IP phone. When the phone status is changed to unavailable, the server stops distributing calls to the IP phone. The IP phone will remain in the unavailable status until the user manually changes the phone status or the ACD auto available timer (if configured) expires. How long the IP phone remains unavailable is configurable by auto-available timer. When the timer expires, the phone status is automatically changed to available. ACD auto available feature depends on support from a SIP server.

You need to configure an ACD key for the user to log into the ACD system. The ACD key LED on the IP phone indicates the ACD status.

Note

SIP-T19P IP phones support ACD feature for BroadSoft server only. For more information, refer to *Yealink IP Phones Deployment Guide for BroadSoft UC-One Environments*.

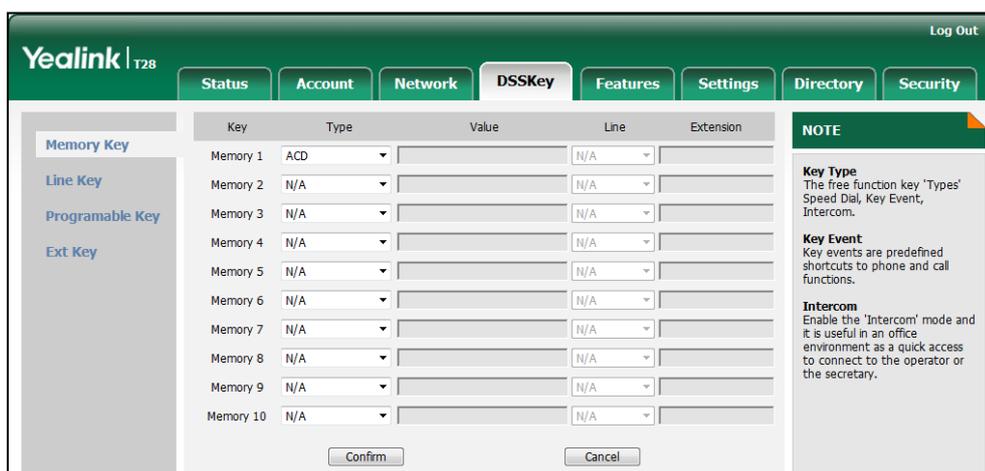
Procedure

ACD 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 419. Configure ACD auto available. For more information, refer to ACD on page 348.
Local	Web User Interface	Assign an ACD key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Configure ACD auto available. Navigate to: http://<phoneIPAddress>/servlet?p=features-acd&q=load
	Phone User Interface	Assign an ACD key.

To configure an ACD key via web user interface:

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

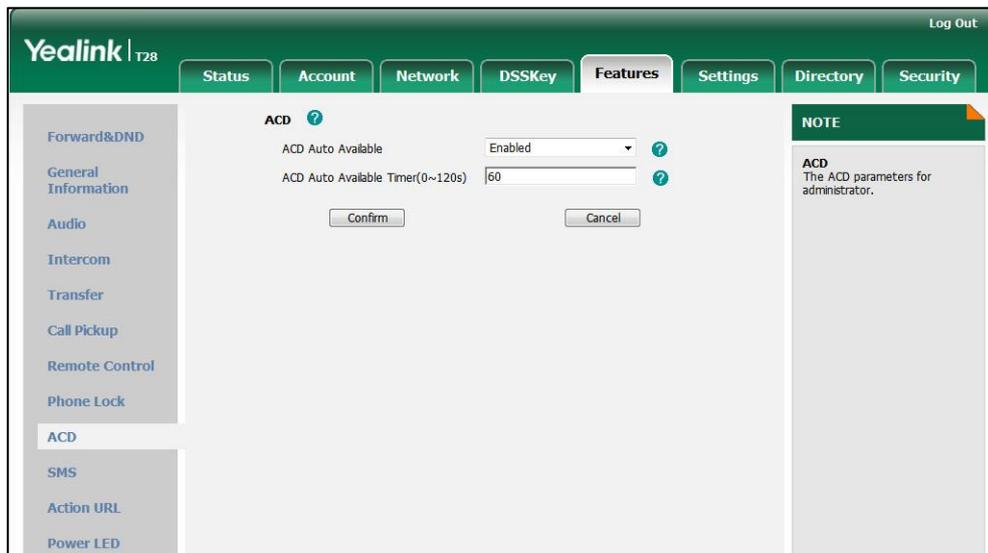


3. Click **Confirm** to accept the change.

To configure ACD auto available via web user interface:

1. Click on **Features->ACD**.
2. Select the desired line from the pull-down list of **ACD Auto Available**.

- Enter the desired time in **ACD Auto Available Timer (0~120s)** field.



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

To configure an ACD key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **ACD** from the **Type** field.
- Press the **Save** soft key to accept the change.

Message Waiting Indicator

Message Waiting Indicator (MWI) informs users of the number of messages waiting in their mailbox without calling the mailbox. IP phones support both audio and visual MWI when receiving new voice messages.

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

The IP phone sends a SUBSCRIBE message to the server for message-summary updates. The server sends a message-summary NOTIFY within the subscription dialog each time the MWI status changes. For solicited MWI, you must enable MWI subscription feature on IP phones. IP phones support subscribing the MWI messages to the account or the voice mail number.

IP phones do not need to subscribe for message-summary updates. The server automatically sends a message-summary NOTIFY in a new dialog each time the MWI status changes.

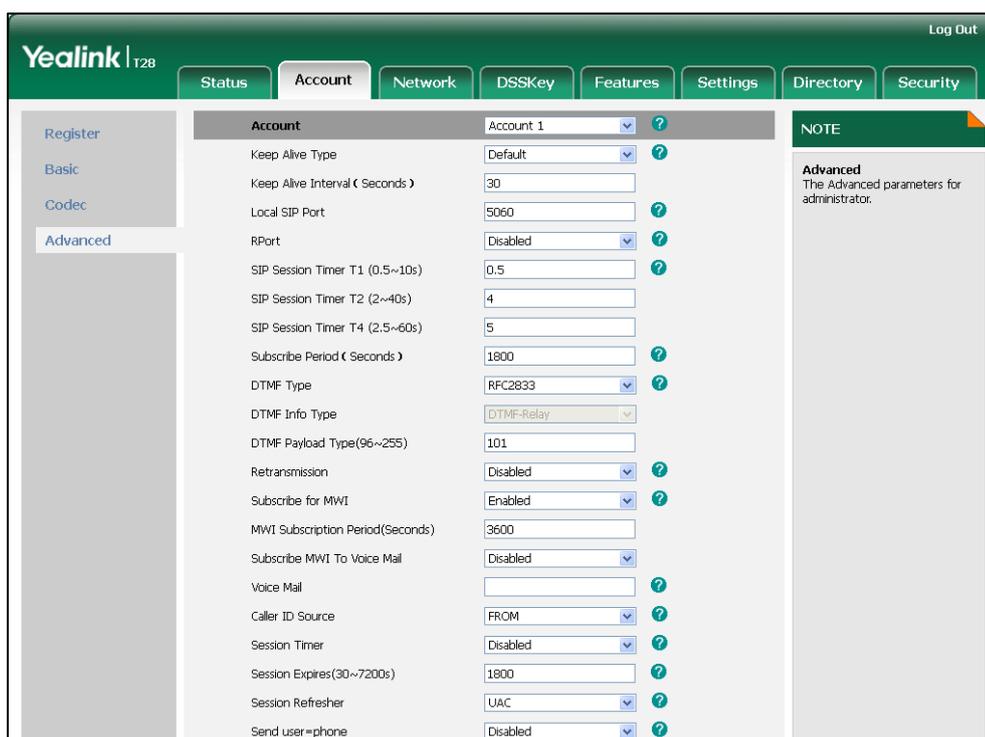
Procedure

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

<p>Configuration File</p>	<p><MAC>.cfg</p>	<p>Configure subscribe for MWI. Configure subscribe MWI to voice mail. For more information, refer to Message Waiting Indicator on page 348.</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure subscribe for MWI. Configure subscribe MWI to voice mail. Navigate to: http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>

To configure subscribe for MWI via web user interface:

1. Click on **Account**.
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 (Seconds)** field.



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

To configure subscribe MWI to voice mail via web user interface:

1. Click on **Account**.
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 MWI To Voice Mail**.
5. Enter the desired voice number in the **Voice Mail** field.

Account	Account 1	
Keep Alive Type	Default	?
Keep Alive Interval (Seconds)	30	
Local SIP Port	5060	?
RPort	Disabled	?
SIP Session Timer T1 (0.5~10s)	0.5	?
SIP Session Timer T2 (2~40s)	4	
SIP Session Timer T4 (2.5~60s)	5	
Subscribe Period (Seconds)	1800	?
DTMF Type	RFC2833	?
DTMF Info Type	DTMF-Relay	
DTMF Payload Type(96~255)	101	
Retransmission	Disabled	?
Subscribe for MWI	Enabled	?
MWI Subscription Period(Seconds)	3600	
Subscribe MWI To Voice Mail	Enabled	?
Voice Mail	1234	?

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

Multicast Paging

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

Sending RTP Stream

Users can send an RTP stream without involving SIP signaling by pressing a configured multicast paging key. A multicast address (IP: Port) should be assigned to the multicast paging key, which is defined to transmit RTP stream to a group of designated IP phones. When the IP phone sends the RTP stream to a pre-configured multicast address, each IP phone preconfigured to listen to the multicast address can receive the RTP stream. When the originator stops sending the RTP stream, the subscribers stop receiving it. This feature is not applicable to SIP-T19P IP phones.

Procedure

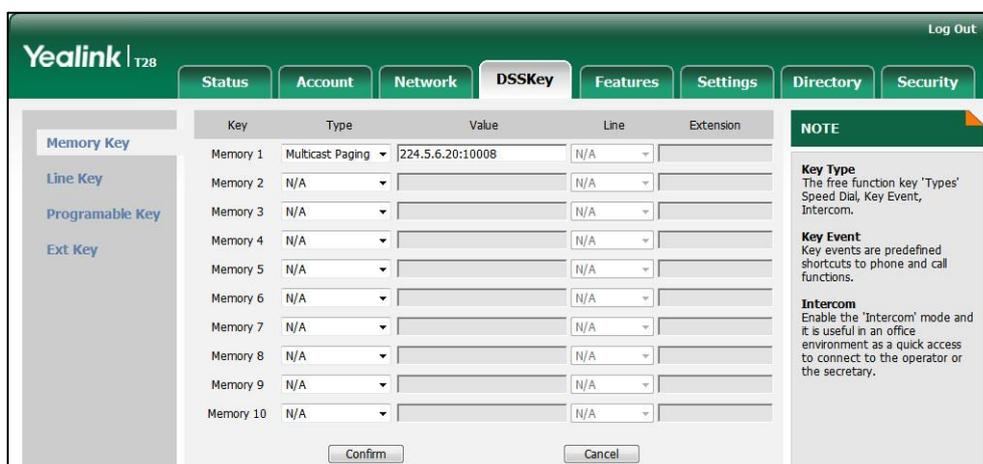
Configuration changes can be performed using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Assign a multicast paging key. For more information, refer to Multicast Paging Key on page 419. Specify a multicast codec for the IP phone to use for multicast RTP. For more information, refer to Sending RTP Stream on page 351.</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Assign a multicast paging key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0 Specify a multicast codec for the IP phone to use for multicast RTP. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	<p>Phone User Interface</p>	<p>Assign a multicast paging key.</p>

To configure a multicast paging key via web user interface:

1. Click on **DSSKey->Memory Key** (or **Line Key**).
2. In the desired DSS key field, select **Multicast Paging** from the pull-down list of **Type**.
3. Enter the multicast IP address and port number in the **Value** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.



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

To configure a codec for multicast paging via web user interface:

1. Click on **Features->General Information**.
2. Select the desired codec from the pull-down list of **Multicast Codec**.

The screenshot shows the Yealink T28 web interface with the 'Features' tab selected. The 'General Information' section is active, displaying various configuration options. The 'Multicast Codec' is currently set to 'G722'. Other settings include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) set to 10, Auto Redial Times (1~300) set to 10, Key As Send set to '#', Multicast Codec set to 'G722', Play Hold Tone (Enabled), Play Hold Tone Delay set to 30, Call Completion (Disabled), Voice Mail Tone (Enable), DHCP Hostname set to 'SIP-T28P', and Reboot In Talking (Disabled). There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' section on the right provides details for Call Waiting, Key As Send, and Hotline Number features.

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

To configure a multicast paging 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 **Multicast Paging** from the **Key Type** field.
5. Enter the multicast IP address and port number in the **Value** field.
6. Press the **Save** soft key to accept the change.

Receiving RTP Stream

IP phones can receive an RTP stream from the pre-configured multicast address(es) without involving SIP signaling, and can handle the incoming multicast paging calls differently depending on the configurations of Paging Barge and Paging Priority Active.

Paging Barge

This parameter defines the priority of the voice call in progress, and decides how the IP phone handles the incoming multicast paging calls when there is already a voice call in progress. If the parameter is configured as disabled, all incoming multicast paging calls

will be automatically ignored. If the parameter is the priority value, the incoming multicast paging calls with higher priority are automatically answered and the ones with lower priority are ignored.

Paging Priority Active

This parameter decides how the IP phone handles the incoming multicast paging calls when there is already a multicast paging call in progress. If the parameter is configured as disabled, the IP phone will automatically ignore all incoming multicast paging calls. If the parameter is configured as enabled, an incoming multicast paging call with higher priority is automatically answered, and the one with lower priority is ignored.

Procedure

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

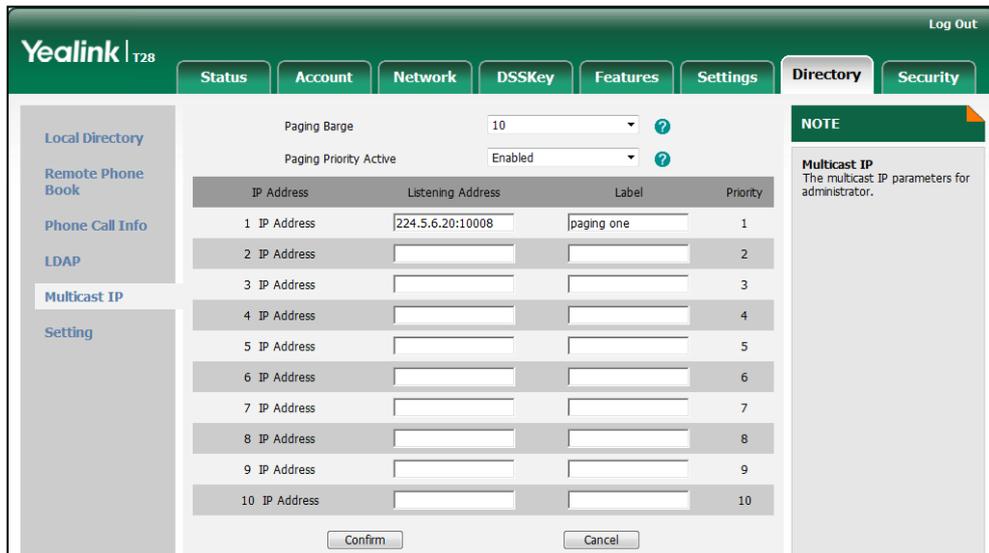
<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the listening multicast address.</p> <p>Configure Paging Barge and Paging Priority Active features.</p> <p>For more information, refer to Receiving RTP Stream on page 352.</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the listening multicast address.</p> <p>Configure Paging Barge and Paging Priority Active features.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load</p>

To configure a listening multicast address via web user interface:

1. Click on **Directory->Multicast IP**.
2. Enter the listening multicast address and port number in the **Listening Address** field.
1 is the highest priority and 10 is the lowest priority.

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

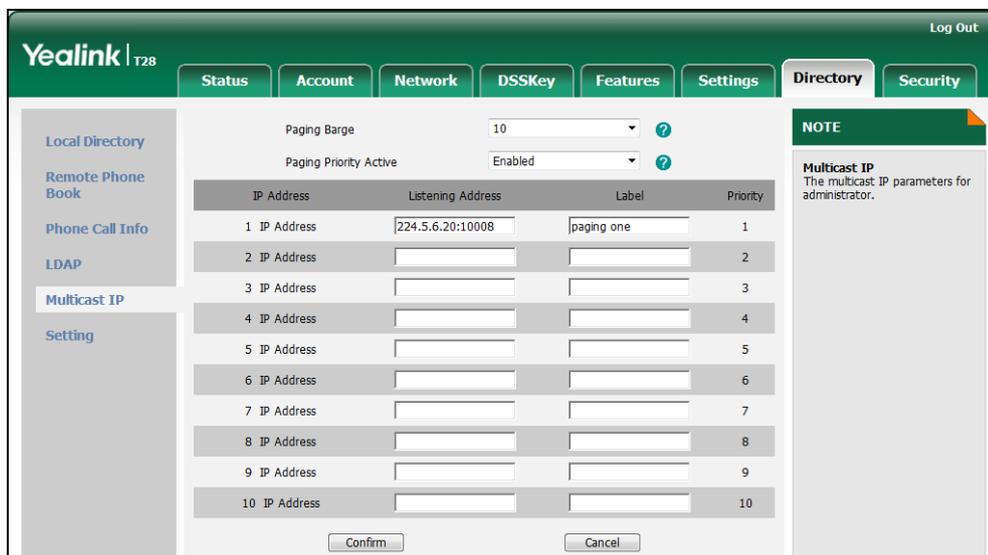
The label will appear on the LCD screen when receiving the RTP multicast.



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

To configure paging barge and paging priority active features via web user interface:

1. Click on **Directory->Multicast IP**.
2. Select the desired value from the pull-down list of **Paging Barge**.
3. Select the desired value from the pull-down list of **Paging Priority Active**.



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

Call Recording

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

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

Note Call recording is not applicable to SIP-T19P IP phones.

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 the specific header "Record: on", and then the recording starts.

Example of a SIP INFO message:

```
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-T28P 2.72.0.1
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 the specific header "Record: off", and then the recording stops.

Example of a SIP INFO message:

```
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-T28P 2.72.0.1
Record: off
Content-Length: 0
```

URL Record

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

Example of an HTTP GET message:

```
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-T28P 2.72.0.1 00:16:65:11:30:68\r\n
```

If the recording is successfully started, the server will respond with a 200 OK message.

Example of a 200 OK message:

```
<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 a 200 OK message.

Example of a 200 OK message:

```
<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 a 200 OK message.

Example of a 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 420. Assign a URL record key. For more information, refer to URL Record Key on page 421.
Local	Web User Interface	Assign a record key and URL record key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0
	Phone User Interface	Assign a record key and URL record key.

To configure a record key via web user interface:

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

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

Key	Type	Value	Line	Extension
Memory 1	Record		N/A	
Memory 2	N/A		N/A	
Memory 3	N/A		N/A	
Memory 4	N/A		N/A	
Memory 5	N/A		N/A	
Memory 6	N/A		N/A	
Memory 7	N/A		N/A	
Memory 8	N/A		N/A	
Memory 9	N/A		N/A	
Memory 10	N/A		N/A	

NOTE

Key Type
The free function key 'Types' Speed Dial, Key Event, Intercom.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

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

To configure a URL record key via web user interface:

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

Key	Type	Value	Line	Extension
Memory 1	URL Record	0.2.1.224/phonerecording.cgi	N/A	
Memory 2	N/A		N/A	
Memory 3	N/A		N/A	
Memory 4	N/A		N/A	
Memory 5	N/A		N/A	
Memory 6	N/A		N/A	
Memory 7	N/A		N/A	
Memory 8	N/A		N/A	
Memory 9	N/A		N/A	
Memory 10	N/A		N/A	

NOTE

Key Type
The free function key 'Types' Speed Dial, Key Event, Intercom.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

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

To configure a record key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **Record** from the **Key Type** field.
- 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 **Value** 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 employees are in the office at the same time, or not in the office for a long time, which means actual personal offices would often be vacant, consuming valuable space and resources.

Hot desking allows a user to clear registration configurations of all accounts on the IP phone, and then 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 422.
Local	Web User Interface	Assign a hot desking key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&q=load&model=0
	Phone User Interface	Assign a hot desking key.

To configure a hot desking key via web user interface:

1. Click on **DSSKey->Memory Keys (Line Key or Programmable Key)**.
SIP-T19P IP phones only support programmable keys and SIP-T22P/T21P/T20P IP phones only support line keys and programmable keys.

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

Key	Type	Value	Line	Extension
Memory 1	Hot Desking		N/A	
Memory 2	N/A		N/A	
Memory 3	N/A		N/A	
Memory 4	N/A		N/A	
Memory 5	N/A		N/A	
Memory 6	N/A		N/A	
Memory 7	N/A		N/A	
Memory 8	N/A		N/A	
Memory 9	N/A		N/A	
Memory 10	N/A		N/A	

NOTE

Key Type
The free function key 'Types' Speed Dial, Key Event, Intercom.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

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

Note You can configure a programable key as a hot desking key on SIP-T19P IP phones only.

To configure a hot desking key via phone user interface:

- Press **Menu->Features->DSS Keys->Memory Keys** (or **Line Keys**).
- Select the desired DSS key.
- Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select **Hot Desking** from the **Key Type** field.
- Press the **Save** soft key to accept the change.

Action URL

Action URL allows IP phones to interact with web server applications by sending an HTTP or HTTPS GET request. You can specify a URL that triggers a GET request when a specified event occurs. Action URL can only be triggered by the pre-defined events (e.g., log on). The valid URL format is: http(s)://IP address of the server/help.xml?.

The following table lists the pre-defined events for action URL.

Event	Description
Setup Completed	When the IP phone completes startup.
Registered	When the IP phone successfully registers an account.
Unregistered	When the IP phone logs off the registered account.
Register Failed	When the IP phone fails to register an account.
Off Hook	When the IP phone is off hook.

Event	Description
On Hook	When the IP phone is on hook.
Incoming Call	When the IP phone receives an incoming call.
Outgoing Call	When the IP phone places a call.
Established	When the IP phone establishes a call.
Terminated	When the IP phone terminates a call.
Open DND	When the IP phone enables the DND mode.
Close DND	When the IP phone disables the DND mode.
Open Always Forward	When the IP phone enables the always forward.
Close Always Forward	When the IP phone disables the always forward.
Open Busy Forward	When the IP phone enables the busy forward.
Close Busy Forward	When the IP phone disables the busy forward.
Open No Answer Forward	When the IP phone enables the no answer forward.
Close No Answer Forward	When the IP phone disables the no answer forward
Transfer Call	When the IP phone transfers a call.
Blind Transfer	When the IP phone blind transfers a call.
Attended Transfer	When the IP phone performs the semi-attended/attended transfer.
Hold	When the IP phone places a call on hold.
UnHold	When the IP phone retrieves a hold call.
Mute	When the IP phone mutes a call.
UnMute	When the IP phone un-mutes a call.
Missed Call	When the IP phone misses a call.
IP Changed	When the IP address of the IP phone changes.
Forward Incoming Call	When the IP phone forwards an incoming call.
Reject Incoming Call	When the IP phone rejects an incoming call.
Answer New-In Call	When the IP phone answers a new call.
Transfer Finished	When the IP phone completes to transfer a call.
Transfer Failed	When the IP phone fails to transfer a call.
Idle To Busy	When the state of the IP phone changes from idle to busy.
Busy To Idle	When the state of phone changes from busy to idle.

An HTTP or HTTPS GET request may contain variable name and variable value, separated by “=”. Each variable value starts with \$ in the query part of the URL. The valid URL format is: `http(s)://IP address of server/help.xml?variable name=$variable`. Variable name can be customized by users, while the variable value is pre-defined. For example, a URL “`http://192.168.1.10/help.xml?mac=$mac`” is specified for the event Mute, \$mac will be dynamically replaced with the MAC address of the IP phone when the IP phone mutes a call.

The following table lists pre-defined variable values.

Variable Value	Description
\$mac	The MAC address of the IP phone
\$ip	The IP address of the IP phone
\$model	The IP phone model
\$firmware	The firmware version of the IP phone
\$active_url	The SIP URI of the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$active_user	The user part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$active_host	The host part of the SIP URI for the current account when the IP phone places a call, receives an incoming call or establishes a call.
\$local	The SIP URI of the caller when the IP phone places a call. The SIP URI of the callee when the IP phone receives an incoming call.
\$remote	The SIP URI of the callee when the IP phone places a call. The SIP URI of the caller when the IP phone receives an incoming call.
\$display_local	The display name of the caller when the IP phone places a call. The display name of the callee when the IP phone receives an incoming call.
\$display_remote	The display name of the callee when the IP phone places a call. The display name of the caller when the IP phone receives an incoming call.

Variable Value	Description
\$call_id	The call-id of the active call.

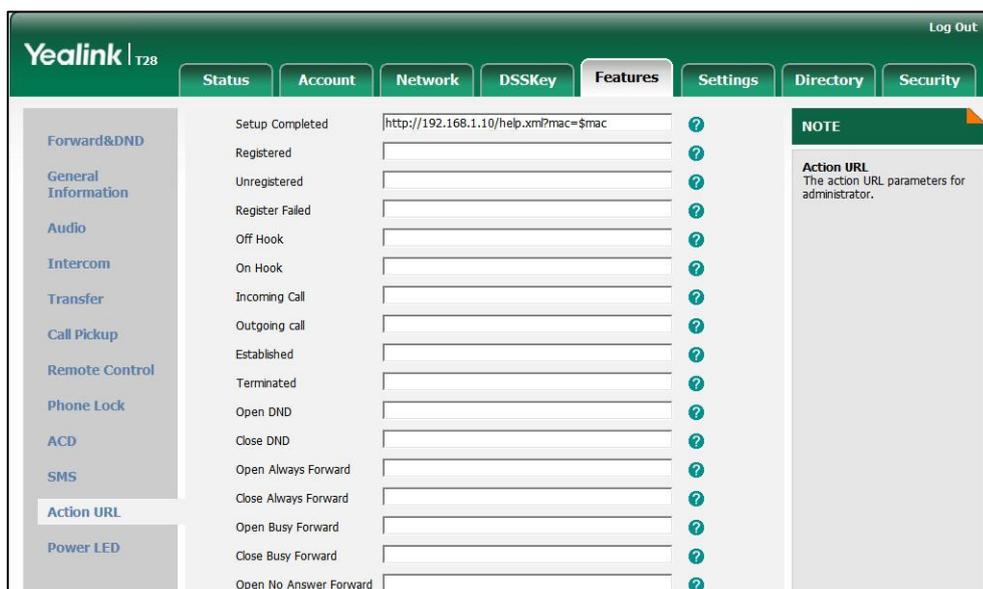
Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure action URL. For more information, refer to Action URL on page 354.
Local	Web User Interface	Configure action URL. Navigate to: http://<phoneIPAddress>/servlet?p=features-actionurl&q=load

To configure action URL via web user interface:

1. Click on **Features->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 IP phones to interact with web server application by receiving and handling an HTTP or HTTPS GET request. When receiving a GET request, the IP phone will perform the specified action and respond with a 200 OK message. A GET request may contain variable named as “key” and variable value,

which are separated by “=”. The valid URI format is: http(s)://phone IP address/servlet?key=variable value.

The following table lists pre-defined variable values:

Variable Value	Phone Action
OK	Press the OK key (For SIPT19P, press ).
ENTER	Press the Enter soft key (Except for SIP-T20P).
SPEAKER	Press the Speakerphone key.
F_TRANSFER	Transfers a call to another party.
VOLUME_UP	Increase the volume.
VOLUME_DOWN	Decrease the volume.
MUTE	Mute the call.
F_HOLD	Place an active call on hold.
X	Cancel actions or reject incoming calls (For SIP-T22P/T21P/T20P, also mute or un-mute calls).
0-9*/POUND	Press the keypad (0-9, * or #).
L1-LX	Press the line keys (Except for SIPT19P, for SIPT28P, X=6, for SIPT26/22P, X=3, for SIPT21P/T20P, X=2).
D1-D10	Press the memory keys (Only for SIPT28/T26P).
F_CONFERENCE	Press the CONF key (Except for SIPT22P/T21P/T19P) or the Conference soft key (Except for SIP-T20P).
F1-F4	Press the soft keys (Except for SIPT20P).
MSG	Press the MESSAGE key.
HEADSET	Press the HEADSET key.
RD	Press the RD key.
UP/DOWN/LEFT/RIGHT	Press the navigation keys.
Reboot	Reboot the IP phone.
AutoP	Perform auto provisioning.
DNDOn	Activate the DND feature.
DNDOff	Deactivate the DND feature.
number=xxx&outgoing_uri=y	Place a call to xxx from SIP URI y.
OFFHOOK	Pick up the handset.

Variable Value	Phone Action
ONHOOK	Hang up the handset.
ANSWER	Answer a call.
Reset	Reset a phone.
ATrans=xxx	Perform a semi-attended/attended transfer to xxx.
BTrans=xxx	Perform a blind transfer to xxx.
CALLEND	End a call.
phonecfg=get[&accounts=x][&dnd=x][&fw=x]	<p>Get firmware version, registration, DND or forward configuration information.</p> <p>The valid value of "x" is 0 or 1, 0 means you do not need to get configuration information. 1 means you want to get configuration information.</p> <p>Note: The valid URI is: http(s)://phone IP address/servlet?phonecfg=get[&accounts=x][&dnd=x][&fw=x]</p>

Note

The variable value is not applicable to all events. For example, the variable value "MUTE" is only applicable when the IP phone is during a call.

When authentication is required, you must enter "p=login&q=login&username=xxx&pwd=yyy&jumpto=URI&" before the variable "key". xxx refers to the login user name and yyy refers to the login password.

For security reasons, IP phones do not receive and handle HTTP/HTTPS GET requests by default. You need to specify the trusted IP address for action URI. When the IP phone receives a GET request from the trusted IP address for the first time, the LCD screen prompts the message "Allow Remote Control?". You can specify one or more trusted IP addresses on the IP phone, or configure the IP phone to receive and handle the URI from any IP address.

Procedure

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

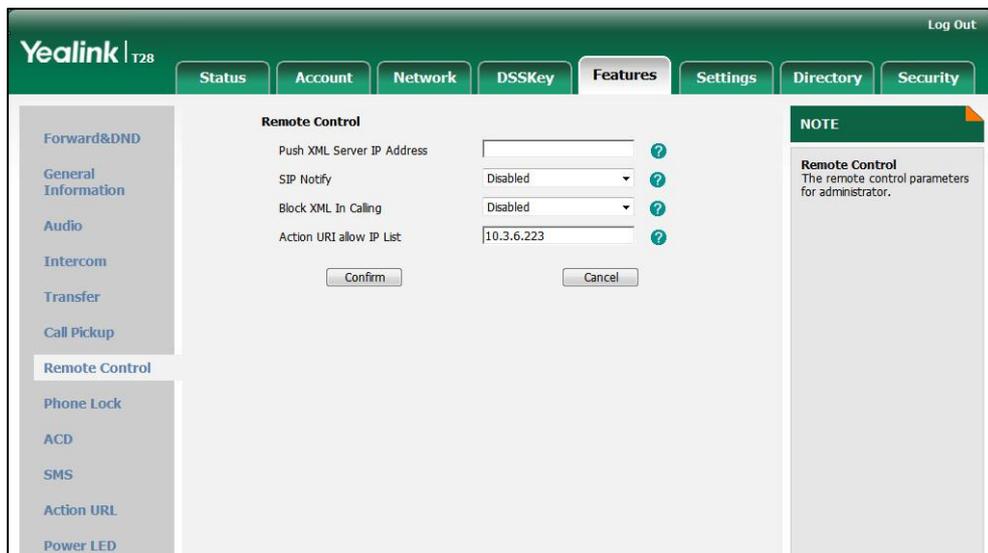
Configuration File	<y0000000000xx>.cfg	<p>Specify the trusted IP address(es) for sending the action URI to the IP phone.</p> <p>For more information, refer to Action URI on page 355.</p>
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<p>Local</p>	<p>Web User Interface</p>	<p>Specify the trusted IP address(es) for sending the action URI to the IP phone.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?&p=features-remotecontrol&q=load</p>
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To configure the trusted IP address(es) for action URI via web user interface:

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

Multiple IP addresses are separated by commas. If you enter “any” in this field, the IP phone can receive and handle GET requests from any IP address. If you leave the field blank, the IP phone cannot receive or handle any HTTP GET request.



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

Server Redundancy

Server redundancy is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

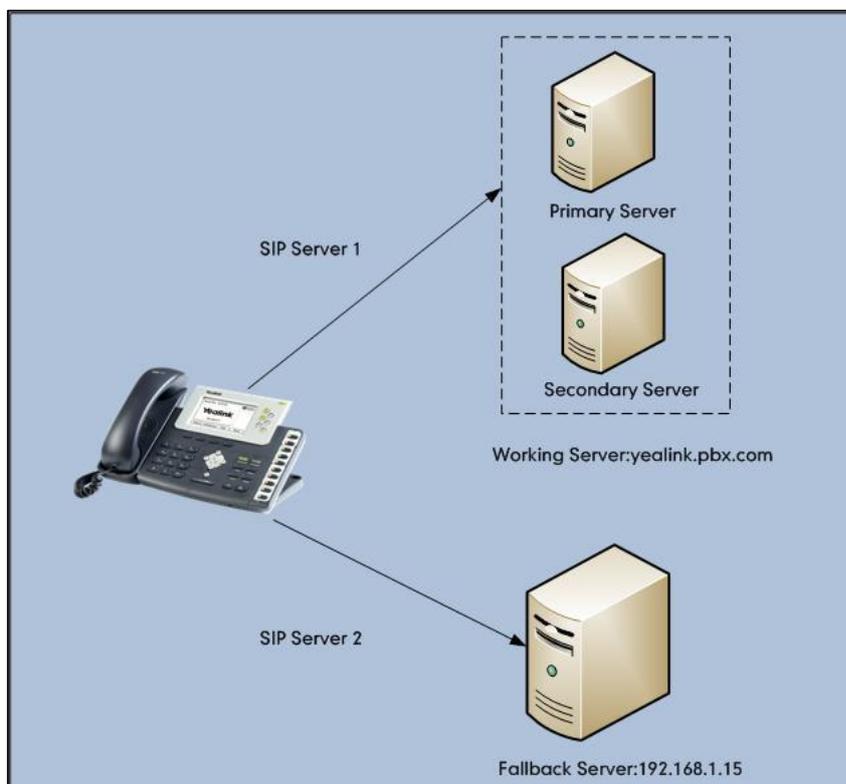
Two types of redundancy are possible. In some cases, a combination of the two may be deployed:

- **Failover:** In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using the DNS mechanism from the primary to the secondary server.

- Fallback:** In this mode, a second less featured call server with SIP capability takes over call control to provide basic calling capability, but without some advanced features offered by the working server (for example, shared line, call recording and MWI). IP phones support configuration of two SIP servers per SIP registration for fallback purpose.

Phone Configuration for Redundancy Implementation

To assist in explaining the redundancy behavior, an illustrative example of how an IP phone may be configured is shown as below. In the example, server redundancy for fallback and failover purposes is deployed. Two separate SIP servers (a working server and a fallback server) are configured for per line registration.



Working Server: Server 1 is configured with the domain name of the working server. For example, yealink.pbx.com. DNS mechanism is used such that the working server is resolved to multiple SIP servers for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server has the highest priority server in a cluster of servers resolved by the DNS server. The secondary server backs up a primary server when the primary server fails and offers the same functionality as the primary server.

Fallback Server: Server 2 is configured with the IP address of the fallback server. For example, 192.168.1.15. A fallback server offers less functionality than the working server.

Phone Registration

Registration methods of the fallback mode include:

- **Concurrent registration:** The IP phone registers to two SIP servers (working server and fallback server) at the same time. In a failure situation, a fallback server can take over the basic calling capability, but without some of the advanced features offered by the working server (default registration method).
- **Successive registration:** The IP phone only registers to one server at a time. The IP phone first registers to the working server. In a failure situation, the IP phone registers to the fallback server.

When registering to the working server, the IP phone must always register to the primary server first except in failover conditions. When the primary server registration is unavailable, the secondary server will serve as the working server.

Procedure

Server redundancy can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the server redundancy on the IP phone. For more information, refer to Server Redundancy on page 356.
Local	Web User Interface	Configure the server redundancy on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

To configure server redundancy for fallback purpose via web user interface:

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Configure registration parameters of the selected account in the corresponding fields.
4. Select the desired value from the pull-down list of **Transport**.

5. Configure parameters of SIP server 1 and SIP server 2 in the corresponding fields.

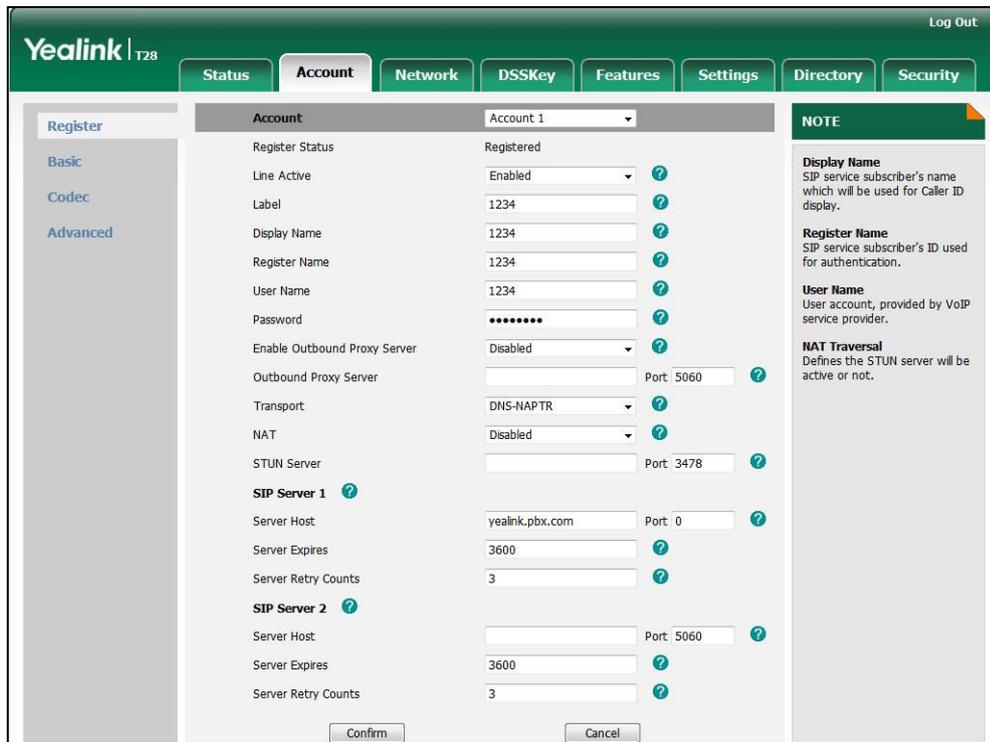
The screenshot shows the Yealink T28 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is selected, and the 'Register' sub-tab is active. The main content area displays configuration fields for 'Account 1'. The fields are organized into sections: 'Register' (Status, Line Active, Label, Display Name, Register Name, User Name, Password, Enable Outbound Proxy Server, Outbound Proxy Server, Transport, NAT, STUN Server), 'SIP Server 1', and 'SIP Server 2'. Each field has a corresponding input type (checkbox, dropdown, text box, or port field) and a help icon. A 'NOTE' section on the right provides definitions for 'Display Name', 'Register Name', 'User Name', and 'NAT Traversal'. At the bottom, there are 'Confirm' and 'Cancel' buttons.

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

To configure server redundancy for failover purpose via web user interface:

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Configure registration parameters of the selected account in the corresponding fields.
4. Select **DNS-NAPTR** from the pull-down list of **Transport**.

- Configure parameters of the SIP server 1 or SIP server 2 in the corresponding fields.
You must set the port of SIP server to 0 for NAPTR, SRV and A queries.



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

Note If the outbound proxy server is required and the transport is set to DNS-NAPTR, you must set the port of outbound proxy server to 0 for NAPTR, SRV and A queries.

SIP Server Domain Name Resolution

If a domain name is configured for a SIP server, the IP address(es) associated with that domain name will be resolved through DNS as specified by RFC 3263. The DNS query involves NAPTR, SRV and A queries, which allows the IP phone to adapt to various deployment environments. The IP phone performs NAPTR query for the NAPTR pointer and transport protocol (UDP, TCP and TLS), the SRV query on the record returned from the NAPTR for the target domain name and the port number, and the A query for the IP addresses.

If an explicit port (except 0) is specified and the transport type is set to DNS-NAPTR, A query will be performed only. If a SIP server port is set to 0 and the transport type is set to DNS-NAPTR, NAPTR and SRV queries will be tried before falling to A query. If no port is found through the DNS query, 5060 will be used.

The following details the procedures of DNS query for the IP phone to resolve the domain name (e.g., yealink.pbx.com) of working server into the IP address, port and transport protocol.

NAPTR (Naming Authority Pointer)

First, the IP phone sends NAPTR query to get the NAPTR pointer and transport protocol.

Example of NAPTR records:

	order	pref	flags	service	regexp	replacement
IN NAPTR	90	50	"s"	"SIP+D2T"	""	_sip._tcp.yealink.pbx.com
IN NAPTR	100	50	"s"	"SIP+D2U"	""	_sip._udp.yealink.pbx.com

Parameters are explained in the following table:

Parameter	Description
order	Specify preferential treatment for the specific record. The order is from lowest to highest, lower order is more preferred.
pref	Specify the preference for processing multiple NAPTR records with the same order value. Lower value is more preferred.
flags	The flag "s" means to perform an SRV lookup.
service	Specify the transport protocols: SIP+D2U: SIP over UDP SIP+D2T: SIP over TCP SIP+D2S: SIP over SCTP SIPS+D2T: SIPS over TCP
regexp	Always empty for SIP services.
replacement	Specify a domain name for the next query.

The IP phone picks the first record, because its order of 90 is lower than 100. The pref parameter is unimportant as there is no other record with order 90. The flag "s" indicates performing the SRV query next. TCP will be used, targeted to a host determined by an SRV query of "_sip._tcp.yealink.pbx.com". If the flag of the NAPTR record returned is empty, the IP phone will perform NAPTR query again according to the previous NAPTR query result.

SRV (Service Location Record)

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

	Priority	Weight	Port	Target
IN SRV	0	1	5060	server1.yealink.pbx.com
IN SRV	0	2	5060	server2.yealink.pbx.com

Parameters are explained in the following table:

Parameter	Description
Priority	Specify preferential treatment for the specific host entry. Lower priority is more preferred.
Weight	When priorities are equal, weight is used to differentiate the preference. The preference is from highest to lowest. Keep the same to load balance.
Port	Identify the port number to be used.
Target	Identify the actual host for an A query.

SRV query returns two records. The two SRV records point to different hosts and have the same priority 0. The weight of the second record is higher than the first one, so the second record will be picked first. The two records also contain a port "5060", the IP phone uses this port. If the Target is not a numeric IP address, the IP phone performs an A query. So in this case, the IP phone uses "server1.yealink.pbx.com" and "server2.yealink.pbx.com" for the A query.

A (Host IP Address)

The IP phone performs an A query for the IP address of each target host name. Example of A records:

```
Server1.yealink.pbx.com IN A 192.168.1.13
```

```
Server2.yealink.pbx.com IN A 192.168.1.14
```

The IP phone picks the IP address "192.168.1.14" first.

Outgoing Call When the Working Server Connection Fails

When a user initiates a call, the phone will go through the following steps to connect the call:

1. Sends the INVITE request to the primary server.
2. If the primary server does not respond correctly to the INVITE, then tries to make the call using the secondary server.
3. If the secondary server is also unavailable, the IP phone will try the fallback server until it either succeeds in making a call or exhausts all servers at which point the call will fail.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below:

- If TCP is used, then the signaling fails if the connection or the send fails.
- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted through all servers in the list and this is the last server, then the signaling fails after the complete UDP timeout defined in RFC 3261.

If it is not the last server in the list, the maximum number of retries depends on the configured retry count.

Procedure

Server redundancy can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transport type on the IP phone. For more information, refer to SIP Server Domain Name Resolution on page 360.
Local	Web User Interface	Configure the transport type on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

LLDP

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol, which allows IP phones to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol, and store the information about other devices. LLDP transmits information as packets called LLDP Data Units (LLDPDUs). An LLDPDU consists of a set of Type-Length-Value (TLV) elements, each of which contains a particular type of information about the device or the port transmitting it.

LLDP-MED (Media Endpoint Discovery)

LLDP-MED is published by the Telecommunications Industry Association (TIA). It is an extension to LLDP that operates between endpoint devices and network connectivity devices. LLDP-MED specifically provides support for voice over IP (VoIP) applications and provides the following capabilities:

- Capabilities Discovery -- allows IP phones to determine the capabilities that the connected switch supports and has enabled.
- Network Policy -- provides voice VLAN configuration to notify IP phones which VLAN to use and QoS-related configuration for voice data. It provides a "plug and play" network environment.
- Power Management -- provides information related to how IP phones are powered, power priority, and how much power IP phones need.
- Inventory Management -- provides a means to effectively manage IP phones and

their attributes such as model number, serial number and software revision.

TLVs supported by IP phones are summarized in the following table:

TLV Type	TLV Name	Description
Mandatory TLVs	Chassis ID	The network address of the IP phone.
	Port ID	The MAC address of the IP phone.
	Time To Live	Seconds until data unit expires.
	End of LLDPDU	Marks end of LLDPDU.
Optional TLVs	System Name	Name assigned to the IP phone. 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 the IP phone. The supported capabilities are Bridge, Telephone and Router. The enabled capabilities are Bridge and Telephone by default.
	Port Description	Description of port that sends data unit. The default value is "WAN PORT".
IEEE Std 802.3 Organizationally Specific TLV	MAC/PHY Configuration/Status	Duplex and bit rate settings of the IP phone. The Auto Negotiation is supported and enabled by default. The advertised capabilities of PMD. Auto-Negotiation is: 100BASE-TX (full duplex mode), 100BASE-TX (half duplex mode), 10BASE-T (full duplex mode), or 10BASE-T (half duplex mode).
TIA Organizationally Specific TLVs	Media Capabilities	The MED device type of the IP phone and the supported LLDP-MED TLV type can be encapsulated in LLDPDU. The supported LLDP-MED TLV types are: LLDP-MED Capabilities, Network Policy, Extended Power via MDI-PD and Inventory.
	Network Policy	Port VLAN ID, application type, L2 priority

TLV Type	TLV Name	Description
		and DSCP value.
	Extended Power-via-MDI	Power type, source, priority and value.
	Inventory – Hardware Revision	Hardware revision of the IP phone.
	Inventory – Firmware Revision	Firmware revision of the IP phone.
	Inventory – Software Revision	Software revision of the IP phone.
	Inventory – Serial Number	Serial number of the IP phone.
	Inventory – Manufacturer Name	Manufacturer name of the IP phone. The default value is “yealink”.
	Inventory – Model Name	Model name of the IP phone.
	Asset ID	Assertion identifier of the IP phone. The default value is “asset”.

Procedure

LLDP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure LLDP. For more information, refer to LLDP on page 356.
Local	Web User Interface	Configure LLDP. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load

To configure LLDP via web user interface:

1. Click on **Network->Advanced**.
2. In the **LLDP** block, select the desired value from the pull-down list of **Active**.
3. Enter the desired time interval in the **Packet Interval (1~3600s)** field.

The screenshot shows the Yealink T28 web interface with the 'Network' tab selected and 'Advanced' sub-tab active. The configuration is as follows:

Section	Parameter	Value
LLDP	Active	Enabled
	Packet Interval (1~3600s)	60
VLAN	WAN Port	Active
	VID (1-4094)	0
	Priority	0
PC Port	Active	Disabled
	VID (1-4094)	0
	Priority	0
DHCP VLAN	Active	Enabled
	Option	132
Port Link	WAN Port Link	Auto Negotiate
	PC Port Link	Auto Negotiate

NOTE
VLAN
 A VLAN is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.
QoS
 When the network capacity is insufficient, QoS could provide priority to users by setting the value.
Local RTP Port
 Define the port for voice transmission.

4. Click **Confirm** to accept the change.
A dialog box pops up to prompt that settings will take effect after a 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 (Internet port and PC port) on the IP phone, the IP phone will tag all packets from these ports with the VLAN ID. The switch receives and forwards the tagged packets to the corresponding VLAN according to the VLAN ID in the tag as described in IEEE Std 802.3.

VLAN on IP phones allows simultaneous access for a regular PC. This feature allows a PC to be daisy chained to an IP phone and the connection for both PC and IP phone to be trunked through the same physical Ethernet cable.

In addition to manual configuration, the IP phone also supports automatic discovery of VLAN via LLDP or DHCP. The assignment takes place in this order: assignment via LLDP, manual configuration, then assignment via DHCP.

VLAN Discovery via DHCP

IP phones support VLAN discovery via DHCP. When the VLAN Discovery method is set to DHCP, the IP phone will examine DHCP option for a valid VLAN ID. The predefined option 132 is used to supply the VLAN ID by default. You can customize the DHCP option used to request the VLAN ID.

Procedure

VLAN can be configured using the configuration files or locally.

Configuration File	<y000000000xx>.cfg	<p>Configure VLAN for the Internet port and PC port manually.</p> <p>Configure DHCP VLAN discovery feature.</p> <p>For more information, refer to VLAN on page 362.</p>
Local	Web User Interface	<p>Configure VLAN for the Internet port and PC port.</p> <p>Configure DHCP VLAN discovery feature.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network-adv&q=load">http://<phoneIPAddress>/servlet?p=network-adv&q=load</p>
	Phone User Interface	<p>Configure VLAN for the Internet port and PC port.</p>

To configure VLAN for Internet port via web user interface:

1. Click on **Network->Advanced**.
2. In the **VLAN** block, select the desired value from the pull-down list of **WAN Port Active**.
3. Enter the VLAN ID in the **VID (1-4094)** field.

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

The screenshot shows the Yealink T28 web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. Under the 'VLAN' section, the 'PC Port Active' checkbox is checked. The 'Priority' dropdown menu is set to '1'. Other settings include: LLDP (Active, Enabled), Packet Interval (60), WAN Port (Active, Enabled, VID 77, Priority 1), PC Port (Active, Disabled, VID 0, Priority 0), DHCP VLAN (Active, Enabled, Option 132), and Port Link (WAN and PC Port Links set to Auto Negotiate). A 'NOTE' sidebar on the right explains VLAN, QoS, and Local RTP Port settings.

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

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

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

To configure VLAN for PC port via web user interface:

- Click on **Network->Advanced**.
- In the **VLAN** block, select the desired value from the pull-down list of **PC Port Active**.
- Enter the VLAN ID in the **VID (1-4094)** field.
- Select the desired value (0-7) from the pull-down list of **Priority**.

This screenshot is identical to the one above, showing the Yealink T28 web interface with the 'VLAN' configuration page. The 'PC Port Active' checkbox is checked, and the 'Priority' dropdown is set to '1'. The interface shows various network settings and a 'NOTE' sidebar.

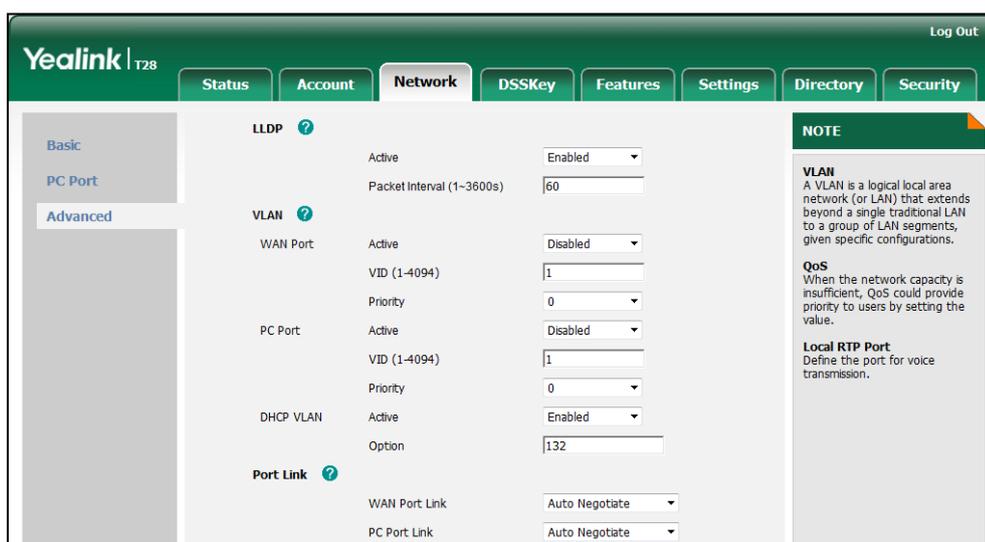
- Click **Confirm** to accept the change.

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

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

To configure DHCP VLAN discovery via web user interface:

1. Click on **Network->Advanced**.
2. In the **VLAN** block, select the desired value from the pull-down list of **DHCP VLAN Active**.
3. Enter the desired option in the **Option** field.
The default option is 132.



4. Click **Confirm** to accept the change.
A dialog box pops up to prompt that settings will take effect after a reboot.
5. 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 (1-4094) 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 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 has become more prevalent due to benefits of scalability, reliability, convenience and security. VPN

provides remote offices or individual users with secure access to their organization's network. There are two types of VPN access: remote-access VPN (connecting an individual device to a network) and site-to-site VPN (connecting two networks together). Remote-access VPN allows employees to access their company's intranet from home or outside the office, and site-to-site VPN allows employees in geographically separated offices to share one cohesive virtual network. VPN can be also classified by the protocols used to tunnel the traffic. It provides security through tunneling protocols: IPSec, SSL, L2TP and PPTP.

IP phones support SSL VPN, which provides remote-access VPN capabilities through SSL. OpenVPN is a full featured SSL VPN software solution that creates secure connections in remote access facilities, designed to work with the TUN/TAP virtual network interface. TUN and TAP are virtual network kernel devices. TAP simulates a link layer device and provides a virtual point-to-point connection, while TUN simulates a network layer device and provides a virtual network segment. IP phones use OpenVPN to achieve VPN feature. To prevent disclosure of private information, tunnel endpoints must authenticate each other before secure VPN tunnel is established. After VPN feature is configured properly on the IP phone, the IP phone acts as a VPN client and uses the certificates to authenticate the VPN server.

To use VPN, the compressed package of VPN-related files should be uploaded to the IP phone in advance. The file format of the compressed package must be *.tar. The related VPN files are: certificates (ca.crt and client.crt), key (client.key) and the configuration file (vpn.cnf) of the VPN client. For more information on how to package a TAR file, refer to *OpenVPN Feature on Yealink IP Phones*.

Note VPN feature is not applicable to SIP-T19P IP phones.

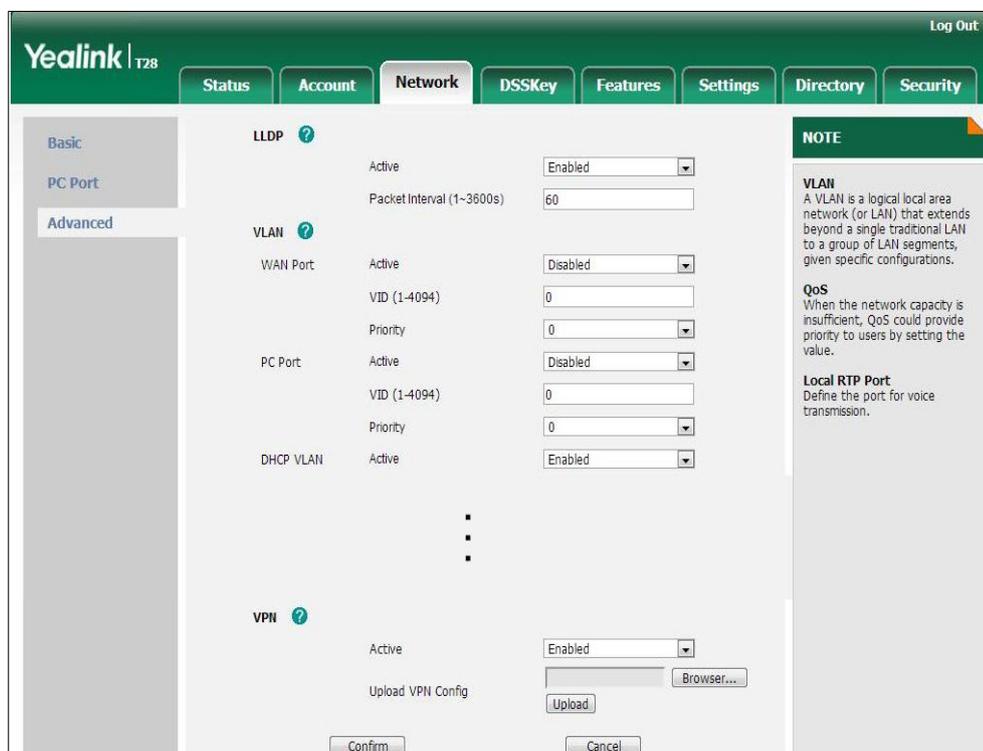
Procedure

VPN can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure VPN feature and upload a TAR file to the IP phone. For more information, refer to VPN on page 365.
Local	Web User Interface	Configure VPN feature and upload a TAR package to the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure VPN feature.

To upload a TAR file and configure VPN via web user interface:

1. Click on **Network->Advanced**.
2. Click **Browse** to locate the TAR file from the local system.
3. Click **Upload** to upload the TAR file.



The web user interface prompts the message "Import config...".

4. In the **VPN** block, select the desired value from the pull-down list of **Active**.
5. Click **Confirm** to accept the change.

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

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

To configure VPN via phone user interface after uploading a TAR file:

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.

You must upload the OpenVPN TAR file using configuration files or via web user interface in advance.

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

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

Quality of Service

Quality of Service (QoS) is the ability to provide different priorities for different packets in the network, allowing the transport of traffic with special requirements. QoS guarantees are important for applications that require fixed bit rate and are delay sensitive when the network capacity is insufficient. There are four major QoS factors to be considered when configuring a modern QoS implementation: bandwidth, delay, jitter and loss.

QoS provides better network service through the following features:

- Supporting dedicated bandwidth
- Improving loss characteristics
- Avoiding and managing network congestion
- Shaping network traffic
- Setting traffic priorities across the network

The Best-Effort service is the default QoS model in IP networks. It provides no guarantees for data delivering, which means delay, jitter, packet loss and bandwidth allocation are unpredictable. Differentiated Services (DiffServ or DS) is the most widely used QoS model. It provides a simple and scalable mechanism for classifying and managing network traffic and providing QoS on modern IP networks. Differentiated Services Code Point (DSCP) is used to define DiffServ classes and stored in the first six bits of the ToS (Type of Service) field. Each router on the network can provide QoS simply based on the DiffServ class. The DSCP value ranges from 0 to 63 with each DSCP specifying a particular per-hop behavior (PHB) applicable to a packet. A PHB refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet.

Four standard PHBs available to construct a DiffServ-enabled network and achieve QoS:

- **Class Selector PHB** -- backwards compatible with IP precedence. Class Selector code points are of the form "xxx000". The first three bits are the IP precedence bits. These class selector PHBs retain almost the same forwarding behavior as nodes that implement IP precedence-based classification and forwarding.
- **Expedited Forwarding PHB** -- the key ingredient in DiffServ model for providing a low-loss, low-latency, low-jitter and assured bandwidth service.
- **Assured Forwarding PHB** -- defines a method by which BAs (Bandwidth Allocations) can be given different forwarding assurances.
- **Default PHB** -- specifies that a packet marked with a DSCP value of "000000" gets the traditional best effort service from a DS-compliant node.

VoIP is extremely bandwidth- and delay-sensitive. QoS is a major issue in VoIP implementations, regarding how to guarantee that packet traffic not be delayed or

dropped due to interference from other lower priority traffic. VoIP can guarantee high-quality QoS only if the voice and the SIP packets are given priority over other kinds of network traffic. IP phones support the DiffServ model of QoS.

Voice QoS

In order to make VoIP transmissions intelligible to receivers, voice packets should not be dropped, excessively delayed, or made to suffer varying delay. DiffServ model can guarantee high-quality voice transmission when the voice packets are configured to a higher DSCP value.

SIP QoS

SIP protocol is used for creating, modifying and terminating two-party or multi-party sessions. To ensure good voice quality, SIP packets emanated from IP phones should be configured with a high transmission priority.

DSCPs for voice and SIP packets can be specified respectively.

Procedure

QoS can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the DSCPs for voice packets and SIP packets. For more information, refer to QoS on page 366 .
Local	Web User Interface	Configure the DSCPs for voice packets and SIP packets. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load

To configure DSCPs for voice packets and SIP packets via web user interface:

1. Click on **Network->Advanced**.
2. Enter the desired value in the **Voice QoS (0~63)** field.

- Enter the desired value in the **SIP QoS (0~63)** field.

The screenshot shows the Yealink T28 web interface with the 'Network' tab selected. The 'Voice QoS' section is expanded, showing the following settings:

Setting	Value
LLDP	Active: Enabled
Packet Interval (1~3600s)	60
VLAN	Active: Disabled
WAN Port	Active: Disabled
VID (1-4094)	0
Priority	0
PC Port	Active: Disabled
VID (1-4094)	0
Priority	0
DHCP VLAN	Active: Enabled
Option	132
Port Link	WAN Port Link: Auto Negotiate
PC Port Link	Auto Negotiate
Voice QoS (0~63)	46
SIP QoS (0~63)	26

A 'NOTE' box on the right side of the interface provides information about VLAN and QoS settings.

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

Network Address Translation

Network Address Translation (NAT) is essentially a translation table that maps public IP address and port combinations to private ones. This reduces the need for a large number of public IP addresses. NAT ensures security since each outgoing or incoming request must first go through a translation process. But in the VoIP environment, NAT breaks end-to-end connectivity.

NAT Traversal

NAT traversal is a general term for techniques that establish and maintain IP connections traversing NAT gateways, typically required for client-to-client networking applications, especially for VoIP deployments. STUN is one of the NAT traversal techniques supported by IP phones.

STUN (Simple Traversal of UDP over NATs)

STUN is a network protocol, used in NAT traversal for applications of real-time voice, video, messaging, and other interactive IP communications. The STUN protocol allows applications to operate behind a NAT to discover the presence of the network address translator, and to obtain the mapped (public) IP address and port number that the NAT has allocated for the UDP connections to remote parties. The protocol requires

assistance from a third-party network server (STUN server) usually located on public Internet. The IP phone can be configured to act as a STUN client, to send exploratory STUN messages to the STUN server. The STUN server uses those messages to determine the public IP address and port used, and then informs the client.

The NAT traversal and STUN server are configurable on a per-line basis.

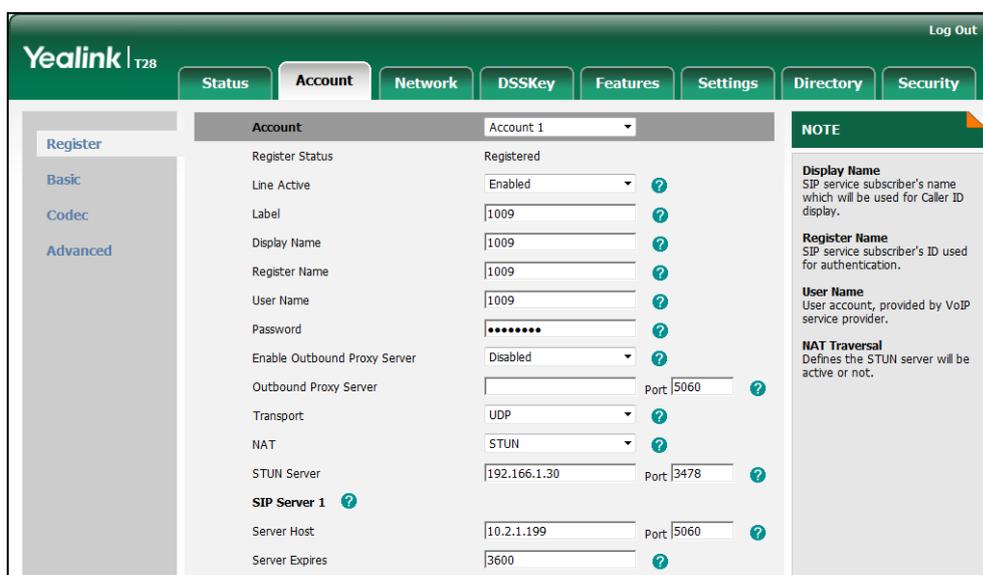
Procedure

NAT traversal and STUN server can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure NAT traversal and STUN server on the IP phone. For more information, refer to Network Address Translation on page 367.
Local	Web User Interface	Configure NAT traversal and STUN server on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

To configure NAT traversal and STUN server via web user interface:

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **STUN** from the pull-down list of **NAT**.
4. Enter the IP address or the domain name of the STUN server 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), part of the IEEE 802.1 group of networking protocols. It offers an authentication mechanism for devices to connect/link to a LAN or WLAN. The 802.1X authentication involves three parties: a supplicant, an authenticator and an authentication server. The supplicant is the IP phone that wishes to attach to the LAN or WLAN. With 802.1X port-based authentication, the IP phone provides credentials, such as user name and password, for the authenticator, and then the authenticator forwards the credentials to the authentication server for verification. If the authentication server determines the credentials are valid, the IP phone is allowed to access resources located on the protected side of the network.

IP phones support protocols EAP-MD5, EAP-TLS, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 for 802.1X authentication.

Procedure

802.1X authentication can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the 802.1X authentication. For more information, refer to 802.1X on page 368.
Local	Web User Interface	Configure the 802.1X authentication. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure the 802.1X authentication.

To configure the 802.1X authentication via web user interface:

1. Click on **Network->Advanced**.
2. In the **802.1x** block, select the desired protocol from the pull-down list of **802.1x Mode**.
 - a) If you select **EAP-MD5**:
 - 1) Enter the user name for authentication in the **Identity** field.

- 2) Enter the password for authentication in the **MD5 Password** field.

The screenshot shows the Yealink T28 web interface with the 'Network' tab selected. The left sidebar has 'Advanced' selected. The main content area is divided into sections: LLDP, 802.1x, and VPN. In the 802.1x section, the '802.1x Mode' is set to 'EAP-MD5', 'Identity' is 'yealink', and 'MD5 Password' is masked with asterisks. There are 'Browse...' buttons for 'CA Certificates' and 'Device Certificates'. The 'LLDP' section has 'Active' set to 'Enabled' and 'Packet Interval (1-3600s)' set to '60'. The 'VPN' section has 'Active' set to 'Disabled' and an 'Upload VPN Config' button. A 'NOTE' box on the right explains VLAN and QoS. At the bottom, there are 'Confirm' and 'Cancel' buttons.

b) If you select EAP-TLS:

- 1) Enter the user name for authentication in the **Identity** field.
- 2) Leave the **MD5 Password** field blank.
- 3) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
- 4) In the **Device Certificates** field, click **Browse** to select the desired client (*.pem or *.cer) certificate from your local system.

5) Click **Upload** to upload the certificates.

The screenshot shows the Yealink T28 Network configuration page. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. The configuration is divided into several sections:

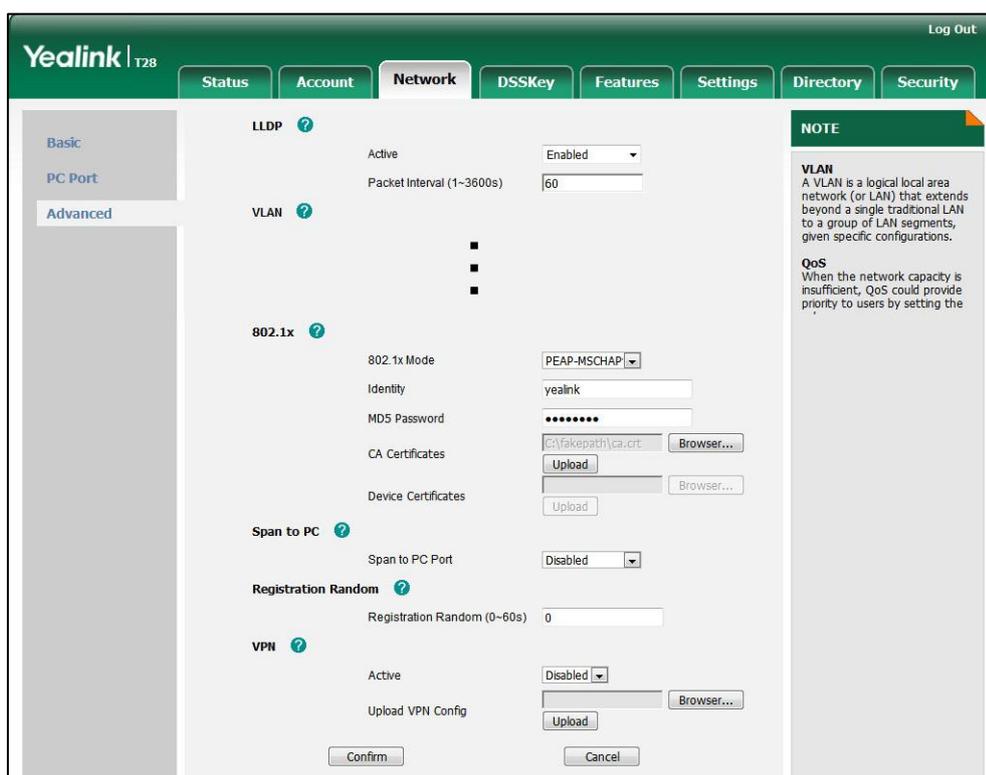
- LLDP:** Active (Enabled), Packet Interval (1-3600s) (60).
- VLAN:** WAN Port (Active, Disabled).
- 802.1x:**
 - 802.1x Mode: EAP-TLS
 - Identity: (empty)
 - MD5 Password: (masked with dots)
 - CA Certificates: C:\fakepath\ca.crt (with Upload and Browser... buttons)
 - Device Certificates: I:\fakepath\client.pem (with Upload and Browser... buttons)
- VPN:** Active (Disabled), Upload VPN Config (with Upload and Browser... buttons).

At the bottom, there are 'Confirm' and 'Cancel' buttons. A 'NOTE' box on the right explains VLAN and QoS.

c) If you select **PEAP-MSCHAPv2**:

- 1) Enter the user name for authentication in the **Identity** field.
- 2) Enter the password for authentication in the **MD5 Password** field.
- 3) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.

4) Click **Upload** to upload the certificate.



d) If you select **EAP-TTLS/EAP-MSCHAPv2**:

- 1) Enter the user name for authentication in the **Identity** field.
- 2) Enter the password for authentication in the **MD5 Password** field.
- 3) In the **CA Certificates** field, click **Browse** to select the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.

- 4) Click **Upload** to upload the certificate.

The screenshot shows the Yealink T28 web interface with the following settings visible:

- LLDP:** Active (Enabled), Packet Interval (1-3600s) (60)
- VLAN:** (Three empty checkboxes)
- 802.1x:**
 - 802.1x Mode: EAP-TTLS/EAP
 - Identity: yealink
 - MD5 Password: [Masked]
 - CA Certificates: C:\fakepath\ca.crt (Upload)
 - Device Certificates: (Upload)
- Span to PC:** Span to PC Port (Disabled)
- Registration Random:** Registration Random (0-60s) (0)
- VPN:** Active (Disabled), Upload VPN Config (Upload)

Buttons: Confirm, Cancel

NOTE:
VLAN
 A VLAN is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.
QoS
 When the network capacity is insufficient, QoS could provide priority to users by setting the...

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

To configure the 802.1X authentication via phone user interface after:

- Press **Menu->Settings->Advanced Settings** (password: admin)
 ->**Network->802.1x Settings**.
- Press **◀** or **▶** , or the **Switch** soft key to select the desired value from the **802.1x Mode** field.
 - If you select **EAP-MD5**:
 - Enter the user name for authentication in the **Identity** field.
 - Enter the password for authentication in the **MD5 Password** field.
 - If you select **EAP-TLS**:
 - Enter the user name for authentication in the **Identity** field.
 - Leave the **MD5 Password** field blank.
 - If you select **PEAP-MSCHAPv2**:
 - Enter the user name for authentication in the **Identity** field.
 - Enter the password for authentication in the **MD5 Password** field.
 - If you select **EAP-TTLS/EAP-MSCHAPv2**:
 - Enter the user name for authentication in the **Identity** field.

- 2) Enter the password for authentication in the **MD5 Password** field.
3. Click **Save** to accept the change.
The IP phone reboots automatically to make the settings effective after a period of time.

TR-069 Device Management

TR-069 is a technical specification defined by the Broadband Forum, which defines a mechanism that encompasses secure auto-configuration of a CPE (Customer-Premises Equipment), and incorporates other CPE management functions into a common framework. TR-069 uses common transport mechanisms (HTTP and HTTPS) for communication between CPE and ACS (Auto Configuration Servers). The HTTP(S) messages contain XML-RPC methods defined in the standard for configuration and management of the CPE.

TR-069 is intended to support a variety of functionalities to manage a collection of CPEs, including the following primary capabilities:

- Auto-configuration and dynamic service provisioning
- Software or firmware image management
- Status and performance monitoring
- Diagnostics

The following table provides a description of RPC methods supported by IP phones.

RPC Method	Description
GetRPCMethods	This method is used to discover the set of methods supported by the CPE.
SetParameterValues	This method is used to modify the value of one or more CPE parameters.
GetParameterValues	This method is used to obtain the value of one or more CPE parameters.
GetParameterNames	This method is used to discover the parameters accessible on a particular CPE.
GetParameterAttributes	This method is used to read the attributes associated with one or more CPE parameters.
SetParameterAttributes	This method is used to modify attributes associated with one or more CPE parameters.
Reboot	This method causes the CPE to reboot.
Download	This method is used to cause the CPE to download a specified file from the designated location.

RPC Method	Description
	File types supported by IP phones are: <ul style="list-style-type: none"> • Firmware Image • Configuration File
Upload	This method is used to cause the CPE to upload a specified file to the designated location. File types supported by IP phones are: <ul style="list-style-type: none"> • Configuration File • Log File
ScheduleInform	This method is used to request the CPE to schedule a one-time Inform method call (separate from its periodic Inform method calls) sometime in the future.
FactoryReset	This method resets the CPE to its factory default state.
TransferComplete	This method informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.
AddObject	This method is used to add a new instance of an object defined on the CPE.
DeleteObject	This method is used to remove a particular instance of an object.

Procedure

TR-069 can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure TR-069 feature. For more information, refer to TR-069 on page 370.
Local	Web User Interface	Configure TR-069 feature. Navigate to: http://<phoneIPAddress>/servl et?p=settings-preference&q=lo ad

To configure TR-069 via web user interface:

1. Click on **Settings->TR069**.
2. Select **Enabled** from the pull-down list of **Enable TR069**.
3. Enter the user name and password authenticated by the ACS in the **ACS Username** and **ACS Password** fields.

4. Enter the URL of the ACS in the **ACS URL** field.
5. Select the desired value from the pull-down list of **Enable Periodic Inform**.
6. Enter the desired time in the **Periodic Inform Interval (seconds)** field.
7. Enter the user name and password authenticated by the IP phone in the **Connection Request Username** and **Connection Request Password** fields.

The screenshot shows the Yealink T28 web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Settings' tab is active, and the 'TR069' configuration page is displayed. The page has a left sidebar with various settings categories like 'Preference', 'Time & Date', 'Upgrade', 'Auto Provision', 'Configuration', 'Dial Plan', 'Voice', 'Ring', 'Tones', 'Softkey Layout', and 'TR069'. The main content area shows the TR069 configuration fields: 'Enable TR069' (Enabled), 'ACS Username' (admin), 'ACS Password' (masked), 'ACS URL' (192.168.1.30), 'Enable Periodic Inform' (Enabled), 'Periodic Inform Interval (seconds)' (60), 'Connection Request Username' (yealink), and 'Connection Request Password' (masked). There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' section on the right states: 'NOTE: TR069 The TR069 parameters for administrator.'

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

IPv6 Support

IPv6 is the next generation network layer protocol, designed as a replacement for the current IPv4 protocol. IPv6 is developed by the Internet Engineering Task Force (IETF) to deal with the long-anticipated problem of IPv4 address exhaustion. IPv6 uses a 128-bit address, consisting of eight groups of four hexadecimal digits separated by colons. VoIP network based on IPv6 can ensure QoS, a set of service requirements to deliver performance guarantee while transporting traffic over the network.

IPv6 Address Assignment Method

Supported IPv6 address assignment methods:

- **Manual Assignment:** An IPv6 address and other configuration parameters (e.g., DNS server) for the IP phone can be statically configured by an administrator.
- **Stateless Address Autoconfiguration (SLAAC):** SLAAC is one of the most convenient methods to assign IP addresses to IPv6 nodes. SLAAC requires no manual configuration of the IP phone, minimal (if any) configuration of routers, and no additional servers. To use IPv6 SLAAC, the IP phone must be connected to a network with at least one IPv6 router connected. This router is configured by the network administrator and sends out Router Advertisement announcements onto the link. These announcements can allow the on-link connected IP phone to configure itself with IPv6 address, as specified in RFC 4862.

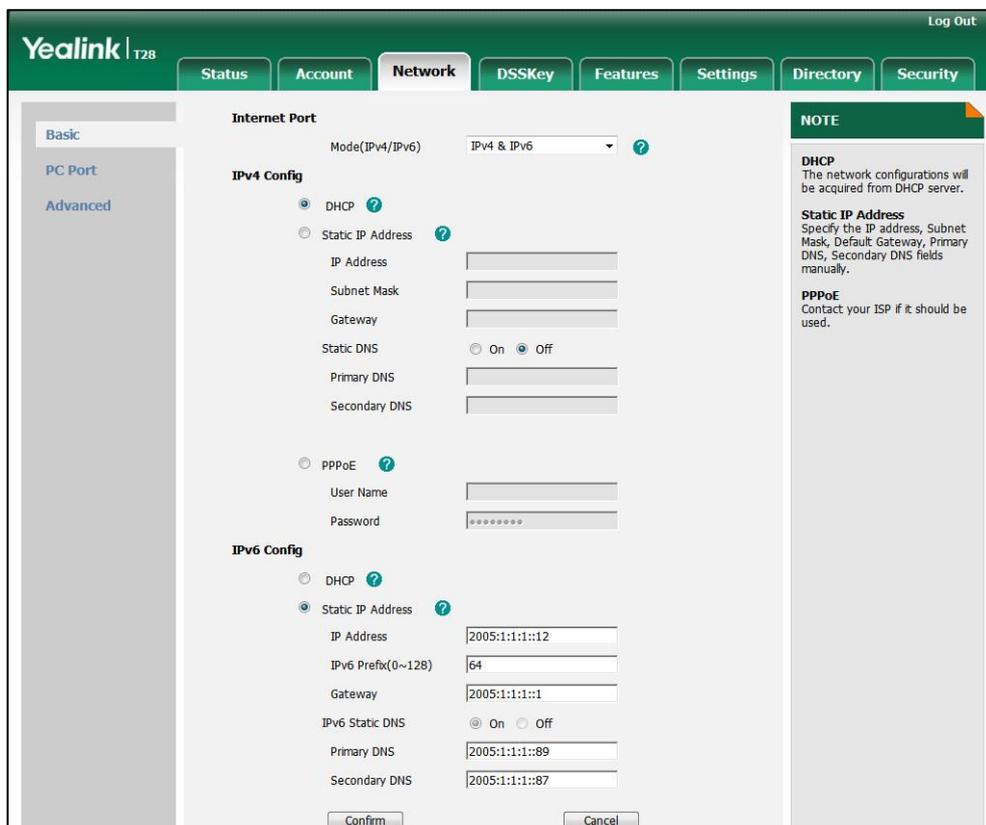
Procedure

IPv6 can be configured using the configuration files or locally.

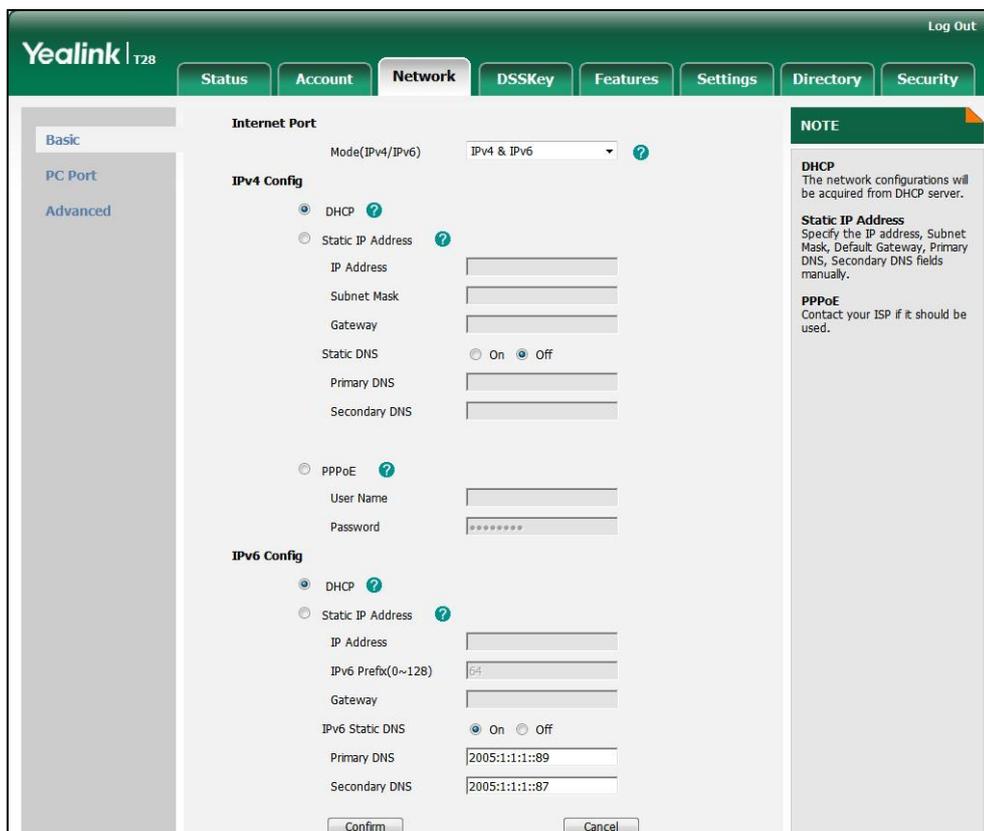
<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the IPv6 address assignment method. For more information, refer to IPv6 on page 372.</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the IPv6 address assignment method. Navigate to: http://<phoneIPAddress>/servlet?&p=network&q=load</p>

To configure IPv6 address assignment method via web user interface:

1. Click on **Network->Basic**.
2. Select the desired address mode (IPv6 or IPv4&IPv6) from the pull-down list of **Mode (IPv4/IPv6)**.
3. In the **IPv6 Config** block, do one of the following.
 - If you mark the **Static IP Address** radio box, configure the IPv6 address and other configuration parameters in the corresponding fields.



- (Optional.) If you mark the **DHCP** radio box, you can configure the static DNS address in the corresponding fields.



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

To configure IPv6 address assignment method via phone user interface:

1. Press **Menu->Settings->Advanced Settings** (password: admin) ->**Network->WAN Port**.
2. Press **←** or **→** to select **IPv4&IPv6** or **IPv6** from the **IP Mode** field.
3. Press **↑** or **↓** to highlight **IPv6** and press the **Enter** soft key.
4. Press **↑** or **↓** to select the desired IPv6 address assignment method.
If you select the **Static IPv6 Client**, configure the IPv6 address and other network parameters in the corresponding fields.
5. Press the **Save** soft key to accept the change
The IP phone reboots automatically to make settings effective after a period of time.

Configuring Audio Features

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

- [Headset Prior](#)
- [Dual Headset](#)
- [Audio Codecs](#)
- [Acoustic Clarity Technology](#)

Headset Prior

Headset prior allows users to use headset preferentially if a headset is physically connected to the IP phone. This feature is especially useful for permanent or full-time headset users.

Procedure

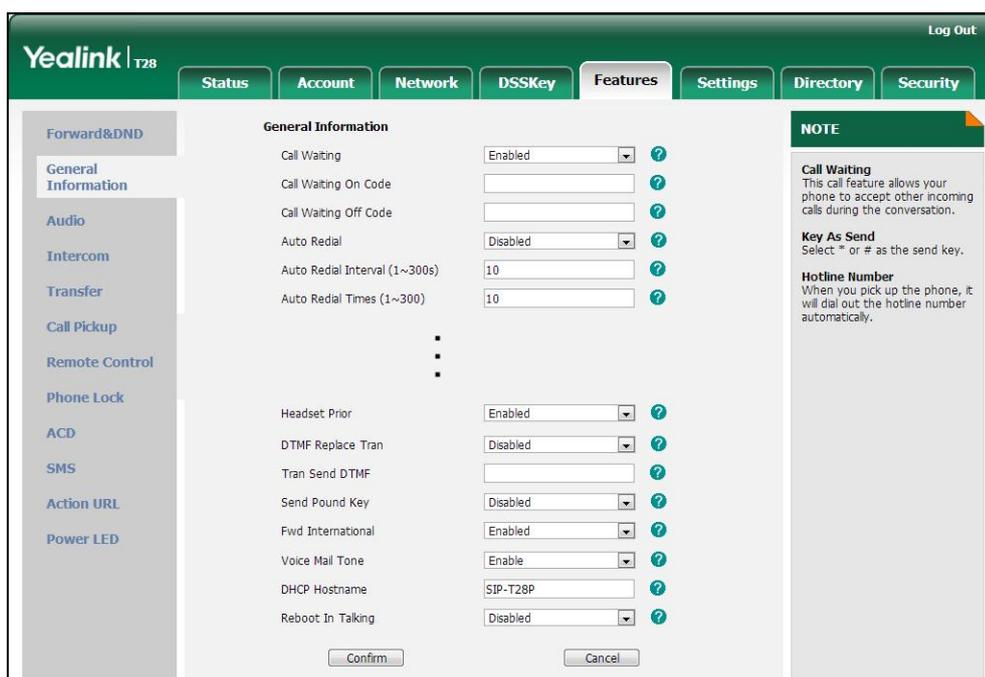
Headset prior can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure headset prior. For more information, refer to Head Prior on page 376.
Local	Web User Interface	Configure headset prior. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

To configure headset prior via web user interface:

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

2. Select the desired value from the pull-down list of **Headset Prior**.



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

Dual Headset

Dual headset allows users to use two headsets on one IP phone. To use this feature, users need to physically connect two headsets to the headset and handset jacks respectively. Once the phone connects to a call, the user with the headset connected to the headset jack has full-duplex capabilities, while the user with the headset connected to the handset jack is only able to listen. This feature is not applicable to SIP-T19P and SIP-T21P IP phones.

Procedure

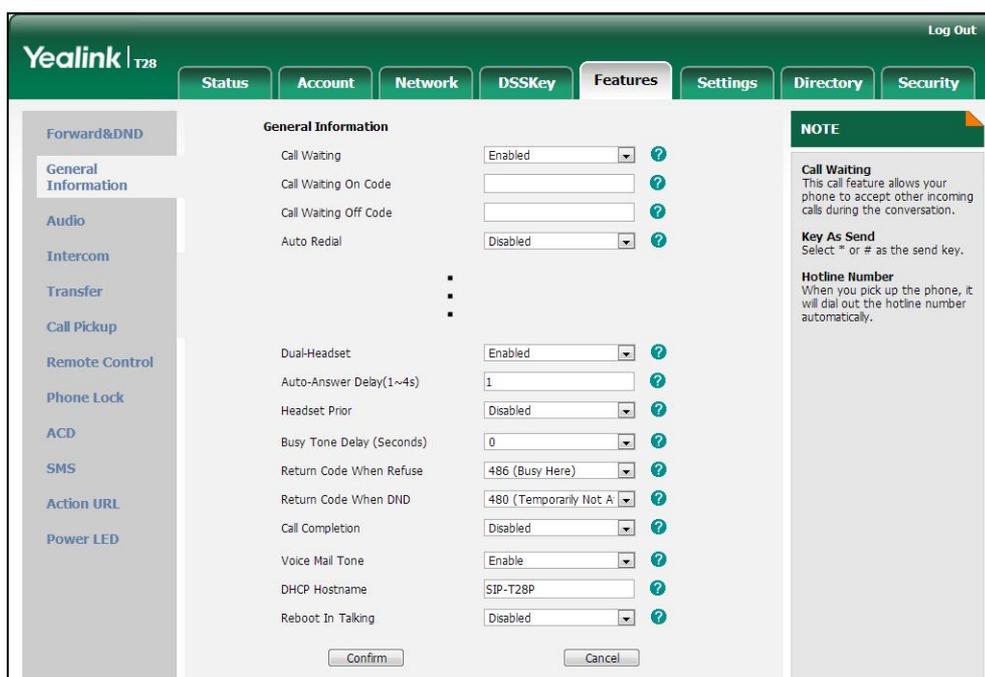
Dual headset can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure dual headset. For more information, refer to Dual Headset on page 376.
Local	Web User Interface	Configure dual headset. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

To configure dual headset via web user interface:

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

- Select the desired value from the pull-down list of **Dual-Headset**.



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

Audio Codecs

CODEC is an abbreviation of COmpress-DECompress, capable of coding or decoding a digital data stream or signal by implementing an algorithm. The object of the algorithm is to represent the high-fidelity audio signal with minimum number of bits while retaining the quality. This can effectively reduce the frame size and the bandwidth required for audio transmission.

The default codecs used on IP phones are summarized in the following table:

Codec	Algorithm	Bit Rate	Sample Rate	Packetization Time
PCMA	G.711 a-law	64 Kbps	8 Ksps	20ms
PCMU	G.711 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, IP phones also support codecs: G723_53, G723_63, G726-16, G726-24, G726-32, G726-40 (Codecs G726-16, G726-24 and G726-40 are not applicable to SIP-T21P and SIP-T19P IP phones). Codecs and priorities of these codecs are configurable on a per-line basis. The attribute "rtptime" is used to define a mapping from RTP payload codes to a codec, clock rate and other encoding parameters.

The corresponding attributes of the codec are listed as follows:

Codec	Configuration Methods	Priority	RTPmap
PCMU	Configuration Files Web User Interface	1	0
PCMA	Configuration Files Web User Interface	2	8
G729	Configuration Files Web User Interface	3	18
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	103
G726-24	Configuration Files Web User Interface	0	104
G726-32	Configuration Files Web User Interface	0	102
G726-40	Configuration Files Web User Interface	0	105
iLBC	Configuration Files Web User Interface	0	106

Packetization Time

Ptime (Packetization Time) is a measurement of the duration (in milliseconds) of the audio data in each RTP packet sent to the destination, and defines how much network bandwidth is used for the RTP stream transfer. Before establishing a conversation, codec and ptime are negotiated through SIP signaling. The valid values of ptime range from 10 to 60, in increments of 10 milliseconds. The default ptime is 20ms. You can also disable the ptime negotiation.

Procedure

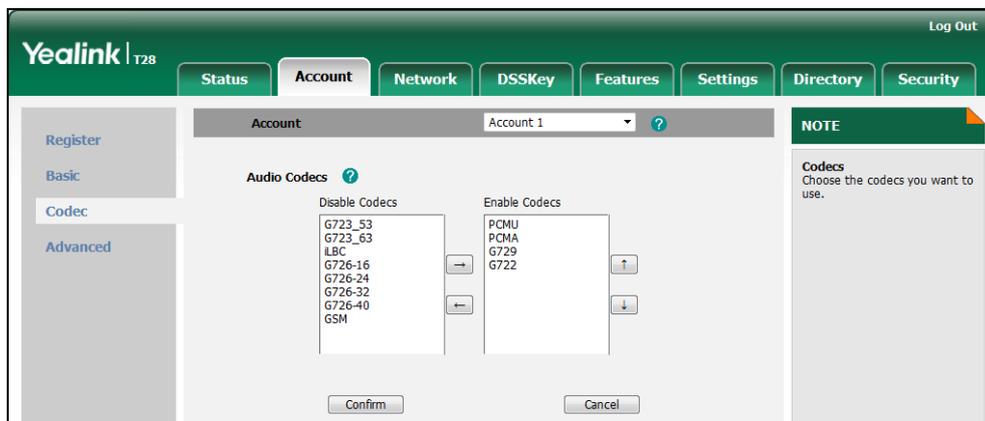
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Configure the codecs to use on a per-line basis.</p> <p>Configure the priority and rtpmap for the enabled codec.</p> <p>For more information, refer to Audio Codecs on page 377.</p> <p>Configure the ptime.</p> <p>For more information, refer to Audio Codecs on page 377.</p>
Local	Web User Interface	<p>Configure the codecs to use and adjust the priority of the enabled codecs on a per-line basis.</p> <p>Configure the ptime.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-codec&q=load&acc=0</p>

To configure the codecs to use and adjust the priority of the enabled codecs on a per-line basis via web user interface:

1. Click on **Account**.
2. Select the desired account from the pull-down list of **Account**.
3. Click on **Codec**.
4. Select the desired codec from the **Disable Codecs** column and then click  .
The selected codec appears in the **Enable Codecs** column.
5. Repeat the step 4 to add more codecs to the **Enable Codecs** column.
6. To remove the codec from the **Enable Codecs** column, select the desired codec and then click  .

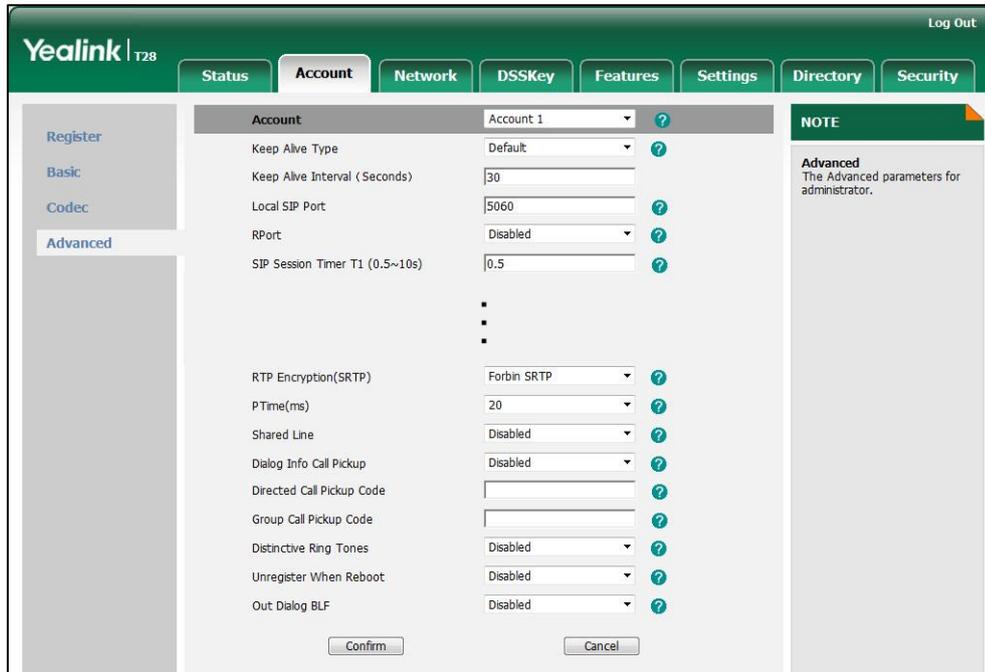
- To adjust the priority of codecs, select the desired codec and then click  or  .



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

To configure the ptime on a per-line basis via web user interface:

- Click on **Account**.
- Select the desired account from the pull-down list of **Account**.
- Click on **Advanced**.
- Select the desired value from the pull-down list of **PTime (ms)**.



- 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. IP phones employ advanced AEC for hands-free operation. Echo cancellation is achieved using the echo canceller.

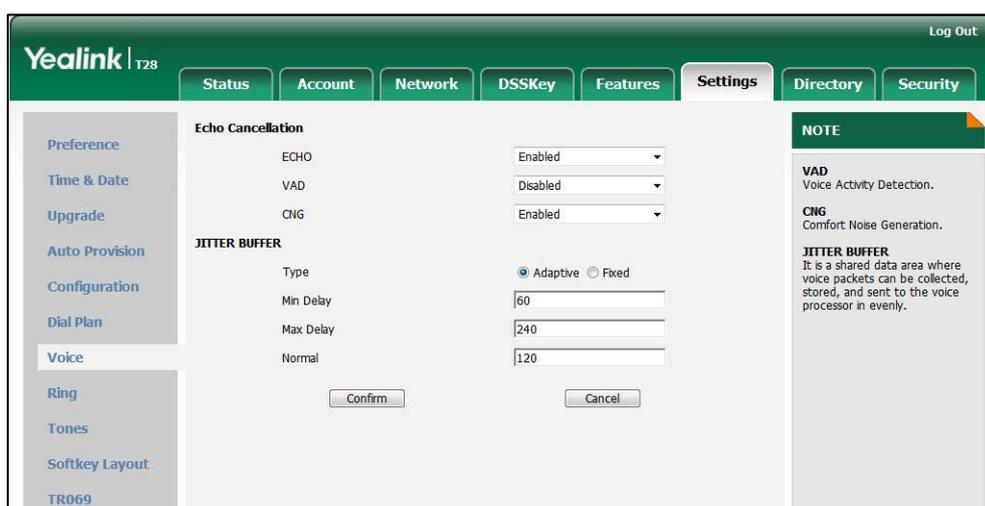
Procedure

AEC can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure AEC. For more information, refer to Acoustic Echo Cancellation on page 381.
Local	Web User Interface	Configure AEC. Navigate to: http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure AEC via web user interface:

1. Click on **Settings->Voice**.
2. Select the desired value from the pull-down list of **ECHO**.



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 deactivate some processes during non-speech section of an audio session. VAD can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth.

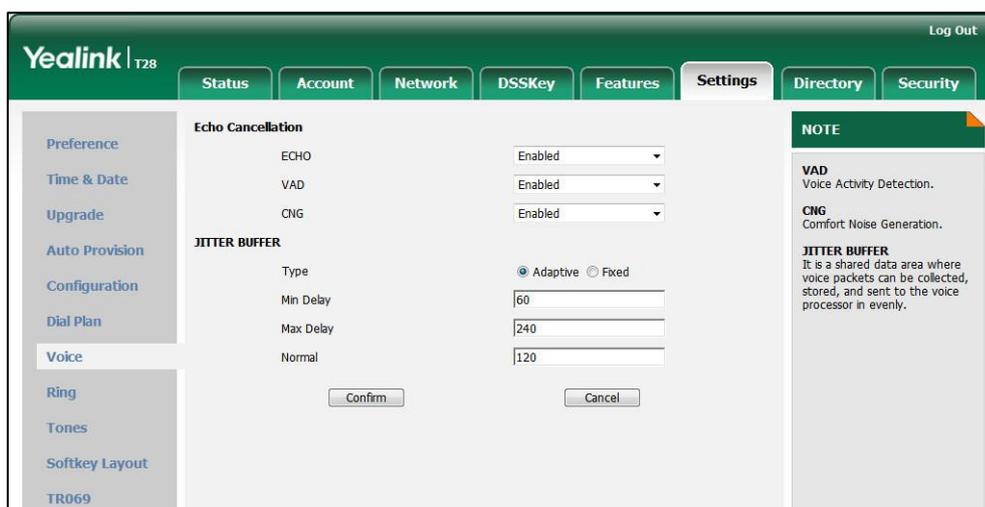
Procedure

VAD can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure VAD. For more information, refer to Voice Activity Detection on page 382.
Local	Web User Interface	Configure VAD. Navigate to: http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure VAD via web user interface:

1. Click on **Settings->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 in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that background sound is consistent throughout the call and the listener does not think the line has released. The purpose of VAD and CNG is to maintain an acceptable perceived QoS while simultaneously keeping transmission costs and bandwidth usage as low as possible.

Procedure

CNG can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure CNG. For more information, refer to Comfort Noise Generation on page 382 .
Local	Web User Interface	Configure CNG. Navigate to: http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure CNG via web user interface:

1. Click on **Settings->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 even intervals. Jitter is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. IP phones support two types of jitter buffers: static and dynamic. A static jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. 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 be also configured on IP phones.

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 382.
Local	Web User Interface	Configure the mode of jitter buffer and the delay time for jitter buffer. Navigate to: http://<phoneIPAddress>/servlet?p=settings-voice&q=load

To configure Jitter Buffer via web user interface:

1. Click on **Settings->Voice**.
2. Mark the desired radio box in the **Type** field.
3. Enter the minimum delay time for adaptive jitter buffer in the **Min Delay** field.
Valid values range from 0 to 300.
4. Enter the maximum delay time for adaptive jitter buffer in the **Max Delay** field.
Valid values range from 0 to 300.

- Enter the fixed delay time for fixed jitter buffer in the **Normal** field.
Valid values range from 0 to 300.

The screenshot shows the Yealink T28 Settings page. The 'Settings' tab is selected, and the 'Voice' section is active. The 'JITTER BUFFER' section is expanded, showing the following configuration:

Section	Parameter	Value
Echo Cancellation	ECHO	Enabled
	VAD	Disabled
	CNG	Enabled
JITTER BUFFER	Type	<input checked="" type="radio"/> Adaptive <input type="radio"/> Fixed
	Min Delay	60
	Max Delay	240
	Normal	120

Buttons for 'Confirm' and 'Cancel' are visible at the bottom of the configuration area. A 'NOTE' section on the right provides additional information:

VAD
Voice Activity Detection.

CNG
Comfort Noise Generation.

JITTER BUFFER
It is a shared data area where voice packets can be collected, stored, and sent to the voice processor in evenly.

- 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

TLS is a commonly-used protocol for providing communications privacy and managing the security of message transmission, allowing IP phones to communicate with other remote parties and connect to the HTTPS URL for provisioning in a way that is designed to prevent eavesdropping and tampering.

TLS protocol is composed of two layers: TLS Record Protocol and TLS Handshake Protocol. The TLS Record Protocol completes the actual data transmission and ensures the integrity and privacy of the data. The TLS Handshake Protocol allows the server and client to authenticate each other and negotiate an encryption algorithm and cryptographic keys before data is exchanged.

The TLS protocol uses asymmetric encryption for authentication of key exchange, symmetric encryption for confidentiality, and message authentication codes for integrity.

- **Symmetric encryption:** For symmetric encryption, the encryption key and the corresponding decryption key can be told by each other. In most cases, the encryption key is the same as the decryption key.
- **Asymmetric encryption:** For asymmetric encryption, each user has a pair of cryptographic keys – a public encryption key and a private decryption key. The information encrypted by the public key can only be decrypted by the corresponding private key and vice versa. Usually, the receiver keeps its private key. The public key is known by the sender, so the sender sends the information encrypted by the known public key, and then the receiver uses the private key to decrypt it.

IP phones support TLS version 1.0. A cipher suite is a named combination of authentication, encryption, and message authentication code (MAC) algorithms used to negotiate the security settings for a network connection using the TLS/SSL network protocol. IP phones support the following cipher suites:

- DHE-RSA-AES256-SHA
- DHE-DSS-AES256-SHA

- AES256-SHA
- EDH-RSA-DES-CBC3-SHA
- EDH-DSS-DES-CBC3-SHA
- DES-CBC3-SHA
- DHE-RSA-AES128-SHA
- DHE-DSS-AES128-SHA
- AES128-SHA
- IDEA-CBC-SHA
- DHE-DSS-RC4-SHA
- RC4-SHA
- RC4-MD5
- EXP1024-DHE-DSS-DES-CBC-SHA
- EXP1024-DES-CBC-SHA
- EDH-RSA-DES-CBC-SHA
- EDH-DSS-DES-CBC-SHA
- DES-CBC-SHA
- EXP1024-DHE-DSS-RC4-SHA
- EXP1024-RC4-SHA
- EXP1024-RC4-MD5
- EXP-EDH-RSA-DES-CBC-SHA
- EXP-EDH-DSS-DES-CBC-SHA
- EXP-DES-CBC-SHA
- EXP-RC4-MD5

The following figure illustrates the TLS messages exchanged between the IP phone and TLS server to establish an encrypted communication channel:

No.	Time	Source	Destination	Protocol	Info
1	0.000000	192.168.3.86	192.168.0.230	SSLV3	Client Hello
2	0.021345	192.168.0.230	192.168.3.86	SSLV3	Server Hello, Certificate, Server Key Exchange, Server Hello done
3	0.954947	192.168.3.86	192.168.0.230	SSLV3	Client Key Exchange, Change Cipher Spec, Encrypted Handshake Message
4	0.970099	192.168.0.230	192.168.3.86	SSLV3	Change Cipher Spec, Encrypted Handshake Message
5	1.012295	192.168.3.86	192.168.0.230	SSLV3	Application Data, Application Data
6	1.013562	192.168.0.230	192.168.3.86	SSLV3	Application Data
7	1.013667	192.168.0.230	192.168.3.86	SSLV3	Application Data

Frame 13: 652 bytes on wire (5216 bits), 652 bytes captured (5216 bits)
 Ethernet II, Src: Vmware_72:c9:2e (00:0c:29:72:c9:2e), Dst: Xiamenye_11:12:b7 (00:15:65:11:12:b7)
 Internet Protocol, Src: 192.168.0.230 (192.168.0.230), Dst: 192.168.3.86 (192.168.3.86)
 Transmission Control Protocol, Src Port: https (443), Dst Port: nmserver (2244), Seq: 1482, Ack: 437, Len: 586
 Secure Socket Layer

Step1: IP phone sends “Client Hello” message proposing SSL options.

Step2: Server responds with “Server Hello” message selecting the SSL options, sends its public key information in “Server Key Exchange” message and concludes its part of the

negotiation with “Server Hello Done” message.

Step3: IP phone sends session key information (encrypted by server’s public key) in the “Client Key Exchange” message.

Step4: Server sends “Change Cipher Spec” message to activate the negotiated options for all future messages it will send.

IP phones can encrypt SIP with TLS, which is called SIPS. When TLS is enabled for an account, the SIP message of this account will be encrypted, and a lock icon appears on the LCD screen after the successful TLS negotiation.

Certificates

The IP phone can serve as a TLS client or a TLS server. The TLS requires the following security certificates to perform the TLS handshake:

- **Trusted Certificate:** When the IP phone requests a TLS connection with a server, the IP phone should verify the certificate sent by the server to decide whether it is trusted based on the trusted certificates list. The IP phone has 30 built-in trusted certificates. You can upload 10 custom certificates at most. The format of the trusted certificate files must be *.pem, *.cer, *.crt and *.der.
- **Server Certificate:** When clients request a TLS connection with the IP phone, the IP phone sends the server certificate to the clients for authentication. The IP phone has two types of built-in server certificates: a unique server certificate and a generic server certificate. You can only upload one server certificate to the IP phone. The old server certificate will be overridden by the new one. The format of the server certificate files must be *.pem and *.cer.
 - **A unique server certificate:** It is unique to an IP phone (based on the MAC address) and issued by the Yealink Certificate Authority (CA).
 - **A generic server certificate:** It issued by the Yealink Certificate Authority (CA). Only if no unique certificate exists, the IP phone may send a generic certificate for authentication.

The IP phone can authenticate the server certificate based on the trusted certificates list. The trusted certificates list and the server certificates list contain the default and custom certificates. You can specify the type of certificates the IP phone accepts: default certificates, custom certificates or all certificates.

Common Name Validation feature enables the IP phone to mandatorily validate the common name of the certificate sent by the connecting server.

Note

In TLS feature, we use the terms trusted and server certificate. These are also known as CA and device certificates.

Firmware upgrade from version 71 to 72 will result in update of the default server certificates.

We strongly recommend that you do not downgrade the firmware. For SIP-T20P/T22P/T26P/T28P IP phones, firmware downgrade will result in damage to SSL certificates.

Procedure

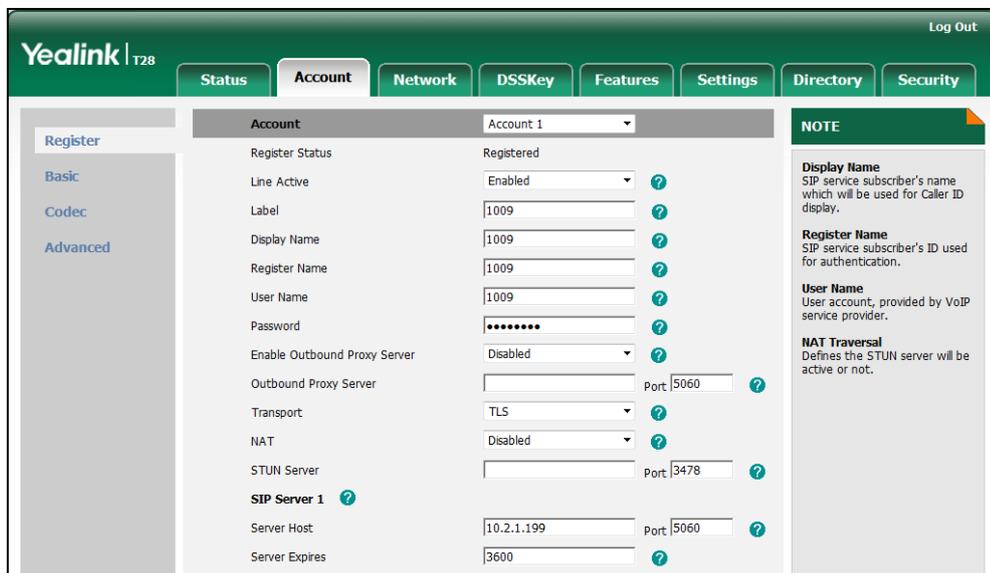
Configuration changes can be performed using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure TLS on a per-line basis. For more information, refer to TLS on page 384 .
	<y0000000000xx>.cfg	Configure trusted certificates feature. Configure server certificates feature. For more information, refer to TLS on page 384 . Upload the trusted certificates. Upload the server certificates. For more information, refer to Uploading Certificates on page 386 .
Local	Web User Interface	Configure TLS on a per-line basis. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0 Configure trusted certificates feature. Upload the trusted certificates. Navigate to: http://<phoneIPAddress>/servl

		<p>et?p=trusted-cert&q=load</p> <p>Configure server certificates feature.</p> <p>Upload the server certificates.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=server-cert&q=load</p>
--	--	---

To configure TLS on a per-line basis via web user interface:

1. Click on **Account->Register**.
2. Select the desired account from the pull-down list of **Account**.
3. Select **TLS** from the pull-down list of **Transport**.

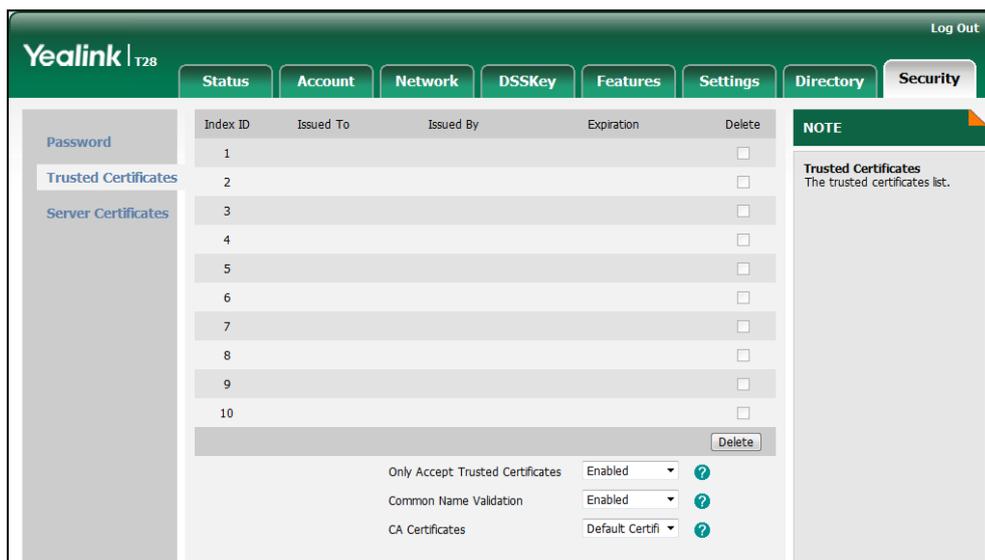


4. Click **Confirm** to accept the change.

To configure the trusted certificates via web user interface:

1. Click on **Security->Trusted Certificates**.

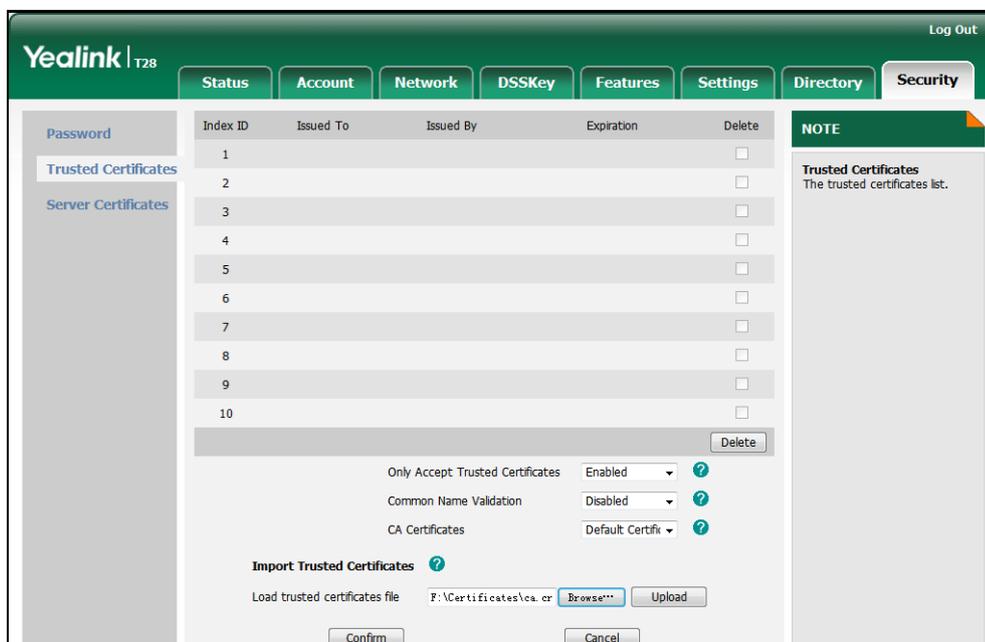
2. Select the desired values from the pull-down lists of **Only Accept Trusted Certificates**, **Common Name Validation** and **CA Certificates**.



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

To upload a trusted certificate via web user interface:

1. Click on **Security->Trusted Certificates**.
2. Click **Browse** to select the certificate (*.pem, *.crt, *.cer or *.der) from your local system.

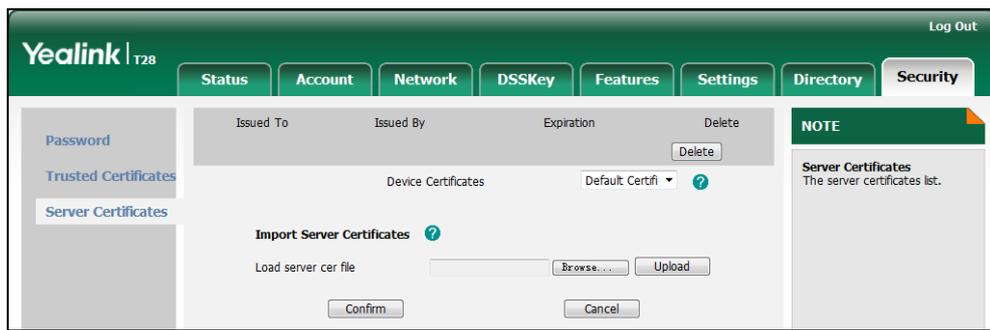


3. Click **Upload** to upload the certificate.

To configure the server certificates via web user interface:

1. Click on **Security->Server Certificates**.

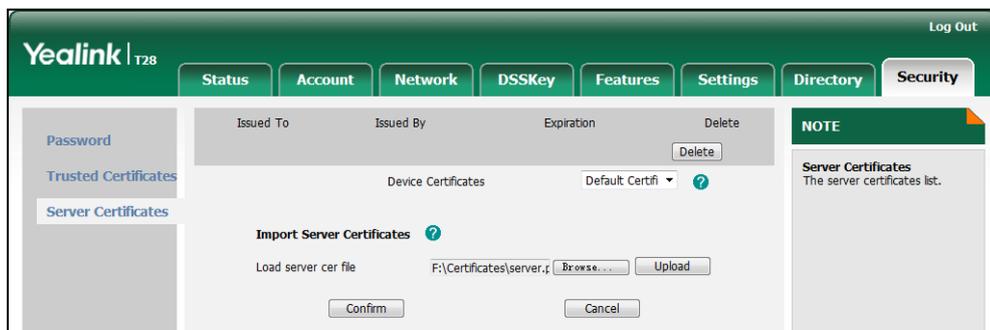
2. Select the desired value from the pull-down list of **Device Certificates**.



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

To upload a server certificate via web user interface:

1. Click on **Security->Server Certificates**.
2. Click **Browse** to select the certificate (*.pem and *.cer) from your local system.



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

A dialog box pops up to prompt "Success: The Server Certificate has been loaded! Rebooting, please wait...".

Secure Real-Time Transport Protocol

Secure Real-Time Transport Protocol (SRTP) encrypts the RTP streams during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call must enable SRTP feature simultaneously. When this feature is enabled on both phones, the type of encryption to utilize for the session is negotiated between the IP phones. This negotiation process is compliant with RFC 4568.

When a user places a call on the enabled SRTP phone, the IP phone sends an INVITE message with the RTP encryption algorithm to the destination phone.

Example of the RTP encryption algorithm carried in the SDP of the INVITE message:

```
m=audio 11780 RTP/SAVP 0 8 18 9 101
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NzFINTUwZDK2OGVIOTc3YzNkYTkWZWVhMTM1YWFj
```

```

a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:NzkyM2FjNzQ2ZDgxYjg0MzQwMGVmMGUxMzdmNWFm
a=crypto:3 F8_128_HMAC_SHA1_80 inline:NDIiMWIzZGE1ZTAwZjA5ZGFhNjQ5YmEANTMzYzA0
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, and then answers the call by responding with a 200 OK message which carries the negotiated RTP encryption algorithm.

Example of the RTP encryption algorithm carried in the SDP of the 200 OK message:

```

m=audio 11780 RTP/SAVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:NGY4OGViMDYzZjQzYTNIOTNkOWRiYzRiMjM0Yzcy
a=sendrecv
a=ptime:20
a=fmtp:101 0-15
    
```

SRTP is configurable on a per-line basis. When SRTP is enabled on both IP phones, RTP streams will be encrypted, and a lock icon appears on the LCD screen of each IP phone after 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 211.

Procedure

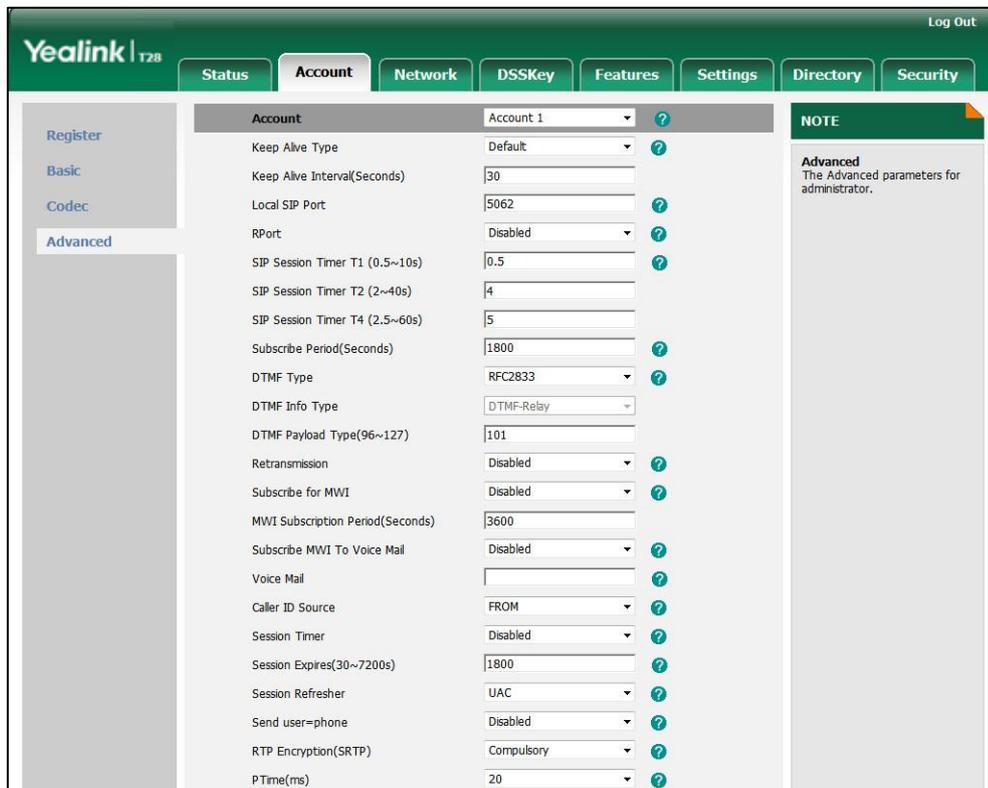
SRTP can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><MAC>.cfg</p>	<p>Configure SRTP feature on a per-line basis. For more information, refer to SRTP on page 387.</p>
----------------------------------	------------------------	---

<p>Local</p>	<p>Web User Interface</p>	<p>Configure SRTP feature on a per-line basis.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>
---------------------	---------------------------	---

To configure SRTP feature via web user interface:

1. Click on **Account**.
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 **RTP Encryption (SRTP)**.



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

Encrypting Configuration Files

Encrypted configuration files can be downloaded from the provisioning server to protect against unauthorized access and tampering of sensitive information (e.g., login passwords, registration information). Yealink provides a configuration encryption tool for encrypting configuration files. The encryption tool encrypts plaintext <y0000000000xx>.cfg and <MAC>.cfg files (one by one or in batch) using 16-character

symmetric keys (the same or different keys for configuration files) and generates encrypted configuration files with the same file name as before. This tool also encrypts the plaintext 16-character symmetric keys using a fixed key, which is the same as the one built in the IP phone, and generates new files named as <xx_Security>.enc (xx indicates the name of the configuration file, for example, y00000000000_Security.enc for y00000000000.cfg file). This tool generates another new file named as Aeskey.txt to store the plaintext 16-character symmetric keys for each configuration file.

For a Microsoft Windows platform, you can use a Yealink-supplied encryption tool "Config_Encrypt_Tool.exe" to encrypt the <y0000000000xx>.cfg and <MAC>.cfg files respectively.

Note

Yealink also provides a configuration encryption tool (yealinkencrypt) for Linux platform if required. For more information, refer to *Yealink Configuration Encryption Tool User Guide*.

For security reasons, administrator should upload encrypted configuration files, <y0000000000xx_Security>.enc and/or <MAC_Security>.enc files to the root directory of the provisioning server. During auto provisioning, the IP phone requests to download <y0000000000xx>.cfg file first. If the downloaded configuration file is encrypted, the phone will request to download <y0000000000xx_Security>.enc file (if enabled) and decrypt it into the plaintext key (e.g., key2) using the built-in key (e.g., key1). Then the IP phone decrypts <y0000000000xx>.cfg file using key2. After decryption, the IP phone resolves configuration files and updates configuration settings onto the IP phone system.

The way the IP phone processes the <MAC>.cfg file is the same to that of the <y0000000000xx>.cfg file.

Procedure to Encrypt Configuration Files

To encrypt the <y0000000000xx>.cfg file:

1. Double click "Config_Encrypt_Tool.exe" to start the application tool.

The screenshot of the main page is shown as below:



When you start the application tool, a file folder named "Encrypted" is created

automatically in the directory where the application tool is located.

2. Click **Browse** to locate configuration file(s) (e.g., y0000000000000.cfg) from your local system in the **Select File(s)** field.

To select multiple configuration files, you can select the first file and then press and hold the **Ctrl** key and select the next files.

3. (Optional.) Click **Browse** to locate the target directory from your local system in the **Target Directory** field.

The tool uses the file folder "Encrypted" as the target directory by default.

4. (Optional.) Mark the desired radio box in the **AES Model** field.

If you mark the **Manual** radio box, you can enter an AES key in the **AES KEY** field or click **Re-Generate** to generate an AES key in the **AES KEY** field. The configuration file(s) will be encrypted using the AES key in the **AES KEY** field.

If you mark the **Auto Generate** radio box, the configuration file(s) will be encrypted using random AES key. The AES keys of configuration files are different.

Note

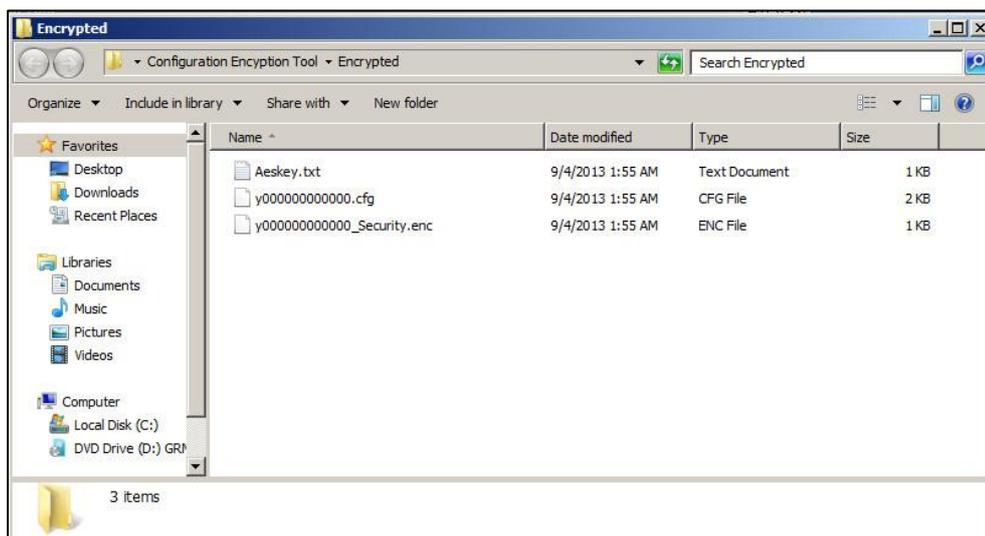
AES keys must be 16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z.

5. Click **Encrypt** to encrypt the configuration file(s).



6. Click **OK**.

The target directory will be automatically opened. You can find the encrypted CFG file(s), encrypted key file(s) and an Aeskey.txt file storing plaintext AES key(s).



Procedure

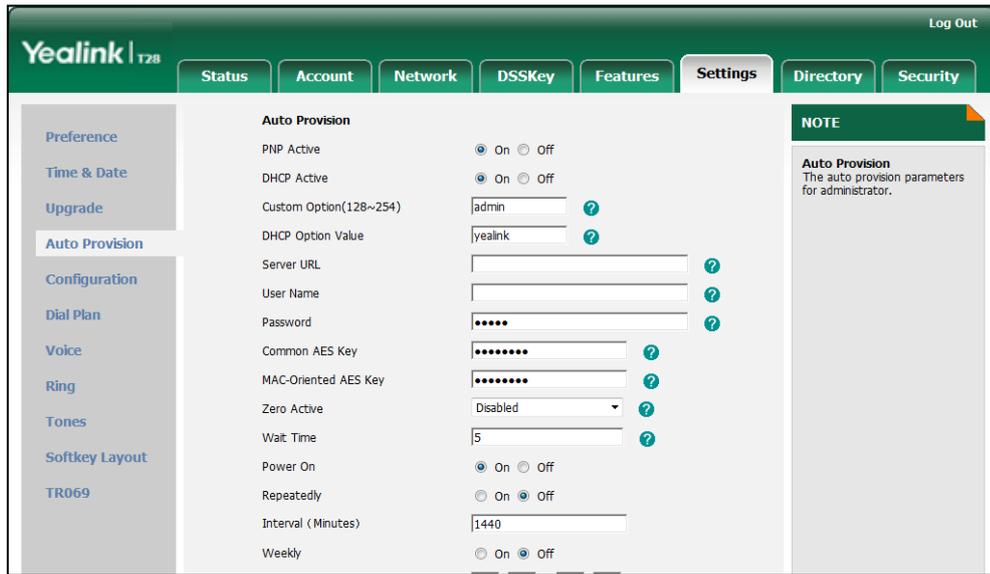
Decryption method can be configured using the configuration files.

Configuration File	<y0000000000xx>.cfg	Configure the decryption method. Configure AES keys. For more information, refer to Configuring Decryption Method on page 387.
Local	Web User Interface	Configure AES keys. Navigate to: http://<phoneIPAddress>/servlet?p=settings-autop&q=load

To configure AES keys via web user interface:

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

- Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.
 AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z.



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

Upgrading Firmware

This chapter provides information about upgrading the IP phone firmware. Two methods of firmware upgrade:

- Manually, from the local system.
- Automatically, from the provisioning server.

The following table lists the associated firmware name for each IP phone model (X is replaced by the actual firmware version).

IP Phone Model	Associated Firmware Name
SIPT28P	2.x.x.x.rom
SIPT26P	6.x.x.x.rom
SIPT22P	7.x.x.x.rom
SIPT21P	34.x.x.x.rom
SIPT20P	9.x.x.x.rom
SIPT19P	31.x.x.x.rom

Note

You can download the latest firmware online:

<http://www.yealink.com/DocumentDownload.aspx?CatelId=142&flag=142>.

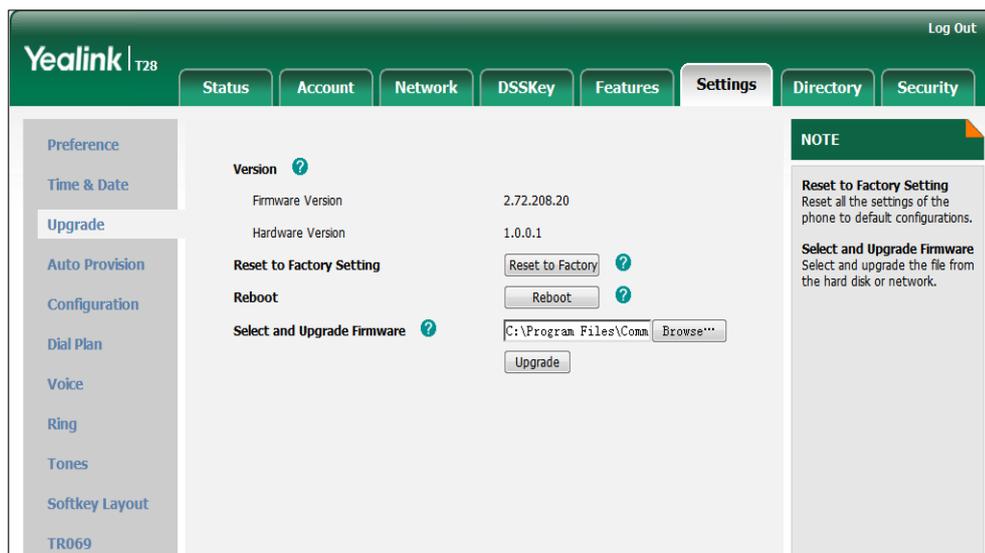
Upgrade via Web User Interface

To manually upgrade firmware via web user interface, you need to store firmware to your local system in advance.

To upgrade firmware manually via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Browse**.
3. Select firmware from the local system.
4. Click **Upgrade**.

A dialog box pops up to prompt “Firmware of the SIP Phone will be updated. It will take 5 minutes to complete. Please don't power off!”.



5. Click **OK** to confirm the upgrade.

Note Do not unplug the network and power cables when the IP phone is upgrading firmware. Do not close and refresh the browser when the IP phone is upgrading firmware via web user interface.

Upgrade Firmware from the Provisioning Server

IP phones support to use FTP, TFTP, HTTP, and HTTPS protocols to download the configuration files and firmware from the provisioning server, and then upgrade firmware automatically.

IP phones can download firmware stored on the provisioning server in one of two ways:

- Check for both configuration files and firmware stored on the provisioning server during startup.
- Automatically check for configuration files and firmware at a fixed interval or specific time.

Method of checking for configuration files and firmware is configurable.

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the way for the IP phone to check for configuration files. Specify the access URL of firmware.</p>
----------------------------------	----------------------------------	---

Resource Files

When configuring particular features, you may need to upload resource files (e.g., local contact directory, remote phone book) to IP phones. The resources files can be local contact directory, remote phone book and so on. Ask Yealink field application engineer for resource file templates. If the resource file is to be used for all IP phones of the same model, the resource file access URL is best specified in the <y0000000000xx>.cfg file. However, if you want to specify the desired phone to use the resource file, the resource file access URL should be specified in the <MAC>.cfg file.

This chapter provides the detailed information on how to customize the following resource files and specify the access URL:

- [Replace Rule Template](#)
- [Dial-now Template](#)
- [Softkey Layout Template](#)
- [Local Contact File](#)
- [Remote XML Phone Book](#)
- [Directory Template](#)
- [Specifying the Access URL of Resource Files](#)

Replace Rule Template

The replace rule template helps with the creation of multiple replace rules. After setup, place the replace rule template to the provisioning server and specify the access URL in the configuration files.

When editing a replace rule template, learn the following:

- <DialRule> indicates the start of a template and </DialRule> indicates the end of a template.
- Create replace rules between <DialRule> and </DialRule>.
- When specifying the desired line(s) to apply the replace rule, the valid values are 0 and line ID. The digit 0 stands for all lines. Multiple line IDs are separated by commas. This is not applicable to SIP-T19P IP phones.
- At most 100 replace rules can be added to the IP phone.
- The expression syntax in the replace rule template is the same as that introduced in the section [Creating Dial Plan](#) on page 32.

Procedure

Use the following procedures to customize a replace rule template.

To customize 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 you leave it blank or enter 0, this replace rule will apply to all lines.

3. Specify the values within double quotes.
4. Place this file to 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

The dial-now template helps with the creation of multiple dial-now rules. After setup, place the dial-now template to the provisioning server and specify the access URL in the configuration files.

When editing a dial-now template, learn the following:

- <DialNow> indicates the start of a template and </DialNow> indicates the end of a template.
- Create dial-now rules between <DialNow> and </DialNow>.
- 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 commas. This is not applicable to SIP-T19P IP phones.
- At most 100 rules can be added to the IP phone.
- The expression syntax in the dial-now rule template is the same as that introduced

in the section [Creating Dial Plan](#) on page 32.

Procedure

Use the following procedures to customize a dial-now template.

To customize 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 you leave it blank or enter 0, this dial-now rule will apply to all lines.

3. Specify the values within double quotes.
4. Place this file to 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

The softkey layout template allows you to assign different soft key layouts to different call states. The call states include CallFailed, CallIn, Connecting, Dialing, RingBack and Talking. After setup, place the templates to the provisioning server and specify the access URL in the configuration files.

When editing a softkey layout template, learn the following:

- <Call States> indicates the start of a template and </Call States> indicates the end of a template. For example, <CallFailed> </CallFailed>.
- <Disable> indicates the start of the disabled soft key list and </Disable> indicates the end of the soft key list, the disabled soft keys are not displayed on the LCD screen.
- Create disabled soft keys between <Disable> and </Disable>.
- <Enable> indicates the start of the enabled soft key list and </Enable> indicates the end of the soft key list, the enabled soft keys are displayed on the LCD screen.
- Create enabled soft keys between <Enable> and </Enable>.

- <Default> indicates the start of the default soft key list and </Default> indicates the end of the default soft key list, the default soft keys are displayed on the LCD screen by default.

Procedure

Use the following procedures to customize a softkey layout template.

To customize a softkey layout template:

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 starts on a separate line:

```
<Key Type=""/>
```

Where:

Key Type="" specifies the enabled soft key (This value cannot be blank).

For each disabled soft key and each default soft key that you want to add, add the same string introduced above.

3. Specify the values within double quotes.
4. Place this file to the provisioning server.

The following is an example of the CallFailed template:

```
<CallFailed>
  <Disable>
    <Key Type="Empty"/>
    <Key Type="Switch"/>
    <Key Type="Cancel"/>
  </Disable>
  <Enable>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Enable>
  <Default>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Default>
</CallFailed>
```

Local Contact File

You can add contacts one by one on the IP phone directly. You can also add multiple contacts at a time and/or share contacts between IP phones using the local contact template file. After setup, place the template file to the provisioning server and specify the access URL of the template file in the configuration files.

When editing a local contact template, learn the following:

- `<root_contact>` indicates the start of a contact list and `</root_contact>` indicates the end of a contact list.
- `<root_group>` indicates the start of a group list and `</root_group>` indicates the end of a group list.
- When specifying a ring tone for a contact or a group, the format of the value must be Auto (the first registered line), Resource:RingN.wav (system ring tone, integer N ranges from 1 to 5) or Custom:Name.wav (custom ring tone).
- When specifying a desired line for a contact, the valid values are 0 and line ID, 0 stands for the first available account. Multiple line IDs are separated by commas.
- At most 5 groups can be added to the IP phone.
- At most 1000 local contacts can be added to the IP phone.

Procedure

Use the following procedures to customize a local contact template file.

To customize a local contact file:

1. Open the template file using an ASCII editor.
2. For each group that you want to add, add the following string to the file. Each starts on a separate line:

```
<group display_name="" ring=""/>
```

Where:

`display_name=""` specifies the name of the group.

`ring=""` specifies the desired ring tone for this group.

3. For each contact that you want to add, add the following string to the file. Each starts on a separate line:

```
<contact display_name="" office_number="" mobile_number="" other_number=""
line="" ring="" group_id_name=""/>
```

Where:

`display_name=""` specifies the name of the contact (This value cannot be blank or duplicated).

`office_number=""` specifies the office number of the contact.

mobile_number="" specifies the mobile number of the contact.

other_number="" specifies the other number of the contact.

line="" specifies the line you want to add this contact to.

ring="" specifies the ring tone for this contact.

group_id_name="" specifies the existing group you want to add the contact to.

4. Specify the values within double quotes.
5. Place this file to the provisioning server.

The following is an example of a local contact file:

```
<root_group>
  <group display_name="Friend" ring=""/>
  <group display_name="Family" ring="Resource:Ring1.wav"/>
</root_group>
<root_contact>
  <contact display_name="John" office_number="1001"
mobile_number="12345678910" other_number="" line="0" ring="Auto"
group_id_name="All Contacts"/>
  <contact display_name="Alice" office_number="1002" mobile_number=""
other_number="" line="1,2" ring="Resource:Ring2.wav"
group_id_name="Friend"/>
</root_contact>
```

Remote XML Phone Book

IP phones can access 5 remote phone books. You can customize the remote XML phone book for IP phones as required. Before specifying the access URL of the remote phone book in the configuration files, you need to create a remote XML phone book and then place it to the provisioning server.

When creating an XML phone book, learn the following:

- <YealinkIPPhoneDirectory> indicates the start of a phone book and </YealinkIPPhoneDirectory> indicates the end of a phone book.
- <DirectoryEntry> indicates the start of a contact and </DirectoryEntry> indicates the end of a contact.

Procedure

Use the following procedures to customize an XML phone book.

Customizing an XML phone book:

1. Open the template file using an ASCII editor.

- For each contact that you want to add, add the following strings to the phone book. Each starts on a separate line:

```
<Name>Mary</Name>
<Telephone>1001</Telephone>
```

Where:

Specify the contact name between <Name> and </Name>.

Specify the contact number between <Telephone> and </Telephone>.

- Specify the values within double quotes.
- Place this file to the provisioning server.

The following is an example of an XML phone book:

```
<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>
```

Note

Yealink supplies a phonebook generation tool to generate a remote XML phone book. For more information, refer to *Yealink Phonebook Generation Tool User Guide*.

Directory Template

Directory provides easy access to frequently used lists. The lists may contain Local Directory, History, Remote Phone Book and LDAP. Users can access the lists by pressing the Directory soft key when the IP phone is idle. After setup, place the directory template to the provisioning server and specify the access URL in the configuration files.

When editing a directory template, learn the following:

- <root_favorite_set> indicates the start of a template and </root_favorite_set> indicates the end of a template.

- Create directory between `<root_favorite_set>` and `</root_favorite_set>`.
- When specifying the display name of the directory list, the valid values are Local Contacts, History, Remote Phone Book (not applicable to SIP-T20P IP phones) and LDAP (not applicable to SIP-T19P and SIP-T20P IP phones).
- When specifying the display priority of the directory list, the valid values are 1, 2, 3 and 4. 1 is the highest priority, 4 is the lowest.
- When enabling or disabling the desired directory list, the valid values are 0 and 1. 0 stands for Disabled, 1 stands for Enabled.

Procedure

Use the following procedures to customize a directory template.

Customizing a directory template:

1. Open the template file using an ASCII editor.
2. For each directory list that you want to configure, add the following string to the file. Each starts on a separate line:

```
<item id_name="" display_name="" priority="" enable="" />
```

Where:

`id_name=""` specifies the existing directory list you want to configure. We do not recommend editing this field.

`display_name=""` specifies the display name of the directory list. We do not recommend editing this field.

`priority=""` specifies the display priority of the directory list.

`enable=""` enables or disables the directory list.

3. Specify the values within double quotes.
4. Place this file to the provisioning server.

The following is an example of a directory template:

```
<root_favorite_set>
  <item id_name="localdirectory" display_name="Local Directory" priority="1"
    enable="1" />
  <item id_name="history" display_name="History" priority="2" enable="0" />
  <item id_name="remotedirectory" display_name="Remote Phone Book"
    priority="3" enable="0" />
  <item id_name="ldap" display_name="LDAP" priority="4" enable="0" />
</root_favorite_set>
```

Super Search Template

The super search template allows you to search for a contact in your desired lists when the phone is in the dialing screen. The lists may contain Local Directory, History, Remote Phone Book and LDAP. After setup, place the super search template to the provisioning server and specify the access URL in the configuration files.

When editing a super search template, learn the following:

- `<root_super_search>` indicates the start of a template and `</root_super_search>` indicates the end of a template.
- Create super search between `<root_super_search>` and `</root_super_search>`.
- When specifying the display name of the directory list, the valid values are Local Contacts, History, Remote Phone Book (not applicable to SIP-T20P IP phones) and LDAP (not applicable to SIP-T19P and SIP-T20P IP phones).
- When specifying the priority of search results, the valid values are 1, 2, 3 and 4. 1 is the highest priority, 4 is the lowest.
- When enabling or disabling the desired directory list, the valid values are 0 and 1. 0 stands for Disabled, 1 stands for Enabled.

Procedure

Use the following procedures to customize a super search template.

Customizing a super search template:

1. Open the template file using an ASCII editor.
2. For each directory list that you want to configure, add the following string to the file. Each starts on a separate line:

```
<item id_name="" display_name="" priority="" enable="" />
```

Where:

`id_name=""` specifies the existing directory list you want to configure. We do not recommend editing this field.

`display_name=""` specifies the display name of the directory list. We do not recommend editing this field.

`priority=""` specifies the priority of search results.

`enable=""` enables or disables the directory list.

3. Specify the values within double quotes.
4. Place this file to the provisioning server.

The following is an example of a super search template:

```
<root_super_search>
  <item id_name="local_directory_search" display_name="Local Contacts"
  priority="1" enable="1" />
  <item id_name="calllog_search" display_name="History" priority="2"
  enable="1" />
  <item id_name="remote_directory_search" display_name="Remote Phone
  Book" priority="3" enable="0" />
  <item id_name="ldap_search" display_name="LDAP" priority="4" enable="0" />
</root_super_search>
```

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 392.
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 393.
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 393.
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 396.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the remote XML phone book. For more information, refer to Access URL of Remote XML Phone Book on page 396.

Configuration File	<y0000000000xx>.cfg	Configure the access URL of the directory template. For more information, refer to Access URL of Directory Template on page 396.
Configuration File	<y0000000000xx>.cfg	Configure the access URL of the super search template. For more information, refer to Access URL of Super Search Template on page 397.

Troubleshooting

This chapter provides an administrator with general information for troubleshooting some common problems that he (or she) may encounter while using IP phones.

Troubleshooting Methods

IP phones can provide feedback in a variety of forms such as log files, packets, status indicators and so on, which can help an administrator more easily find the system problem and fix it.

The following are helpful for better understanding and resolving the working status of the IP phone.

- [Viewing Log Files](#)
- [Capturing Packets](#)
- [Enabling Watch Dog Feature](#)
- [Getting Information from Status Indicators](#)
- [Analyzing Configuration File](#)

Viewing Log Files

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 also specify the severity level of the log to be reported to a log file. The default system log level is 3 (Changes to this parameter via web user interface require a reboot).

In the configuration files, you can use the following parameters to configure system log settings:

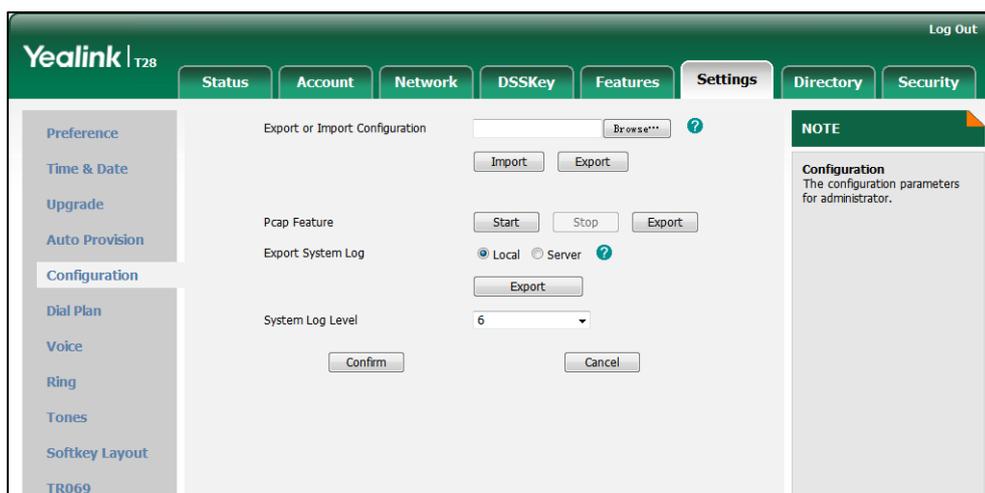
- **syslog.server** -- Specify the IP address of the syslog server to which the log will be exported.
- **syslog.log_level** -- Specify the system log level.

For more information on the system log setting parameters, refer to [Log Settings](#) on page 397.

To configure the level of the system log via web user interface:

1. Click on **Settings->Configuration**.

2. Select the desired level from the pull-down list of **System Log Level**.



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

A dialog box pops up to prompt "Do you want to restart your machine?". The configuration will take effect after a reboot.

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

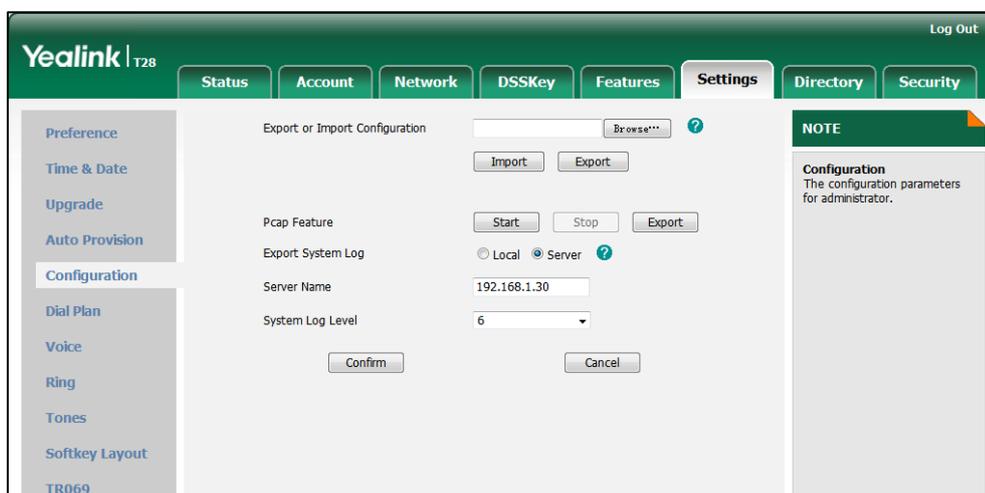
After a reboot, the system log level is set as 6, the administrator debug level.

Note

Administrator level debugging may make some sensitive information accessible (e.g., password-dial number), we recommend that you reset the system log level to 3 after having the syslog file provided.

To configure the phone to export the system log to a syslog server via web user interface:

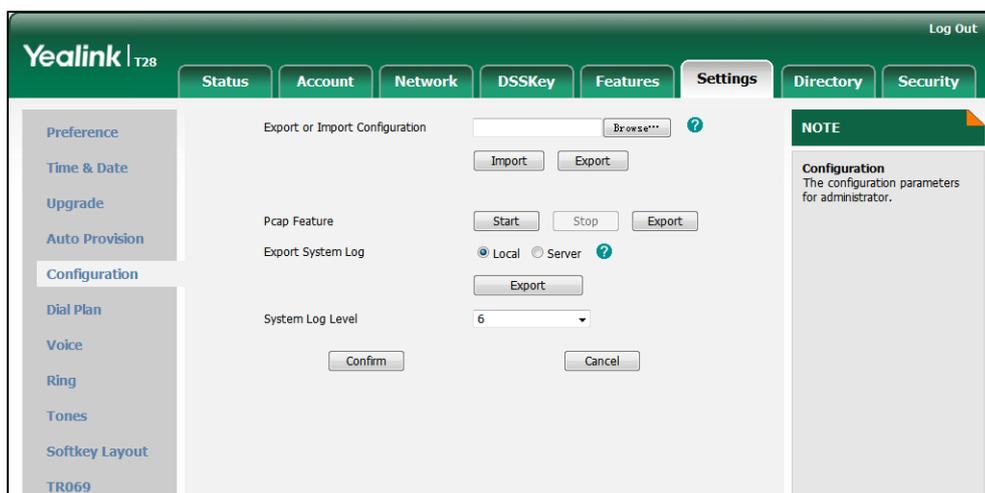
1. Click on **Settings->Configuration**.
2. Mark the **Server** radio box in the **Export System Log** field.
3. Enter the IP address or domain name of the syslog server in the **Server Name** field.



4. Click **Confirm** to accept the change.
A dialog box pops up to prompt “Do you want to restart your machine?”. The configuration will take effect after a reboot.
5. Click **OK** to reboot the IP phone.
The system log will be exported successfully to the desired syslog server after a reboot.
6. Reproduce the issue.

To export a log file to the local system via web user interface:

1. Click on **Settings->Configuration**.
2. Mark the **Local** radio box in the **Export System Log** field.
3. Reproduce the issue.
4. Click **Export** to open file download window, and then save the file to your local system.



The following figure shows a portion of a log file:

```

496 root      8876 SW  /yealink/bin/ggsvca_ipp
497 root      8876 SW  /yealink/bin/ggsvca_ipp
498 root      8876 SW  /yealink/bin/ggsvca_ipp
499 root      8876 SW  /yealink/bin/ggsvca_ipp
500 root      8876 SW  /yealink/bin/ggsvca_ipp
501 root      8876 SW  /yealink/bin/ggsvca_ipp
507 root      16424 SW  /yealink/bin/Screen.exe
508 root      10344 SW  /yealink/bin/sipServer.exx
509 root      10344 SW  /yealink/bin/sipServer.exx
515 root      16424 SW  /yealink/bin/Screen.exe
517 root      16424 SW  /yealink/bin/Screen.exe
519 root      10344 SW  /yealink/bin/sipServer.exx
521 root      16424 SW  /yealink/bin/Screen.exe
522 root      16424 SW  /yealink/bin/Screen.exe
523 root      16424 SW  /yealink/bin/Screen.exe
524 root      10344 SW  /yealink/bin/sipServer.exx
525 root      SW< [IRQ 45]
526 root      10344 SW  /yealink/bin/sipServer.exx
527 root      16424 SW  /yealink/bin/Screen.exe
528 root      16424 SW  /yealink/bin/Screen.exe
529 root      16424 SW  /yealink/bin/Screen.exe
1147 root     1788 SWN  sleep 1000
1227 root     10120 SWN  ConfigManApp.com
1228 root     4624 SW  /yealink/bin/mini_httpd -p 80 -d /yealink/html -c cgi
1229 root     2812 SWN  sh -c cd /tmp;ifconfig >> Messages;ps >> Messages;tar
1230 root     2812 RWN  ps
Feb 29 06:01:09 mini_httpd[388]: mini_httpd.c(1510):child process 1227 exit!
Feb 29 06:01:12 mini_httpd[1232]: mini_httpd.c(1997):path:/cgi-bin/ConfigManApp.com, query:Id=27
Feb 29 06:01:12 mini_httpd[388]: mini_httpd.c(1510):child process 1232 exit!
Feb 29 06:01:12 mini_httpd[1233]: mini_httpd.c(1997):path:/cgi-bin/ConfigManApp.com, query:Id=27
Feb 29 06:01:12 mini_httpd[388]: mini_httpd.c(1510):child process 1233 exit!
Feb 29 06:01:12 mini_httpd[1234]: mini_httpd.c(1997):path:/cgi-bin/ConfigManApp.com, query:Id=27
Feb 29 06:01:12 mini_httpd[388]: mini_httpd.c(1510):child process 1234 exit!

```

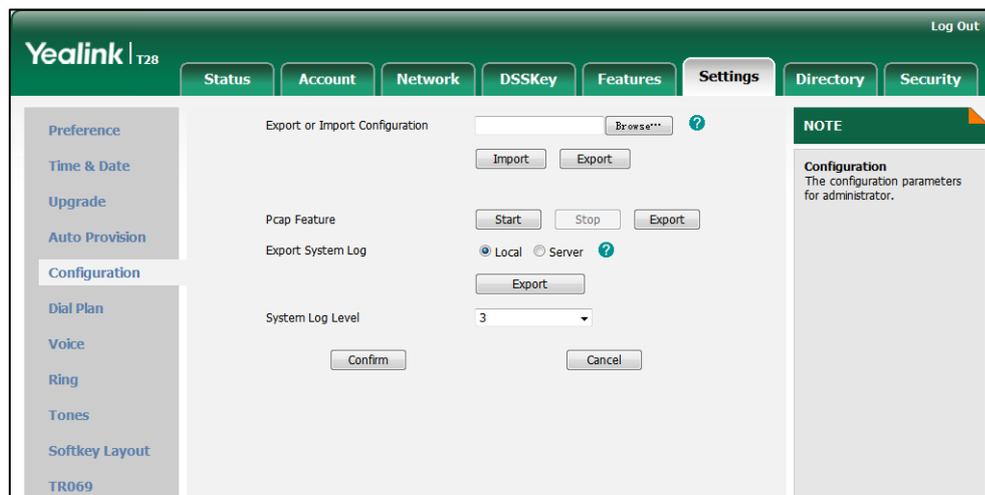
Capturing Packets

You can capture packet in two ways: capturing the packet via web user interface or using the Ethernet software. You can analyze the packet captured for troubleshooting purpose.

To capture packets via web user interface:

1. Click on **Settings->Configuration**.
2. Click **Start** to start capturing signal traffic.
3. Reproduce the issue to get stack traces.
4. Click **Stop** to stop capturing.

5. Click **Export** to open the file download window, and then save the file to your local system.



To capture packets using the Ethernet software:

Connect the Internet port of the IP phone and the PC to the same HUB, and then use Sniffer, Ethereal or Wireshark software to capture the signal traffic.

Enabling Watch Dog Feature

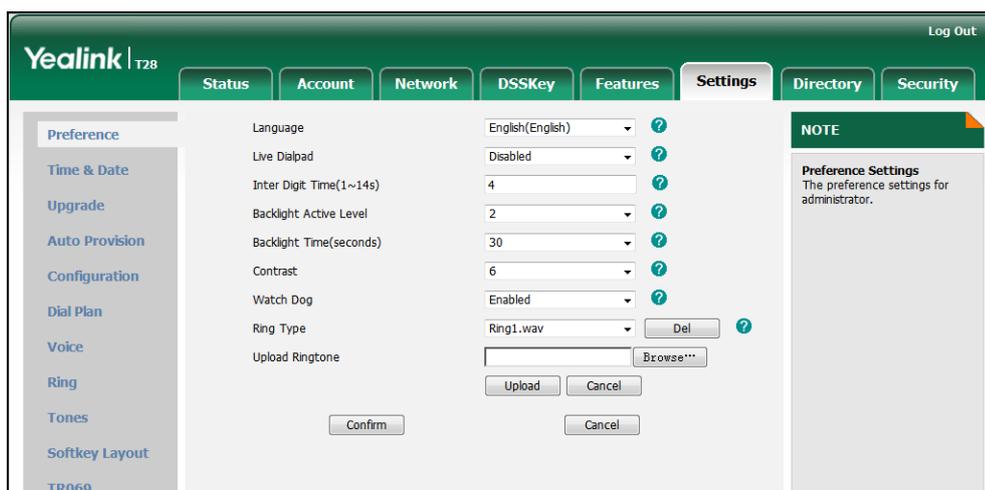
The IP phone provides a troubleshooting feature called “Watch Dog”, which helps you monitor the IP phone status and provides the ability to get stack traces from the last time the IP phone failed. If Watch Dog feature is enabled, the IP phone will automatically reboot when it detects a fatal failure. This feature can be configured using the configuration files or via web user interface.

You can use the “watch_dog.enable” parameter to configure watch dog feature in the configuration files. For more information, refer to [Watch Dog](#) on page 398.

To configure watch dog feature via web user interface:

1. Click on **Settings->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

Status indicators may consist of the power LED, MESSAGE key LED, line key indicator, headset key indicator and the on-screen icon.

The following shows two examples of obtaining the phone information from status indicators:

- If a LINK failure of the IP phone is detected, a prompting message "Network Unavailable" and the icon  will appear on the LCD screen.
- If a voice mail is received, the MESSAGE key LED illuminates.

For more information on the icons, refer to [Reading Icons](#) on page 20.

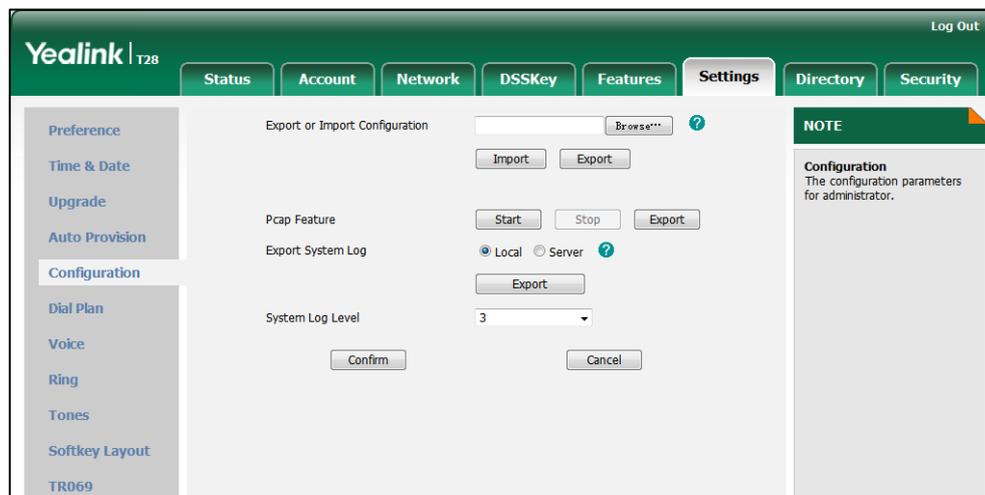
Analyzing Configuration File

Wrong configurations may have an impact on your phone use. You can export configuration file to check the current configuration of the IP phone and troubleshoot if necessary.

To export configuration file via web user interface:

1. Click on **Settings->Configuration**.

- In the **Export or Import Configuration** block, click **Export** to open the file download window, and then save the file to your local system.



Troubleshooting Solutions

This section describes solutions to common issues that may occur while using the IP phone. Upon encountering a scenario not listed in this section, contact your Yealink reseller for further support.

Why is the LCD screen blank?

Do one of the following:

- Ensure that the IP phone is properly plugged into a functional AC outlet.
- Ensure that the IP phone is plugged into a socket controlled by a switch that is on.
- If the IP phone is plugged into a power strip, try plugging it directly into a wall outlet.
- If your phone is PoE powered, ensure that you are using a PoE-compliant switch or hub.

Why doesn't the IP phone get an IP address?

Do one of the following:

- Ensure that the Ethernet cable is plugged into the Internet port on the IP phone and the Ethernet cable is not loose.
- Ensure that the Ethernet cable is not damaged.
- Ensure that the IP address and related network parameters are set correctly.
- Ensure that your network switch or hub is operational.

Why does the IP phone display “No Service”?

The LCD screen prompts “No Service” message when there is no available SIP account on the IP phone.

Do one of the following:

- Ensure that an account is actively registered on the IP phone at the path **Menu->Status->More->Accounts**.
- Ensure that the SIP account parameters have been configured correctly.

How do I find the basic information of the IP phone?

Press the **OK** key when the IP phone is idle to check the basic information (e.g., IP address, MAC address and firmware version).

Why doesn't the IP phone upgrade firmware successfully?

Do one of the following:

- Ensure that the target firmware is not the same as the current firmware.
- Ensure that the target firmware is applicable to the IP phone model.
- Ensure that the current or the target firmware is not protected.
- Ensure that the power is on and the network is available in the process of upgrading.
- Ensure that the web browser is not closed or refreshed when upgrading firmware via web user interface.

Why doesn't the IP phone display time and date correctly?

Check if the IP phone is configured to obtain the time and date from the NTP server automatically. If your phone is unable to access the NTP server, configure the time and date manually.

Why do I get poor sound quality during a call?

If you have poor sound quality/acoustics like intermittent voice, low volume, echo or other noises, the possible reasons could be:

- Users are seated too far out of recommended microphone range and sound faint, or are seated too close to sensitive microphones and cause echo.
- Intermittent voice is mainly caused by packet loss, due to network congestion, and

jitter, due to message recombination of transmission or receiving equipment (e.g., timeout handling, retransmission mechanism, buffer under run).

- Noisy equipment, such as a computer or a fan, may cause voice interference. Turn off any noisy equipment.
- Line issues can also cause this problem; disconnect the old line and redial the call to ensure another line may provide better connection.

What is the difference between a remote phone book and a local phone book?

A remote phone book is placed on a server, while a local phone book is placed on the IP phone flash. A remote phone book can be used by everyone that can access the server, while a local phone book can only be used by a specific phone. A remote phone book is always used as a central phone book for a company; each employee can load it to obtain the real-time data from the same server.

What is the difference among user name, register name and display name?

Both user name and register name are defined by the server. User name identifies the account, while register name matched with a password is for authentication purposes. Display name is the caller ID that will be displayed on the callee's phone LCD screen. Server configurations may override the local ones.

How to reboot the IP phone remotely?

IP phones support remote reboot by a SIP NOTIFY message with "Event: check-sync" header. When receiving a NOTIFY message with the parameter "reboot=true", the IP phone reboots immediately. The NOTIFY message is formed as shown:

```
NOTIFY sip:<user>@<dsthost> SIP/2.0
To: sip:<user>@<dsthost>
From: sip:sipsak@<srchost>
CSeq: 10 NOTIFY
Call-ID: 1234@<srchost>
Event: check-sync;reboot=true
```

Why does the IP phone use DOB format logo file instead of popular BMP, JPG and so on?

The IP phone only uses logo file in DOB format, as the DOB format file has a high compression ratio (the size of the uncompressed file compared to that of the compressed file) and can be stored in smaller space. Tools for converting BMP format to DOB format are available. For more information, refer to *Yealink SIP-T2 Series T19P T4 Series IP Phones Auto Provisioning Guide*.

How to increase or decrease the volume?

Press the volume key to increase or decrease the ringer volume when the phone is idle, or to adjust the volume of engaged audio device (handset, speakerphone or headset) when there is an active call in progress.

What will happen if I connect both PoE cable and power adapter?

Which has the higher priority?

IP phones manufactured before February 2010 will use the power adapter preferentially, while those made later will use PoE preferentially.

What is auto provisioning?

Auto provisioning refers to the update of IP phones, including update on configuration parameters, local phone book, firmware and so on. You can use auto provisioning on a single phone, but it makes more sense in mass deployment.

What is PnP?

Plug and Play (PnP) is a method for IP phones to acquire the provisioning server address. With PnP enabled, the IP phone broadcasts the PnP SUBSCRIBE message to obtain a provisioning server address during startup. Any SIP server recognizing the message will respond with the preconfigured provisioning server address, so the IP phone will be able to download the CFG files from the provisioning server. PnP depends on support from a SIP server.

Why doesn't the IP phone update the configuration?

Do one of the following:

- Ensure that the configuration is set correctly.
- Reboot the IP phone. Some configurations require a reboot to take effect.
- Ensure that the configuration is applicable to the IP phone model.
- The configuration may depend on support from a server.

What do "on code" and "off code" mean?

They are codes that the IP phone sends to the server when a certain action takes place. On code is used to activate a feature on the server side, while off code is used to deactivate a feature on the server side.

For example, if you set the Always Forward on code to be *78 (may vary on different servers), and the target number to be 201. When you enable Always Forward on the IP phone, the IP phone sends *78201 to the server, and then the server will enable Always Forward feature on the server side, hence being able to get the right status of the extension.

How to solve the IP conflict problem?

Do one of the following:

- Reset another available IP address for the IP phone.
- Check network configuration via phone user interface at the path **Menu->Settings->Advanced Settings->Network->WAN Port->IPv4**. If Static IP Client is selected, select DHCP IP Client instead.

How to reset the IP phone to factory configurations?

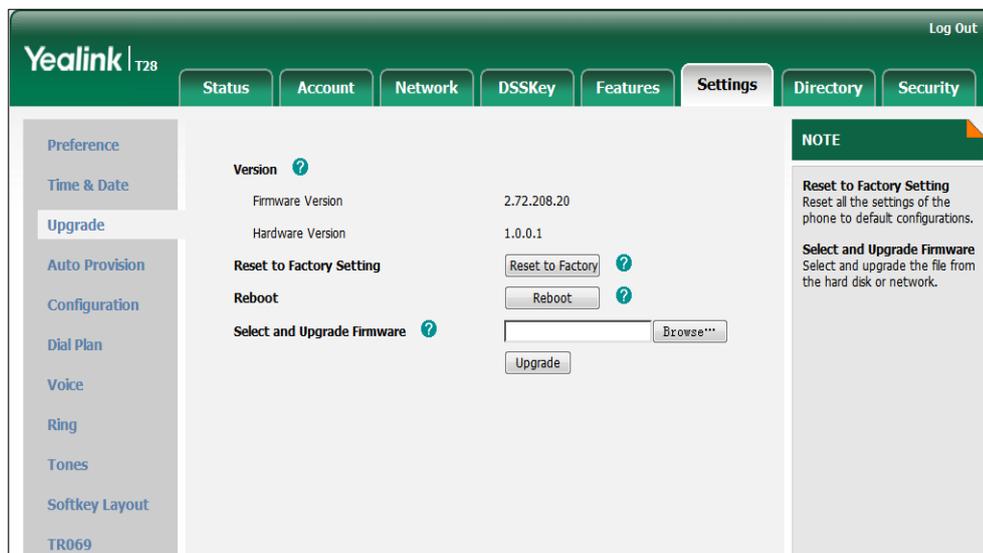
Reset your phone to factory configurations after you have tried all troubleshooting suggestions but do not solve the problem. Note that all custom settings will be overwritten after resetting.

To reset the IP phone via web user interface:

1. Click on **Settings->Upgrade**.

2. Click **Reset to Factory Reset** in the **Reset to Factory Setting** field.

The web user interface prompts the message "Do you want to reset to factory?".



3. Click **OK** to confirm the resetting.

The phone will be reset to factory successfully after startup.

Note Reset of your phone may take a few minutes. Do not power off until the phone starts up successfully.

How to restore the administrator password?

Factory reset can restore the original password. All custom settings will be overwritten after reset.

What are the main differences among SIP-T28P, IP-T26P, SIP-T22P, SIP-T21P, SIP-T20P and SIP-T19P IP phones?

Phone Model	LCD	Logo Display	Line Key	Memory Key	SMS	XML Browser
SIP-T28P	320*160 pixel	236*82 pixel	6	10	Support	Support
SIP-T26P	132*64 pixel	132*64 pixel	3	10	Support	Support
SIP-T22P	132*64 pixel	132*64 pixel	3	/	Support	Support

Phone Model	LCD	Logo Display	Line Key	Memory Key	SMS	XML Browser
SIP-T21P	132*64 pixel	132*64 pixel	2	/	Support	Support
SIP-T20P	3-line (2*15 characters and an icon line)	Text log	2	/	/	Support (Non UI)
SIP-T19P	132*64 pixel	132*64 pixel	/	/	Support	Support

Appendix

Appendix A: Glossary

802.1x--an IEEE Standard for port-based Network Access Control (PNAC). It is a part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.

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.

EAP-TLS (Extensible Authentication Protocol-Transport Layer Security) --provides for mutual authentication, integrity-protected cipher suite negotiation between two endpoints.

PEAP-MSCHAPv2 (Protected Extensible Authentication Protocol-Microsoft Challenge Handshake Authentication Protocol version 2) --provides for mutual authentication, but does not require a client certificate on the IP phone.

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.

MIB (Management Information Base)--a virtual database used for managing the entities in a communications network.

OID (Object Identifier)--assigned to an individual object within a MIB.

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	Spain(Madrid)
+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)
+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
+04:30	Afghanistan
+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 configuration parameters in the configuration files for each feature. The configuration files are <y0000000000xx>.cfg and <MAC>.cfg.

Setting Parameters in Configuration Files

You can set parameters in the configuration files to configure 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 for a specific 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-	Configuration File
network.internet_port.type	<MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-PPPoE 2-Static IP Address
Example	network.internet_port.type= 0

Parameter-	Configuration File
network.static_dns_enable	<y0000000000xx>.cfg
Description	Enables or disables the phone to use manually configured static IPv4 DNS when the

	parameter "network.internet_port.type" is set to 0 (DHCP). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	network.static_dns_enable= 0

Static Network Settings

Parameter- network.internet_port.type	Configuration File <MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-PPPoE 2-Static IP Address
Example	network.internet_port.type = 2

Parameter- network.ip_address_mode	Configuration File <MAC>.cfg
Description	Configures the IP address mode. IP phones support to use the IPv4 address only, the IPv6 address only or both IPv4 and IPv6 addresses. Note: If you change this parameter, the IP phone will reboot to make the change take effect.

Format	Integer
Default Value	0
Range	Valid values are: 0-IPv4 1-IPv6 2-IPv4&IPv6
Example	network.ip_address_mode = 0

Parameter- network.internet_port.ip	Configuration File <MAC>.cfg
Description	Configures the IP address when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.ip = 192.168.1.20

Parameter- network.internet_port.mask	Configuration File <MAC>.cfg
Description	Configures the subnet mask when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Subnet Mask
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.mask = 255.255.255.0

Parameter-	Configuration File
network.internet_port.gateway	<MAC>.cfg
Description	Configures the default gateway when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.internet_port.gateway = 192.168.1.254

Parameter-	Configuration File
network.primary_dns	<MAC>.cfg
Description	Configures the primary DNS server when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.primary_dns = 202.101.103.55

Parameter-	Configuration File
network.secondary_dns	<MAC>.cfg
Description	Configures the secondary DNS server when the Internet port type is configured as Static IP Address and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take

	effect.
Format	IPv4 Address
Default Value	Blank
Range	Not Applicable
Example	network.secondary_dns = 202.101.103.54

PPPoE

Parameter-	Configuration File
network.internet_port.type	<MAC>.cfg
Description	Configures the Internet port type. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-PPPoE 2-Static IP Address
Example	network.internet_port.type= 1

Parameter-	Configuration File
network.pppoe.user	<y0000000000xx>.cfg
Description	Configures the PPPoE user name when the Internet port type is configured as PPPoE and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	network.pppoe.user = xmyealink

Parameter-	Configuration File
network.pppoe.password	<y0000000000xx>.cfg
Description	Configures the PPPoE password when the Internet port type is configured as PPPoE and the IP address mode is configured as IPv4 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	network.pppoe.password = yealink123

Internet and PC Ports Transmission Methods

Internet Port Transmission Method

Parameter-	Configuration File
network.internet_port.speed_duplex	<y0000000000xx>.cfg
Description	Configures the transmission method of Internet port. Note: We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-Auto negotiate 1-Full duplex, 10Mbps 2-Full duplex, 100Mbps 3-Half duplex, 10Mbps 4-Half duplex, 100Mbps
Example	network.internet_port.speed_duplex = 0

PC Port Transmission Method

Parameter-	Configuration File
network.pc_port.speed_duplex	<y0000000000xx>.cfg
Description	Configures the transmission method of PC port. Note: We recommend that you do not change this parameter. If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-Auto negotiate 1-Full duplex, 10Mbps 2-Full duplex, 100Mbps 3-Half duplex, 10Mbps 4-Half duplex, 100Mbps
Example	network.pc_port.speed_duplex = 0

PC Port Mode

Parameter-	Configuration File
network.PC_port.enable	<y0000000000xx>.cfg
Description	Enables or disables the PC port. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Auto Negotiation
Example	network.PC_port.enable = 1

Parameter-	Configuration File
network.bridge_mode	<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. It is not applicable to SIP-T19P and SIP-T21P IP phones. SIP-T19P and SIP-T21P IP phones only support bridge mode for PC connection.
Format	Integer
Default Value	1
Range	Valid values are: 0-Router 1-Bridge
Example	network.bridge_mode = 1

Parameter-	Configuration File
network.pc_port.ip	<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. It is not applicable to SIP-T19P and SIP-T21P IP phones.
Format	IP Address
Default Value	10.0.0.1
Range	Not Applicable
Example	network.pc_port.ip = 10.0.0.1

Parameter-	Configuration File
network.pc_port.mask	<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. It is not applicable to SIP-T19P and SIP-T21P IP phones.
Format	IP Address
Default Value	255.255.255.0
Range	Not Applicable

Example	network.pc_port.mask = 255.255.255.0
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Parameter-	Configuration File
network.pc_port.dhcp_server	<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. It is not applicable to SIPT19P and SIPT21P IP phones.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	network.pc_port.dhcp_server = 1

Parameter-	Configuration File
network.dhcp.start_ip	<y0000000000xx>.cfg
Description	Configures 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. It is not applicable to SIPT19P and SIPT21P IP phones.
Format	IP Address
Default Value	10.0.0.10
Range	Not Applicable
Example	network.dhcp.start_ip = 10.0.0.10

Parameter-	Configuration File
network.dhcp.end_ip	<y0000000000xx>.cfg
Description	Configures 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. It is not applicable to SIP-T19P and SIP-T21P IP phones.
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-	Configuration File
dialplan.replace.prefix.X	<y0000000000xx>.cfg
Description	Configures the string you want to replace. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.prefix.1 = 123

Parameter-	Configuration File
dialplan.replace.replace.X	<y0000000000xx>.cfg
Description	Configures the alternate string instead of what the user enters. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.replace.replace.1 = 1

Parameter-	Configuration File
dialplan.replace.line_id.X	<y0000000000xx>.cfg
Description	Configures the desired line to apply this replace rule. The digit 0 stands for all lines. X ranges from 1 to 100. Note: Multiple line IDs are separated by commas. It is not applicable to SIP-T19P IP phones.
Format	Integer
Default Value	Blank (for all lines)
Range	Valid values are: 0 to 6 (for SIP-T28P) 0 to 3 (for SIP-T26P/T22P) 0 to 2 (for SIP-T21P/T20P)
Example	dialplan.replace.line_id.1 = 1,2

Dial-now

Parameter-	Configuration File
dialplan.dialnow.rule.X	<y0000000000xx>.cfg
Description	Configures the string used to match the numbers entered by the user. When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the numbers without pressing the send key. X ranges from 1 to 100.
Format	String
Default Value	Blank
Range	String within 511 characters
Example	dialplan.dialnow.rule.1 = 123

Parameter-	Configuration File
dialplan.dialnow.line_id.X	<y0000000000xx>.cfg
Description	Configures the desired line to apply this dial-now rule. The digit 0 stands for all lines. X ranges from 1 to 100. Note: Multiple line IDs are separated by

	commas. It is not applicable to SIP-T19P IP phones.
Format	Integer
Default Value	Blank (for all lines)
Range	Valid values are: 0 to 6 (for SIP-T28P) 0 to 3 (for SIP-T26P/T22P) 0 to 2 (for SIP-T21P/T20P)
Example	dialplan.dialnow.line_id.1 = 1,2

Parameter-	Configuration File
phone_setting.dialnow_delay	<y0000000000xx>.cfg
Description	Configures the delay time (in seconds) for the dial-now rule. When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the entered number after the specified delay time.
Format	Integer
Default Value	1
Range	1 to 14
Example	phone_setting.dialnow_delay = 1

Area Code

Parameter-	Configuration File
dialplan.area_code.code	<y0000000000xx>.cfg
Description	Configures the area code to add before the entered numbers.
Format	String
Default Value	Blank
Range	String within 16 characters
Example	dialplan.area_code.code = 010

Parameter-	Configuration File
dialplan.area_code.min_len	<y0000000000xx>.cfg
Description	Configures the minimum length of the entered numbers.
Format	Integer
Default Value	1
Range	1 to 15
Example	dialplan.area_code.min_len = 1

Parameter-	Configuration File
dialplan.area_code.max_len	<y0000000000xx>.cfg
Description	Configures the maximum length of the entered numbers. Note: The value must be larger than the minimum length.
Format	Integer
Default Value	15
Range	1 to 15
Example	dialplan.area_code.max_len = 15

Parameter-	Configuration File
dialplan.area_code.line_id	<y0000000000xx>.cfg
Description	Configures the desired line to apply this area code rule. The digit 0 stands for all lines. Note: Multiple line IDs are separated by commas. It is not applicable to SIP-T19P IP phones.
Format	Integer
Default Value	Blank (for all lines)
Range	Valid values are: 0 to 6 (for SIP-T28P) 0 to 3 (for SIP-T26P/T22P) 0 to 2 (for SIP-T21P/T20P)
Example	dialplan.area_code.line_id = 1,2

Block Out

Parameter-	Configuration File
dialplan.block_out.number.X	<y0000000000xx>.cfg
Description	Configures the block out numbers. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	dialplan.block_out.number.1 = 1234

Parameter-	Configuration File
dialplan.block_out.line_id.X	<y0000000000xx>.cfg
Description	Configures the desired line to apply this block out rule. The digit 0 stands for all lines. X ranges from 1 to 10. Note: Multiple line IDs are separated by commas. It is not applicable to SIP-T19P IP phones.
Format	Integer
Default Value	Blank (for all lines)
Range	Valid values are: 0 to 6 (for SIP-T28P) 0 to 3 (for SIP-T26P/T22P) 0 to 2 (for SIP-T21P/T20P)
Example	dialplan.block_out.line_id.1 = 1,2,3

Power Indicator LED

Parameter-	Configuration File
phone_setting.common_power_led_enable	<y0000000000xx>.cfg
Description	Enables or disables the power indicator LED to be turned on.
Format	Boolean
Default Value	1

Range	Valid values are: 0 -Disabled (power indicator LED is off) 1 -Enabled (power indicator LED is solid green)
Example	phone_setting.common_power_led_enable = 1

Parameter- phone_setting.ring_power_led_flash_enable	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the power indicator LED to flash when the phone receives an incoming call.</p> <p>If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".</p>
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled (power indicator LED does not flash) 1 -Enabled (power indicator LED fast flashes (300ms) green)
Example	phone_setting.ring_power_led_flash_enable = 1

Parameter- phone_setting.mail_power_led_flash_enable	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the power indicator LED to flash when the phone receives a voice mail or a text message.</p> <p>If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".</p>
Format	Boolean

Default Value	0
Range	Valid values are: 0 -Disabled (power indicator LED does not flash) 1 -Enabled (power indicator LED slow flashes (1000ms) green)
Example	phone_setting.mail_power_led_flash_enable = 0

Parameter- phone_setting.mute_power_led_flash_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the power indicator LED to flash when a call is mute. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled (power indicator LED does not flash) 1 -Enabled (power indicator LED fast flashes (300ms) green)
Example	phone_setting.mute_power_led_flash_enable = 1

Parameter- phone_setting.hold_and_held_power_led_flash_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the power indicator LED to flash when a call is placed on hold or is held. If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".

Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled (power indicator LED does not flash)</p> <p>1-Enabled (power indicator LED fast flashes (500ms) green)</p>
Example	phone_setting.hold_and_held_power_led_flash_enable = 0

Parameter- phone_setting.talk_and_dial_power_led_enable	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the power indicator LED to be turned on when the phone is busy.</p> <p>If it is set to 0, the status of the power indicator LED is determined by the value of the parameter "phone_setting.common_power_led_enable".</p>
Format	Boolean
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled (power indicator LED is off)</p> <p>1-Enabled (power indicator LED is solid green)</p>
Example	phone_setting.talk_and_dial_power_led_enable = 1

Contrast

Parameter- phone_setting.contrast	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the contrast of the LCD screen.</p> <p>For SIPT28P IP phones, it configures the LCD's contrast of the IP phone and connected EXP39.</p> <p>For SIPT26P IP phones, it configures the LCD's contrast of the connected EXP39 only.</p>

	<p>For SIP-T19P and SIP-T21P IP phones, it configures the LCD's contrast of the IP phone only.</p> <p>Note: We recommend that you set the contrast of the LCD screen to 6 as a more comfortable level. It is only applicable to SIP-T19P, SIP-T21P and SIP-T28P IP phones, and EXP39 connected to SIP-T26P and SIP-T28P IP phones.</p>
Format	Integer
Default Value	6
Range	1 to 10
Example	phone_setting.contrast = 6

Backlight

Parameter- phone_setting.active_backlight_level	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the backlight idle intensity used to adjust the backlight intensity of the LCD screen. Level 3 is the brightest.</p> <p>Note: It is only applicable to SIP-T28P IP phones and the connected EXP39.</p>
Format	Integer
Default Value	2
Range	1 to 3
Example	phone_setting.active_backlight_level = 2

Parameter- phone_setting.backlight_time	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the delay time to turn off the backlight when the IP phone is inactive.</p> <p>If it is set to 60 (60s), the LCD backlight is turned off when the IP phone is inactive for 60 seconds.</p> <p>Note: It is not applicable to SIP-T19P, SIP-T20P</p>

	and SIPT21P IP phones.
Format	Integer
Default Value	30
Range	<p>Valid values are:</p> <p>0-Always off</p> <p>1-Always on</p> <p>15-15s</p> <p>30-30s</p> <p>60-60s</p> <p>120-120s</p> <p>300-300s</p> <p>600-600s</p> <p>1800-1800s</p>
Example	phone_setting.backlight_time = 30

User Password

Parameter- security.user_password	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the password of the user for web server access.</p> <p>The IP phone uses "user" as the default user password.</p> <p>Note: IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.</p>
Format	username:new password
Default Value	user
Range	String within 32 characters
Example	security.user_password = user:password123

Administrator Password

Parameter- security.user_password	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the password of the administrator for web server access.</p> <p>The IP phone uses "admin" as the default</p>

	administrator password. Note: IP phones support ASCII characters 32-126(0x20-0x7E) only in passwords.
Format	administrator username:new password
Default Value	admin
Range	String within 32 characters
Example	security.user_password = admin:password000

Phone Lock

Parameter-	Configuration File
phone_setting.lock	<y0000000000xx>.cfg
Description	<p>Configures the type of phone lock.</p> <p>Menu Key: The Menu soft key and MESSAGE key are locked (For SIP-T20P, the MENU key is locked).</p> <p>Function Keys: MESSAGE, RD, CONF, HOLD, MUTE, TRAN, OK, X, navigation keys, soft keys, line keys and memory keys are locked (For SIP-T22P/T21P, CONF, HOLD, MUTE and memory keys do not exist; For SIP-T20P, the MUTE key, soft keys and memory keys do not exist, but the additional MENU and Directory keys are locked; For SIP-T19P, CONF, HOLD, OK, X, memory keys and line keys do not exist, but the additional √ key is locked).</p> <p>All Keys: All keys are locked except the volume key. You are only allowed to dial emergency numbers, reject incoming calls by pressing the X key, answer incoming calls by lifting the handset, pressing the Speakerphone key, the HEADSET key or the OK key, place an active call on hold by pressing the Hold soft key or the HOLD key, resume the held call by pressing the Resume soft key or the HOLD key, and end the call by hanging up the handset, pressing the Speakerphone key or pressing the X key (For SIP-T22P/T21P, HOLD key does not exist; For SIP-T20P, soft keys do not exist. For SIP-T19P, HOLD and X keys do not exist).</p>

	If it is set to 0 (Disabled), 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
Example	phone_setting.lock = 1

Parameter- phone_setting.phone_lock.unlock_pin	Configuration File <y0000000000xx>.cfg
Description	Configures a new unlock PIN. Once the IP phone is locked, you can use the default password "123" to unlock it.
Format	numeric characters
Default Value	123
Range	characters within 15 digits
Example	phone_setting.phone_lock.unlock_pin = 123

Parameter- phone_setting.phone_lock.lock_time_out	Configuration File <y0000000000xx>.cfg
Description	Configures the IP phone to automatically lock the keypad after a delay time (in seconds). If it is set to 0 (0s), the keypad will not be locked automatically. In this case, you need to long press the pound key to lock the keypad. Note: This parameter works only if the IP phone lock type is preset.
Format	Integer
Default Value	0
Range	0 to 3600
Example	phone_setting.phone_lock.lock_time_out = 8

Time and Date

Parameter- local_time.manual_time_enable	Configuration File <MAC>.cfg
Description	Configures the phone to obtain time from NTP server or manual settings.
Format	Integer
Default Value	1
Range	Valid values are: 0-Manual 1-NTP
Example	local_time.manual_time_enable = 1

NTP Server

Parameter- local_time.manual_ntp_srv_prior	Configuration File <MAC>.cfg
Description	Enables or disables the phone to use manually configured NTP server preferentially.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled (use the NTP server obtained by DHCP preferentially) 1-Enabled
Example	local_time.manual_ntp_srv_prior = 0

Parameter- local_time.ntp_server1	Configuration File <MAC>.cfg
Description	Configures the IP address or the domain name of the primary NTP server.
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	String within 99 characters

Example	local_time.ntp_server1 = cn.pool.ntp.org
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Parameter-	Configuration File
local_time.ntp_server2	<MAC>.cfg
Description	Configures the IP address or the domain name of the secondary NTP server. If the primary NTP server is not configured or cannot be accessed, the IP phone will request the time and date from the secondary NTP server.
Format	IP Address or Domain Name
Default Value	cn.pool.ntp.org
Range	String within 99 characters
Example	local_time.ntp_server2 = cn.pool.ntp.org

Parameter-	Configuration File
local_time.interval	<MAC>.cfg
Description	Configures the IP phone to update time and date from the NTP server at regular intervals (in seconds).
Format	Integer
Default Value	1000
Range	15 to 86400
Example	local_time.interval = 1000

Time Zone

Parameter-	Configuration File
local_time.time_zone	<MAC>.cfg
Description	Configures the time zone. For more available time zones, refer to Appendix B: Time Zones on page 257.
Format	String
Default Value	+8
Range	-11 to +13
Example	local_time.time_zone = +8

Parameter-	Configuration File
local_time.time_zone_name	<MAC>.cfg
Description	Configures the desired time zone name. For more available time zone names, refer to Appendix B: Time Zones on page 257.
Format	String
Default Value	China(Beijing)
Range	String within 32 characters
Example	local_time.time_zone_name = China(Beijing)

DST

Parameter-	Configuration File
local_time.summer_time	<MAC>.cfg
Description	Enables or disables Daylight Saving Time (DST) feature.
Format	Integer
Default Value	2
Range	Valid values are: 0-Disabled 1-Enabled 2-Automatic
Example	local_time.summer_time = 2

Parameter-	Configuration File
local_time.dst_time_type	<MAC>.cfg
Description	Configures the DST type. Note: It works only if the parameter "local_time.summer_time" is set to 1 (Enabled).
Format	Integer
Default Value	0
Range	Valid values are: 0-By Date 1-By Week
Example	local_time.dst_time_type = 0

Parameter-	Configuration File
local_time.start_time	<MAC>.cfg
Description	<p>Configures the time to start DST.</p> <p>If "local_time.dst_time_type" is set to 0 (By Date), use the mapping: MM: 1=Jan, 2=Feb,..., 12=Dec DD:1=the first day in a month,..., 31= the last day in a month HH:0=1am, 1=2am,..., 23=12pm</p> <p>If "local_time.dst_time_type" is set to 1 (By Week), use the mapping: Month: 1=Jan, 2=Feb,..., 12=Dec Week of Month: 1=the first week in a month,..., 5=the last week in a month Day of Week: 1=Mon, 2=Tues,..., 7=Sun Hour of Day: 0=1am, 1=2am,..., 23=12pm</p> <p>Note: It works only if the parameter "local_time.summer_time" is set to 1 (Enabled).</p>
Format	<p>The value formats are:</p> <ul style="list-style-type: none"> • MM/DD/HH (For By Date) • Month/Week of Month/Day of Week/Hour of Day (For By Week)
Default Value	1/1/0
Range	1 to 12/1 to 31/0 to 23 (for By Date) 1 to 12/1 to 5/1 to 7/0 to 23 (for By Week)
Example	local_time.start_time = 1/1/0

Parameter-	Configuration File
local_time.end_time	<MAC>.cfg
Description	<p>Configures the time to end DST.</p> <p>If "local_time.dst_time_type" is set to 0 (By Date), use the mapping: MM: 1=Jan, 2=Feb,..., 12=Dec DD:1=the first day in a month,..., 31= the last day in a month HH:0=1am, 1=2am,..., 23=12pm</p>

	<p>If "local_time.dst_time_type" is set to 1 (By Week), use the mapping:</p> <p>Month: 1=Jan, 2=Feb,..., 12=Dec</p> <p>Week of Month: 1=the first week in a month,..., 5=the last week in a month</p> <p>Day of Week: 1=Mon, 2=Tues,..., 7=Sun</p> <p>Hour of Day: 0=1am, 1=2am,..., 23=12pm</p> <p>Note: It works only if the parameter "local_time.summer_time" is set to 1 (Enabled).</p>
Format	<p>The value formats are:</p> <ul style="list-style-type: none"> • MM/DD/HH (For By Date) • Month/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 = 12/31/23

Parameter-	Configuration File
local_time.dhcp_time	<MAC>.cfg
Description	<p>Enables or disables the phone to update time with the offset time obtained from the DHCP server.</p> <p>Note: It is only available to offset from GMT 0.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	local_time.dhcp_time = 0

Parameter-	Configuration File
local_time.offset_time	<MAC>.cfg
Description	Configures the offset time (in minutes).
Format	Integer
Default Value	Blank
Range	-300 to +300
Example	local_time.offset_time = 120

Time Format

Parameter-	Configuration File
local_time.time_format	<MAC>.cfg
Description	Configures the time format. If it is set to 0 (12 Hour), the time display will use 12 hour format. If it is set to 1 (24 Hour), the time display will use 24 hour format.
Format	Integer
Default Value	1
Range	Valid values are: 0-12 Hour 1-24 Hour
Example	local_time.time_format = 1

Date Format

Parameter-	Configuration File
local_time.date_format	<MAC>.cfg
Description	Configures the date format. IP phones support various date formats. You can change the desired format according to your requirement.
Format	Integer
Default Value	0 Note: For SIP-T20P IP phones, the default value is 7.
Range	For SIP-T28P/T26P/T21P/T22P/T19P IP phones:

	<p>Valid values are:</p> <p>0-WWW MMM DD 1-DD-MMM-YY 2-YYYY-MM-DD 3-DD/MM/YYYY 4-MM/DD/YY 5-DD MMM YYYY 6-WWW DD MMM</p> <p>For SIP-T20P IP phones:</p> <p>7-MM DD YY 8-DD MM YY 9-YY MM DD</p> <p>Note: “WWW” represents the abbreviation of the week, “DD” represents a two-digit day, “MMM” represents the first three letters of the month, “YYYY” represents a four-digit year, and “YY” represents a two-digit year which is not displayed on the LCD screen of SIP-T20P IP phones.</p>
Example	local_time.date_format = 0

Language

Parameter-	Configuration File
gui_lang.url	<y0000000000xx>.cfg
Description	<p>Configures the access URL of the language pack.</p> <p>Note: The language packs you load are dependent on available language packs from the provisioning server. You can download the language pack to the phone user interface only.</p>
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<p>The following example uses HTTP to download the language pack “lang+English.txt” from the provisioning server 192.168.10.25.</p> <p>gui_lang.url =</p>

	http://192.168.10.25/lang+English.txt
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Parameter-	Configuration File
lang.gui	<y0000000000xx>.cfg
Description	Configures the language used on the phone user interface.
Format	String
Default Value	English
Range	<p>Valid values are:</p> <p>English</p> <p>Chinese_S (only applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>Chinese_T (only applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>German</p> <p>French</p> <p>Italian</p> <p>Portuguese</p> <p>Polish</p> <p>Spanish</p> <p>Turkish</p>
Example	lang.gui = English

Parameter-	Configuration File
lang.wui	<y0000000000xx>.cfg
Description	<p>Configures the language used on the web user interface.</p> <p>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.</p>
Format	String
Default Value	Blank
Range	Valid values are:

	<p>English</p> <p>Chinese_S (only applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>German</p> <p>French (not applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>Italian</p> <p>Portuguese (not applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>Spanish (not applicable to SIP-T19P and SIP-T21P IP phones)</p> <p>Turkish</p>
Example	lang.wui = English

Logo Customization

Parameter-	Configuration File
phone_setting.lcd_logo.mode	<y0000000000xx>.cfg
Description	<p>Configures the logo mode of the LCD screen.</p> <p>If it is set to 0 (Disabled), the IP phone is not allowed to display a logo.</p> <p>If it is set to 1 (System logo), the LCD screen will display the system logo.</p> <p>If it is set to 2 (Custom logo), the LCD screen will display the custom logo (you need to upload a custom logo file to the phone).</p> <p>For SIPT20P IP phones:</p> <p>Enables or disables a text logo.</p> <p>If it is set to 0 (Disabled), the IP phone is not allowed to display a text logo.</p> <p>If it is set to 1 (Enabled), the LCD screen will display the custom text logo.</p>
Format	Integer
Default Value	0 Note: For SIP-T28 IP phones, the default value is 1.
Range	Valid values are: 0-Disabled

	<p>1-System logo 2-Custom logo</p> <p>Note: For SIP-T28 IP phones, valid values are 1(System logo) and 2(Custom logo). For SIP-T20P IP phones, valid values are 0(Disabled) and 1(Enabled).</p>
Example	phone_setting.lcd_logo.mode = 1

Parameter- lcd_logo.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of custom logo file. Note: It is not applicable to SIP-T20P IP phones.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	The following example uses HTTP to download the custom logo file (logo.dob) from the provisioning server 192.168.10.25. lcd_logo.url = http://192.168.10.25/logo.dob

Parameter- phone_setting.lcd_logo.text	Configuration File <y0000000000xx>.cfg
Description	Configures a text logo. Note: It is only applicable to SIP-T20P IP phones.
Format	String
Default Value	Yealink
Range	String within 15 characters
Example	phone_setting.lcd_logo.text = Yealink

Key as Send

Parameter- features.key_as_send	Configuration File <y0000000000xx>.cfg
Description	Configures the "#" or "*" key as the send key.

	<p>If it is set to 0 (Disabled), neither “#” nor “*” can be used as a send key.</p> <p>If it is set to 1 (# key), the pound key is used as the send key.</p> <p>If it is set to 2 (* key), the asterisk key is used as the send key.</p>
Format	Integer
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-# key</p> <p>2-* key</p>
Example	features.key_as_send = 1

Parameter-	Configuration File
features.key_tone	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to play a tone when a user presses a key.</p> <p>If it is set to 1 (Enabled), the IP phone will play a tone when a user presses a key.</p>
Format	Boolean
Default Value	1
Range	<p>0-Disabled</p> <p>1-Enabled</p>
Example	features.key_tone = 1

Parameter-	Configuration File
features.send_key_tone	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to play a tone when a user presses a send key.</p> <p>If it is set to 1 (Enabled), the IP phone will play a tone when a user presses a send key.</p> <p>Note: It works only if the parameter “features.key_tone” is set to 1 (Enabled).</p>
Format	Boolean
Default Value	1

Range	Valid values are: 0-Disabled 1-Enabled
Example	features.send_key_tone = 1

Hotline

Parameter-	Configuration File
features.hotline_number	<y0000000000xx>.cfg
Description	Configures the hotline number. It configures a number that the IP phone automatically dials out when lifting the handset, pressing the speakerphone key or the line key. Leaving it blank disables hotline feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	features.hotline_number = 3601

Parameter-	Configuration File
features.hotline_delay	<y0000000000xx>.cfg
Description	Configures the waiting time (in seconds) the IP phone automatically dials out the hotline number. If it is set to 0 (0s), the IP phone will immediately dial out the preconfigured hotline number when you lift the handset, press the speakerphone key or press the line key. If it is set to a value greater than 0, the IP phone will wait the specified seconds before dialing out the predefined hotline number when you lift the handset, press the speakerphone key or press the line key.
Format	Integer
Default Value	4
Range	0 to 10

Example	features.hotline_delay = 4
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Call Log

Parameter-	Configuration File
features.save_call_history	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to save call log. If it is set to 0 (Disabled), the IP phone cannot log the placed calls, received calls, missed calls and the forwarded calls in the call log lists.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.save_call_history = 1

Missed Call Log

Parameter-	Configuration File
account.X.missed_callog	<MAC>.cfg
Description	Enables or disables missed call log feature for account X. If it is set to 0 (Disabled), there is no indicator displaying on the LCD screen, the IP phone does not log the missed call in the Missed Calls list. If it is set to 1 (Enabled), a prompt message "<number> New Missed Call(s)" along with an indicator icon is displayed on the IP phone idle screen when the IP phone misses calls. X ranges from 1 to 6.
Format	Boolean
Default Value	1
Range	Valid values are:

	0-Disabled 1-Enabled
Example	account.1.missed_calllog = 1

Live Dialpad

Parameter-	Configuration File
phone_setting.predial_autodial	<y0000000000xx>.cfg
Description	Enables or disables live dialpad feature. If it is set to 1 (Enabled), the IP phone will automatically dial out the entered phone number without having to press any key.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	phone_setting.predial_autodial = 1

Parameter-	Configuration File
phone_setting.inter_digit_time	<y0000000000xx>.cfg
Description	Configures the time (in seconds) for the phone to automatically dial out the entered digits without pressing any other key. Note: It works only if the parameter "phone_setting.predial_autodial" is set to 1 (Enabled).
Format	Integer
Default Value	4
Range	1 to 14
Example	phone_setting.inter_digit_time = 4

Call Waiting

Parameter-	Configuration File
call_waiting.enable	<y0000000000xx>.cfg
Description	<p>Enables or disables call waiting feature.</p> <p>If it is set to 0 (Disabled), a new incoming call is automatically rejected by the IP phone with a busy message while during a call.</p> <p>If it is set to 1 (Enabled), the LCD screen will present a new incoming call while during a call.</p>
Format	Boolean
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	call_waiting.enable = 1

Parameter-	Configuration File
call_waiting.tone	<y0000000000xx>.cfg
Description	<p>Enables or disables the playing of a call waiting tone when the IP phone receives an incoming call during a call.</p> <p>If it is set to 1 (Enabled), the IP phone will perform an audible indicator when receiving a new incoming call during a call.</p> <p>Note: It works only if the parameter "call_waiting.enable" is set to 1 (Enabled).</p>
Format	Boolean
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	call_waiting.tone = 1

Parameter-	Configuration File
call_waiting.on_code	<y0000000000xx>.cfg
Description	Configures the call waiting on code to activate the server-side call waiting feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.on_code = *72

Parameter-	Configuration File
call_waiting.off_code	<y0000000000xx>.cfg
Description	Configures the call waiting off code to deactivate the server-side call waiting feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	call_waiting.off_code = *73

Auto Redial

Parameter-	Configuration File
auto_redial.enable	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to automatically redial the called number when it is busy. If it is set to 1 (Enabled), the IP phone will dial the previous dialed out number automatically when the dialed number is busy.
Format	Boolean
Default Value	0
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	auto_redial.enable = 1

Parameter-	Configuration File
auto_redial.interval	<y0000000000xx>.cfg
Description	Configures the interval (in seconds) for the IP phone to wait between redials. The IP phone redials the dialed number at regular intervals till the callee answers the call.
Format	Integer
Default Value	10
Range	1 to 300
Example	auto_redial.interval = 10

Parameter-	Configuration File
auto_redial.times	<y0000000000xx>.cfg
Description	Configures the redial times for the IP phone. The IP phone tries to redial the dialed number as many times as configured till the callee answers the call.
Format	Integer
Default Value	10
Range	1 to 300
Example	auto_redial.times = 10

Auto Answer

Parameter-	Configuration File
account.X.auto_answer	<MAC>.cfg
Description	Enables or disables auto answer feature for account X. If it is set to 1 (Enabled), the IP phone can automatically answer an incoming call. X ranges from 1 to 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	Valid values are: 0-Disabled 1-Enabled
Example	account.1.auto_answer = 1

Parameter- features.auto_answer_delay	Configuration File <y0000000000xx>.cfg
Description	Configures the delay time (in seconds) before the phone automatically answers an incoming call.
Format	Integer
Default Value	1
Range	1 to 4
Example	features.auto_answer_delay = 1

Call Completion

Parameter- features.call_completion_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables call completion feature. If a user places a call and the callee is temporarily not available to answer the call, call completion feature allows notifying the user when the callee becomes available to receive a call. If it is set to 1 (Enabled), the caller is notified when the callee becomes available to receive a call.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled

Example	features.call_completion_enable = 1
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Anonymous Call

Parameter-	Configuration File
account.X.anonymous_call	<MAC>.cfg
Description	<p>Enables or disables anonymous call feature for account X.</p> <p>If it is set to 1 (Enabled), the IP phone will block its identity from showing up to the callee when placing a call. The callee's phone LCD screen presents anonymous instead of the caller's identity.</p> <p>X ranges from 1 to 6.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	account.1.anonymous_call = 1

Parameter-	Configuration File
account.X.send_anonymous_code	<MAC>.cfg
Description	<p>Configures the phone to send anonymous on/off code to activate/deactivate the server-side anonymous call feature for account X.</p> <p>If it is set to 0 (Off Code), the IP phone will send anonymous off code to deactivate the server-side anonymous call feature.</p> <p>If it is set to 1 (On Code), the IP phone will send anonymous on code to activate the server-side anonymous call feature.</p> <p>X ranges from 1 to 6.</p>
Format	Boolean
Default Value	0

Range	Valid values are: 0-Off Code 1-On Code
Example	account.1.send_anonymous_code = 0

Parameter- account.X.anonymous_call_oncode	Configuration File <MAC>.cfg
Description	Configures the anonymous call on code to activate the server-side anonymous call feature for account X. X ranges from 1 to 6. Note: It works only if the parameter "account.X.send_anonymous_code" is set to 1 (Enabled).
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_oncode = *72

Parameter- account.X.anonymous_call_offcode	Configuration File <MAC>.cfg
Description	Configures the anonymous call off code to deactivate the server-side anonymous call feature for account X. X ranges from 1 to 6. Note: It works only if the parameter "account.X.send_anonymous_code" is set to 1 (Enabled).
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_call_offcode = *73

Anonymous Call Rejection

Parameter- account.X.reject_anonymous_call	Configuration File <MAC>.cfg
Description	Enables or disables anonymous call rejection feature for account X. If it is set to 1 (Enabled), the IP phone will automatically reject incoming calls from users enabled anonymous call feature. The anonymous user's phone LCD screen presents "Anonymity Disallowed". X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.reject_anonymous_call = 1

Parameter- account.X.anonymous_reject_oncode	Configuration File <MAC>.cfg
Description	Configures the anonymous call rejection on code to activate the server-side anonymous call rejection feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_oncode = *74

Parameter- account.X.anonymous_reject_offcode	Configuration File <MAC>.cfg
Description	Configures the anonymous call rejection off

	code to deactivate the server-side anonymous call rejection feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.anonymous_reject_offcode = *75

Do Not Disturb

Return Message When DND

Parameter-	Configuration File
features.dnd_refuse_code	<y0000000000xx>.cfg
Description	Configures a return code and reason of SIP response messages when rejecting an incoming call by DND. A specific reason is displayed on the caller's phone LCD screen. If it is set to 486 (Busy here), the caller's phone LCD screen will display the reason "Busy here" when the callee enables DND feature.
Format	Integer
Default Value	480
Range	Valid values are: 404 -No Found 480 -Temporarily not available 486 -Busy here
Example	features.dnd_refuse_code = 480

DND Mode

Parameter-	Configuration File
features.dnd_mode	<y0000000000xx>.cfg
Description	Configures the DND mode for the IP phone. If it is set to 0 (Phone), DND feature is effective for the IP phone. If it is set to 1 (Custom), you can configure DND feature for each account.

Format	Integer
Default Value	0
Range	Valid values are: 0-Phone 1-Custom
Example	features.dnd_mode = 0

DND in Phone Mode

Parameter-	Configuration File
features.dnd.enable	<y0000000000xx>.cfg
Description	Enables or disables DND feature. If it is set to 1 (Enabled), the IP phone will reject incoming calls on all accounts.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.dnd.enable = 1

Parameter-	Configuration File
features.dnd.on_code	<y0000000000xx>.cfg
Description	Configures the DND on code to activate the server-side DND feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	features.dnd.on_code = *71

Parameter-	Configuration File
features.dnd.off_code	<y0000000000xx>.cfg
Description	Configures the DND off code to deactivate the server-side DND feature.
Format	String

Default Value	Blank
Range	String within 32 characters
Example	features.dnd.off_code = *72

DND in Custom Mode

Parameter-	Configuration File
account.X.dnd.enable	<MAC>.cfg
Description	Enables or disables DND feature for account X. If it is set to 1 (Enabled), the IP phone will reject incoming calls on account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.dnd.enable = 1

Parameter-	Configuration File
account.X.dnd.on_code	<MAC>.cfg
Description	Configures the DND on code to activate the server-side DND feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.dnd.on_code = *73

Parameter-	Configuration File
account.X.dnd.off_code	<MAC>.cfg
Description	Configures the DND off code to deactivate the server-side DND feature for account X. X ranges from 1 to 6.
Format	String

Default Value	Blank
Range	String within 32 characters
Example	account.1.dnd.off_code = *74

Busy Tone Delay

Parameter-	Configuration File
features.busy_tone_delay	<y0000000000xx>.cfg
Description	Configures 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 it is 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 = 0

Return Code When Refuse

Parameter-	Configuration File
features.normal_refuse_code	<y0000000000xx>.cfg
Description	Configures a return code and reason of SIP response messages when rejecting an incoming call. A specific reason is displayed on the caller's phone LCD screen. If it is set to 486 (Busy here), the caller's phone LCD screen will display the 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 = 486

180 Ring Workaround

Parameter-	Configuration File
phone_setting.is_deal180	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message.</p> <p>If it is set to 1 (Enabled), the IP phone will resume and play the local ringback tone upon a subsequent 180 message received.</p>
Format	Boolean
Default Value	1
Range	Valid values are: 0 -Disabled 1 -Enabled
Example	phone_setting.is_deal180 = 1

Use Outbound Proxy in Dialog

Parameter-	Configuration File
sip.use_out_bound_in_dialog	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to send the SIP requests to the outbound proxy server.</p> <p>If it is set to 1 (Enabled), all the SIP request messages from the IP phone will be forced to send to the outbound proxy server.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p>
Format	Boolean

Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	sip.use_out_bound_in_dialog = 1

SIP Session Timer

Parameter-	Configuration File
account.X.advanced.timer_t1	<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
Range	0.5 to 10
Example	account.1.advanced.timer_t1 = 0.5

Parameter-	Configuration File
account.X.advanced.timer_t2	<MAC>.cfg
Description	Configures the session timer T2 (in seconds) for account X. T2 represents the maximum retransmit interval for non-INVITE requests and INVITE responses. X ranges from 1 to 6.
Format	Float
Default Value	4
Range	2 to 40
Example	account.1.advanced.timer_t2 = 4

Parameter-	Configuration File
account.X.advanced.timer_t4	<MAC>.cfg
Description	Configures the session timer of T4 (in seconds) for account X. T4 represents the maximum duration a message will remain in the network. X ranges from 1 to 6.
Format	Float
Default Value	5
Range	2.5 to 60
Example	account.1.advanced.timer_t4 = 5

Session Timer

Parameter-	Configuration File
account.X.session_timer.enable	<MAC>.cfg
Description	Enables or disables the session timer for account X. If it is set to 1 (Enabled), IP phone will send periodic re-INVITE requests to refresh the session during a call. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.session_timer.enable = 1

Parameter-	Configuration File
account.X.session_timer.expires	<MAC>.cfg
Description	Configures the IP phone to refresh the session during a call at regular intervals (in seconds) for account X. If it is set to 1800 (1800s), the IP phone will

	refresh the session during a call before 1800 seconds. X ranges from 1 to 6.
Format	Integer
Default Value	1800
Range	30 to 7200
Example	account.1.session_timer.expires = 1800

Parameter-	Configuration File
account.X.session_timer.refresher	<MAC>.cfg
Description	Configures the session timer refresher for account X. If it is set to 0 (UAC), refreshing the session is performed by the IP phone. If it is set to 1 (UAS), refreshing the session is performed by a SIP server. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	Valid values are: 0-UAC 1-UAS
Example	account.1.session_timer.refresher = 0

Call Hold

Parameter-	Configuration File
features.play_hold_tone.enable	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to play a tone when there is a hold call on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled

	1-Enabled
Example	features.play_hold_tone.enable = 1

Parameter-	Configuration File
features.play_hold_tone.delay	<y0000000000xx>.cfg
Description	<p>Configures the interval (in seconds) at which the IP phone plays a hold tone.</p> <p>If it is set to 30 (30s), the IP phone will play a hold tone every 30 seconds when there is a hold call on the IP phone.</p> <p>Note: It works only if the parameter "features.play_hold_tone.enable" is set to 1 (Enabled).</p>
Format	Integer
Default Value	30
Range	3 to 3600
Example	features.play_hold_tone.delay = 30

Parameter-	Configuration File
sip.rfc2543_hold	<y0000000000xx>.cfg
Description	<p>Configures whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.</p> <p>If it is set to 0 (Disabled), SDP media direction attributes (such as a=sendonly) per RFC 3264 is used when placing a call on hold.</p> <p>If it is set to 1 (Enabled), SDP media connection address c=0.0.0.0 per RFC 2543 is used when placing a call on hold.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	sip.rfc2543_hold = 0

Call Forward

Call Forward Mode

Parameter-	Configuration File
features.fwd_mode	<y0000000000xx>.cfg
Description	Configures the call forward mode for the IP phone. If it is set to 0 (Phone), call forward feature is effective for the IP phone. If it is set to 1 (Custom), you can configure call forward feature for each account.
Format	Integer
Default Value	0
Range	Valid values are: 0-Phone 1-Custom
Example	features.fwd_mode = 0

Call Forward in Phone Mode

Always Forward

Parameter-	Configuration File
forward.always.enable	< y0000000000xx >.cfg
Description	Enables or disables always forward feature. If it is set to 1 (Enabled), incoming calls are forwarded to the destination number immediately.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	forward.always.enable = 1

Parameter- forward.always.target	Configuration File < y0000000000xx >.cfg
Description	Configures the destination number of the always forward.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.always.target = 3601

Parameter- forward.always.on_code	Configuration File < y0000000000xx >.cfg
Description	Configures the always forward on code to activate the server-side always forward feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.always.on_code = *72

Parameter- forward.always.off_code	Configuration File < y0000000000xx >.cfg
Description	Configures the always forward off code to deactivate the server-side always forward feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.always.off_code = *73

Busy Forward

Parameter- forward.busy.enable	Configuration File < y0000000000xx >.cfg
Description	Enables or disables busy forward feature. If it is set to 1 (Enabled), incoming calls are

	forwarded to the destination number when the callee is busy.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	forward.busy.enable = 1

Parameter- forward.busy.target	Configuration File < y0000000000xx >.cfg
Description	Configures the destination number of the busy forward.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.busy.target = 3602

Parameter- forward.busy.on_code	Configuration File < y0000000000xx >.cfg
Description	Configures the busy forward on code to activate the server-side busy forward feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.busy.on_code = *74

Parameter- forward.busy.off_code	Configuration File < y0000000000xx >.cfg
Description	Configures the busy forward off code to deactivate the server-side busy forward feature.
Format	String

Default Value	Blank
Range	String within 32 characters
Example	forward.busy.off_code = *75

No Answer Forward

Parameter-	Configuration File
forward.no_answer.enable	< y0000000000xx >.cfg
Description	Enables or disables no answer forward feature. If it is set to 1 (Enabled), incoming calls are forward to the destination number after a period of ring time.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	forward.no_answer.enable = 1

Parameter-	Configuration File
forward.no_answer.target	< y0000000000xx >.cfg
Description	Configures the destination number of the no answer forward.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.target = 3603

Parameter-	Configuration File
forward.no_answer.timeout	< y0000000000xx >.cfg
Description	Configures ring times (N) to wait before forwarding incoming calls. Incoming calls will be forwarded when not answered after N*6 seconds.
Format	Integer

Default Value	2
Range	0 to 20
Example	forward.no_answer.timeout = 2

Parameter- forward.no_answer.on_code	Configuration File < y0000000000xx >.cfg
Description	Configures the no answer forward on code to activate the server-side no answer forward feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.on_code = *76

Parameter- forward.no_answer.off_code	Configuration File < y0000000000xx >.cfg
Description	Configures the no answer forward off code to deactivate the server-side no answer forward feature.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	forward.no_answer.off_code = *77

Call Forward in Custom Mode

Always Forward

Parameter- account.X.always_fwd.enable	Configuration File <MAC>.cfg
Description	Enables or disables always forward feature for account X. If it is set to 1 (Enabled), incoming calls to the account X are forwarded to the destination number immediately. X ranges from 1 to 6.

Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.always_fwd.enable = 1

Parameter- account.X.always_fwd.target	Configuration File <MAC>.cfg
Description	Configures the destination number of the always forward for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.always_fwd.target = 3601

Parameter- account.X.always_fwd.on_code	Configuration File <MAC>.cfg
Description	Configures the always forward on code to activate the server-side always forward feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.always_fwd.on_code = *72

Parameter- account.X.always_fwd.off_code	Configuration File <MAC>.cfg
Description	Configures the always forward off code to deactivate the server-side always forward feature for account X. X ranges from 1 to 6.

Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.off_code = *73

Busy Forward

Parameter- account.X.busy_fwd.enable	Configuration File <MAC>.cfg
Description	Enables or disables busy forward feature for account X. If it is set to 1 (Enabled), incoming calls to the account X are forwarded to the destination number when the callee is busy. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.busy_fwd.enable = 1

Parameter- account.X.busy_fwd.target	Configuration File <MAC>.cfg
Description	Configures the destination number of the busy forward for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.target = 3602

Parameter- account.X.busy_fwd.on_code	Configuration File <MAC>.cfg
Description	Configures the busy forward on code to

	activate the server-side busy forward feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.on_code = *74

Parameter-	Configuration File
account.X.busy_fwd.off_code	<MAC>.cfg
Description	Configures the busy forward off code to deactivate the server-side busy forward feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.busy_fwd.off_code = *75

No Answer Forward

Parameter-	Configuration File
account.X.timeout_fwd.enable	<MAC>.cfg
Description	Enables or disables no answer forward feature for account X. If it is set to 1 (Enabled), incoming calls to the account X are forward to the destination number after a period of ring time. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.timeout_fwd.enable = 1

Parameter-	Configuration File
account.X.timeout_fwd.target	<MAC>.cfg
Description	Configures the destination number of the no answer forward for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.timeout_fwd.target = 3603

Parameter-	Configuration File
account.X.timeout_fwd.timeout	<MAC>.cfg
Description	Configures ring times (N) to wait before forwarding incoming calls for account X. Incoming calls will be forwarded when not answered after N*6 seconds X ranges from 1 to 6.
Format	Integer
Default Value	2
Range	0 to 20
Example	account.1.timeout_fwd.timeout = 2

Parameter-	Configuration File
account.X.timeout_fwd.on_code	<MAC>.cfg
Description	Configures the no answer forward on code to activate the server-side no answer forward feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.timeout_fwd.on_code = *76

Parameter-	Configuration File
account.X.timeout_fwd.off_code	<MAC>.cfg
Description	Configures the no answer forward off code to activate the server-side no answer forward feature for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.timeout_fwd.off_code = *77

Fwd International

Parameter-	Configuration File
forward.international.enable	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to forward an incoming call to an international phone number.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	forward.international.enable = 1

Call Transfer

Parameter-	Configuration File
transfer.blind_tran_on_hook_enable	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to complete the blind transfer through on-hook.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled

	1-Enabled
Example	transfer.blind_tran_on_hook_enable = 1

Parameter- transfer.on_hook_trans_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to complete the semi-attended transfer or the attended transfer through on-hook.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	transfer.on_hook_trans_enable = 1

Parameter- transfer.semi_attend_tran_enable	Configuration File <y0000000000xx>.cfg
Description	Configures whether to display the missed call prompt on the destination party's phone when performing a semi-attended transfer.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Enabled 1-Disabled
Example	transfer.semi_attend_tran_enable = 1

Network Conference

Parameter- account.X.conf_type	Configuration File <MAC>.cfg
Description	Configures the conference type for account X. If it is set to 0 (Local Conference), conferences are set up on the IP phone

	locally. If it is set to 2 (Network Conference), conferences are set up by the server. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	Valid values are: 0-Local Conference 2-Network Conference
Example	account.1.conf_type = 0

Parameter-	Configuration File
account.X.conf_uri	<MAC>.cfg
Description	Configures 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	SIP URI
Default Value	Blank
Range	SIP URI within 511 characters
Example	account.1.conf_uri = conference@example.com

Transfer on Conference Hang Up

Parameter-	Configuration File
transfer.tran_others_after_conf_enable	<y0000000000xx>.cfg
Description	Enables or disables 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	Valid values are: 0-Disabled 1-Enabled
Example	transfer.tran_others_after_conf_enable = 1

Directed Call Pickup

Phone Basis

Parameter- features.pickup.direct_pickup_enable	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to display the DPickup soft key when the IP phone is off-hook. Note: It is not applicable to SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.pickup.direct_pickup_enable = 1

Parameter- features.pickup.direct_pickup_code	Configuration File <MAC>.cfg
Description	Configures the directed call pickup code on a phone basis. Note: The directed call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.
Format	String
Default Value	Blank

Range	String within 32 characters
Example	features.pickup.direct_pickup_code = *97

Per-line Basis

Parameter-	Configuration File
account.X.direct_pickup_code	<MAC>.cfg
Description	Configures the directed call pickup code on a per-line basis. X ranges from 1 to 6. Note: The directed call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.direct_pickup_code = *68

Group Call Pickup

Phone Basis

Parameter-	Configuration File
features.pickup.group_pickup_enable	<MAC>.cfg
Description	Enables or disables the IP phone to display the GPickup soft key when the IP phone is off-hook. Note: It is not applicable to SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.pickup.group_pickup_enable = 1

Parameter-	Configuration File
features.pickup.group_pickup_code	<MAC>.cfg
Description	Configures the group call pickup code on a phone basis. Note: The group call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	features.pickup.group_pickup_code = *98

Per-line Basis

Parameter-	Configuration File
account.X.group_pickup_code	<MAC>.cfg
Description	Configures the group call pickup code on a per-line basis. X ranges from 1 to 6. Note: The group call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	account.1.group_pickup_code = *69

Dialog Info Call Pickup

Parameter-	Configuration File
account.X.dialoginfo_callpickup	<MAC>.cfg
Description	Configures Dialog Info Call Pickup feature for account X. If it is set to 1 (Enabled), call pickup is implemented through SIP signals. X ranges from 1 to 6.

	Note: It is not applicable to SIP-T19P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.dialoginfo_callpickup = 1

Web Server Type

Parameter-	Configuration File
wui.http_enable	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to access its web user interface using HTTP protocol. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	wui.http_enable = 1

Parameter-	Configuration File
network.port.http	<y0000000000xx>.cfg
Description	Configures the HTTP port used to access the web user interface of the IP phone. The default HTTP port is 80. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	80

Range	1 to 65535
Example	network.port.http = 80

Parameter- wui.https_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to access its web user interface using HTTPS protocol. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	wui.https_enable = 1

Parameter- network.port.https	Configuration File <y0000000000xx>.cfg
Description	Configures the HTTPS port used to access the web user interface of the IP phone. The default HTTPS port is 443. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	443
Range	1 to 65535
Example	network.port.https = 443

Calling Line Identification Presentation

Parameter- account.X.cid_source	Configuration File <MAC>.cfg
Description	Configures the presentation of the caller

	<p>identity for account X.</p> <p>0-FROM (Derives the name and number of the caller from the "From" header).</p> <p>1-PAI (Derives the name and number of the caller from the "PAI" header. If the server does not send the "PAI" header, displays "anonymity" on the callee's phone).</p> <p>2-PAI-FROM (Derives the name and number of the caller from the "PAI" header preferentially. If the server does not send the "PAI" header, derives from the "From" header).</p> <p>3-RPID-PAI-FROM</p> <p>4-PAI-RPID-FROM</p> <p>5-RPID-FROM</p> <p>X ranges from 1 to 6.</p>
Format	Integer
Default Value	0
Range	0 to 5
Example	account.1.cid_source = 0

Connected Line Identification Presentation

Parameter-	Configuration File
account.X.cp_source	<MAC>.cfg
Description	<p>Configures the presentation of the callee's identity for account X.</p> <p>0-PAI-RPID (Derives the name and number of the callee from the "PAI" header preferentially. If the server does not send the "PAI" header, derives from the "RPID" header).</p> <p>1-Dialed Digits (Preferentially displays the dialed digits on the caller's phone).</p> <p>2-RFC 4916 (Derives the name and number of the callee from "From" header in the Update message).</p> <p>When the RFC 4916 is enabled on the IP phone, the caller sends the SIP request message which contains the from-change</p>

	<p>tag in the Supported header. The caller then receives an UPDATE message from the callee, and displays the identity in the From header.</p> <p>X ranges from 1 to 6.</p>
Format	Integer
Default Value	0
Range	0 to 2
Example	account.1.cp_source = 0

DTMF

Parameter-	Configuration File
account.X.dtmf.type	<MAC>.cfg
Description	<p>Configures the DTMF type for account X.</p> <p>If it is set to 0 (INBAND), DTMF digits are transmitted in the voice band.</p> <p>If it is set to 1 (RFC 2833), DTMF digits are transmitted by RTP Events compliant to RFC 2833.</p> <p>If it is set to 2 (SIP INFO), DTMF digits are transmitted by the SIP INFO messages.</p> <p>If it is set to 3 (AUTO or SIP INFO), the IP phone negotiates with the other end to use INBAND or RFC 2833, if there is no negotiation, using SIP INFO by default.</p> <p>X ranges from 1 to 6.</p>
Format	Integer
Default Value	1
Range	<p>Valid values are:</p> <p>0-INBAND</p> <p>1-RFC 2833</p> <p>2-SIP INFO</p> <p>3-AUTO or SIP INFO</p>
Example	account.1.dtmf.type = 1

Parameter-	Configuration File
account.X.dtmf.dtmf_payload	<MAC>.cfg
Description	Configures the RFC 2833 payload type. X ranges from 1 to 6.
Format	Integer
Default Value	101
Range	96 to 127
Example	account.1.dtmf.dtmf_payload = 101

Parameter-	Configuration File
account.X.dtmf.info_type	<MAC>.cfg
Description	Configures the DTMF info type when the DTMF type is configured as "SIP INFO", "AUTO or 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 = 0

Parameter-	Configuration File
features.dtmf.repetition	<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 = 3

Suppress DTMF Display

Parameter- features.dtmf.hide	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to suppress the display of DTMF digits. If it is set to 1 (Enabled), the DTMF digits are displayed as asterisks.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.dtmf.hide = 1

Parameter- features.dtmf.hide_delay	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to display the DTMF digits for a short period before displaying asterisks. Note: It works only if the parameter "features.dtmf.hide" is set to 1 (Enabled). It is not applicable to SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.dtmf.hide_delay = 1

Transfer via DTMF

Parameter- features.dtmf.replace_tran	Configuration File <y0000000000xx>.cfg
Description	Enables or disables transfer via DTMF

	<p>feature.</p> <p>If it is set to 0 (Disabled), the IP phone will perform the transfer as normal when pressing the transfer key during a call.</p> <p>If it is set to 1 (Enabled), the IP phone will transmit the specified DTMF digits to the server for completing call transfer when pressing the transfer key during a call.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	features.dtmf.replace_tran = 1

Parameter-	Configuration File
features.dtmf.transfer	<y0000000000xx>.cfg
Description	<p>Configures the DTMF digits to be transmitted to complete the transfer.</p> <p>Note: It works only if the parameter "features.dtmf.replace_tran" is set to 1 (Enabled).</p>
Format	String
Default Value	Blank
Range	<p>Valid values are: 0-9, *, # and A-D.</p> <p>String within 32 characters</p>
Example	features.dtmf.transfer = 123

Incoming Intercom calls

Parameter-	Configuration File
features.intercom.allow	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to automatically answer an incoming intercom call.</p> <p>If it is set to 0 (Disabled), the IP phone will reject incoming intercom calls and sends a</p>

	<p>busy signal to the caller.</p> <p>If it is set to 1 (Enabled), the IP phone will automatically answer an incoming intercom call.</p>
Format	Boolean
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	features.intercom.allow = 1

Parameter- features.intercom.mute	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to mute the microphone when answering an intercom call.</p> <p>If it is set to 0 (Disabled), the microphone is un-muted for incoming calls.</p> <p>If it is set to 1 (Enabled), the microphone is muted for intercom calls.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	features.intercom.mute = 1

Parameter- features.intercom.tone	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to play a warning tone when receiving an intercom call.</p> <p>If it is set to 0 (Disabled), the IP phone will automatically answer the intercom call without a warning tone.</p> <p>If it is set to 1 (Enabled), the IP phone will</p>

	play a warning tone to alert you before answering the intercom call.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.intercom.tone = 1

Parameter-	Configuration File
features.intercom.barge	<y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to automatically answer an incoming intercom call while there is already an active call on the IP phone.</p> <p>If it is set to 0 (Disabled), the IP phone will handle an incoming intercom call like a waiting call while there is already an active call on the IP phone.</p> <p>If it is set to 1 (Enabled), the IP phone will automatically answer the intercom call while there is already an active call on the IP phone and place the active call on hold.</p>
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.intercom.barge = 1

Distinctive Ring Tones

Parameter-	Configuration File
features.alert_info_tone	<y0000000000xx>.cfg
Description	Enables and disables the IP phone to map the keywords in the Alert-info header to the specified Bellcore ring tones.

Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.alert_info_tone = 1

Parameter- account.X.alert_info_url_enable	Configuration File <MAC>.cfg
Description	Enables or disables distinctive ring tones feature for account X. If it is set to 1 (Enabled), the IP phone will try to download the WAV ring tone file from the URL and then play the remote ring tone when the Alert-Info header contains a remote URL. X ranges from 1 to 6.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.alert_info_url_enable = 1

Parameter- distinctive_ring_tones.alert_info.X .text	Configuration File <y0000000000xx>.cfg
Description	Configures the texts to map the keywords contained in the SIP header. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	distinctive_ring_tones.alert_info.1.text = family

Parameter- distinctive_ring_tones.alert_info.X .ringer	Configuration File <y0000000000xx>.cfg
Description	Configures the desired ring tones for each text. The value ranges from 1 to 5, the digit stands for the appropriate ring tone. X ranges from 1 to 10.
Format	Integer
Default Value	1
Range	Valid values are: 1-Ring1.wav 2-Ring2.wav 3-Ring3.wav 4-Ring4.wav 5-Ring5.wav
Example	distinctive_ring_tones.alert_info.1.ringer = 1

Tones

Parameter- voice.tone.country	Configuration File <y0000000000xx>.cfg
Description	Configures the country tone for the IP phone.
Format	String
Default Value	Custom
Range	Valid values are: <ul style="list-style-type: none"> • Custom • Australia • Austria • Brazil • Belgium • China • Czech • Denmark • Finland • France

	<ul style="list-style-type: none"> • Germany • Great Britain • Greece • Hungary • Lithuania • India • Italy • Japan • Mexico • New Zealand • Netherlands • Norway • Portugal • Spain • Switzerland • Sweden • Russia • United States • Chile • Czech ETSI
Example	voice.tone.country = Custom

Parameter- voice.tone.dial voice.tone.ring voice.tone.busy voice.tone.congestion voice.tone.callwaiting voice.tone.dialrecall voice.tone.info voice.tone.stutter voice.tone.message (not applicable to SIP-T20P IP phones) voice.tone.autoanswer	Configuration File <y0000000000xx>.cfg
Description	Configures the tone for each condition. tonelist = element[,element] [,element]... Where element = [!]Freq1[+Freq2][+Freq3][+Freq4] /Duration Freq : the frequency of the tone (ranges from 200 to 7000 Hz). If it is set to 0 (0Hz), it means the tone is not played. A tone is comprised of

	<p>at most four different frequencies.</p> <p>Duration: the time duration (in milliseconds, ranges from 0 to 30000ms) of the ring tone.</p> <p>You can configure at most eight different tones for one condition, and separate tones by commas (e.g., 250/200, !0/1000, 200+300/500, 600+700+800+1000/2000). The exclamation point (!) can be added optionally, which means these tones are only played once.</p> <p>Note: It works only if the parameter "voice.tone.country" is set to Custom.</p>
Format	Refer to the introduction above
Default Value	Blank
Range	Not Applicable
Example	voice.tone.dial = 800+200/1000, 0/100, 500/1200, 500+600+950+1500/5000

Remote Phone Book

Parameter-	Configuration File
remote_phonebook.data.X.url	<y0000000000xx>.cfg
Description	<p>Configures the access URL of the remote XML phone book.</p> <p>X ranges from 1 to 5.</p> <p>Note: It is not applicable to SIP-T20P IP phones.</p>
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml

Parameter-	Configuration File
remote_phonebook.data.X.name	<y0000000000xx>.cfg
Description	Configures the display name of the remote phone book item.

	Note: It is not applicable to SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.data.1.name = yl01

Parameter- remote_phonebook.display_name	Configuration File <y0000000000xx>.cfg
Description	Configures the display name of the remote phone book. If you leave it blank, Remote Phone Book is displayed on the LCD screen at the path Menu->Directory. Note: It is not applicable to SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	remote_phonebook.display_name = Remote Phone Book

Parameter- features.remote_phonebook.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform a remote phone book search when receiving an incoming call. Note: It is not applicable to SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.remote_phonebook.enable = 1

Parameter- features.remote_phonebook.flash_time	Configuration File <y0000000000xx>.cfg
Description	Configures how often to refresh the local cache of the remote phone book. If it is set to 3600 (3600s), the IP phone will refresh the local cache of the remote phone book every 3600 seconds. Note: It is not applicable to SIP-T20P IP phones.
Format	Integer
Default Value	21600
Range	3600 to 2592000
Example	features.remote_phonebook.flash_time = 21600

LDAP

Parameter- ldap.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables LDAP feature on the IP phone. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	ldap.enable =1

Parameter- ldap.name_filter	Configuration File <y0000000000xx>.cfg
Description	Configures 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. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
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 LCD screen.

Parameter-	Configuration File
ldap.number_filter	<y0000000000xx>.cfg
Description	Configures 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. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
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 LCD screen.

Parameter-	Configuration File
ldap.host	<y0000000000xx>.cfg
Description	Configures the IP address or domain name of the LDAP server. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 99 characters
Example	ldap.host = 192.168.1.20

Parameter-	Configuration File
ldap.port	<y0000000000xx>.cfg
Description	Configures the LDAP server port. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Integer
Default Value	389
Range	1 to 65535
Example	ldap.port = 389

Parameter-	Configuration File
ldap.base	<y0000000000xx>.cfg
Description	Configures the LDAP search base which corresponds to the location in the LDAP phone book from which the LDAP search request begins. The search base narrows the search scope and decreases directory search time. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	ldap.base = dc=yealink,dc=cn

Parameter-	Configuration File
ldap.user	<y0000000000xx>.cfg
Description	<p>Configures the user name uses to login the LDAP server.</p> <p>This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the user name to access the LDAP server.</p> <p>Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.</p>
Format	String
Default Value	Blank
Range	String within 99 characters
Example	ldap.user = cn=manager,dc=yealink,dc=cn

Parameter-	Configuration File
ldap.password	<y0000000000xx>.cfg
Description	<p>Configures the password to login the LDAP server.</p> <p>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.</p> <p>Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.</p>
Format	String
Default Value	Blank
Range	String within 99 characters
Example	ldap.password = secret

Parameter-	Configuration File
ldap.max_hits	<y0000000000xx>.cfg
Description	<p>Configures the maximum number of search results to be returned by the LDAP server. If the value of the "Max.Hits" is blank, the</p>

	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. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Integer
Default Value	50
Range	1 to 32000
Example	ldap.max_hits = 50

Parameter-	Configuration File
ldap.name_attr	<y0000000000xx>.cfg
Description	Configures the name attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple name attributes separated by spaces. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	ldap.name_attr = cn sn

Parameter-	Configuration File
ldap.numb_attr	<y0000000000xx>.cfg
Description	Configures the number attributes of each record to be returned by the LDAP server. It compresses the search results. You can configure multiple number attributes separated by spaces. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	String
Default Value	Blank

Range	String within 99 characters
Example	ldap.numb_attr = telephoneNumber

Parameter-	Configuration File
ldap.display_name	<y0000000000xx>.cfg
Description	Configures the display name of the contact record displayed on the LCD screen. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones. The value must start with “%” symbol.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	ldap.display_name = %cn The cn of the contact record is displayed on the LCD screen.

Parameter-	Configuration File
ldap.version	<y0000000000xx>.cfg
Description	Configures the LDAP protocol version supported by the IP phone. Make sure the protocol value corresponds with the version assigned on the LDAP server. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Integer
Default Value	3
Range	2 or 3
Example	ldap.version = 3

Parameter-	Configuration File
ldap.call_in_lookup	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform an LDAP search when receiving an incoming call.

	Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	ldap.call_in_lookup = 1

Parameter-	Configuration File
ldap.ldap_sort	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to sort the search results in alphabetical order or numerical order. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	ldap.ldap_sort = 1

BLF

Visual and Audio Alert for BLF Pickup

Parameter-	Configuration File
features.pickup.blf_visual_enable	<MAC>.cfg
Description	Enables or disables the IP phone to display a visual prompt when the monitored user receives an incoming call. Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.
Format	Boolean

Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.pickup.blf_visual_enable = 1

Parameter-	Configuration File
features.pickup.blf_audio_enable	<MAC>.cfg
Description	Enables or disables the IP phone to play an alert tone when the monitored user receives an incoming call. Note: It is not applicable to SIP-T19P IP phones.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.pickup.blf_audio_enable = 1

BLF LED Mode

Parameter-	Configuration File
features.blf_led_mode	<y0000000000xx>.cfg
Description	It configures BLF LED mode and provides four kinds of definition for the BLF key LED status. For more information, refer to Busy Lamp Field on page 142. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Default Value	0
Range	0 to 3
Example	features.blf_led_mode = 1

Music on Hold

Parameter-	Configuration File
account.X.music_server_uri	<MAC>.cfg
Description	Configures the Music on Hold server address. Examples for valid values: <10.1.3.165>, 10.1.3.165, sip:moh@sip.com, <sip:moh@sip.com>, <yealink.com> or yealink.com. X ranges from 1 to 6. Note: The DNS query in this parameter only supports A query.
Format	String
Default Value	Blank
Range	String within 256 characters
Example	account.1.music_server_uri = <10.1.3.165>

ACD

Parameter-	Configuration File
account.X.acd.enable	<MAC>.cfg
Description	Enables or disables ACD feature for account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	account.1.acd.enable = 1

Parameter-	Configuration File
account.X.acd.available	<MAC>.cfg
Description	Enables or disables the IP phone to display the available and unavailable soft keys after the phone logs into the ACD system for

	account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	account.1.acd.available = 1

Parameter- acd.auto_available	Configuration File <y0000000000xx>.cfg
Description	Enables or disables ACD auto available feature. If it is set to 1 (Enabled), the IP phone will automatically change the phone status to available.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	acd.auto_available = 1

Parameter- acd.auto_available_timer	Configuration File <y0000000000xx>.cfg
Description	Configures the length of time (in seconds) before the IP phone state is automatically changed to available. Note: It works only if the parameter "acd.auto_available" is set to 1 (Enabled).
Format	Integer
Default Value	60
Value	0 to 120
Example	acd.auto_available_timer = 60

Message Waiting Indicator

Parameter- account.X.subscribe_mwi	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to subscribe the message waiting indicator to the account for account X. If it is set to 1 (Enabled), the IP phone will send a SUBSCRIBE message to the server for message-summary updates. X 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 SUBSCRIBE 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

Parameter-	Configuration File
voice_mail.number.X	<MAC>.cfg
Description	Configures the voice mail number for account X. X ranges from 1 to 6.
Format	String
Default Value	Blank
Value	String within 99 characters
Example	voice_mail.number.1 = 1234

Parameter-	Configuration File
account.X.subscribe_mwi_to_vm	<MAC>.cfg
Description	Enables or disables the IP phone to subscribe the message waiting indicator to the voice mail number for account X. X ranges from 1 to 6. Note: It works only if the parameters "account.X.subscribe_mwi" is set to 1 (Enabled) and "voice_mail.number.X" is configured.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	account.1.subscribe_mwi_to_vm = 0

Sending RTP Stream

Parameter-	Configuration File
multicast.codec	<y0000000000xx>.cfg
Description	Configures a multicast codec for the IP phone to use to send an RTP stream. Note: It is not applicable to SIP-T19P IP phones.
Format	string

Default Value	G722
Range	<p>Valid values are:</p> <ul style="list-style-type: none"> • PCMU • PCMA • G729 • G722 • G726-16 (not applicable to SIP-T21P) • G726-24 (not applicable to SIP-T21P) • G726-32 • G726-40 (not applicable to SIP-T21P) • G723_53
Example	multicast.codec = G722

Receiving RTP Stream

Parameter- multicast.receive_priority.enable	Configuration File <y0000000000xx>.cfg
Description	<p>Enables or disables the IP phone to handle the incoming multicast paging calls when there is an active multicast paging call on the IP phone.</p> <p>If it is set to 1 (Enabled), the IP phone will answer the incoming multicast paging call with a higher priority and ignore that with a lower priority.</p>
Format	Boolean
Default Value	1
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	multicast.receive_priority.enable = 1

Parameter- multicast.receive_priority.priority	Configuration File < y0000000000xx >.cfg
Description	Configures the priority of multicast paging calls.

	1 is the highest priority, 10 is the lowest priority. If it is set to 0, all incoming multicast paging calls will be automatically ignored.
Format	Integer
Default Value	10
Range	0 to 10
Example	multicast.receive_priority.priority = 10

Parameter-	Configuration File
multicast.listen_address.X.label	< y0000000000xx >.cfg
Description	Configures the label to be displayed on the LCD screen when receiving the RTP multicast. X ranges from 1 to 10.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	multicast.listen_address.1.label = Paging1

Parameter-	Configuration File
multicast.listen_address.X.ip_address	< y0000000000xx >.cfg
Description	Configures the multicast address and port number that the IP phone listens to. X ranges from 1 to 10. Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Format	String
Default Value	Blank
Range	Not Applicable
Example	multicast.listen_address.1.ip_address = 224.5.6.20:10008

Action URL

Parameter-	Configuration File
action_url.setup_completed	<y0000000000xx>.cfg
action_url.registered	
action_url.unregistered	
action_url.register_failed	
action_url.off_hook	
action_url.on_hook	
action_url.incoming_call	
action_url.outgoing_call	
action_url.call_established	
action_url.dnd_on	
action_url.dnd_off	
action_url.always_fwd_on	
action_url.always_fwd_off	
action_url.busy_fwd_on	
action_url.busy_fwd_off	
action_url.no_answer_fwd_on	
action_url.no_answer_fwd_off	
action_url.transfer_call	
action_url.blind_transfer_call	
action_url.attended_transfer_call	
action_url.hold	
action_url.unhold	
action_url.mute	
action_url.unmute	
action_url.missed_call	
action_url.call_terminated	
action_url.busy_to_idle	
action_url.idle_to_busy	
action_url.ip_change	
action_url.forward_incoming_call	
action_url.reject_incoming_call	
action_url.answer_new_incoming_call	
action_url.transfer_finished	

action_url.transfer_failed	
Description	<p>Configures the URL for the predefined event.</p> <p>The value format is: http(s)://IP address of server/help.xml? variable name=variable value.</p> <p>Valid variable values are:</p> <ul style="list-style-type: none"> • \$mac • \$ip • \$model • \$firmware • \$active_url • \$active_user • \$active_host • \$local • \$remote • \$display_local • \$display_remote • \$call_id
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	<pre>action_url.mute = http://192.168.0.20/help.xml?model=\$model</pre>

Action URI

Parameter- features.action_uri_limit_ip	Configuration File <y0000000000xx>.cfg
Description	<p>Configures the address(es) from which Action URI will be accepted.</p> <p>For discontinuous IP addresses, multiple IP addresses are separated by commas.</p> <p>For continuous IP addresses, the format likes *.*.* and the "*" stands for the values 0~255.</p>

	<p>For example: 10.10.*.* stands for the IP addresses that range from 10.10.0.0 to 10.10.255.255.</p> <p>If left blank, the IP phone cannot receive or handle any HTTP GET request.</p> <p>If it is set to "any", the IP phone will accept and handle HTTP GET requests from any IP address.</p>
Format	IP Address or any
Default Value	Blank
Range	String within 511 characters
Example	features.action_uri_limit_ip = any

Server Redundancy

Parameter- account.X.sip_server.Y.address	Configuration File <MAC>.cfg
Description	<p>Configures the IP address or domain name of the SIP server Y for account X.</p> <p>X ranges from 1 to 6.</p> <p>Y ranges from 1 to 2.</p>
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 256 characters
Example	account.1.sip_server.1.address = yealink.pbx.com

Parameter- account.X.sip_server.Y.port	Configuration File <MAC>.cfg
Description	<p>Configures the port of the SIP server Y for account X.</p> <p>X ranges from 1 to 6.</p> <p>Y ranges from 1 to 2.</p>
Format	Integer
Default Value	5060

Range	0 to 65535
Example	account.1.sip_server.1.port = 5060

Parameter- account.X.sip_server.Y.expires	Configuration File <MAC>.cfg
Description	Configures the registration expires (in seconds) of the SIP server Y for account X. X ranges from 1 to 6. Y ranges from 1 to 2.
Format	Integer
Default Value	3600
Range	30 to 2147483647
Example	account.1.sip_server.1.expires = 3600

Parameter- account.X.sip_server.Y.retry_counts	Configuration File <MAC>.cfg
Description	Configures the retry times for the IP phone to resend requests when the SIP server Y does not respond correctly for account X. X ranges from 1 to 6. Y ranges from 1 to 2.
Format	Integer
Default Value	3
Range	0 to 20
Example	account.1.sip_server.1.retry_counts = 3

Fallback Mode

Parameter- account.X.fallback.redundancy_type	Configuration File <MAC>.cfg
Description	Configures the registration mode for the IP phone in fallback mode. X ranges from 1 to 6.
Format	Integer
Default Value	0

Range	Valid values are: 0-Concurrent registration 1-Successive registration
Example	account.1.fallback.redundancy_type = 0

Parameter- account.X.fallback.timeout	Configuration File <MAC>.cfg
Description	Configures the time interval (in seconds) for the IP phone to detect whether the working server is available by sending the registration request after the fallback server takes over call control. It is only applicable to successive registration mode. X ranges from 1 to 6.
Format	Integer
Default Value	120
Range	10 to 2147483647
Example	account.1.fallback.timeout = 120

Failover Mode

Parameter- account.X.sip_server.Y.failback_mode	Configuration File <MAC>.cfg
Description	Configures the mode for the IP phone to retry the primary server in failover mode. X ranges from 1 to 6. Y ranges from 1 to 2.
Format	Integer
Default Value	0
Range	Valid values are: 0-newRequests: all requests are sent to the primary server first, regardless of the last used server. 1-DNSTTL: the IP phone retries to send requests to the primary server after the

	<p>timeout equal to the DNSTTL configured for the server that the IP phone is registered to.</p> <p>2-registration: the IP phone retries to send REGISTER requests to the primary server when registration renewal.</p> <p>3-duration: the IP phone retries to send requests to the primary server after the timeout defined by the <code>account.X.sip_server.Y.failback_timeout</code> parameter.</p>
Example	<code>account.1.sip_server.1.failback_mode = 0</code>

Parameter- <code>account.X.sip_server.Y.failback_timeout</code>	Configuration File <MAC>.cfg
Description	<p>Configures the timeout (in seconds) for the IP phone to retry to send requests to the primary server after failing over to the current working server when the parameter "account.X.sip_server.Y.failback_mode" is set to 3 (duration).</p> <p>If you set the parameter to 0, the IP phone will not send requests to the primary server until a failover event occurs with the current working server.</p> <p>X ranges from 1 to 6. Y ranges from 1 to 2.</p>
Format	Integer
Default Value	3600
Range	0, 60 to 65535
Example	<code>account.1.sip_server.1.failback_timeout = 3600</code>

Parameter- <code>account.X.sip_server.Y.register_on_enable</code>	Configuration File <MAC>.cfg
Description	Enables or disables the IP phone to register to the secondary server before sending

	requests to the secondary server in the failover mode. X ranges from 1 to 6. Y ranges from 1 to 2.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.sip_server.1.register_on_enable = 0

SIP Server Domain Name Resolution

Parameter-	Configuration File
account.X.transport	<MAC>.cfg
Description	Configures the transport type for account X. If the parameter is set to 3 (DNS-NAPTR) and no server port is given, the IP phone performs the DNS NAPTR and SRV queries for the service type and port. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	Valid values are: 0-UDP 1-TCP 2-TLS 3-DNS-NAPTR
Example	account.1.transport = 3

Parameter-	Configuration File
account.X.naptr_build	<MAC>.cfg
Description	Configures UDP SRV query or TCP/TLS SRV query for the IP phone to be performed when no result is returned from NAPTR

	query. X ranges from 1 to 6.
Format	Integer
Default Value	0
Range	Valid values are: 0-UDP 1-TCP or TLS.
Example	account.1.naptr_build = 0

LLDP

Parameter- network.lldp.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables LLDP feature on the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	network.lldp.enable = 1

Parameter- network.lldp.packet_interval	Configuration File <y0000000000xx>.cfg
Description	Configures the amount of time (in seconds) between the transmissions of LLDP packet. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It works only if the parameter "network.lldp.enable" is set to 1 (Enabled).
Format	Integer
Default Value	60

Range	1 to 3600
Example	network.lldp.packet_interval = 60

VLAN

Internet Port

Parameter-	Configuration File
network.vlan.internet_port_enable	<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	Valid values are: 0-Disabled 1-Enabled
Example	network.vlan.internet_port_enable = 1

Parameter-	Configuration File
network.vlan.internet_port_vid	<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	1 to 4094
Example	network.vlan.internet_port_vid = 1

Parameter-	Configuration File
network.vlan.internet_port_priority	<y0000000000xx>.cfg
Description	Configures the priority value used for passing VLAN packets. 7 is the highest priority, 0 is the lowest priority. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	0 to 7
Example	network.vlan.internet_port_priority = 0

PC Port

Parameter-	Configuration File
network.vlan.pc_port_enable	<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	Valid values are: 0-Disabled 1-Enabled
Example	network.vlan.pc_port_enable = 1

Parameter-	Configuration File
network.vlan.pc_port_vid	<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	1 to 4094
Example	network.vlan.pc_port_vid = 1

Parameter- network.vlan.pc_port_priority	Configuration File <y0000000000xx>.cfg
Description	Configures the priority value used for passing VLAN packets. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	0 to 7
Example	network.vlan.pc_port_priority = 0

DHCP VLAN Discovery

Parameter- network.vlan.dhcp_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables DHCP VLAN discovery feature on the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	network.vlan.dhcp_enable = 1

Parameter- network.vlan.dhcp_option	Configuration File <y0000000000xx>.cfg
Description	Configures the DHCP option used to

	request the VLAN ID. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	132
Range	128 to 254
Example	network.vlan.dhcp_option = 132

VPN

Parameter-	Configuration File
network.vpn_enable	<y0000000000xx>.cfg
Description	Enables or disables VPN feature on the IP phone. Note: It is not applicable to SIP-T19P IP phones. If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	network.vpn_enable = 1

Parameter-	Configuration File
openvpn.url	<y0000000000xx>.cfg
Description	Configures the access URL of the OpenVPN TAR package. Note: It is not applicable to SIP-T19P IP phones.
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	openvpn.url = http://192.168.10.25/OpenVPN.tar
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QoS

Parameter- network.qos.rtpptos	Configuration File <y0000000000xx>.cfg
Description	Configures the DSCP for voice packets. The default DSCP value for RTP packets is 46 (Expedited Forwarding). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	46
Range	0 to 63
Example	network.qos.rtpptos = 46

Parameter- network.qos.signaltos	Configuration File <y0000000000xx>.cfg
Description	Configures the DSCP for SIP packets. The default DSCP value for SIP packets is 26 (Assured Forwarding). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	26
Range	0 to 63
Example	network.qos.signaltos = 26

Network Address Translation

Parameter-	Configuration File
account.X.nat.nat_traversal	<MAC>.cfg
Description	Enables or disables the NAT traversal for account X. X ranges from 1 to 6.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	account.1.nat.nat_traversal = 0

Parameter-	Configuration File
account.X.nat.stun_server	<MAC>.cfg
Description	Configures 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	String within 99 characters
Example	account.1.nat.stun_server = 218.107.220.201

Parameter-	Configuration File
account.X.nat.stun_port	<MAC>.cfg
Description	Configures the port of the STUN server. X ranges from 1 to 6.
Format	Integer
Default Value	3478
Range	1024 to 65000
Example	account.1.nat.stun_port = 3478

802.1X

Parameter- network.802_1x.mode	Configuration File <y0000000000xx>.cfg
Description	Configures the types of the 802.1X authentication to use on the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-Disabled 1-EAP-MD5 2-EAP-TLS 3-PEAP-MSCHAPv2 4-EAP-TTLS/EAP-MSCHAPv2
Example	network.802_1x.mode = 1

Parameter- network.802_1x.identity	Configuration File <y0000000000xx>.cfg
Description	Configures the identity used for authenticating the IP phone. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	network.802_1x.identity = admin

Parameter- network.802_1x.md5_password	Configuration File <y0000000000xx>.cfg
Description	Configures the password used for authenticating the IP phone.

	Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to EAP-MD5, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 protocols.
Format	String
Default Value	Blank
Range	String within 32 characters
Example	network.802_1x.md5_password = admin123

Parameter- network.802_1x.root_cert_url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the CA certificate used for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to EAP-TLS, PEAP-MSCHAPv2 and EAP-TTLS/EAP-MSCHAPv2 protocols. The format of the certificate must be *.pem, *.crt, *.cer or *.der.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	network.802_1x.root_cert_url = http://192.168.1.10/ca.pem

Parameter- network.802_1x.client_cert_url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the device certificate used for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to the EAP-TLS protocol. The format of the certificate must be *.pem or *.cer.

Format	URL
Default Value	Blank
Range	String within 511 characters
Example	network.802_1x.client_cert_url = http://192.168.1.10/ client.pem

TR-069

Parameter- managementserver.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables TR-069 feature on the IP phone.
Format	Integer
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	managementserver.enable = 1

Parameter- managementserver.username	Configuration File <y0000000000xx>.cfg
Description	Configures the user name to authenticate with the ACS. This string is set to the empty string if no authentication is required.
Format	String
Default Value	Blank
Range	String within 128 characters
Example	managementserver.username = user1

Parameter- managementserver.password	Configuration File <y0000000000xx>.cfg
Description	Configures the password to authenticate with the ACS. This string is set to the empty string if no authentication is required.

Format	String
Default Value	Blank
Range	String within 64 characters
Example	managementserver.password = pwd123

Parameter- managementserver.url	Configuration File <y0000000000xx>.cfg
Description	Configures the URL of the ACS.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	managementserver.url = http://192.168.1.20/acs/

Parameter- managementserver.connection_re quest_username	Configuration File <y0000000000xx>.cfg
Description	Configures the user name for the IP phone to authenticate the incoming connection requests.
Format	String
Default Value	Blank
Range	String within 128 characters
Example	managementserver.connection_request_username = acsuser

Parameter- managementserver.connection_re quest_password	Configuration File <y0000000000xx>.cfg
Description	Configures the password for the IP phone to authenticate the incoming connection requests.
Format	String
Default Value	Blank

Range	String within 64 characters
Example	managementserver.connection_request_password = acspwd

Parameter- managementserver.periodic_inform_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to periodically report its configuration information to the ACS.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	managementserver.periodic_inform_enable = 1

Parameter- managementserver.periodic_inform_interval	Configuration File <y0000000000xx>.cfg
Description	Configures the interval (in seconds) to report its configuration information to the ACS.
Format	Integer
Default Value	60
Range	5 to 4294967295
Example	managementserver.periodic_inform_interval = 60

IPv6

Parameter- network.ip_address_mode	Configuration File <MAC>.cfg
Description	Configures the IP address mode. Note: If you change this parameter, the IP

	phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-IPv4 1-IPv6 2-IPv4&IPv6
Example	network.ip_address_mode = 1

Parameter- network.ipv6_internet_port.type	Configuration File <MAC>.cfg
Description	Configures the IPv6 address assignment method. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-DHCP 1-Static IP Address
Example	network.ipv6_internet_port.type = 0

Parameter- network.ipv6_static_dns_enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the phone to use manually configured static IPv6 DNS when the parameter "network.ipv6_internet_port.type" is set to 0 (DHCP). Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0

Range	Valid values are: 0-Disabled 1-Enabled
Example	network.ipv6_static_dns_enable= 0

Parameter-	Configuration File
network.ipv6_internet_port.ip	<MAC>.cfg
Description	Configures the IPv6 address when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_internet_port.ip = 2026:1234:1:1:215:65ff:fe1f:caa

Parameter-	Configuration File
network.ipv6_prefix	<MAC>.cfg
Description	Configures the prefix of the IPv6 address when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	64
Range	0 to 128
Example	network.ipv6_prefix = 64

Parameter-	Configuration File
network.ipv6_internet_port.gateway	<MAC>.cfg
Description	Configures the gateway when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_internet_port.gateway = 3036:1:1:c3c7:c11c:5447:23a6:255

Parameter-	Configuration File
network.ipv6_primary_dns	<MAC>.cfg
Description	Configures the primary DNS server when the IPv6 address assignment method is configured as Static IP Address and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_primary_dns = 3036:1:1:c3c7: c11c:5447:23a6:256

Parameter-	Configuration File
network.ipv6_secondary_dns	<MAC>.cfg
Description	Configures the secondary DNS server when the IPv6 address assignment method is configured as Static IP Address

	and the IP address mode is configured as IPv6 or IPv4&IPv6. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IPv6 Address
Default Value	Blank
Range	Not Applicable
Example	network.ipv6_secondary_dns = 2026:1234:1:1:c3c7:c11c:5447:23a6

Audio Feature Parameters

Head Prior

Parameter-	Configuration File
features.headset_prior	<y0000000000xx>.cfg
Description	Enables or disables headset prior feature. If it is set to 1 (enabled), a user needs to press the HEADSET key to activate the headset mode. The headset mode will not be deactivated until the user presses the HEADSET key again.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	features.headset_prior = 1

Dual Headset

Parameter-	Configuration File
features.headset_training	<y0000000000xx>.cfg
Description	Enables or disables dual headset feature. If it is set to 1 (Enabled), users can use two

	<p>headsets on one phone. When the IP phone joins in a call, the users with the headset connected to the headset jack have a full-duplex conversation, while the users with the headset connected to the handset jack are only allowed to listen to.</p> <p>Note: It is not applicable to SIP-T19P and SIP-T21P IP phones.</p>
Format	Boolean
Default Value	0
Range	<p>Valid values are:</p> <p>0-Disabled</p> <p>1-Enabled</p>
Example	features.headset_training = 1

Audio Codecs

Parameter- account.X.codec.Y.enable	Configuration File <MAC>.cfg
Description	<p>Enables or disables the IP phone to use the specific codec for account X.</p> <p>X ranges from 1 to 6.</p> <p>Y ranges from 1 to 11.</p>
Format	Boolean
Default Value	<p>For SIP-T20P/T22P/T26P/T28P IP phones:</p> <p>When Y=1, the default value is 1;</p> <p>When Y=2, the default value is 1;</p> <p>When Y=3, the default value is 0;</p> <p>When Y=4, the default value is 0;</p> <p>When Y=5, the default value is 1;</p> <p>When Y=6, the default value is 1;</p> <p>When Y=7, the default value is 0;</p> <p>When Y=8, the default value is 0;</p> <p>When Y=9, the default value is 0;</p> <p>When Y=10, the default value is 0;</p> <p>When Y=11, the default value is 0.</p>

	<p>For SIP-T19P/T21P IP phones:</p> <p>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.</p>
Range	<p>Valid values are:</p> <p>0-Disabled 1-Enabled</p>
Example	account.1.codec.1.enable = 1

Parameter- account.X.codec.Y.payload_type	Configuration File <MAC>.cfg
Description	<p>Configures the codec for account X to use. X ranges from 1 to 6. Y ranges from 1 to 11.</p>
Format	String
Default Value	<p>For SIP-T20P/T22P/T26P/T28P IP phones:</p> <p>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=7, the default value is iLBC; 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.</p> <p>For SIP-T19P/T21P IP phones:</p> <p>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;</p>

	<p>When Y=4, the default value is G723_63;</p> <p>When Y=5, the default value is G729;</p> <p>When Y=6, the default value is G722;</p> <p>When Y=7, the default value is iLBC;</p> <p>When Y=8, the default value is G726-32.</p>
Range	<p>Valid values are:</p> <p>PCMU</p> <p>PCMA</p> <p>G729</p> <p>G722</p> <p>G723_53</p> <p>G723_63</p> <p>G726-16</p> <p>G726-24</p> <p>G726-32</p> <p>G726-40</p> <p>iLBC</p>
Example	<p>account.1.codec.1.payload_type = PCMU</p>

Parameter- account.X.codec.Y.priority	Configuration File <MAC>.cfg
Description	<p>Configures the priority for the codec.</p> <p>X ranges from 1 to 6.</p> <p>Y ranges from 1 to 11.</p>
Format	Integer
Default Value	<p>For SIP-T20P/T22P/T26P/T28P IP phones:</p> <p>When Y=1, the default value is 1;</p> <p>When Y=2, the default value is 2;</p> <p>When Y=3, the default value is 0;</p> <p>When Y=4, the default value is 0;</p> <p>When Y=5, the default value is 3;</p> <p>When Y=6, the default value is 4;</p> <p>When Y=7, the default value is 0;</p> <p>When Y=8, the default value is 0;</p> <p>When Y=9, the default value is 0;</p>

	<p>When Y=10, the default value is 0; When Y=11, the default value is 0.</p> <p>For SIP-T19P/T21P IP phones:</p> <p>When Y=1, the default value is 1; When Y=2, the default value is 2; When Y=3, the default value is 0; 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.</p>
Range	0 to 10
Example	account.1.codec.1.priority = 1

Parameter- account.X.codec.Y.rtpmap	Configuration File <MAC>.cfg
Description	<p>Configures the rtpmap. X ranges from 1 to 6. Y ranges from 1 to 11.</p>
Format	Integer
Default Value	<p>For SIP-T20P/T22P/T26P/T28P IP phones:</p> <p>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 106; When Y=8, the default value is 103; When Y=9, the default value is 104; When Y=10, the default value is 102; When Y=11, the default value is 105.</p> <p>For SIP-T19P/T21P IP phones:</p> <p>When Y=1, the default value is 0; When Y=2, the default value is 8;</p>

	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 106; When Y=8, the default value is 102.
Range	0 to 127
Example	account.1.codec.1.rtpmap = 0

Ptime

Parameter-	Configuration File
account.X.ptime	<MAC>.cfg
Description	Configures 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 = 20

Acoustic Echo Cancellation

Parameter-	Configuration File
voice.echo_cancellation	<y0000000000xx>.cfg
Description	Enables or disables AEC feature on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	voice.echo_cancellation = 1

Voice Activity Detection

Parameter-	Configuration File
voice.vad	<y0000000000xx>.cfg
Description	Enables or disables VAD feature on the IP phone.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	voice.vad = 1

Comfort Noise Generation

Parameter-	Configuration File
voice.cng	<y0000000000xx>.cfg
Description	Enables or disables CNG feature on the IP phone.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	voice.cng = 1

Jitter Buffer

Parameter-	Configuration File
voice.jib.adaptive	<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 for jitter buffer. Note: It works only if the parameter "voice.jib.adaptive" is set to 1 (Adaptive).
Format	Integer
Default Value	60
Range	0 to 400
Example	voice.jib.min = 60

Parameter- voice.jib.max	Configuration File <y0000000000xx>.cfg
Description	Configures the maximum delay time for jitter buffer. Note: It works only if the parameter "voice.jib.adaptive" is set to 1 (Adaptive).
Format	Integer
Default Value	240
Range	0 to 400
Example	voice.jib.max = 300

Parameter- voice.jib.normal	Configuration File <y0000000000xx>.cfg
Description	Configures the fixed delay time for jitter buffer. Note: It works only if the parameter "voice.jib.adaptive" is set to 0 (Fixed).
Format	Integer
Default Value	120

Range	0 to 400
Example	voice.jib.mormal = 120

Security Feature Parameters

TLS

Parameter-	Configuration File
account.X.transport	<MAC>.cfg
Description	Configures the transport type for account X. If it is set to 2 (TLS), the SIP message of this account will be encrypted after the successful TLS negotiation. X ranges from 1 to 6.
Format	Integer
Default Value	0 (UDP)
Range	Valid values are: 0-UDP 1-TCP 2-TLS 3-DNS-NAPTR
Example	account.1.transport = 2

Parameter-	Configuration File
security.trust_certificates	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to authenticate the connecting server based on the trusted certificates list. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled

	1-Enabled
Example	security.trust_certificates = 1

Parameter- security.ca_cert	Configuration File <y0000000000xx>.cfg
Description	Configures the type of certificates the IP phone used to authenticate the connecting server. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	2
Range	Valid values are: 0-Default certificates 1-Custom certificates 2-All certificates
Example	security.ca_cert = 2

Parameter- security.cn_validation	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to mandatorily validate the CommonName or SubjectAltName of the certificate sent by the connecting server. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	security.cn_validation = 0

Parameter-	Configuration File
security.dev_cert	<y0000000000 xx>.cfg
Description	Configures the type of certificates the IP phone sends for authentication. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	Valid values are: 0-Default certificates 1-Custom certificates
Example	security.dev_cert = 0

Uploading Certificates

Parameter-	Configuration File
trusted_certificates.url	<y0000000000xx>.cfg
Description	Configures the access URL of the certificate used to authenticate the connecting server. Note: The certificate you want to upload must be in *.pem, *.crt, *.cer or *.der format.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	trusted_certificates.url = http://192.168.1.20/tc.crt

Parameter-	Configuration File
server_certificates.url	<y0000000000xx>.cfg
Description	Configures the access URL of the certificate the IP phone sends for authentication. Note: The certificate you want to upload must be in *.pem or *.cer format.
Format	URL

Default Value	Blank
Range	String within 511 characters
Example	server_certificates.url = http://192.168.1.20/ca.pem

SRTP

Parameter-	Configuration File
account.X.srtp_encryption	<MAC>.cfg
Description	Configures whether to use voice encryption service. If it is set to 1 (Optional), the IP phone will negotiate with the other IP phone what type of encryption to utilize for the session. If it is set to 2 (Compulsory), the IP phone is forced to use SRTP during a call. X ranges from 1 to 6.
Format	Integer
Default Value	0
Value	Valid values are: 0-Disabled 1-Optional 2-Compulsory
Example	account.1.srtp_encryption = 0

Configuring Decryption Method

Parameter-	Configuration File
auto_provision.aes_key_in_file	<y0000000000xx>.cfg
Description	Enables or disables the IP phone to decrypt configuration files using the encrypted AES keys. If it is set to 1 (Enabled), the IP phone will download <y0000000000xx_Security>.enc and <MAC_Security>.enc files during auto provisioning, and then decrypts these files into the plaintext keys (e.g., key2, key3) respectively using the phone built-in key

	(e.g., key1). The IP phone then decrypts the encrypted configuration files using corresponding key (e.g., key2, key3). If it is set to 0 (Disabled), the IP phone will decrypt the encrypted configuration files using plaintext AES keys configured on the IP phone.
Format	Boolean
Default Value	0
Value	Valid values are: 0-Disabled 1-Enabled
Example	auto_provision.aes_key_in_file = 0

Parameter- auto_provision.aes_key_16.com	Configuration File <y0000000000xx>.cfg
Description	Configures the plaintext AES key which is used to decrypt the <y0000000000xx>.cfg file. Note: It works only if the parameter "auto_provision.aes_key_in_file" is set to 0 (Disabled).
Format	String
Default Value	Blank
Range	16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z
Example	auto_provision.aes_key_16.com = 0123456789abcdef

Parameter- auto_provision.aes_key_16.mac	Configuration File <y0000000000xx>.cfg
Description	Configures the plaintext AES key which is used to decrypt the <MAC>.cfg file. Note: It works only if the parameter "auto_provision.aes_key_in_file" is set to 0 (Disabled).
Format	String

Default Value	Blank
Range	16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z
Example	auto_provision.aes_key_16.mac = 0123456789abmins

Parameter- auto_provision.update_file_mode	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to update encrypted configuration settings only during auto provisioning.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	auto_provision.update_file_mode = 0

Upgrading Firmware

Parameter- firmware.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of firmware. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	firmware.url = http://192.168.1.20/2.72.0.1.rom

Parameter- auto_provision.power_on	Configuration File <y0000000000xx>.cfg
Description	Enables or disables the IP phone to perform

	an auto provisioning process when powered on.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	auto_provision.power_on = 1

Parameter- auto_provision.repeat.enable	Configuration File < y0000000000xx >.cfg
Description	Enables or disables the IP phone to check new configuration repeatedly.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	auto_provision.repeat.enable =0

Parameter- auto_provision.repeat.minutes	Configuration File < y0000000000xx >.cfg
Description	Configures the interval (in minutes) for the IP phone to check new configuration. Note: It works only if the parameter "auto_provision.repeat.enable" is set to 1(Enabled).
Format	Integer
Default Value	1440
Range	1 to 43200
Example	auto_provision.repeat.minutes = 1000

Parameter-	Configuration File
auto_provision.weekly.enable	< y0000000000xx >.cfg
Description	Enables or disables the IP phone to check new configuration weekly.
Format	Boolean
Default Value	0
Range	Valid values are: 0-Disabled 1-Enabled
Example	auto_provision.weekly.enable =0

Parameter-	Configuration File
auto_provision.weekly.begin_time	< y0000000000xx >.cfg
Description	Configures the begin time of the day for the IP phone to check new configuration weekly. Note: It works only if the parameter "auto_provision.weekly.enable" is set to 1(Enabled).
Format	Time
Default Value	00:00
Range	00:00 to 23:59
Example	auto_provision.weekly.begin_time = 01:30

Parameter-	Configuration File
auto_provision.weekly.end_time	< y0000000000xx >.cfg
Description	Configures the end time of the day for the IP phone to check new configuration weekly. Note: It works only if the parameter "auto_provision.weekly.enable" is set to 1(Enabled).
Format	Time
Default Value	00:00
Range	00:00 to 23:59

Example	auto_provision.weekly.end_time = 21:30
Parameter- auto_provision.weekly.dayofweek	Configuration File < y0000000000xx >.cfg
Description	Configures the days of the week for the IP phone to check new configuration weekly.
Format	Integer
Default Value	0123456
Range	Valid values are: 0-Sunday 1-Monday 2-Tuesday 3-Wednesday 4-Thursday 5-Friday 6-Saturday
Example	auto_provision.weekly.dayofweek = 0123456

Resource Files

Access URL of Replace Rule Template

Parameter- dialplan_replace_rule.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the replace rule template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	dialplan_replace_rule.url = http://192.168.10.25/dialplan.xml

Access URL of Dial-now Template

Parameter- dialplan_dialnow.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the dial-now template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	dialplan_dialnow.url = http://192.168.10.25/dialnow.xml

Access URL of Softkey Layout Template

Parameter- custom_softkey_call_failed.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the CallFailed state.
Format	URL
Default Value	Blank
Range	String within 511 characters
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	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the CallIn state.
Format	URL

Default Value	Blank
Range	String within 511 characters
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	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Connecting state.
Format	URL
Default Value	Blank
Range	String within 511 characters
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	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Dialing state.
Format	URL
Default Value	Blank
Range	String within 511 characters
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
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Parameter-	Configuration File
custom_softkey_ring_back.url	<y0000000000xx>.cfg
Description	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the RingBack state.
Format	URL
Default Value	Blank
Range	String within 511 characters
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-	Configuration File
custom_softkey_talking.url	<y0000000000xx>.cfg
Description	Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Talking state.
Format	URL
Default Value	Blank
Range	String within 511 characters
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-	Configuration File
local_contact.data.url	<y0000000000xx>.cfg
Description	Configures the access URL of the local contact file.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	local_contact.data.url = http://192.168.10.25/contact.xml

Access URL of Remote XML Phone Book

Parameter-	Configuration File
remote_phonebook.data.X.url	<y0000000000xx>.cfg
Description	Configures the access URL of the remote XML phone book. X ranges from 1 to 5.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml

Access URL of Directory Template

Parameter-	Configuration File
directory_setting.url	<y0000000000xx>.cfg
Description	Configures the access URL of the directory template.
Format	URL
Default Value	Blank
Range	String within 511 characters

Example	directory_setting.url = http://192.168.1.20/favorite_setting.xml
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Access URL of Super Search Template

Parameter- super_search.url	Configuration File <y0000000000xx>.cfg
Description	Configures the access URL of the super search template.
Format	URL
Default Value	Blank
Range	String within 511 characters
Example	super_search.url = http://192.168.1.20/super_search.xml

Troubleshooting

Log Settings

Parameter- syslog.mode	Configuration File <y0000000000xx>.cfg
Description	Configures the syslog mode. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	0
Range	0-Local 1-Server
Example	syslog.mode = 1

Parameter- syslog.server	Configuration File <y0000000000xx>.cfg
Description	Configures the IP address or domain name of the syslog server where to export the log

	files. Note: It works only if the parameter "syslog.mode" is set to 1 (Server). If you change this parameter, the IP phone will reboot to make the change take effect.
Format	IP Address or Domain Name
Default Value	Blank
Range	String within 99 characters
Example	syslog.server = 192.168.1.50

Parameter- syslog.log_level	Configuration File <y0000000000xx>.cfg
Description	Configures the severity level of the logs to be reported to a log file. Note: If you change this parameter, the IP phone will reboot to make the change take effect.
Format	Integer
Default Value	3
Range	0 to 6
Example	syslog.log_level = 3

Watch Dog

Parameter- watch_dog.enable	Configuration File <y0000000000xx>.cfg
Description	Enables or disables Watch Dog feature.
Format	Boolean
Default Value	1
Range	Valid values are: 0-Disabled 1-Enabled
Example	watch_dog.enable = 1

Configuring DSS Key

This section provides the DSS key parameters you can configure on IP phones. DSS key consists of memory key, line key and programable key. The following table lists the number of DSS keys you can configure for each phone model:

Phone Model	Line Key	Memory Key	Programable Key
SIP-T28P	6	10	14
SIP-T26P	3	10	14
SIP-T22P	3	/	13
SIP-T21P	2	/	11
SIP-T20P	2	/	9
SIP-T19P	/	/	11

Note The programable key takes effect only if the IP phone is idle.

DSS key can be assigned with various key features. The parameters of the DSS key are detailed in the following:

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	<p>Configures key feature for the DSS key.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p> <p>For memory keys:</p> <p>Valid types are:</p> <ul style="list-style-type: none"> N/A

	<ul style="list-style-type: none"> • Conference • Forward • Transfer • Hold • DND • Call Return • SMS • Directed Pickup • Call Park • DTMF • Voice Mail • Speed Dial • Intercom • Line • BLF • URL • Group Listening • XML Group • Group Pickup • Multicast Paging • Record • XML Browser • URL Record • LDAP • Prefix • Zero Touch • ACD • Local Group • Custom Button • Keypad Lock • Directory <p>For line keys:</p> <p>Valid types are:</p> <ul style="list-style-type: none"> • Conference • Forward • Transfer • Hold
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	<ul style="list-style-type: none">• DND• Call Return• SMS (not applicable to SIP-T20P IP phones)• Directed Pickup• Call Park• DTMF• Voice Mail• Speed Dial• Intercom• Line• BLF• Group Listening• XML Group (not applicable to SIP-T20P IP phones)• Group Pickup• Multicast Paging• Record• XML Browser• Hot Desking• URL Record• LDAP (not applicable to SIP-T20P IP phones)• Prefix• Zero Touch• ACD• Local Group• Custom Button• Keypad Lock• Directory <p>For programmable keys:</p> <p>Valid types are:</p> <ul style="list-style-type: none">• N/A• Forward• DND• Call Return• SMS (not applicable to SIP-T20P IP phones)• Directed Pickup• Speed Dial
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	<ul style="list-style-type: none"> • XML Group (not applicable to SIP-T19P IP phones) • Group Pickup • XML Browser • History • Menu • Switch Account (not applicable to SIP-T19P IP phones) • New SMS (not applicable to SIP-T20P IP phones) • Status • Hot Desking (only applicable to SIP-T19P IP phones) • LDAP (not applicable to SIP-T19P and SIP-T20P IP phones) • Prefix (not applicable to SIP-T20P IP phones) • Zero Touch • Local Directory • Local Group • XML Directory (not applicable to SIP-T20P IP phones) • Keypad Lock • Directory
Format	Integer
Default Value	<p>For the memory key, the default value is 0 (N/A).</p> <p>For the line key, the default value is 15 (Line).</p> <p>For the programmable key,</p> <p>when x=1, the default value is 28.</p> <p>when x=2, the default value is 61.</p> <p>when x=3, the default value is 5.</p> <p>when x=4, the default value is 30.</p> <p>when x=5, the default value is 28.</p> <p>when x=6, the default value is 29.</p> <p>when x=7, the default value is 31.</p> <p>when x=8, the default value is 31.</p> <p>when x=9, the default value is 33.</p> <p>when x=10/11/12/13, the default value is 0.</p>

	when x=14, the default value is 2.
Range	<p>Valid values are:</p> <ul style="list-style-type: none"> 0-N/A 1-Conference 2-Forward 3-Transfer 4-Hold 5-DND 7-Call Return 8-SMS 9-Directed Pickup 10-Call Park 11-DTMF 12-Voice Mail 13-Speed Dial 14-Intercom 15-Line 16-BLF 17-URL 18-Group Listening 22-XML Group 23-Group Pickup 24-Multicast Paging 25-Record 27-XML Browser 28-History 30-Menu 31-Switch Account 32-New SMS 33-Status 34-Hot Desking 35-URL Record 38-LDAP 40-Prefix 41-Zero Touch 42-ACD 43-Local Directory

	<p>45-Local Group</p> <p>47-XML Directory</p> <p>49-Custom Button</p> <p>50-Keypad Lock</p> <p>61-Directory</p>
Example	memorykey.1.type = 8

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Parameter- programablekey.X.line	
Description	<p>Configures the desired line to apply the key feature.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p> <p>When assigning the following features, you do not need to configure this parameter:</p> <ul style="list-style-type: none"> • DTMF • Prefix • XML Browser • LDAP (not applicable to SIP-T19P and SIP-T20P) • Conference • Forward • Hold • DND • Call Return • SMS (not applicable to SIP-T20P) • Record

	<ul style="list-style-type: none"> • URL Record • Multicast Paging • Group Listening • Local Group • XML Group (not applicable to SIP-T20P) • ACD • Hot Desking • Zero Touch • URL (not applicable to SIP-T20P) • Keypad Lock • Directory
Format	Integer
Default Value	<p>For the memory key and programable key, the default value is not applicable.</p> <p>For the line key, when x=1, the default value is 1.</p> <p style="padding-left: 40px;">When x=2, the default value is 2.</p> <p style="padding-left: 40px;">...</p> <p style="padding-left: 40px;">When x=6, the default value is 6.</p>
Range	<p>Valid values are:</p> <p>1 to 6 (for SIP-T28P)</p> <p>1 to 3 (for SIP-T26P/T22P)</p> <p>1 to 2 (for SIP-T21P/T20P)</p> <p>1 (for SIP-T19P)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>6-Line 6</p>
Example	memorykey.1.line = 2

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Parameter- programablekey.X.value	
Description	Configures the value for some key features.

	<p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p>
Format	String
Default Value	Blank
Range	String within 99 characters
Example	<p>When you assign the Speed Dial to the memory key, this parameter is used to specify the number you want to dial out.</p> <p>memorykey.1.value = 1001</p>

Parameter- memorykey.X.pickup_value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.pickup_value	
Description	<p>Configures the pickup code for BLF feature.</p> <p>This parameter is only applicable to BLF feature.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p>
Format	String
Default Value	Blank
Range	String within 256 characters
Example	memorykey.1.pickup_value = *88

Parameter- memorykey.X.xml_phonebook	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.xml_phonebook	
Parameter- programablekey.X.xml_phonebook	

Description	<p>Configures the desired group or remote phone book when multiple groups or remote phone books are configured on the IP phone.</p> <p>This parameter is only applicable to Local Group/XML Group features.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programmable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p> <p>When the key feature is configured as Local Group, valid values are:</p> <p>0-All contacts</p> <p>1-First local group</p> <p>...</p> <p>5-Fifth local group</p> <p>When the key feature is configured as XML Group (remote phone book), valid values are:</p> <p>0-First XML group</p> <p>1-Second XML group</p> <p>...</p> <p>4-Fifth XML group</p>
Format	Integer
Default Value	0
Range	0 to 5
Example	Configures the second remote phone book. memorykey.1.xml_phonebook = 1

Keypad Lock Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	<p>Configures a DSS key as a keypad lock key on the IP phone.</p> <p>The digit 50 stands for the key type Keypad Lock.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p>
Format	Integer
Value	50
Example	memorykey.1.type = 50

DND Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	<p>Configures a DSS key as a DND key on the IP phone.</p> <p>The digit 5 stands for the key type DND.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p>

	For the programmable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	Integer
Value	5
Example	memorykey.1.type = 5

Directed Call Pickup Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programmablekey.X.type	
Description	<p>Configures a DSS key as a directed call pickup key on the IP phone.</p> <p>The digit 9 stands for the key type Directed Pickup.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programmable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p>
Format	Integer
Value	9
Example	memorykey.1.type = 9

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Parameter- programablekey.X.line	
Description	Configures the desired line to apply the directed call pickup key. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	Integer
Range	Valid values are: 1 to 6 (for SIP-T28P) 1 to 3 (for SIP-T26P/T22P) 1 to 2 (for SIP-T21P/T20P) 1 (for SIP-T19P) 1-Line 1 2-Line 2 ... 6-Line 6
Example	memorykey.1.line = 1

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Parameter- programablekey.X.value	
Description	Configures the directed call pickup feature code followed by the monitored extension. For the memory key, x ranges from 1 to 10.

	For the line key, x ranges from 1 to 6. For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	String
Range	String within 99 characters
Example	memorykey.1.value = *971001

Group Call Pickup Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	Configures a DSS key as a group call pickup key on the IP phone. The digit 23 stands for the key type Group Pickup . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	Integer
Value	23
Example	memorykey.1.type = 23

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Parameter- programablekey.X.line	
Description	<p>Configures the desired line to apply the group call pickup key.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p>
Format	Integer
Range	<p>Valid values are:</p> <p>1 to 6 (for SIP-T28P)</p> <p>1 to 3 (for SIP-T26P/T22P)</p> <p>1 to 2 (for SIP-T21P/T20P)</p> <p>1 (for SIP-T19P)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>6-Line 6</p>
Example	memorykey.1.line = 1

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Parameter- programablekey.X.value	
Description	<p>Configures the group call pickup feature code.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p>

	For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	String
Range	String within 99 characters
Example	memorykey.1.value = *98

Call Return Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	Configures a DSS key as a call return key on the IP phone. The digit 7 stands for the key type Call Return . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. For the programable key, x ranges from 1 to 14 (For SIP-T19P IP phones, x=1-9, 13, 14; For SIP-T20P IP phones, x=5-12, 14; For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).
Format	Integer
Value	7
Example	memorykey.1.type = 7

Call Park Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as a call park key on the IP phone. The digit 10 stands for the key type Call Park . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	10
Example	memorykey.1.type = 10

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Description	Configures the desired line to apply the call park key. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Range	Valid values are: 1 to 6 (for SIP-T28P) 1 to 3 (for SIP-T26P/T22P) 1 to 2 (for SIP-T21P/T20P) 1-Line 1 2-Line 2 ... 6-Line 6
Example	memorykey.1.line = 1

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Description	Configures the call park feature code. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	String
Range	String within 99 characters
Example	memorykey.1.value = *99

Intercom Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as an intercom key. The digit 14 stands for the key type Intercom . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	14
Example	memorykey.1.type = 14

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Description	Configures the desired line to apply the intercom key. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6.

	Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Range	Valid values are: 1 to 6 (for SIP-T28P) 1 to 3 (for SIP-T26P/T22P) 1 to 2 (for SIP-T21P/T20P) 1-Line 1 2-Line 2 ... 6-Line 6
Example	memorykey.1.line = 1

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Description	Configures the intercom number. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	String
Range	String within 99 characters
Example	memorykey.1.value = 1008

LDAP Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	Configures a DSS key as an LDAP key on the IP phone. The digit 38 stands for the key type LDAP . For the memory key, x ranges from 1 to 10.

	<p>For the line key, x ranges from 1 to 6.</p> <p>For the programmable key, x ranges from 1 to 14. (For SIP-T21P IP phones, x=1-10, 14; For SIP-T22P IP phones, x=1-10, 12-14; For SIP-T26P/T28P IP phones, x ranges from 1 to 14).</p> <p>Note: It is not applicable to SIP-T19P and SIP-T20P IP phones.</p>
Format	Integer
Value	38
Example	memorykey.1.type = 38

BLF Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	<p>Configures a DSS key as a BLF key on the IP phone.</p> <p>The digit 16 stands for the key type BLF.</p> <p>For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6.</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p>
Format	Integer
Value	16
Example	memorykey.1.type = 16

Parameter- memorykey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.line	
Description	<p>Configures the desired line to apply the BLF key.</p> <p>For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6.</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p>
Format	Integer

Range	<p>Valid values are:</p> <p>1 to 6 (for SIP-T28P) 1 to 3 (for SIP-T26P/T22P) 1 to 2 (for SIP-T21P/T20P)</p> <p>1-Line 1 2-Line 2 ... 6-Line 6</p>
Example	memorykey.1.line = 1

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Description	<p>Configures the number of the monitored user. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6.</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p>
Format	String
Range	String within 99 characters
Example	memorykey.1.value = 1008

Parameter- memorykey.X.pickup_value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.pickup_value	
Description	<p>Configures the pickup code for BLF feature. This parameter only applies to BLF feature. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6.</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p>
Format	String
Default Value	Blank
Range	String within 256 characters
Example	memorykey.1.pickup_value = *88

ACD Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as an ACD key on the IP phone. The digit 42 stands for the key type ACD . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	42
Example	memorykey.1.type = 42

Multicast Paging Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as a multicast paging key on the IP phone. The digit 24 stands for the key type Multicast Paging . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	24
Example	memorykey.1.type = 24

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Description	Configures the multicast IP address and port number. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones. The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.
Format	IP Address
Range	224.0.0.0 to 239.255.255.255
Example	memorykey.1.value = 224.5.5.6:10008

Record Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as a record key on the IP phone. The digit 25 stands for the key type Record . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	25
Example	memorykey.1.type = 25

URL Record Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Description	Configures a DSS key as a URL record key on the IP phone. The digit 35 stands for the key type URL Record . For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	Integer
Value	35
Example	memorykey.1.type = 35

Parameter- memorykey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.value	
Description	Configures the URL to record a call. For the memory key, x ranges from 1 to 10. For the line key, x ranges from 1 to 6. Note: It is not applicable to SIP-T19P IP phones.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	memorykey.1.value = http://10.1.2.224/phonerecording.cgi

Hot Desking Key

Parameter- memorykey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- linekey.X.type	
Parameter- programablekey.X.type	
Description	<p>Configures a DSS key as a hot desking key on the IP phone.</p> <p>The digit 34 stands for the key type Hot Desking.</p> <p>For the memory key, x ranges from 1 to 10.</p> <p>For the line key, x ranges from 1 to 6.</p> <p>For the programable key, x=1-9, 13, 14.</p> <p>Note: You can configure a programable key as a hot desking key on SIPT19P IP phones only.</p>
Format	Integer
Value	34
Example	memorykey.1.type = 34

Appendix D: SIP (Session Initiation Protocol)

This section describes how Yealink 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 1889—RTP Media control

- RFC 2112—Multipart MIME
- RFC 2246—The TLS Protocol Version 1.0
- RFC 2327—SDP: Session Description Protocol
- RFC 2543—SIP: Session Initiation Protocol
- RFC 2616—Hypertext Transfer Protocol -- HTTP/1.1
- RFC 2617—Http Authentication: Basic and Digest access authentication
- RFC 2782—A DNS RR for specifying the location of services (DNS SRV)
- RFC 2806—URLs for Telephone Calls
- RFC 2833—RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC2915—The Naming Authority Pointer (NAPTR) DNS Resource Record
- RFC 3087—Control of Service Context using SIP Request-URI
- 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 3266—Support for IPv6 in Session Description Protocol (SDP)
- RFC 3310—HTTP Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC 3311—The Session Initiation Protocol (SIP) UPDATE Method
- RFC 3312—Integration of Resource Management and SIP
- RFC 3313—Private SIP Extensions for Media Authorization
- RFC 3323—A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3324—Requirements for Network Asserted Identity
- RFC 3325—SIP Asserted Identity
- RFC 3326—The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC 3361—DHCP-for-IPv4 Option for SIP Servers
- RFC 3372—SIP for Telephones (SIP-T): Context and Architectures
- RFC 3420—Internet Media Type message/sipfrag
- RFC 3428—Session Initiation Protocol (SIP) Extension for Instant Messaging
- RFC 3455—Private Header (P-Header) Extensions to the SIP for the 3GPP
- RFC 3486—Compressing the Session Initiation Protocol (SIP)
- RFC 3489—STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
- RFC 3515—The Session Initiation Protocol (SIP) Refer Method

- RFC 3550—RTP , RTCP, IETF RFC 3550
- RFC 3556—Session Description Protocol (SDP) Bandwidth Modifiers for RTCP Bandwidth
- RFC 3581—An Extension to the SIP for Symmetric Response Routing
- RFC 3608—SIP Extension Header Field for Service Route Discovery During Registration
- RFC 3665—Session Initiation Protocol (SIP) Basic Call Flow Examples
- RFC 3666—SIP Public Switched Telephone Network (PSTN) Call Flows.
- RFC 3680—SIP Event Package for Registrations
- RFC 3702—Authentication, Authorization, and Accounting Requirements for the SIP
- RFC 3711—The Secure Real-time Transport Protocol (SRTP)
- 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 3890—A Transport Independent Bandwidth Modifier for the SDP
- RFC 3891—The Session Initiation Protocol (SIP) “Replaces” Header
- RFC 3892—The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3959—The Early Session Disposition Type for SIP
- RFC 3960—Early Media and Ringing Tone Generation in SIP
- RFC3966—The tel URI for telephone number
- 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 4244—An Extension to the SIP for Request History Information
- RFC 4317—Session Description Protocol (SDP) Offer/Answer Examples
- RFC 4353—A Framework for Conferencing with the SIP
- RFC 4475—Session Initiation Protocol (SIP) Torture
- RFC 4485—Guidelines for Authors of Extensions to the SIP
- RFC 4504—SIP Telephony Device Requirements and Configuration
- RFC 4566—SDP: Session Description Protocol.

- RFC 4568—Session Description Protocol (SDP) Security Descriptions for Media Streams
- RFC 4575—A SIP Event Package for Conference State
- RFC 4579—SIP Call Control - Conferencing for User Agents
- RFC 4662—A SIP Event Notification Extension for Resource Lists
- RFC 5009—P-Early-Media Header
- RFC 5079—Rejecting Anonymous Requests in SIP
- RFC 5359—Session Initiation Protocol Service Examples
- RFC 5589—Session Initiation Protocol (SIP) Call Control - Transfer
- draft-levy-sip-diversion-04.txt—Diversion Indication in SIP
- draft-ietf-sip-cc-transfer-05.txt—SIP Call Control - Transfer
- 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-ietf-sipping-cc-conferencing-03.txt—SIP Call Control - Conferencing for User Agents

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	Yealink IP phones support mid-call changes such as placing a call on hold as signaled by a new INVITE that contains an existing Call-ID.
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	

Method	Supported	Notes
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	
Expires	Yes	
From	Yes	
Max-Forwards	Yes	
Min-SE	Yes	
P-Asserted-Identity	Yes	

Method	Supported	Notes
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	
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	
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	

4xx Response	Supported	Notes
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:

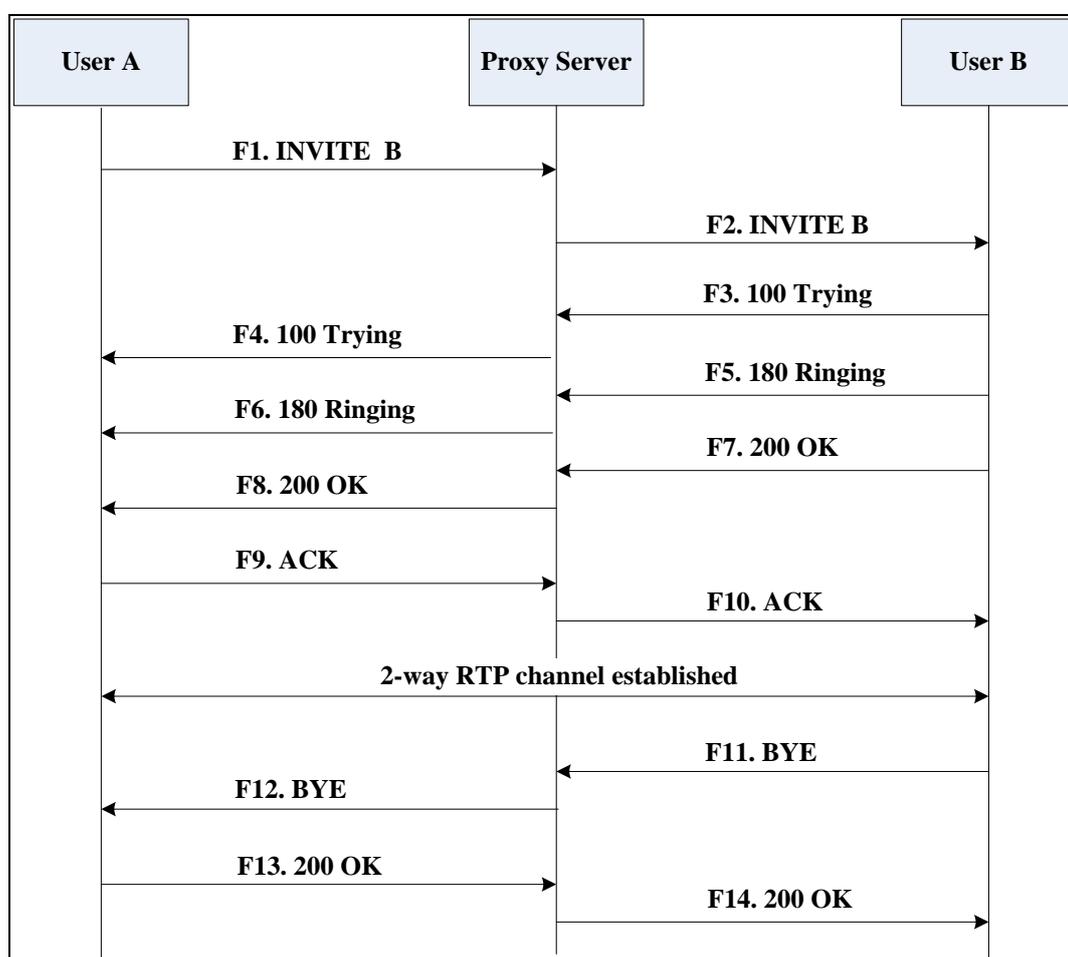
- SIP 1xx—Informational Responses
- SIP 2xx—Successful Responses
- SIP 3xx—Redirection Responses
- SIP 4xx—Client Failure Responses
- SIP 5xx—Server Failure Responses
- SIP 6xx—Global Failure Responses

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 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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of User B is inserted

Step	Action	Description
		in the Request-URI field. <ul style="list-style-type: none"> • 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.
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

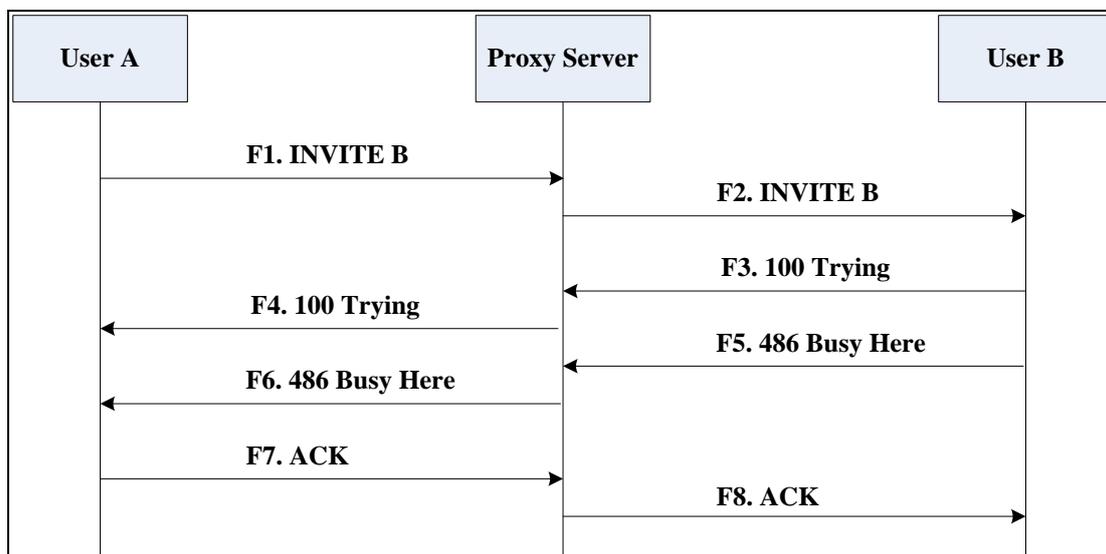
Step	Action	Description
		response notifies User A that the connection has been made.
F9	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F10	ACK—Proxy Server to User B	The proxy server 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 caused by the called user's being busy. In this scenario, the two end users are User A and User B. User A and User B are located at 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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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

Step	Action	Description
		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.
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.

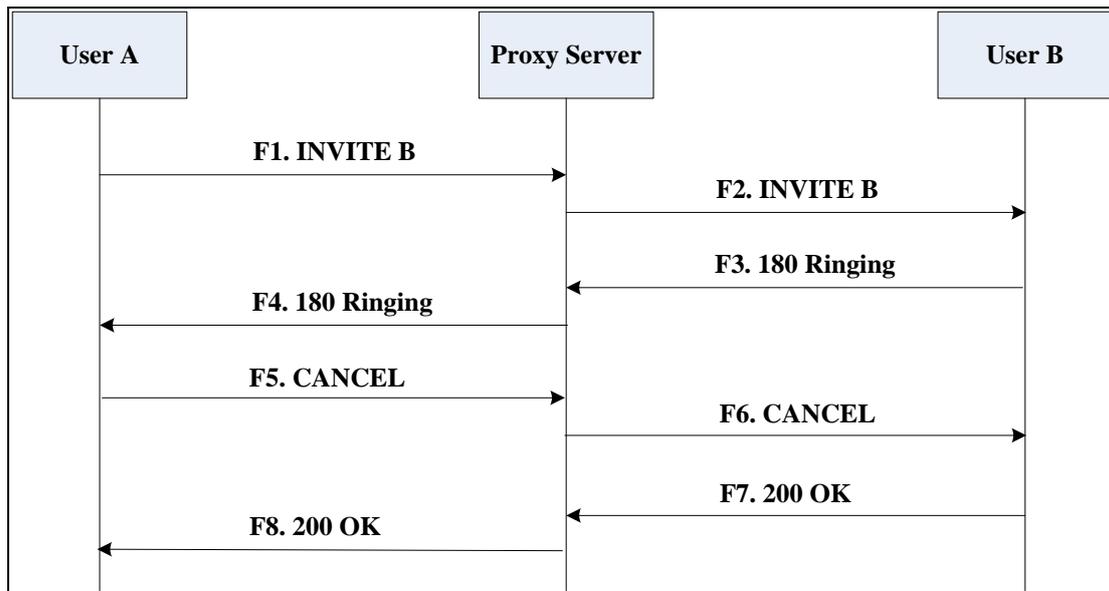
Unsuccessful Call Setup—Called User Does Not Answer

The following figure illustrates the scenario of an unsuccessful call caused by the called user's no answering. In this scenario, the two end users are User A and User B. User A and User B are located at 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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

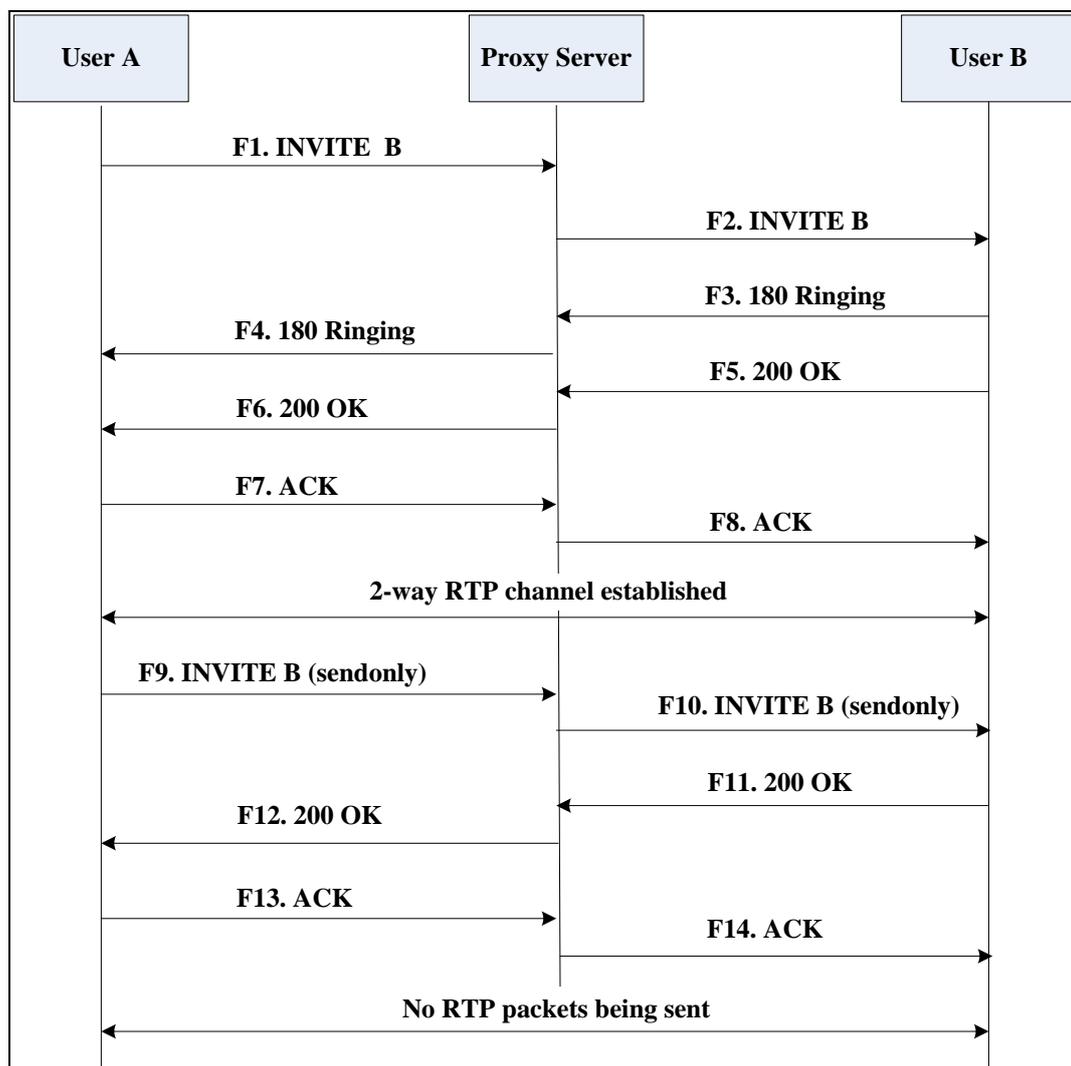
Successful Call Setup and Call Hold

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 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 places User B on hold.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

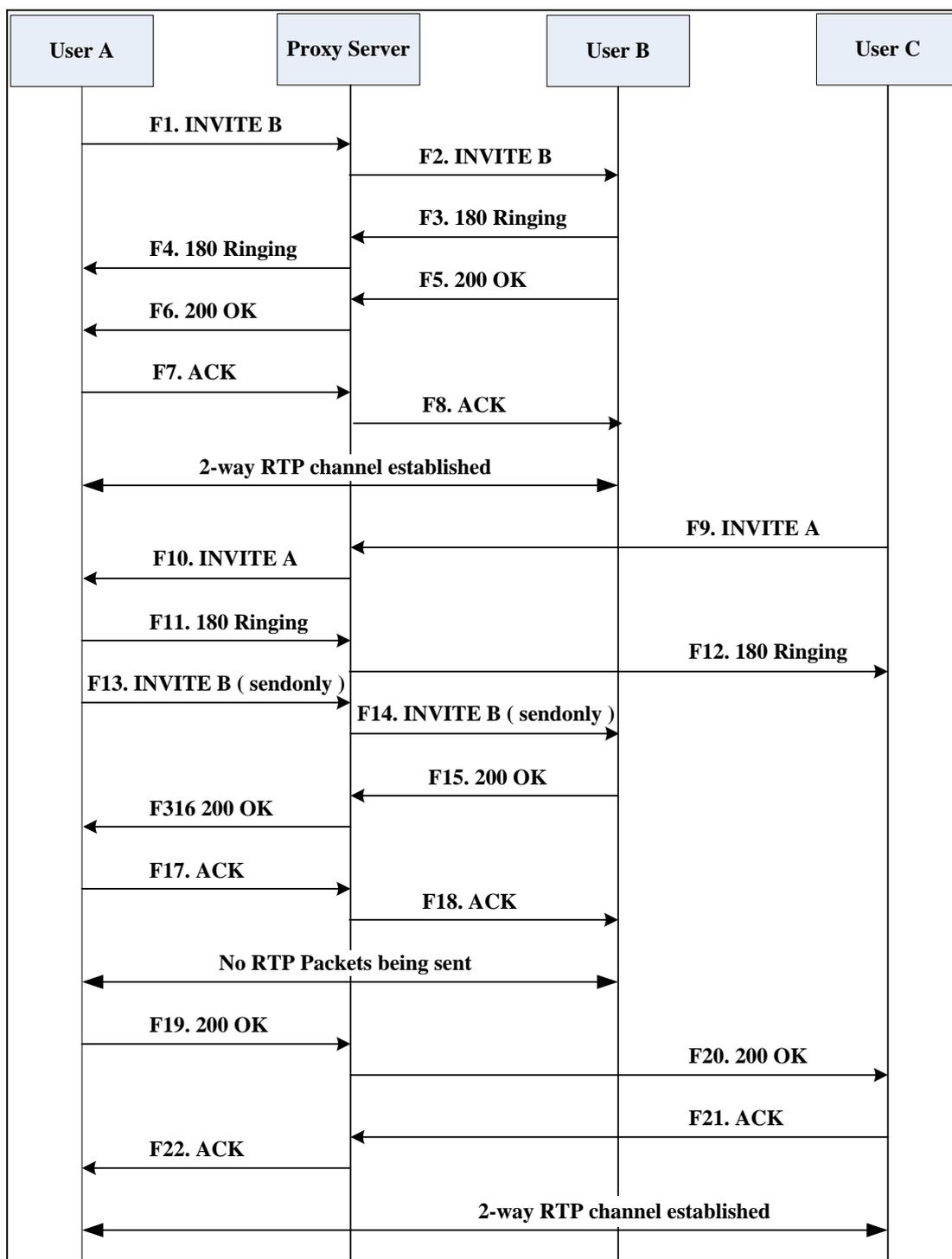
Step	Action	Description
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy 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 placed 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 two parties are in a call, one of the participants receives and answers an incoming call from a third party. 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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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 placed 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

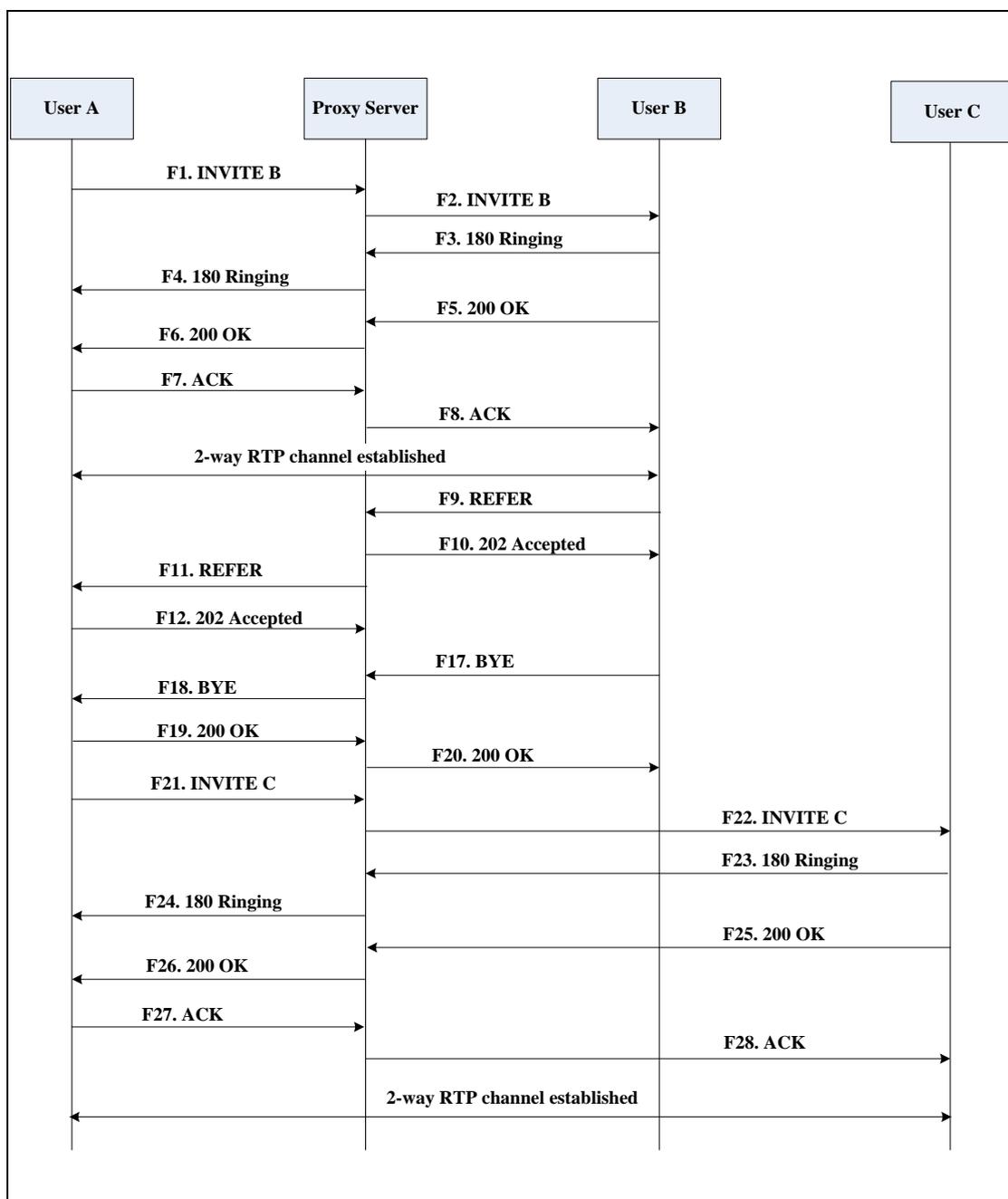
The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to a third party without consultation. This is called a blind transfer. In this call flow scenario, the end users are User A, User B, and 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.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

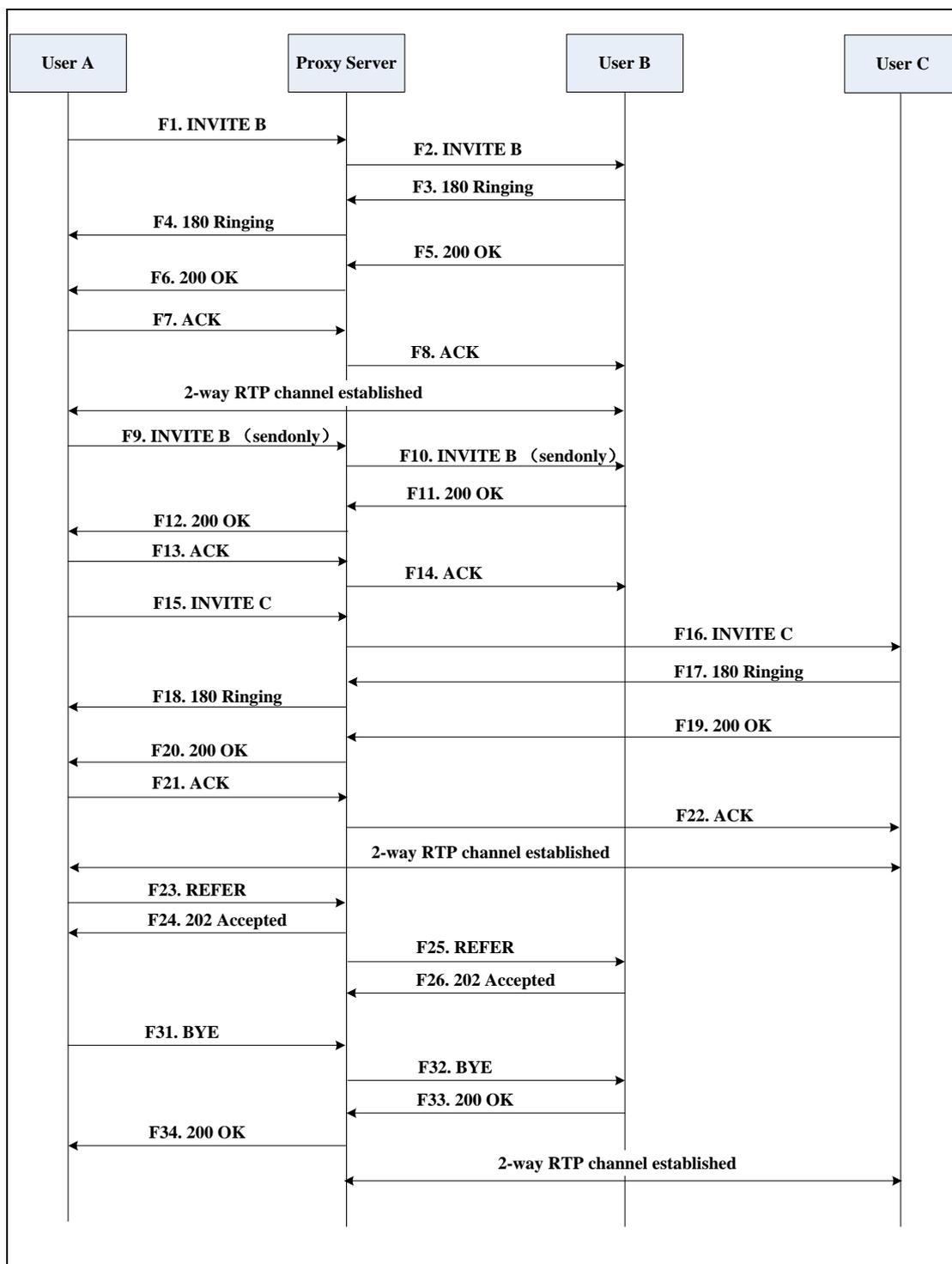
Call Transfer with Consultation

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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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 placed 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 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.

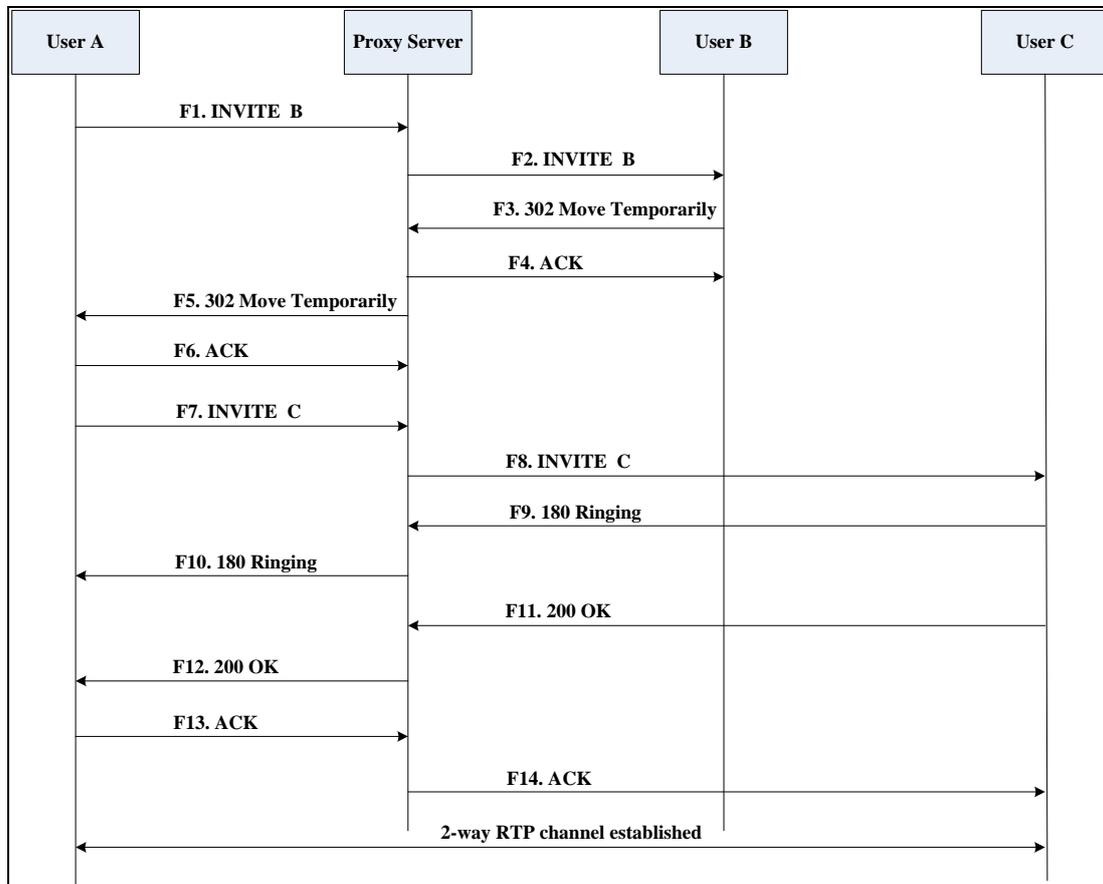
Always Call Forward

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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

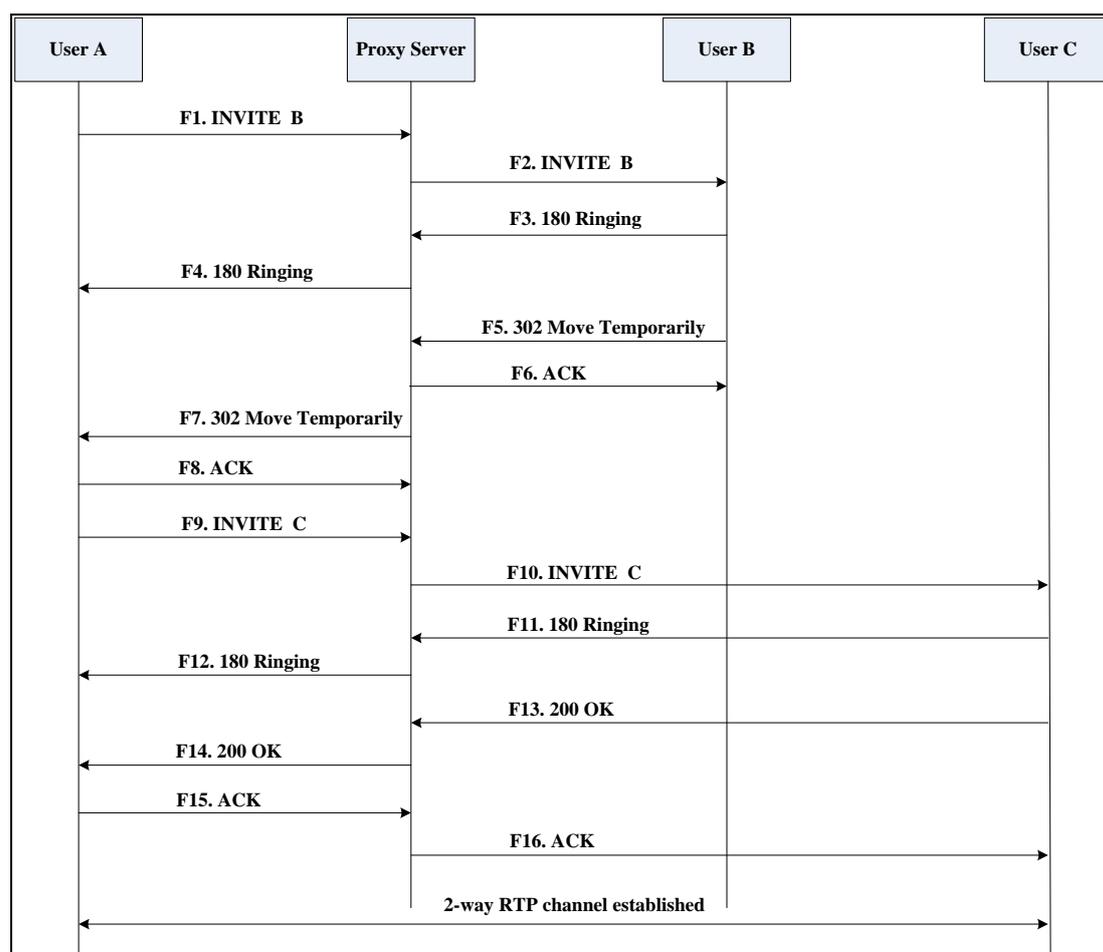
Busy Call Forward

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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

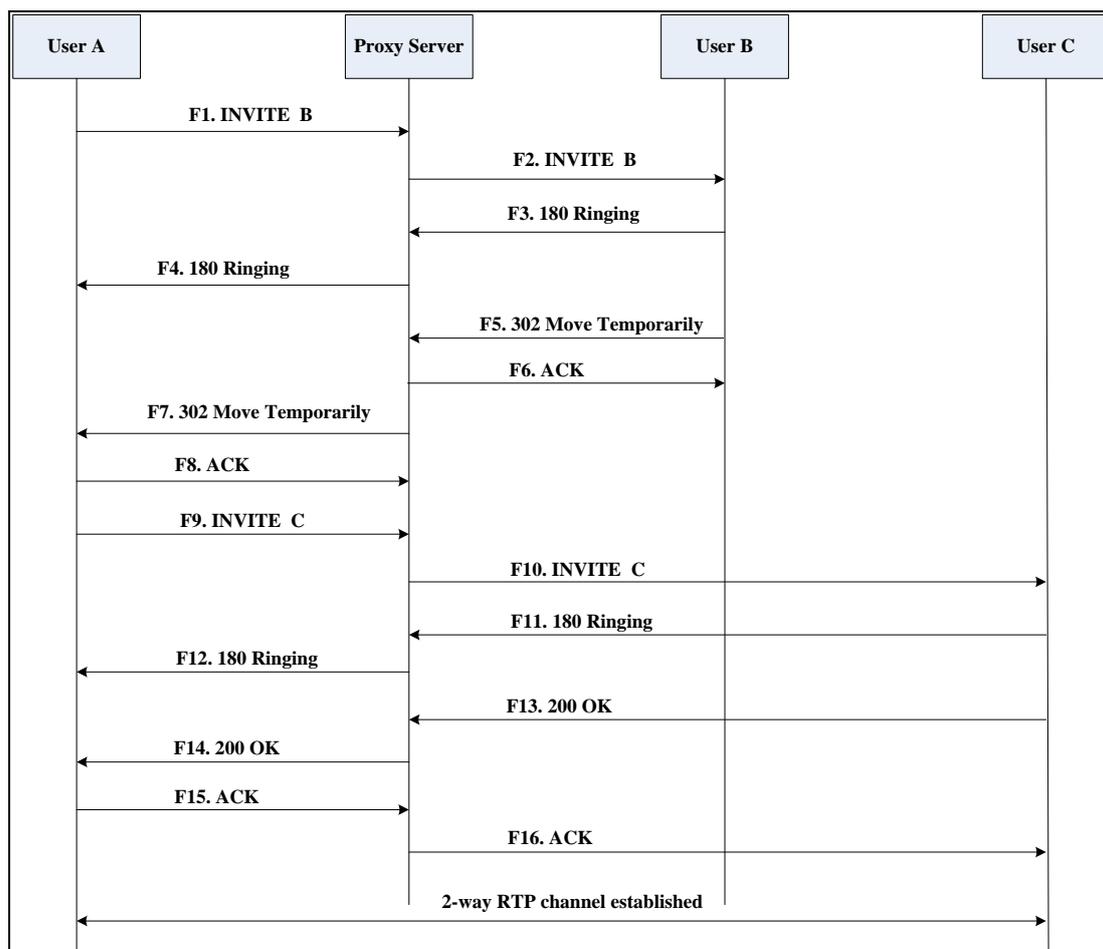
No Answer Call Forward

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	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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.

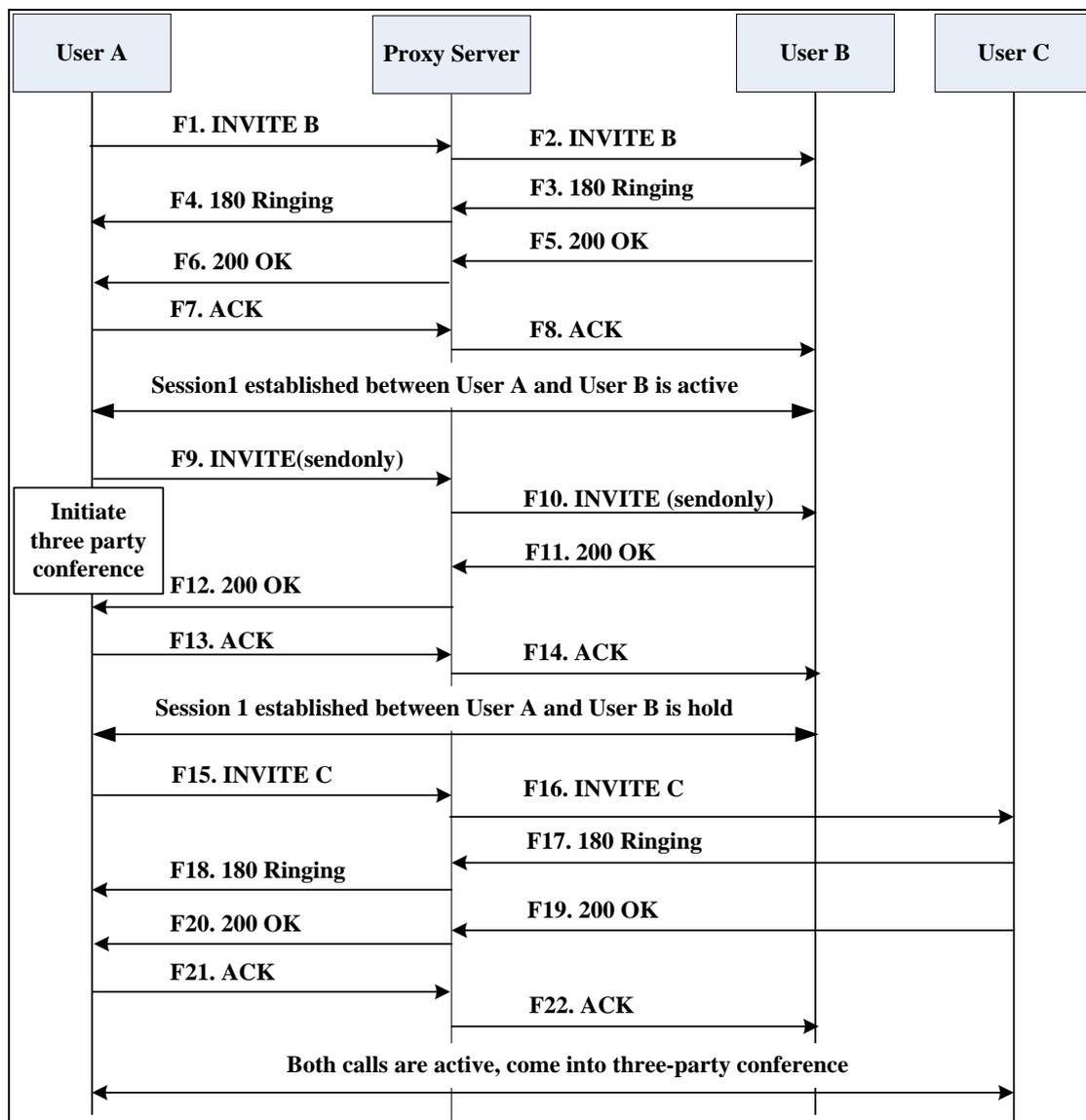
Call Conference

The following figure illustrates successful 3-way calling between Yealink 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 places 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.



Step	Action	Description
F1	INVITE—User A to Proxy Server	<p>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.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • 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 placed 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 beginning 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 required. The settings in the <y0000000000xx>.cfg file will be overridden by settings in the <MAC>.cfg file.

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 =
```

```
phone_setting.dialnow_delay =
dialplan.replace.prefix.1 =
dialplan.replace.replace.1 =
dialplan.replace.line_id.1 =
dialplan.item.1 =
```

#Time Settings

```
local_time.time_zone =
local_time.time_zone_name =
local_time.ntp_server1 =
local_time.ntp_server2 =
local_time.interval =
local_time.dhcp_time =
```

#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

```
features.play_hold_tone.enable =
features.play_hold_tone.delay =
sip.rfc2543_hold =
```

#Hotline

```
features.hotline_number =
features.hotline_delay =
```

#Web Server Type

```
wui.http_enable =
network.port.http =
wui.https_enable =
network.port.https =
```

#DTMF Suppression

```
features.dtmf.hide =
features.dtmf.hide_delay =
```

#Call Forward

In Phone Mode

```
features.fwd_mode = 0
forward.always.enable =
forward.always.target =
forward.always.on_code =
forward.always.off_code =
forward.busy.enable =
forward.busy.target =
forward.busy.on_code =
forward.busy.off_code =
forward.no_answer.enable =
forward.no_answer.target =
forward.no_answer.timeout =
forward.no_answer.on_code =
forward.no_answer.off_code =
```

#In Custom Mode

```
features.fwd_mode = 1
account.1.always_fwd.enable =
account.1.always_fwd.target =
account.1.always_fwd.on_code =
account.1.busy_fwd.off_code =
account.1.busy_fwd.enable =
account.1.busy_fwd.target =
```

```
account.1.busy_fwd.on_code =
account.1.busy_fwd.off_code =
account.1.timeout_fwd.enable =
account.1.timeout_fwd.target =
account.1.timeout_fwd.timeout =
account.1.timeout_fwd.on_code =
account.1.timeout_fwd.off_code =
```

#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.info =
voice.tone.stutter =
voice.tone.message =
voice.tone.autoanswer =
```

#Remote Phone Book

```
features.remote_phonebook.enable =
features.remote_phonebook.flash_time =
```

#LDAP

```
ldap.enable =  
ldap.name_filter =  
ldap.number_filter =  
ldap.host =  
ldap.port =  
ldap.base =  
ldap.user =  
ldap.password =  
ldap.max_hits =  
ldap.name_attr =  
ldap.numb_attr =  
ldap.display_name =  
ldap.version =  
ldap.call_in_lookup =  
ldap.ldap_sort =
```

#Action URL

```
action_url.setup_completed =  
action_url.registered =  
action_url.unregistered =  
action_url.register_failed =  
action_url.off_hook =  
action_url.on_hook =  
action_url.incoming_call =  
action_url.outgoing_call =  
action_url.call_established =  
action_url.dnd_on =  
action_url.dnd_off =  
action_url.always_fwd_on =  
action_url.always_fwd_off =  
action_url.busy_fwd_on =  
action_url.busy_fwd_off =  
action_url.no_answer_fwd_on =  
action_url.no_answer_fwd_off =  
action_url.transfer_call =  
action_url.blind_transfer_call =  
action_url.attended_transfer_call =  
action_url.hold =  
action_url.unhold =  
action_url.mute =  
action_url.unmute =  
action_url.missed_call =  
action_url.call_terminated =
```

```
action_url.busy_to_idle =  
action_url.idle_to_busy =  
action_url.ip_change =  
action_url.forward_incoming_call =  
action_url.reject_incoming_call =  
action_url.answer_new_incoming_call =  
action_url.transfer_finished =  
action_url.transfer_failed =
```

#Access URL of Resource Files

```
dialplan_dialnow.url =  
dialplan_replace_rule.url =  
local_contact.data.url =  
remote_phonebook.data.1.url =  
directory_setting.url =  
super_search.url =
```


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