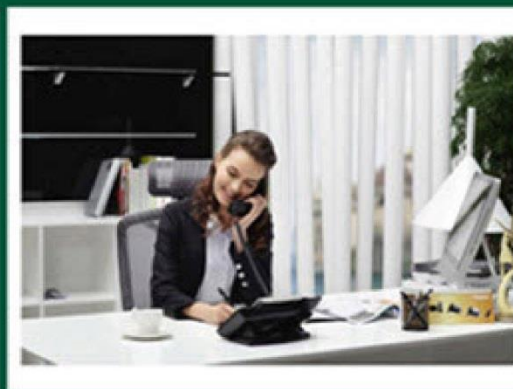


Yealink



Ultra-elegant SIP-T4X IP Phones Administrator Guide

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2. This device must accept any interference received, including interference that may cause undesired operation.

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Note: This device is tested and complies with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experienced radio/TV technician for help.

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The original GPL license, source code of components licensed under GPL and used in Yealink products can be downloaded online:

<http://www.yealink.com/GPLOpenSource.aspx?BaseInfoCatId=293&NewsCatId=293&CatId=293>.

About This Guide

The guide is intended for administrators who need to properly configure, customize, manage, and troubleshoot the IP phone system rather than the end-users. It provides details on the functionality and configuration of SIP-T4X 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 the IP networking and prior knowledge of IP telephony concepts are necessary.

Documentations

This guide covers SIP-T48G, T46G, T42G and T41P IP phones. The following related documents for SIP-T4X IP phones are available:

- Quick Start Guides, which describe how to assemble IP phones and configure the most basic features available on IP phones.
- User Guides, which describe basic and advanced features available on IP phones.
- Auto Provisioning Deployment Guide, which describes how to provision IP phones using the configuration files.
- Configuration Conversion Tool User Guide, which describes how to convert and encrypt the configuration files using the Configuration Conversion Tool.
- <y000000000xx>.cfg and <MAC>.cfg template configuration files.
- IP Phones Deployment Guide for BroadSoft UC-One Environments, which describes how to configure the 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 the firmware version 73 or higher. The firmware format is like x.x.x.x.rom. The second x from left should be greater than or equal to 73 (e.g., the firmware version of SIP-T46G IP phone: 28.73.0.40.rom). This administrator guide includes the following chapters:

- Chapter 1, "[Product Overview](#)" describes SIP components and SIP IP phones.
- Chapter 2, "[Getting Started](#)" describes how to install and connect IP phones and configuration methods.
- Chapter 3, "[Configuring Basic Features](#)" describes how to configure basic features

on IP phones.

- Chapter 4, "[Configuring Advanced Features](#)" describes how to configure advanced features on IP phones.
- Chapter 5, "[Configuring Audio Features](#)" describes how to configure audio features on IP phones.
- Chapter 6, "[Configuring Security Features](#)" describes how to configure security features on 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 sample configuration files.

Summary of Changes

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

Changes for Release 73, Guide Version 73.40

The following section is new for this version:

- [Hide Features Access Code](#) on page 303

Major updates have occurred to the following sections:

- [Physical Features of SIP-T4X IP Phones](#) on page 4
- [Configuration Files](#) on page 17
- [ReCall](#) on page 232
- [Distinctive Ring Tones](#) on page 260
- [BLF List](#) on page 297
- [Static DNS Cache](#) on page 371
- [Voice Quality Monitoring](#) on page 426
- [Appendix C: Configuring DSS Key](#) on page 514
- [Appendix B: Time Zones](#) on page 511

Changes for Release 73, Guide Version 73.16

The following sections are new for this version:

- [Notification Popups](#) on page 52

- [Call Display](#) on page 62
- [Input Method Customization](#) on page 99
- [Off Hook Hot Line Dialing](#) on page 129
- [Feature Key Synchronization](#) on page 211
- [BLF List](#) on page 297
- [Capturing the Current Screen of the Phone](#) on page 358
- [Voice Quality Monitoring](#) on page 426

Major updates have occurred to the following sections:

- [Configuration Files](#) on page 17
- [DHCP](#) on page 21
- [Configuring Basic Network Parameters](#) on page 21
- [Upgrading Firmware](#) on page 40
- [Phone Lock](#) on page 72
- [Language](#) on page 90
- [Anonymous Call Rejection](#) on page 158
- [DTMF](#) on page 241
- [Distinctive Ring Tones](#) on page 260
- [Remote Phone Book](#) on page 273
- [LDAP](#) on page 277
- [Message Waiting Indicator](#) on page 310
- [Multicast Paging](#) on page 316
- [VLAN](#) on page 383
- [802.1X Authentication](#) on page 399
- [IPv6 Support](#) on page 414
- [Transport Layer Security](#) on page 461
- [Secure Real-Time Transport Protocol](#) on page 471
- [Encrypting Configuration Files](#) on page 474
- [Analyzing Configuration Files](#) on page 506

Changes for Release 72.0, Guide Version 72.3

The following sections are new for this version:

- [Provisioning Server](#) on page 18
- [Static DNS Cache](#) on page 371
- [Background Noise Suppression](#) on page 454

- [Automatic Gain Control](#) on page 454

Major updates have occurred to the following section:

- [Configuration Methods](#) on page 16
- [Call Hold](#) on page 181
- [Audio Codecs](#) on page 445
- [Acoustic Clarity Technology](#) on page 452
- [Resource Files](#) on page 481

Changes for Release 72.0, Guide Version 72.2

This version is updated to incorporate SIP-T48G IP phones as one of the T4X device models. The following sections are new for this version:

- [Directory](#) on page 131
- [Search Source List in Dialing](#) on page 133

Major updates have occurred to the following section:

- [SIP IP Phone Models](#) on page 3
- [Connecting the IP Phones](#) on page 9
- [Reading Icons](#) on page 14
- [PPPoE](#) on page 32
- [Wallpaper](#) on page 56
- [Phone Lock](#) on page 72
- [Language](#) on page 90
- [Logo Customization](#) on page 103
- [Distinctive Ring Tones](#) on page 260
- [Busy Lamp Field](#) on page 289
- [Automatic Call Distribution](#) on page 305
- [Directory Template](#) on page 485
- [Super Search Template](#) on page 486

Changes for Release 72.0, Guide Version 72.1

The following sections are new for this version:

- [Power Indicator LED](#) on page 48
- [Contrast](#) on page 55

Major updates have occurred to the following sections:

- [DHCP](#) on page 21
- [Backlight](#) on page 59
- [Time and Date](#) on page 78
- [Key as Send](#) on page 111
- [Anonymous Call](#) on page 154
- [Busy Lamp Field](#) on page 289
- [Action URL](#) on page 337
- [IPv6 Support](#) on page 414
- [Transport Layer Security](#) on page 461

Changes for Release 71.0, Guide Version 71.181

Major updates have occurred to the following section:

- [Language](#) on page 90

Changes for Release 71.0, Guide Version 71.180

Major updates have occurred to the following sections:

- [Language](#) on page 90
- [Anonymous Call](#) on page 154

Changes for Release 71.0, Guide Version 71.171

Major updates have occurred to the following sections:

- [Backlight](#) on page 59
- [Language](#) on page 90
- [Logo Customization](#) on page 103
- [Anonymous Call](#) on page 154
- [Action URL](#) on page 337
- [Action URI](#) on page 354
- [Audio Codecs](#) on page 445

Changes for Release 71.0, Guide Version 71.170

Major updates have occurred to the following sections:

- [Language](#) on page 90

- [Auto Answer](#) on page 149
- [Audio Codecs](#) on page 445
- [Encrypting Configuration Files](#) on page 474

Changes for Release 71.0, Guide Version 71.150

This version is updated to incorporate T41P as one of the T4X device models. The following section is new for this version:

- [Logo Customization](#) on page 103

Major updates have occurred to the following sections:

- [SIP IP Phone Models](#) on page 3
- [Configuring Transmission Methods of the Internet Port and PC Port](#) on page 35
- [Language](#) on page 90
- [Remote Phone Book](#) on page 273
- [Server Redundancy](#) on page 359
- [Audio Codecs](#) on page 445
- [Transport Layer Security](#) on page 461
- [Secure Real-Time Transport Protocol](#) on page 471

Changes for Release 71.0, Guide Version 71.80

This version is updated to incorporate T42G as one of the T4X device models. The following section is new for this version:

- [SIP IP Phone Models](#) on page 3

Major updates have occurred to the following sections:

- [Reading Icons](#) on page 14
- [PPPoE](#) on page 32
- [Upgrading Firmware](#) on page 40
- [Backlight](#) on page 59
- [Language](#) on page 90
- [Call Completion](#) on page 152
- [TR-069 Device Management](#) on page 409
- [IPv6 Support](#) on page 414
- [Audio Codecs](#) on page 445
- [Appendix C: Configuring DSS Key](#) on page 514

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Product Overview

This chapter contains the following information about SIP-T4X 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 the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP provides capabilities to:

- Determine the location of the target endpoint -- SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target endpoint -- Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between endpoints. Conferences are established using only the media capabilities that can be supported by all endpoints.
- Determine the availability of the target endpoint -- A call cannot be completed because the target endpoint is unavailable. SIP determines whether the called party is already on the IP phone or 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 the following roles:

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

User Agent Client (UAC)

The UAC is an application that initiates up to six feasible SIP requests to the UAS. The six requests issued by the UAC are: INVITE, ACK, OPTIONS, BYE, CANCEL and REGISTER. When the SIP session is being initiated by the UAC SIP component, the UAC determines the information essential for the request, which is the protocol, the port and the IP address of the UAS to which the request is being sent. This information can be dynamic and 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 the SIP-T4X IP phone family. SIP-T4X 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. SIP-T4X 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.

SIP-T4X IP phones provide a powerful and flexible IP communication solution for Ethernet TCP/IP networks, delivering excellent voice quality. The high-resolution graphic display provides content in multiple languages for system status, call log and directory access. SIP-T4X IP phones also support advanced functionalities, including LDAP, Busy Lamp Field, Sever Redundancy and Network Conference.

The following IP phone models are described:

- SIP-T48G
- SIP-T46G
- SIP-T42G
- SIP-T41P

SIP-T4X IP phones comply with the SIP standard (RFC 3261), and they can only be used within a network that supports this type of phone.

In order to operate as SIP endpoints in your network successfully, SIP-T4X IP phones must meet the following requirements:

- A working IP network is established.
- VoIP gateways are configured for SIP.
- The latest (or compatible) firmware of SIP-T4X IP phones is available.

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

Physical Features of SIP-T4X IP Phones

This section lists the available physical features of SIPT4X IP phones.

SIPT48G



Physical Features:

- 7" 800 x 480 pixel color touch screen with backlight
- 24 bit depth color
- 16 VoIP accounts, Broadsoft Validated/Asterisk® Compatible
- HD Voice: HD Codec, HD Handset, HD Speaker
- 26 keys including 7 feature keys
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 1*RJ12 (6P6C) expansion module port
- 4 LEDs: 1*power, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)
- Built-in USB port, support Bluetooth headset

SIPT46G



Physical Features:

- 4.3" 480 x 272 pixel color display with backlight
- 24 bit depth color
- 16 VoIP accounts, Broadsoft Validated/Asterisk® Compatible
- HD Voice: HD Codec, HD Handset, HD Speaker
- 40 keys including 10 line keys
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 1*RJ12 (6P6C) expansion module port
- 14 LEDs: 1*power, 10*line, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)
- Built-in USB port, support Bluetooth headset

SIPT42G



Physical Features:

- 192 x 64 graphic LCD
- 12 VoIP accounts, Broadsoft Validated/Asterisk® Compatible
- HD Voice: HD Codec, HD Handset, HD Speaker
- 34 keys including 6 line keys
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100/1000Mbps Ethernet ports
- 1*RJ12 (6P6C) EHS36 headset adapter port
- 10 LEDs: 1*power, 6*line, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)

SIPT41P



Physical Features:

- 192 x 64 graphic LCD
- 6 VoIP accounts, Broadsoft Validated/Asterisk® Compatible
- HD Voice: HD Codec, HD Handset, HD Speaker
- 34 keys including 6 line keys
- 1*RJ9 (4P4C) handset port
- 1*RJ9 (4P4C) headset port
- 2*RJ45 10/100Mbps Ethernet ports
- 1*RJ12 (6P6C) EHS36 headset adapter port
- 10 LEDs: 1*power, 6*line, 1*mute, 1*headset, 1*speakerphone
- Power adapter: AC 100~240V input and DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)

Key Features of SIP-T4X IP Phones

In addition to physical features introduced above, SIP-T4X IP phones also support the following key features when running the latest firmware:

- **Phone Features**
 - **Call Options:** call waiting, call hold, call mute, call forward, call transfer, call pickup, conference.
 - **Basic Features:** DND, auto redial, live dialpad, dial plan, hotline, caller

identity, auto answer.

- **Advanced Features:** BLF, server redundancy, distinctive ring tones, remote phone book, LDAP, 802.1X authentication.

- **Codecs and Voice Features**

- Wideband codec: G.722
- Narrowband codec: G.711 (A/μ), G.723, G.726, G.729, iLBC.
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

- **Network Features**

- SIP v1 (RFC2543), v2 (RFC3261)
- IPv4/IPv6 support
- NAT Traversal: STUN mode
- DTMF: INBAND, RFC2833, SIP INFO
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: Static/DHCP/PPPoE (PPPoE is for SIP-T48G/T46G only)
- TFTP/DHCP client
- HTTP/HTTPS server
- DNS client
- NAT/DHCP server

- **Management**

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

- **Security**

- HTTPS (server/client)
- SRTP (RFC3711)
- Transport Layer Security (TLS)
- VLAN (802.1q), QoS
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES encryption
- Phone lock for personal privacy protection (not applicable to SIP-T48G)
- Admin/User configuration mode

Getting Started

This chapter provides basic information and installation instructions of SIP-T4X IP phones.

This chapter provides the following sections:

- [Connecting the IP Phones](#)
- [Initialization Process Overview](#)
- [Verifying Startup](#)
- [Reading Icons](#)
- [Configuration Methods](#)
- [Provisioning Server](#)
- [Configuring Basic Network Parameters](#)
- [Upgrading Firmware](#)

Connecting the IP Phones

This section introduces how to install SIP-T4X IP phones with the components in packaging contents.

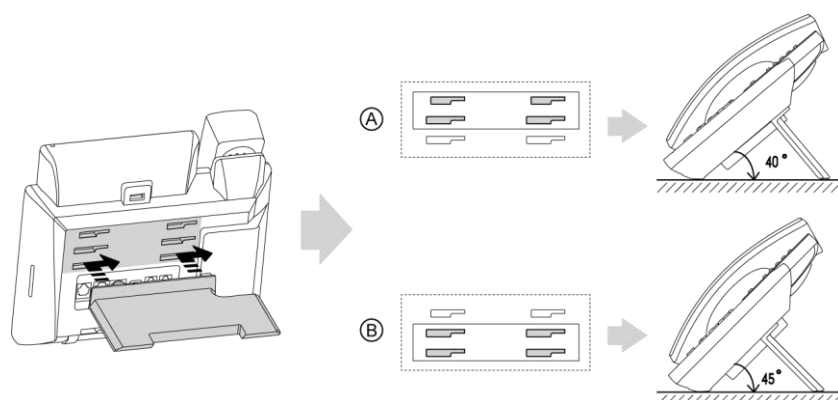
1. Attach the stand and optional wall mount bracket
2. Connect the handset and optional headset
3. Connect the network and power

Note

A headset, wall mount bracket and power adapter are not included in packaging contents.

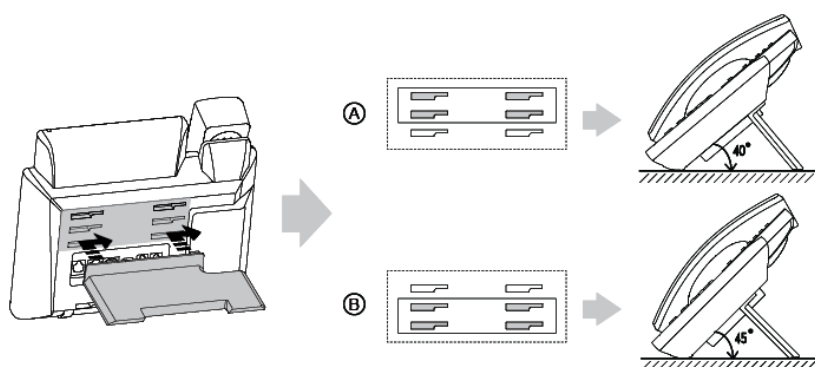
- 1) **Attach the stand and optional wall mount bracket:**

For SIP-T46G:

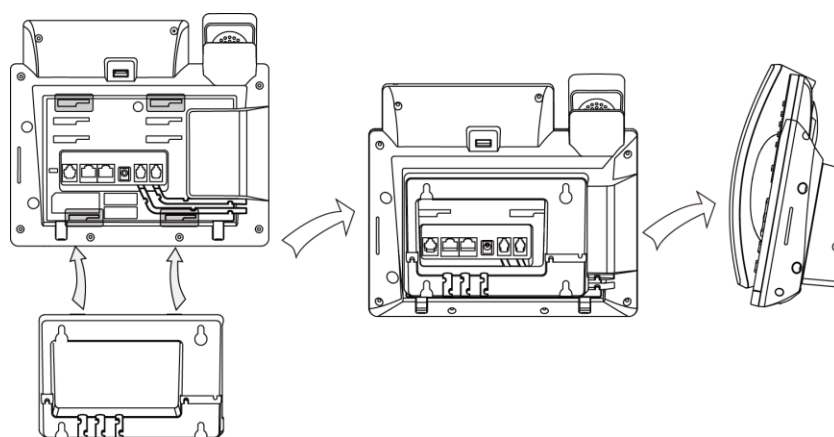


Desk Mount Method

For SIP-T48G/T42G/T41P:



Desk Mount Method



Wall Mount Method (Optional)

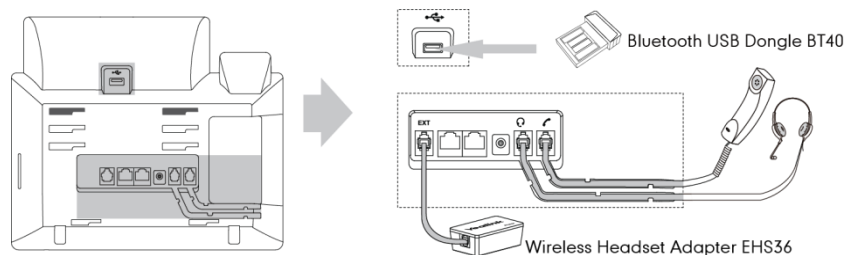
Note

The top two slots on SIP-T48G IP phones are plugged up by silica gel. You need to pull out silica gel before attaching the wall mount bracket.

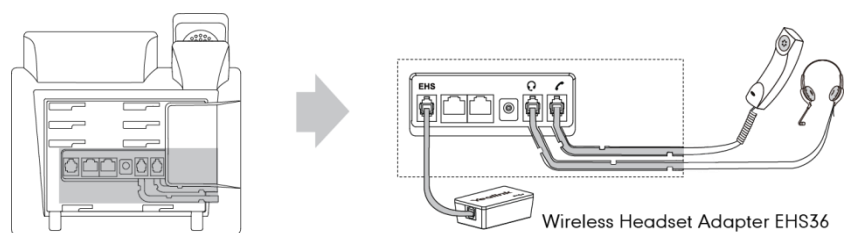
For more information on how to mount the IP phone to a wall, refer to *Yealink Wall Mount Quick Installation Guide for SIP-T4X IP Phones*.

2) Connect the handset, optional headset and Bluetooth headset:

For SIP-T48G/T46G:



For SIP-T42G/T41P:



Note

Wireless headset adapter EHS36 and Bluetooth USB dongle should be purchased separately.

For more information on how to use the EHS36 on the IP phone, refer to *Yealink EHS36 User Guide*.

Bluetooth can only be used on the SIP-T48G and SIP-T46G IP phones. For more information on how to use the Bluetooth on SIP-T48G/T46G IP phones, refer to *Yealink Bluetooth USB Dongle BT40 User Guide*.

EXT port on SIP-T48G/T46G IP phones can also be used to connect the expansion module EXP40. For more information on how to connect the EXP40, refer to *Yealink EXP40 User Guide*.

3) Connect the network and power:

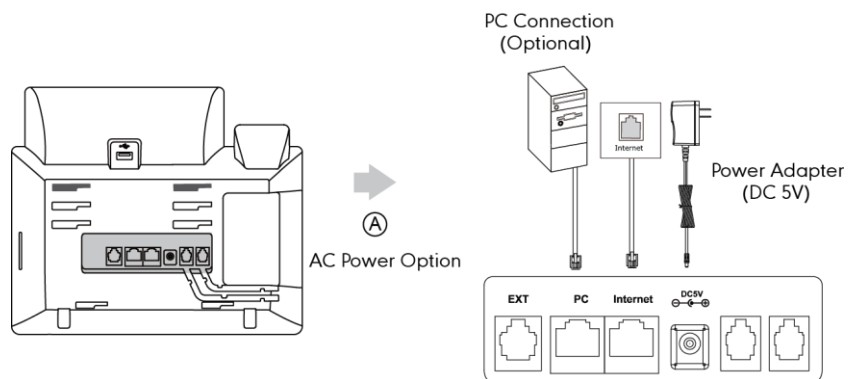
- AC power
- Power over Ethernet (PoE)

AC Power

To connect the AC power and network:

1. Connect the DC plug of the power adapter to the DC5V port on IP phones 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 IP phones and the one on the wall or switch/hub device port.

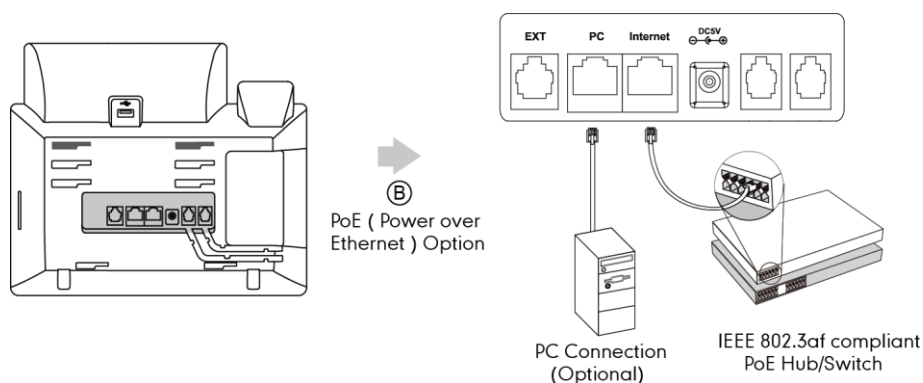


Power over Ethernet

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 IP phones 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 IP phone to the power adapter. Make sure the switch/hub is PoE-compliant.

IP phones 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 IP phones are updating firmware and configurations.

Initialization Process Overview

The initialization process of IP phones is responsible for network connectivity and operation of IP phones 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 proceed:

Loading the ROM file

The ROM file resides in the flash memory of IP phones. IP phones come from the factory with a ROM file preloaded. During initialization, IP phones run a bootstrap loader that loads and executes the ROM file.

Configuring the VLAN

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

Querying the DHCP (Dynamic Host Configuration Protocol) Server

IP phones are capable of querying a DHCP server. DHCP is enabled on IP phones by default. The following network parameters can be obtained from the DHCP server during initialization:

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

You need to configure the network parameters of IP phones manually if any of them is not provided by the DHCP server. For more information on configuring network parameters manually, refer to [Configuring Network Parameters Manually](#) on page 26.

Contacting the auto provisioning server

SIP-T4X IP phones support the FTP, TFTP, HTTP, and HTTPS protocols for auto provisioning and are configured by default to use TFTP protocol. If IP phones are configured to obtain configurations from the TFTP server, they will connect to the TFTP server and download the configuration file(s) during startup. IP phones will be able to resolve and apply the configurations written in the configuration file(s). If IP phones do not obtain the configurations from the TFTP server, IP phones will use the configurations stored in the flash memory.

Updating firmware

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

Downloading the resource files

In addition to configuration file(s), IP phones 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 red.
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
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.

Reading Icons

Icons associated with different features may appear on the LCD screen. The following table provides a description for each icon on SIP-T4X IP phone models.

T48G	T46G	T42G/T41P	Description
			Network is unavailable
			Registered successfully
			Register failed
 (Flashing)			Registering

T48G	T46G	T42G/T41P	Description
			Hands-free speakerphone mode
			Handset mode
			Headset mode
			Voice Mail
/		/	Text Message
			Auto Answer
		DND	Do Not Disturb
			Call Forward
			Call Hold
			Call Mute
			Ringer volume is 0
/			Phone Lock
			Multi-lingual lowercase letters input mode
			Multi-lingual uppercase letters input mode
			Alphanumeric input mode
			Numeric input mode
			Multi-lingual uppercase and lowercase letters input mode
			Received Calls
			Placed Calls
			Missed Calls
			Forwarded Calls

T48G	T46G	T42G/T41P	Description
			Recording box is full
			A call cannot be recorded
			Recording starts successfully
			Recording cannot be started
			Recording cannot be stopped
		VPN	VPN is enabled
		/	Bluetooth mode is on
		/	Bluetooth headset is both paired and connected
/		/	Conference
			The default contact icon
		/	The default caller photo

Configuration Methods

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

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

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

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

Phone User Interface

An administrator or a user can configure and use IP phones via phone user interface.

Specific features access is restricted to the administrator. These specific features are password protected by default. The default password is “admin” (case-sensitive). Not all features are available on phone user interface. For more information, refer to Yealink phone-specific user guide, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

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). Most features are available for configuring via 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 65.

Configuration Files

An administrator can deploy and maintain a mass of IP phones using configuration files. The configuration files consist of:

- Common CFG file
- MAC-Oriented CFG file
- MAC-local CFG file

Common CFG file

A Common CFG file contains parameters that affect the basic operation of the IP phone, such as language and volume. It will be effectual for all IP phones of the same model. The common CFG file has a fixed name for each IP phone model. The name of the Common CFG file for each SIP-T4X IP phone model is:

- SIP-T48G: y000000000035.cfg
- SIP-T46G: y000000000028.cfg
- SIP-T42G: y000000000029.cfg
- SIP-T41P: y000000000036.cfg

MAC-Oriented CFG file

A MAC-Oriented CFG file contains parameters unique to a particular phone. It will only be effectual for a specific IP phone. The MAC-Oriented CFG file is named after the MAC address of the IP phone. For example, if the MAC address of an IP phone is 001565113af8, the name of the MAC-Oriented CFG file must be 001565113af8.cfg.

MAC-local CFG file

A MAC-local CFG file contains changes that users make via web user interface and phone user interface. It will only be effectual for a specific IP phone. The MAC-local CFG

file is named after the MAC address of the IP phone. This file is stored locally on the IP phone and can also be uploaded to the provisioning server.

The MAC-local CFG file enables the phone to protect personalized settings. For more information on how to protect personalized settings, refer to the section *Specific Scenarios-Protect Personalized Settings* in *Yealink SIP-T2_Series_T4_Series_IP_Phones_Auto_Provisioning_Guide*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Central Provisioning

IP phones can be centrally provisioned from a provisioning server using the configuration files (<y000000000xx>.cfg and <MAC>.cfg). You can use a text-based editing application to edit configuration files, and then store configuration files to a provisioning server. For more information on the provisioning server, refer to [Provisioning Server](#) on page 18.

IP phones can obtain the provisioning server address during startup. 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_T4_Series_IP_Phones_Auto_Provisioning_Guide*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

When modifying parameters, learn the following:

- Parameters in configuration files override those stored in IP phones' flash memory.
- The .cfg extension of the configuration files must be in lowercase.
- Each line in a configuration file must use the following format and adhere to the following rules:

```
variable-name = value
```

- Associate only one value with one variable.
- Separate variable name and value with equal sign.
- Set only one variable per line.
- Put the variable and value on the same line, and do not break the line.
- Comment the variable on a separated line. Use the pound (#) delimiter to distinguish the comments.

Provisioning Server

Supported Provisioning Protocols

IP phones perform the auto provisioning function of downloading configuration files,

downloading resource files and upgrading firmware. The transfer protocol is used to download files from the provisioning server. IP phones support several transport protocols for provisioning, including FTP, TFTP, HTTP, and HTTPS protocols. And you can specify the transport protocol in the provisioning server address, for example, `http://xxxxxxx`. If not specified, the TFTP protocol is used. The provisioning server address can be IP address, domain name or URL. If a user name and password are specified as part of the provisioning server address, for example, `http://user:pwd@server/dir`, they will be used only if the server supports them.

Note

A URL should contain forward slashes instead of back slashes and should not contain spaces. Escape characters are not supported.

If a user name and password are not specified as part of the provisioning server address, the User Name and Password of the provisioning server configured on the IP phone will be used.

There are two types of FTP methods—active and passive. IP phones are not compatible with active FTP.

Setting up the Provisioning Server

The provisioning server can be on the local LAN or anywhere on the Internet. Use the following procedure as a recommendation if this is your first provisioning server setup. For more information on how to set up a provisioning server, refer to *Yealink_SIP-T2_Series_T4_Series_IP_Phones_Auto_Provisioning_Guide*.

To set up the provisioning server:

1. Install a provisioning server application or locate a suitable existing server.
2. Create an account and home directory.
3. Set security permissions for the account.
4. Create configuration files and edit them as desired.
5. Copy the configuration files and resource files to the provisioning server.

For more information on how to deploy IP phones using configuration files, refer to [Deploying Phones from the Provisioning Server](#) on page 19.

Note

Typically all phones are configured with the same server account, but the server account provides a means of conveniently partitioning the configuration. Give each account a unique home directory on the server and change the configuration on a per-account basis.

Deploying Phones from the Provisioning Server

The parameters in the new downloaded configuration files will override the duplicate parameters in files downloaded earlier. During auto provisioning, IP phones download

the common configuration file first, and then the MAC-oriented file. Therefore any parameter in the MAC-oriented configuration file will override the same one in the common configuration file.

Yealink supplies configuration files for each phone model, which is delivered with the IP phone firmware. The configuration files, supplied with each firmware release, must be used with that release. Otherwise, configurations may not take effect, and the IP phone will behave without exception. Before you configure parameters in the configuration files, Yealink recommends that you create new configuration files containing only those parameters that require changes.

To deploy IP phones from the provisioning server:

1. Create per-phone configuration files by performing the following steps:
 - a) Obtain a list of phone MAC addresses (the bar code label on the back of the IP phone or on the outside of the box).
 - b) Create per-phone <MAC>.cfg files by using the MAC-Oriented CFG file from the distribution as templates.
 - c) Edit the parameters in the file as desired.
2. Create new common configuration files by performing the following steps:
 - a) Create <y0000000000xx>.cfg files by using the Common CFG file from the distribution as templates.
 - b) Edit the parameters in the file as desired.
3. Copy configuration files to the home directory of the provisioning server.
4. Reboot IP phones to trigger the auto provisioning process.

IP phones discover the provisioning server address, and then download the configuration files from the provisioning server.

For more information on configuration files, refer to [Configuration Files](#) on page 17. For more information on encrypting configuration files, refer to [Encrypting Configuration Files](#) on page 474.

During the auto provisioning process, the IP phone supports the following methods to discover the provisioning server address:

- **Zero Touch:** Zero Touch feature guides you to configure network settings and the provisioning server address via phone user interface after startup.
- **PnP:** PnP feature allows IP phones to discover the provisioning server address by broadcasting the PnP SUBSCRIBE message during startup.
- **DHCP:** DHCP option can be used to provide the address or URL of the provisioning server to IP phones. When the IP phone requests an IP address using the DHCP protocol, the resulting response may contain option 66 or the custom option (if configured) that contains the provisioning server address.
- **Static:** You can manually configure the server address via phone user interface or web user interface.

For more information on the above methods, refer to *Yealink SIP-T2_Series_T4_Series_IP_Phones_Auto_Provisioning_Guide*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatelId=142&flag=142>.

Configuring Basic Network Parameters

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

Note This section mainly introduces IPv4 network parameters. IP phones also support IPv6. For more information on IPv6, refer to [IPv6 Support](#) on page 414.

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 DHCP is used, 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 the specific values in the corresponding options.

The following table lists the 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.

Parameter	DHCP Option	Description
Time Server	4	Specify a list of time servers available to the client.
Domain Name Server	6	Specify a list of domain name servers available to the client.
Log Server	7	Specify a list of MIT-LCS UDP servers available to the client.
Host Name	12	Specify the name of the client.
Domain Server	15	Specify the domain name that client should use when resolving hostnames via DNS.
Broadcast Address	28	Specify the broadcast address in use on the client's subnet.
Network Time Protocol Servers	42	Specify a list of the NTP servers available to the client by IP address.
Vendor-Specific Information	43	Identify the vendor-specific information.
Vendor Class Identifier	60	Identify the vendor type.
TFTP Server Name	66	Identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options.
Bootfile Name	67	Identify a bootfile when the 'file' field in the DHCP header has been used for DHCP options.

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

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

Procedure

DHCP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure DHCP on the IP phone. Parameter: network.internet_port.type Configure static DNS address when DHCP is used. Parameters: network.primary_dns network.secondary_dns
	<y0000000000xx>.cfg	Configure the IP phone to use manually configured static IPv4 DNS. Parameters: network.static_dns_enable
Local	Web User Interface	Configure DHCP on the IP phone. Configure static DNS address when DHCP is used. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network&q=load">http://<phoneIPAddress>/servlet?p=network&q=load
	Phone User Interface	Configure DHCP on the IP phone. Configure static DNS address when DHCP is used.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.internet_port.type	0, 1 or 2	0
<p>Description: Configures the Internet (WAN) port type for IPv4 when the IP address mode is configured as IPv4 or IPv4&IPv6.</p> <p>0-DHCP 1-PPPoE (not applicable to SIP-T42G/T41P) 2-Static IP Address</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take</p>		

Parameters	Permitted Values	Default
<p>effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type</p>		
network.static_dns_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to use manually configured static IPv4 DNS when the Internet (WAN) port type for IPv4 is configured as DHCP.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin)->Network->WAN Port->IPv4->Static DNS</p>		
network.primary_dns	IPv4 Address	Blank
<p>Description: Configures the primary IPv4 DNS server when the static IPv4 DNS is enabled.</p> <p>Example: network.primary_dns = 202.101.103.55</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Primary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Static DNS (Enabled)->Primary DNS</p>		
network.secondary_dns	IPv4 Address	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the secondary IPv4 DNS server when the static IPv4 DNS is enabled.</p> <p>Example: network.secondary_dns = 202.101.103.54</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Secondary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Static DNS (Enabled)->Secondary DNS</p>		

To configure DHCP via web user interface:

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

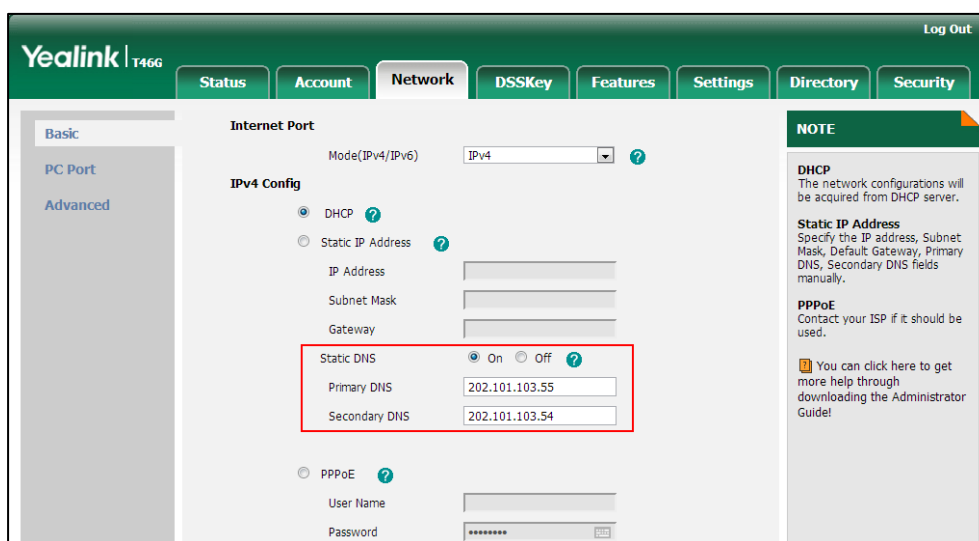
The screenshot shows the Yealink T46G web user interface. The 'Network' tab is selected, and the 'IPv4 Config' section is visible. The 'DHCP' radio button is selected and highlighted with a red box. The 'Static IP Address' radio button is unselected. The 'Static DNS' radio button is also unselected. The 'PPPoE' radio button is unselected. The 'Internet Port' section shows 'Mode(IPv4/IPv6)' set to 'IPv4'. The 'IPv4 Config' section includes fields for 'IP Address', 'Subnet Mask', 'Gateway', 'Primary DNS', and 'Secondary DNS'. The 'Static DNS' section has 'On' and 'Off' radio buttons, with 'Off' selected. The 'PPPoE' section includes 'User Name' and 'Password' fields. A 'NOTE' section on the right provides information about DHCP, Static IP Address, and PPPoE.

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

To configure DHCP via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4**.
2. Press **◀** or **▶** , or the **Switch** soft key to select the **DHCP** from the **Type** field.
3. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

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

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4->Static DNS**.
2. Press **◀** or **▶** , or the **Switch** soft key to select **Enabled** from the **Static DNS** field.
3. Enter the desired values in the **Primary DNS** and **Secondary DNS** fields respectively.
4. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

Configuring Network Parameters Manually

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

- IP Address

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

Procedure

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

Configuration File	<MAC>.cfg	Configure network parameters of the IP phone manually. Parameters: network.internet_port.type network.ip_address_mode network.internet_port.ip network.internet_port.mask network.internet_port.gateway network.primary_dns network.secondary_dns
Local	Web User Interface	Configure network parameters of the IP phone manually. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=network&q=load">http://<phoneIPAddress>/servlet?p=network&q=load
	Phone User Interface	Configure network parameters of the IP phone manually.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.internet_port.type	0, 1 or 2	0
<p>Description: Configures the Internet (WAN) port type for IPv4 when the IP address mode is configured as IPv4 or IPv4&IPv6.</p> <p>0-DHCP 1-PPPoE (not applicable to SIP-T42G/T41P) 2-Static IP Address</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: Network->Basic->IPv4 Config</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type</p>		
network.ip_address_mode	0, 1 or 2	0
<p>Description: Configures the IP address mode.</p> <p>0-IPv4 1-IPv6 2-IPv4&IPv6</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->Internet Port->Mode (IPv4/IPv6)</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IP Address Mode</p>		
network.internet_port.ip	IPv4 Address	Blank
<p>Description: Configures the IPv4 address when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet (WAN) port type for IPv4 is configured as Static IP Address.</p> <p>Example: network.internet_port.ip = 192.168.1.20</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->IP Address</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(Static IP)->IP Address</p>		
network.internet_port.mask	Subnet Mask	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the IPv4 subnet mask when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet (WAN) port type for IPv4 is configured as Static IP Address.</p> <p>Example: network.internet_port.mask = 255.255.255.0</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Subnet Mask</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(Static IP)->Subnet Mask</p>		
network.internet_port.gateway	IPv4 Address	Blank
<p>Description: Configures the IPv4 default gateway when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet (WAN) port type for IPv4 is configured as Static IP Address.</p> <p>Example: network.internet_port.gateway = 192.168.1.254</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Gateway</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(Static IP)->Gateway</p>		
network.primary_dns	IPv4 Address	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the primary IPv4 DNS server when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet (WAN) port type for IPv4 is configured as Static IP Address.</p> <p>Example: network.primary_dns = 202.101.103.55</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Primary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(Static IP)->Primary DNS</p>		
network.secondary_dns	IPv4 Address	Blank
<p>Description: Configures the secondary IPv4 DNS server when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet (WAN) port type for IPv4 is configured as Static IP Address.</p> <p>Example: network.secondary_dns = 202.101.103.54</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->Static IP Address->Secondary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(Static IP)->Secondary DNS</p>		

To configure the IP address mode via web user interface:

1. Click on **Network->Basic**.

- Select the desired value from the pull-down list of **Mode (IPv4/IPv6)**.

The screenshot shows the Yealink T466 web interface. The 'Network' tab is selected. Under 'Internet Port', the 'Mode (IPv4/IPv6)' dropdown menu is highlighted with a red box and set to 'IPv4 & IPv6'. Below this, the 'IPv4 Config' section is visible, with the 'DHCP' radio button selected. The 'Static IP Address' section includes fields for IP Address, Subnet Mask, Gateway, Primary DNS (202.101.103.55), and Secondary DNS (202.101.103.54). The 'PPPoE' section has fields for User Name and Password. A 'NOTE' box on the right provides information about DHCP, Static IP Address, and PPPoE configurations.

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





To configure a static IPv4 address via web user interface:

- Click on **Network->Basic**.
- In the **IPv4 Config** block, mark the **Static IP Address** radio box.
- Enter the IP address, subnet mask, default gateway, primary DNS and secondary DNS in the corresponding fields.

The screenshot shows the Yealink T466 web interface. The 'Network' tab is selected. Under 'Internet Port', the 'Mode (IPv4/IPv6)' dropdown menu is set to 'IPv4 & IPv6'. Below this, the 'IPv4 Config' section is visible, with the 'Static IP Address' radio button selected. The 'Static IP Address' section includes fields for IP Address (192.168.1.20), Subnet Mask (255.255.255.0), Gateway (192.168.1.254), Primary DNS (202.101.103.55), and Secondary DNS (202.101.103.54). The 'PPPoE' section has fields for User Name (a) and Password. A 'NOTE' box on the right provides information about DHCP, Static IP Address, and PPPoE configurations.



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

To configure the IP address mode via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port**.
2. Press  or  to highlight the **IP Address Mode** field.
3. Press  or  to select **IPv4** or **IPv4&IPv6** from the **IP Address Mode** field.
4. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

To configure a static IPv4 address via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4**.
2. Press  or  , or the **Switch** soft key to select the **Static IP** from the **Type** field.
3. Enter the desired values in the **IP Address, Subnet Mask, Gateway, Primary DNS** and **Secondary DNS** fields respectively.
4. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

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 Internet port of the IP phone. Contact your ISP for the PPPoE user name and password. PPPoE is not applicable to SIP-T42G and SIP-T41P IP phones.

Procedure

PPPoE can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure PPPoE on the IP phone. Parameters: network.internet_port.type
	<y0000000000xx>.cfg	Configure the user name and password for PPPoE on the IP phone. Parameters: network.pppoe.user network.pppoe.password
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.

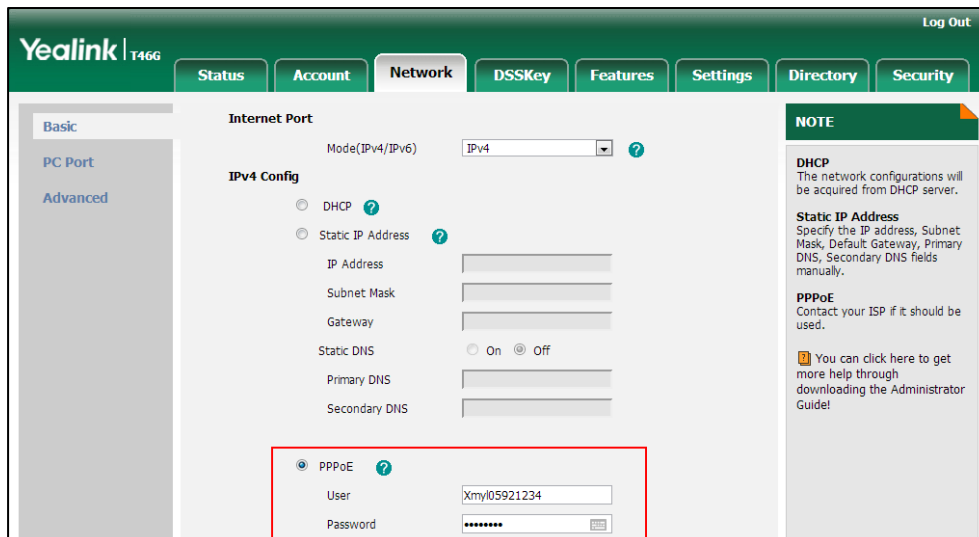
Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.internet_port.type	0, 1 or 2	0
<p>Description: Configures the Internet (WAN) port type for IPv4 when the IP address mode is configured as IPv4 or IPv4&IPv6.</p> <p>0-DHCP 1-PPPoE (not applicable to SIP-T42G/T41P) 2-Static IP Address</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type</p>		
network.pppoe.user	String within 32 characters	Blank
<p>Description: Configures the user name for PPPoE connection when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet port type is configured as PPPoE.</p> <p>Example: network.pppoe.user = Xmyl05921234</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->PPPoE->User</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(PPPoE) ->PPPoE User</p>		
network.pppoe.password	String within 99 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the password for PPPoE connection when the IP address mode is configured as IPv4 or IPv4&IPv6, and the Internet port type is configured as PPPoE.</p> <p>Example: network.pppoe.password = yealink123</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv4 Config->PPPoE->>Password</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv4->Type(PPPoE) ->PPPoE Password</p>		

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 the corresponding fields.



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

To configure PPPoE via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv4**.
2. Press **←** or **→** , or the **Switch** soft key to select the **PPPoE** from the **Type** field.
3. Enter the user name and password in the corresponding fields.

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

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

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

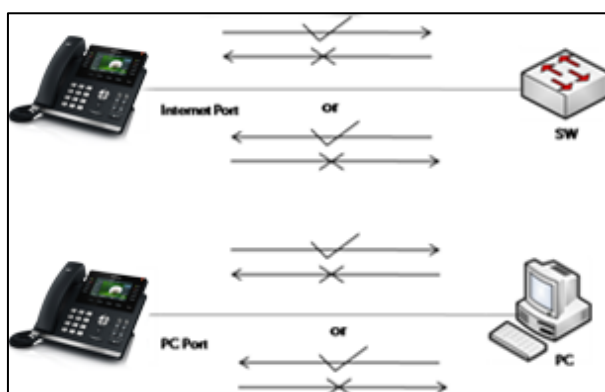
Auto-negotiation is configured for both Internet and PC ports on the IP phone by default.

Auto-negotiation

Auto-negotiation means that all 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 IP phones to auto-negotiate during the transmission.

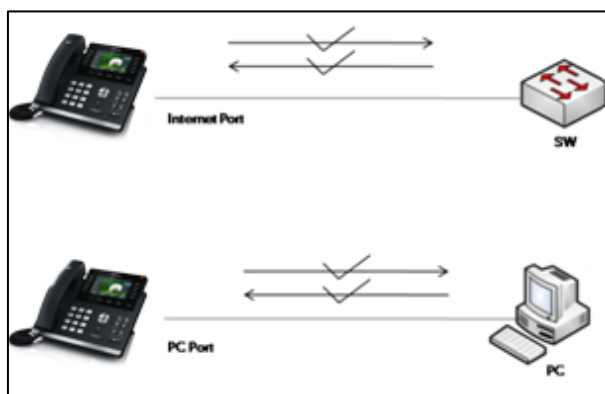
Half-duplex

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 IP phones 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 IP phones to transmit in 10Mbps, 100Mbps or 1000Mbps (1000Mbps is not applicable to SIPT41P).



Procedure

The transmission method of Ethernet port can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the transmission methods of Ethernet ports. Parameters: network.internet_port.speed_duplex network.pc_port.speed_duplex
Local	Web User Interface	Configure the transmission method of Ethernet port. Navigate to: <a href="http://<phoneIPAddress>/servlet?parameter=network-adv&q=load">http://<phoneIPAddress>/servlet?parameter=network-adv&q=load

Details of Configuration Parameters:

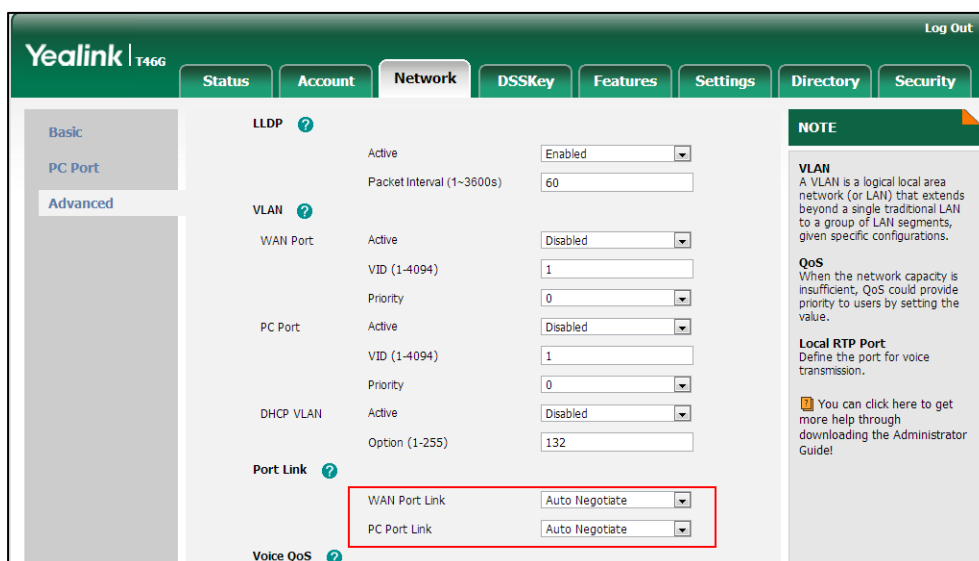
Parameters	Permitted Values	Default
network.internet_port.speed_duplex	0, 1, 2, 3, 4 or 5	0

Parameters	Permitted Values	Default
<p>Description: Configures the transmission method and speed of the Internet (WAN) port.</p> <p>0-Auto negotiate 1-Full duplex, 10Mbps 2-Full duplex, 100Mbps 3-Half duplex, 10Mbps 4-Half duplex, 100Mbps 5-Full duplex, 1000Mbps (not applicable to SIP-T41P)</p> <p>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.</p> <p>Web User Interface: Network->Advanced->Port Link->WAN Port Link</p> <p>Phone User Interface: None</p>		
network.pc_port.speed_duplex	0, 1, 2, 3, 4 or 5	0
<p>Description: Configures the transmission method and speed of the PC (LAN) port.</p> <p>0-Auto negotiate 1-Full duplex, 10Mbps 2-Full duplex, 100Mbps 3-Half duplex, 10Mbps 4-Half duplex, 100Mbps 5-Full duplex, 1000Mbps (not applicable to SIP-T41P)</p> <p>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.</p> <p>Web User Interface: Network->Advanced->Port Link->PC Port Link</p> <p>Phone User Interface: None</p>		

To configure the transmission method of Ethernet port 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

The PC port on the back of the IP phone is used to connect a PC. You can enable or disable the PC (LAN) port on SIP-T4X IP phones via web user interface or using configuration files.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the PC port. Parameters: network.PC_port.enable
Local	Web User Interface	Configure the PC port. Navigate to: http://<phoneIPAddress>/servlet ?p=network-pcport&q=load

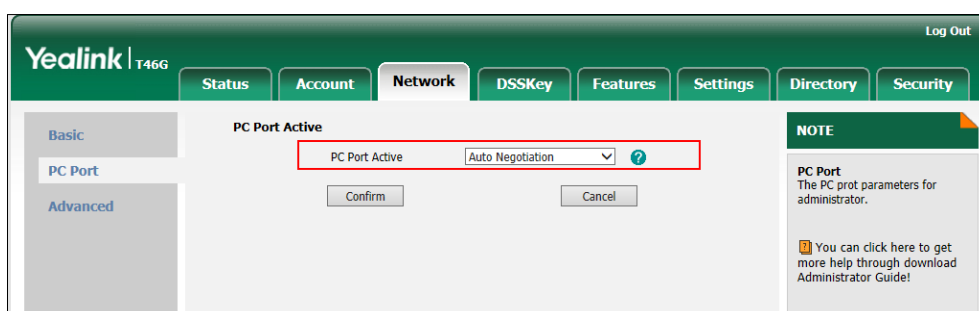
Details of Configuration Parameters:

Parameter	Permitted Values	Default
network.PC_port.enable	0 or 1	1

Parameter	Permitted Values	Default
<p>Description: Enables or disables the PC (LAN) port.</p> <p>0-Disabled</p> <p>1-Auto Negotiation</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->PC Port->PC Port Active</p> <p>Phone User Interface: None</p>		

To enable the PC port via web user interface:

1. Click on **Network->PC Port**.
2. Select **Auto Negotiation** from the pull-down list of **PC Port Active**.



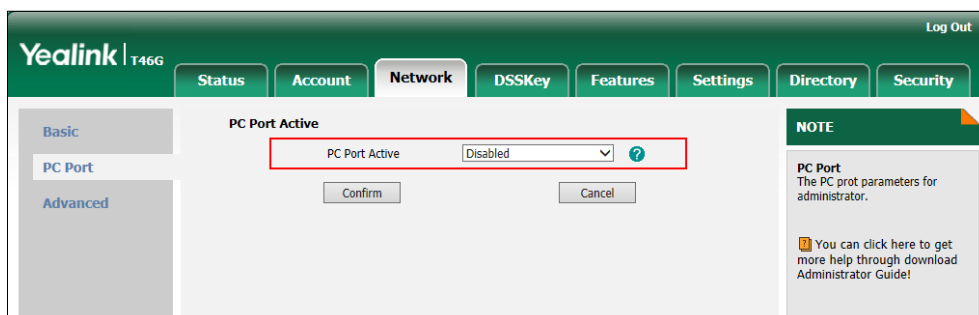
3. Click **Confirm** to accept the change.

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

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

To disable the PC port via web user interface:

1. Click on **Network->PC Port**.
2. Select **Disabled** from the pull-down list of **PC Port Active**.



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 phone.

Upgrading Firmware

This section provides information on 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	Firmware Name Example
SIP-T48G	35.x.x.x.rom	35.73.0.40.rom
SIP-T46G	28.x.x.x.rom	28.73.0.40.rom
SIP-T42G	29.x.x.x.rom	29.73.0.40.rom
SIP-T41P	36.x.x.x.rom	36.73.0.40.rom

Note

You can download the latest firmware online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Do not unplug the network and power cables when the IP phone is upgrading firmware.

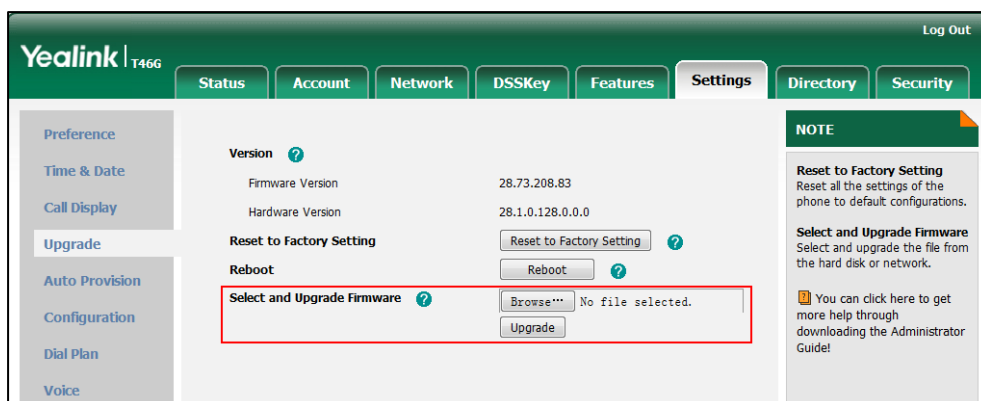
Upgrade via Web User Interface

To manually upgrade firmware via web user interface, you need to store the firmware to the local system in advance.

To upgrade firmware manually via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Browse**.
3. Locate the 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 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 using the 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 booting up.
- 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.

Configuration File	<code><y0000000000xx>.cfg</code>	Configure the way for the IP phone to check for configuration files. Parameters: auto_provision.power_on auto_provision.repeat.enable auto_provision.repeat.minutes auto_provision.weekly.enable auto_provision.weekly.begin_time
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		<p>auto_provision.weekly.end_time auto_provision.weekly.dayofweek Specify the access URL of firmware. Parameter: firmware.url</p>
Local	Web User Interface	<p>Configure the way for the IP phone to check for configuration files. Navigate to: http://<phoneIPAddress>/servlet?parameter=settings-autop&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
auto_provision.power_on	0 or 1	1
<p>Description: Enables or disables the IP phone to perform an auto provisioning process when powered on. 0-Disabled 1-Enabled Web User Interface: Settings->Auto Provision->Power On Phone User Interface: None</p>		
auto_provision.repeat.enable	0 or 1	0
<p>Description: Enables or disables the IP phone to perform an auto provisioning process repeatedly. 0-Disabled 1-Enabled Web User Interface: Settings->Auto provision->Repeatedly Phone User Interface: None</p>		

Parameters	Permitted Values	Default
auto_provision.repeat.minutes	Integer from 1 to 43200	1440
<p>Description: Configures the interval (in minutes) for the IP phone to perform an auto provisioning process repeatedly.</p> <p>Note: It works only if the value of the parameter "auto_provision.repeat.enable" is set to 1(Enabled).</p> <p>Web User Interface: Settings->Auto provision->Interval (Minutes)</p> <p>Phone User Interface: None</p>		
auto_provision.weekly.enable	0 or 1	0
<p>Description: Enables or disables the IP phone to perform an auto provisioning process weekly.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Auto provision->Weekly</p> <p>Phone User Interface: None</p>		
auto_provision.weekly.begin_time	Time from 00:00 to 23:59	00:00
<p>Description: Configures the begin time of the day for the IP phone to perform an auto provisioning process weekly.</p> <p>Note: It works only if the value of the parameter "auto_provision.weekly.enable" is set to 1(Enabled).</p> <p>Web User Interface: Settings->Auto provision->Time</p> <p>Phone User Interface: None</p>		
auto_provision.weekly.end_time	Time from 00:00 to 23:59	00:00

Parameters	Permitted Values	Default
<p>Description: Configures the end time of the day for the IP phone to perform an auto provisioning process weekly.</p> <p>Note: It works only if the value of the parameter "auto_provision.weekly.enable" is set to 1(Enabled).</p> <p>Web User Interface: Settings->Auto provision->Time</p> <p>Phone User Interface: None</p>		
<p>auto_provision.weekly.dayofweek</p>	<p>0,1,2,3,4,5,6 or a combination of these digits</p>	<p>0123456</p>
<p>Description: Configures the days of the week for the IP phone to perform an auto provisioning process weekly.</p> <p>0-Sunday 1-Monday 2-Tuesday 3-Wednesday 4-Thursday 5-Friday 6-Saturday</p> <p>Example: auto_provision.weekly.dayofweek = 01 means the IP phone will perform an auto provisioning process every Sunday and Monday.</p> <p>Note: It works only if the value of the parameter "auto_provision.weekly.enable" is set to 1(Enabled). The old parameters "auto_provision.weekly.mask" is also applicable to SIP-T4X IP phones.</p> <p>Web User Interface: Settings->Auto provision->Day of Week</p> <p>Phone User Interface: None</p>		
<p>firmware.url</p>	<p>URL within 511 characters</p>	<p>Blank</p>

Parameters	Permitted Values	Default
<p>Description: Configures the access URL of the firmware file.</p> <p>Example: firmware.url = http://192.168.1.20/2.73.0.40.rom</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Settings->Upgrade->Select and Upgrade Firmware</p> <p>Phone User Interface: None</p>		

To configure the way for the IP phone to check for new configuration files via web user interface:

1. Click on **Settings->Auto Provision**.
2. Make the desired change.

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

When the "Power On" is set to **On**, the IP phone will check configuration files stored on the provisioning server during startup and then will download firmware from the server.

Configuring Basic Features

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

- [Power Indicator LED](#)
- [Notification Popups](#)
- [Contrast](#)
- [Wallpaper](#)
- [Backlight](#)
- [Call Display](#)
- [Web Server Type](#)
- [User Password](#)
- [Administrator Password](#)
- [Phone Lock](#)
- [Time and Date](#)
- [Language](#)
- [Input Method Customization](#)
- [Logo Customization](#)
- [Softkey Layout](#)
- [Key as Send](#)
- [Dial Plan](#)
- [Hotline](#)
- [Off Hook Hot Line Dialing](#)
- [Directory](#)
- [Search Source List in Dialing](#)
- [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](#)
- [Feature Key Synchronization](#)
- [Transfer on Conference Hang Up](#)
- [Directed Call Pickup](#)
- [Group Call Pickup](#)
- [Dialog-Info Call Pickup](#)
- [ReCall](#)
- [Call Park](#)
- [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.

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.

Mute Power Light Flash

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

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.

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.

Note

Power indicator LED feature is only applicable to IP phones running firmware version 72 or later.

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the power indicator LED.</p> <p>Parameters:</p> <p>phone_setting.common_power_led_enable</p> <p>phone_setting.ring_power_led_flash_enable</p> <p>phone_setting.mail_power_led_flash_enable</p> <p>phone_setting.mute_power_led_flash_enable</p> <p>phone_setting.hold_and_held_power_led_flash_enable</p> <p>phone_setting.talk_and_dial_power_led_enable</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the power indicator LED.</p> <p>Navigate to:</p>

		http://<phoneIPAddress>/servlet?p=features-powerled&q=load
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Details of Configuration Parameters:

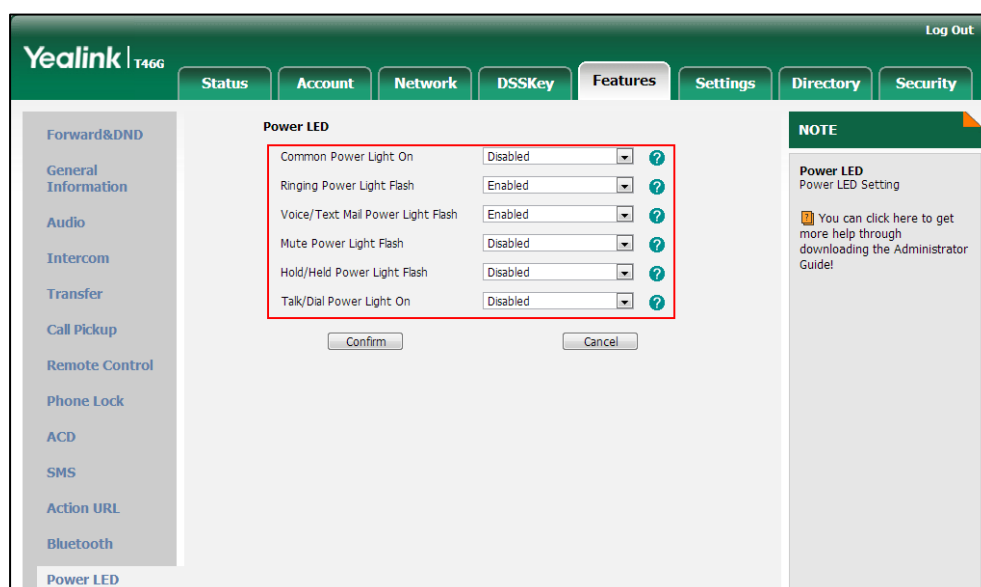
Parameters	Permitted Values	Default
phone_setting.common_power_led_enable	0 or 1	0
<p>Description: Enables or disables the power indicator LED to be turned on. 0-Disabled (power indicator LED is off) 1-Enabled (power indicator LED is solid red) Note: The old parameter "features.power_led_on" is also applicable to IP phones and "features.idle_talk_power_led_flash_enable" is also applicable to SIP-T4X IP phones. Web User Interface: Features->Power LED->Common Power Light On Phone User Interface: None</p>		
phone_setting.ring_power_led_flash_enable	0 or 1	1
<p>Description: Enables or disables the power indicator LED to flash when the IP phone receives an incoming call. 0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (300ms) red) Web User Interface: Features->Power LED->Ringing Power Light Flash Phone User Interface: None</p>		
phone_setting.mail_power_led_flash_enable	0 or 1	1
<p>Description: Enables or disables the power indicator LED to flash when the IP phone receives a voice mail or a text message. 0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED slow flashes (1000ms) red)</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: Features->Power LED->Voice/Text Mail Power Light Flash</p> <p>Phone User Interface: None</p>		
phone_setting.mute_power_led_flash_enable	0 or 1	0
<p>Description: Enables or disables the power indicator LED to flash when a call is mute.</p> <p>0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (300ms) red)</p> <p>Web User Interface: Features->Power LED->Mute Power Light Flash</p> <p>Phone User Interface: None</p>		
phone_setting.hold_and_held_power_led_flash_enable	0 or 1	0
<p>Description: Enables or disables the power indicator LED to flash when a call is placed on hold or is held.</p> <p>0-Disabled (power indicator LED does not flash) 1-Enabled (power indicator LED fast flashes (500ms) red)</p> <p>Web User Interface: Features->Power LED->Hold/Held Power Light Flash</p> <p>Phone User Interface: None</p>		
phone_setting.talk_and_dial_power_led_enable	0 or 1	0
<p>Description: Enables or disables the power indicator LED to be turned on when the IP phone is busy.</p> <p>0-Disabled (power indicator LED is off) 1-Enabled (power indicator LED is solid red)</p> <p>Web User Interface: Features->Power LED->Talk/Dial Power Light On</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		

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 **Ringing 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.

Notification Popups

Notification popups feature allows the IP phone to display the pop-up message when it misses a call, forwards an incoming call to other party or receives a new voice mail or a new text message.

Note

Notification popups feature is applicable to IP phones running firmware version 73 or later.

Procedure

Notification popups can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure notification popups. Parameters: features.voice_mail_popup.enable features.missed_call_popup.enable features.forward_call_popup.enable features.text_message_popup.enable
Local	Web User Interface	Configure notification popups. Navigate to: http://<phoneIPAddress>/servlet?p=features-notifypop&q=load

Details of Configuration Parameters:

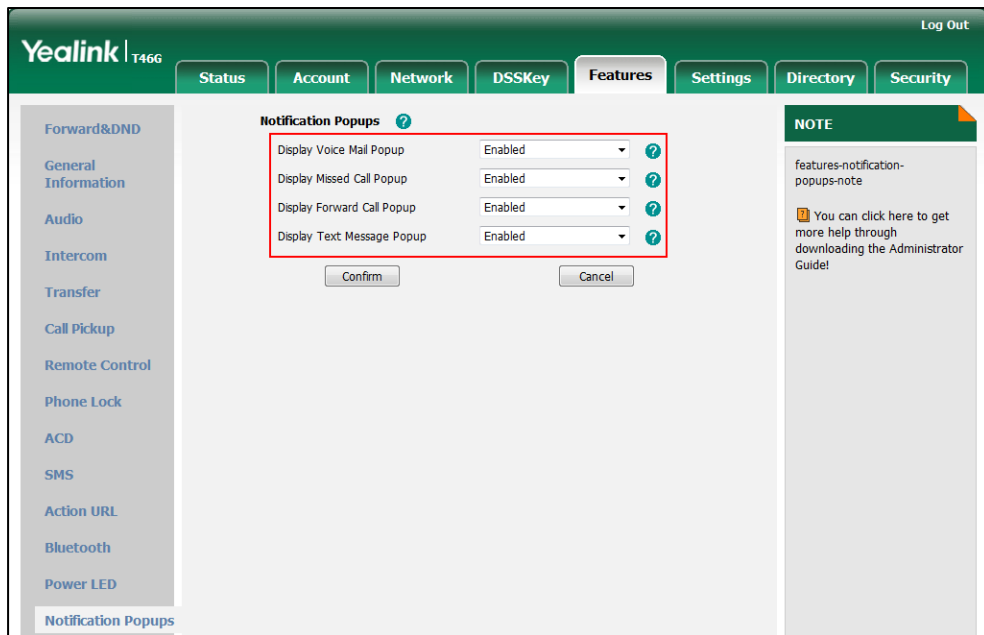
Parameters	Permitted Values	Default
features.voice_mail_popup.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to display the pop-up message box when it receives a new voice mail.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If the voice mail pop-up message box disappears, it won't pop up again unless the user receives a new voice mail or the user re-registers the account that has unread voice mail(s). It is only applicable to IP phones running firmware version 73 or later.</p> <p>Web User Interface: Features->Notification Popups->Display Voice Mail Popup</p> <p>Phone User Interface: None</p>		
features.missed_call_popup.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to display the pop-up message box when it misses a call.</p> <p>0-Disabled 1-Enabled</p>		

Parameters	Permitted Values	Default
<p>Note: It is only applicable to IP phones running firmware version 73 or later.</p> <p>Web User Interface: Features->Notification Popups->Display Missed Call Popup</p> <p>Phone User Interface: None</p>		
features.forward_call_popup.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to display the pop-up message box when it forwards an incoming call to other party.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It is only applicable to IP phones running firmware version 73 or later.</p> <p>Web User Interface: Features->Notification Popups->Display Forward Call Popup</p> <p>Phone User Interface: None</p>		
features.text_message_popup.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to display the pop-up message box when it receives a new text message.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It is only applicable to SIP-T46G IP phones running firmware version 73 or later.</p> <p>Web User Interface: Features->Notification Popups->Display Text Message Popup</p> <p>Phone User Interface: None</p>		

To configure the notification popups via web user interface:

1. Click on **Features->Notification Popups**.
2. Select the desired value from the pull-down list of **Display Voice Mail Popup**.
3. Select the desired value from the pull-down list of **Display Missed Call Popup**.
4. Select the desired value from the pull-down list of **Display Forward Call Popup**.

- Select the desired value from the pull-down list of **Display Text Message Popup**.



- 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. You can only configure the LCD's contrast of the expansion module EXP40 connected to SIP-T48G and T46G IP phones. Make sure the expansion module has been connected to the IP phone before adjustment.

Procedure



Contrast can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configures the LCD's contrast of EXP40 connected to SIP-T48G/T46G IP phones. Parameter: phone_setting.contrast
Local	Phone User Interface	Configures the LCD's contrast of EXP40 connected to SIP-T48G/T46G IP phones.

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.contrast	Integer from 1 to 10	6
<p>Description: Configures the LCD's contrast of EXP40 connected to SIP-T48G/T46G IP phones.</p> <p>Note: We recommend that you set the contrast of the LCD screen to 6 as a more comfortable level.</p> <p>Web User Interface: None</p> <p>Phone User Interface: Menu->Basic->Display->Contrast</p>		

To configure contrast via phone user interface (only applicable to EXP40 connected to SIP-T48G/T46G IP phones):

1. Press **Menu->Basic->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.

Wallpaper

Wallpaper is an image used as the background of the IP phone idle screen. Users can select an image from IP phone's built-in background or customize wallpaper from personal pictures. To set the custom wallpaper as the IP phone background, you need to upload the custom wallpaper to the IP phone in advance. The wallpaper is not applicable to SIP-T42G and SIP-T41P IP phones.

The wallpaper image format must meet the following:

Phone Model	Format	Resolution	Single File Size	Total File Size
SIP-T46G	.jpg/.png/.bmp	<=480*272	<=5MB	<=20MB
SIP-T48G	.jpg/.png/.bmp	<=800*480	<=5MB	<=20MB

Procedure

Wallpaper can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configures the wallpaper displayed on the IP phone. Parameter: phone_setting.backgrounds Specify the access URL of the custom wallpaper. Parameter: wallpaper_upload.url
Local	Web User Interface	Upload the custom wallpaper. Change the wallpaper via web user interface. Navigate to: http://<phoneIPAddress>/servlet?p=settings-preference&q=load
	Phone User Interface	Change the wallpaper via phone user interface.

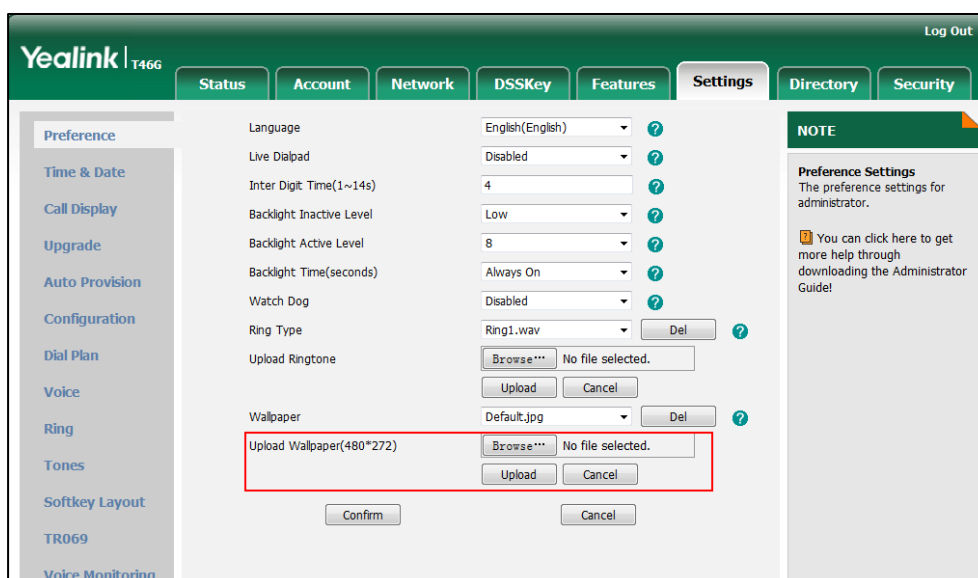
Details of the Configuration Parameter:

Parameters	Permitted Values	Default
phone_setting.backgrounds	Refer to the following content	Resource: Default.png
<p>Description: Configures the wallpaper displayed on the IP phone.</p> <p>Example: To set a phone built-in picture (e.g., 1.png) to be wallpaper, the value format is: phone_setting.backgrounds = Resource:1.png To configure a custom picture (e.g., custom1.png) to be wallpaper, the value format is: Config:custom1.png</p> <p>Permitted Values: Resource:X (Valid values of X are: Default.png,1.png,2.png,3.png,4.png,5.png, 6.png, 7.png,8.png or 9.png) or Config:wallpaper name</p> <p>Note: It is only applicable to SIP-T48G/T46G IP phones.</p>		

Parameters	Permitted Values	Default
Web User Interface: Settings->Preference->Wallpaper Phone User Interface: Menu->Basic->Display->Wallpaper		
wallpaper_upload.url	URL within 511 characters	Blank
Description: Configures the access URL of the wallpaper image. Example: wallpaper_upload.url = http://192.168.10.25/wallpaper.jpg Note: It is only applicable to SIP-T48G/T46G IP phones. Web User Interface: Settings->Preference->Upload Wallpaper (480*272) Phone User Interface: None		

To upload custom wallpaper via web user interface:

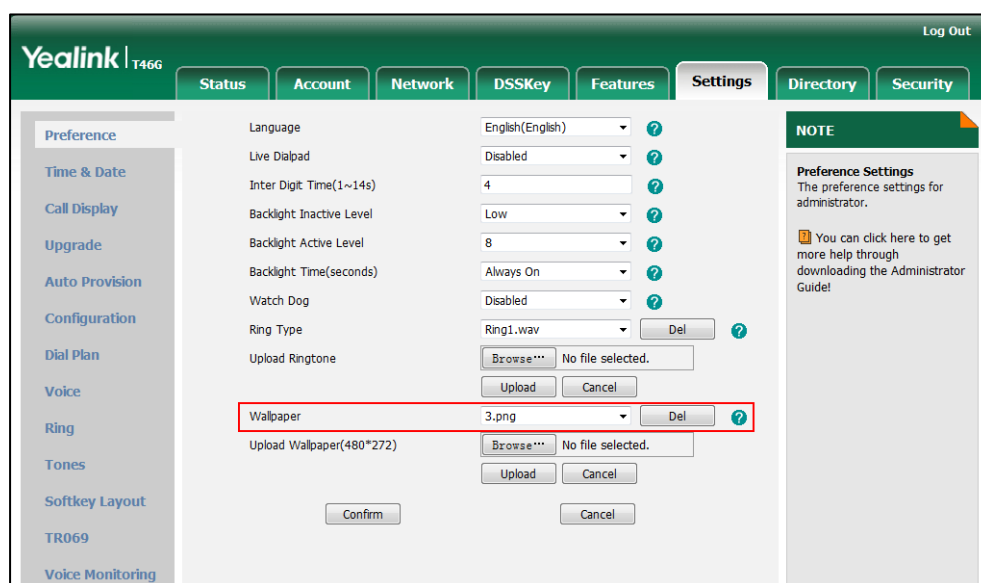
1. Click on **Settings->Preference**.
2. In the **Upload Wallpaper** field, click **Browse** to locate the wallpaper image from your local system.
3. Click **Upload** to upload the file.



4. Click **Confirm** to accept the change.
 The custom wallpaper appears in the pull-down list of **Wallpaper**.

To change the wallpaper via web user interface:

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



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

To change the wallpaper via phone user interface:

1. Press **Menu->Basic->Display->Wallpaper**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired wallpaper.
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 change the backlight when the IP phone is inactive. Backlight turns off quickly if a short backlight time is configured, this may not give users enough time to read messages. Backlight time is applicable to SIP-T4X IP phones and EXP40 connected to SIP-T48G/T46G IP phones.

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

- **Always On:** Backlight is turned on permanently.
- **15s, 30s, 60s, 120s, 300s, 600s or 1800s:** Backlight is turned off or turned dusky when the IP phone is inactive after a preset period of time. It is automatically turned on if the status of the IP phone changes or any key is pressed.

Backlight Active Level is used to adjust the backlight intensity of the LCD screen when the phone is active. Backlight Inactive Level is used to adjust the backlight intensity of the LCD screen when the phone is inactive. Backlight Active Level is only applicable to

SIP-T48G and SIP-T46G IP phones and the connected EXP40. Backlight Inactive Level is only applicable to SIP-T48G and SIP-T46G IP phones.

Procedure

Backlight can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cf g	Configure the backlight of the LCD screen. Parameters: phone_setting.active_backlight_level phone_setting.inactive_backlight_level phone_setting.backlight_time
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.

Details of Configuration Parameters:

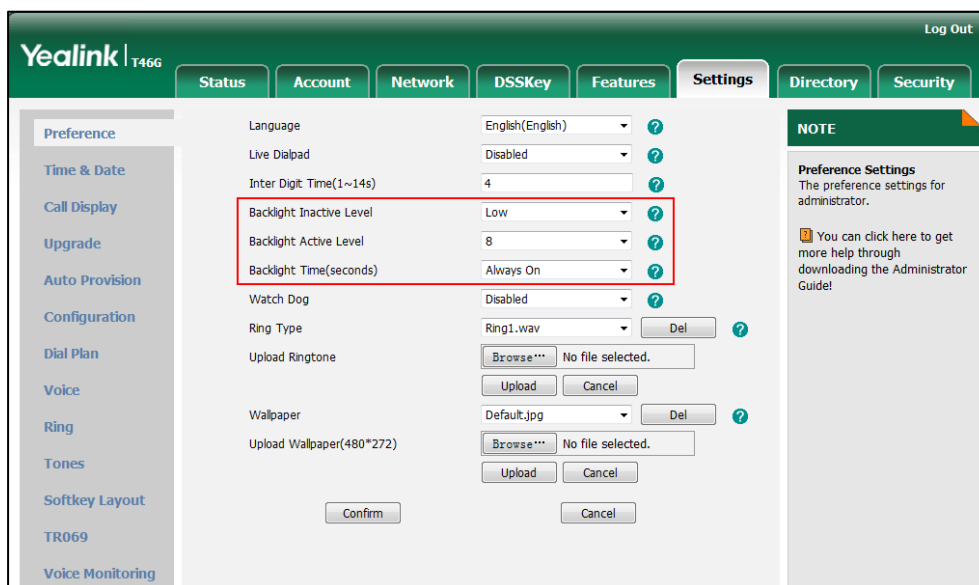
Parameters	Permitted Values	Default
phone_setting.active_backlight_level	Integer from 1 to 10	8
<p>Description: Configures the intensity of the LCD screen when the phone is active. 10 is the highest intensity.</p> <p>Note: It is only applicable to SIP-T48G/T46G IP phones and the connected EXP40.</p> <p>Web User Interface: Settings->Preference->Backlight Active Level</p> <p>Phone User Interface: Menu->Basic->Display->Backlight->Backlight Active Level</p>		
phone_setting.inactive_backlight_level	0 or 1	1
<p>Description: Configures the intensity of the LCD screen when the phone is inactive. 0-Off</p>		

Parameters	Permitted Values	Default
<p>1-Low</p> <p>Note: It is only applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: Settings->Preference->Backlight Inactive Level</p> <p>Phone User Interface: Menu->Basic->Display->Backlight->Backlight Inactive Level</p>		
phone_setting.backlight_time	0, 15, 30, 60, 120, 300, 600 or 1800	0
<p>Description: Configures the delay time (in seconds) to change the intensity of the LCD screen when the IP phone is inactive.</p> <p>0-Always on 15-15s 30-30s 60-60s 120-120s 300-300s 600-600s 1800-1800s</p> <p>If it is set to 60 (60s), the intensity of the LCD screen will be changed when the IP phone is inactive for 60 seconds.</p> <p>Web User Interface: Settings->Preference->Backlight Time(seconds)</p> <p>Phone User Interface: Menu->Basic->Display->Backlight->Backlight Time</p>		

To configure the backlight via web user interface:

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

- Select the desired value from the pull-down list of **Backlight Time (seconds)**.



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

To configure the backlight via phone user interface:

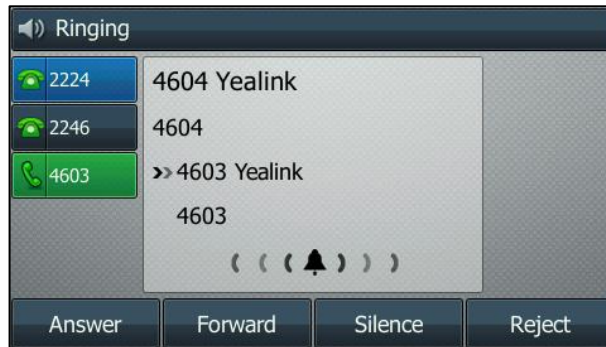
- Press **Menu->Basic->Display->Backlight**.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired level from the **Backlight Active Level** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Backlight Inactive Level** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired time from the **Backlight Time** field.
- Press the **Save** soft key to accept the change.

Call Display

Display contact photo allows the IP phone to present the contact avatar when it receives an incoming call, dials an outgoing call or engages in a call. Display contact photo feature is only applicable to SIP-T48G/SIP-T46G IP phones.

Display Called Party Information allows the IP phone to present the callee identity in addition to the presentation of caller identity when it receives an incoming call.

The following figure shows an incoming call from Yealink 4604 to Yealink 4603 on the SIP-T46G IP phone.



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

Note SIP-T42G/T41P IP phones have a limited display (up to three lines) due to their smaller screen size.

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure display contact photo feature.</p> <p>Parameter: phone_setting.contact_photo_display.enable</p> <p>Configure display called party information feature.</p> <p>Parameter: phone_setting.called_party_info_display.enable</p> <p>Configure the caller information display method.</p> <p>Parameter: phone_setting.call_info_display_method</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure call display feature.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=settings-calldisplay&q=lo</p>

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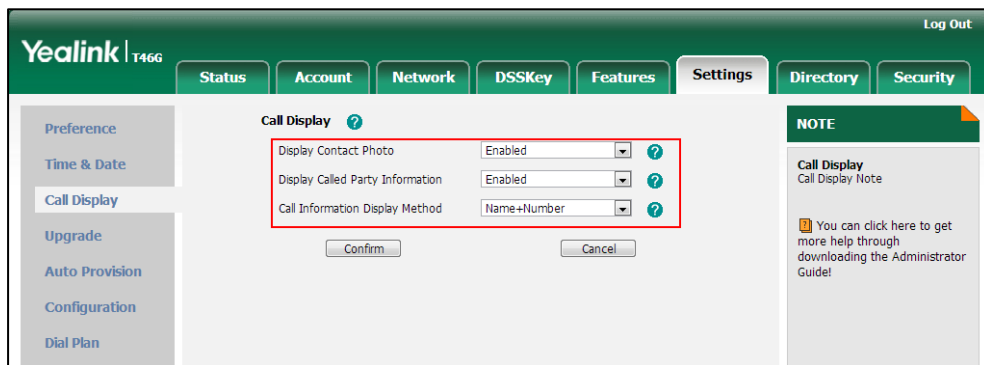
Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.contact_photo_display.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to display contact avatar when it receives an incoming call, dials an outgoing call or engages in a call.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It is only applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: Settings->Call Display->Display Contact Photo</p> <p>Phone User Interface: None</p>		
phone_setting.called_party_info_display.enable	0 or 1	0
<p>Description: Enables or disables the IP phone to present the callee identity when it receives an incoming call.</p> <p>Web User Interface: Settings->Call Display->Display Called Party Information</p> <p>Phone User Interface: None</p>		
phone_setting.call_info_display_method	Integer from 0 to 4	0
<p>Description: Configures the call information display method when the IP phone receives an incoming call, dials an outgoing call or engages in a call.</p> <p>0-Name+Number 1-Number+Name 2-Number 3-Name 4-Full Contact Info (display name<sip:xxx@domain.com>)</p>		

Parameters	Permitted Values	Default
Web User Interface: Settings->Call Display->Call Information Display Method Phone User Interface: None		

To configure call display features via web user interface:

1. Click on **Settings->Call Display**.
2. Select the desired value from the pull-down list of **Display Contact Photo**.
3. Select the desired value from the pull-down list of **Display Called Party Information**.
4. Select the desired value from the pull-down list of **Call Information Display Method**.



5. Click **Confirm** 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 the pages returned by the web server. Both the HTTP and HTTPS port numbers are configurable.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Specify the web access type, HTTP port and HTTPS port. Parameters: wui.http_enable network.port.http wui.https_enable
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		network.port.https
Local	Web User Interface	Specify the web access type, HTTP port and HTTPS port. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Specify the web access type, HTTP port and HTTPS port.

Details of Configuration Parameters:

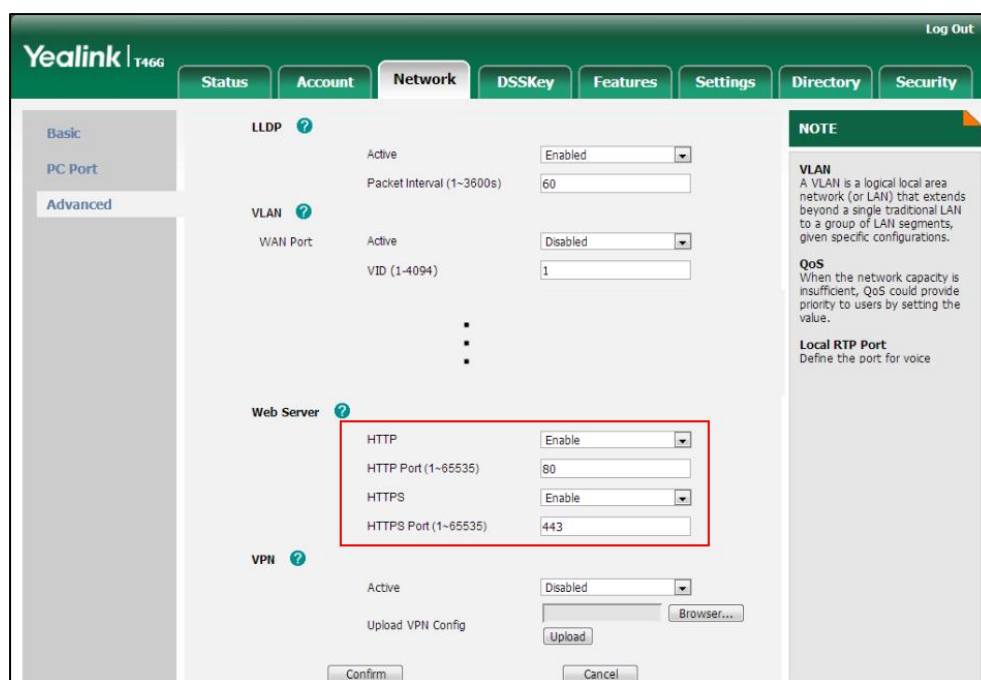
Parameters	Permitted Values	Default
wui.http_enable	0 or 1	1
<p>Description: Enables or disables the user to access web user interface of the IP phone using HTTP protocol. 0-Disabled 1-Enabled Note: If you change this parameter, the IP phone will reboot to make the change take effect. Web User Interface: Network->Advanced->Web Server->HTTP Phone User Interface: Menu->Advanced (default password: admin)->Network->Webserver Type ->HTTP Status</p>		
network.port.http	Integer from 1 to 65535	80
<p>Description: Configures the HTTP port for the user to access web user interface of the IP phone using the HTTP protocol. The default HTTP port is 80. Note: If you change this parameter, the IP phone will reboot to make the change take effect. Web User Interface: Network->Advanced->Web Server->HTTP Port (1~65535) Phone User Interface: Menu->Advanced (default password: admin)->Network->Webserver Type ->HTTP</p>		

Parameters	Permitted Values	Default
Port		
wui.https_enable	0 or 1	1
<p>Description: Enables or disables the user to access web user interface of the IP phone using HTTPS protocol.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->Web Server->HTTPS</p> <p>Phone User Interface: Menu->Advanced (default password: admin)->Network->Webserver Type ->HTTPS Status</p>		
network.port.https	Integer from 1 to 65535	443
<p>Description: Configures the HTTPS port for the user to access web user interface of the IP phone using the HTTPS protocol.</p> <p>The default HTTPS port is 443.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->Web Server->HTTPS Port (1~65535)</p> <p>Phone User Interface: Menu->Advanced (default password: admin)->Network->Webserver Type ->HTTPS Port</p>		

To configure the web server type via web user interface:

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

The default HTTPS port is 443.



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

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

7. Click **OK** to reboot the phone.

To configure the web server type via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->Webserver Type**.
2. Press **←** or **→** , or the **Switch** soft key to select the desired value in the **HTTP Status** field.
3. Enter the HTTP port in the **HTTP Port** field.
4. Press **←** or **→** , or the **Switch** soft key to select the desired value in the **HTTPS Status** field.
5. Enter the HTTPS port in the **HTTPS Port** field.
6. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

User Password

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

A user or an administrator can change the user password. The default user password is "user". For security reasons, the user or the 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. Parameter: security.user_password
Local	Web User Interface	Change the user password of the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=security&q=load

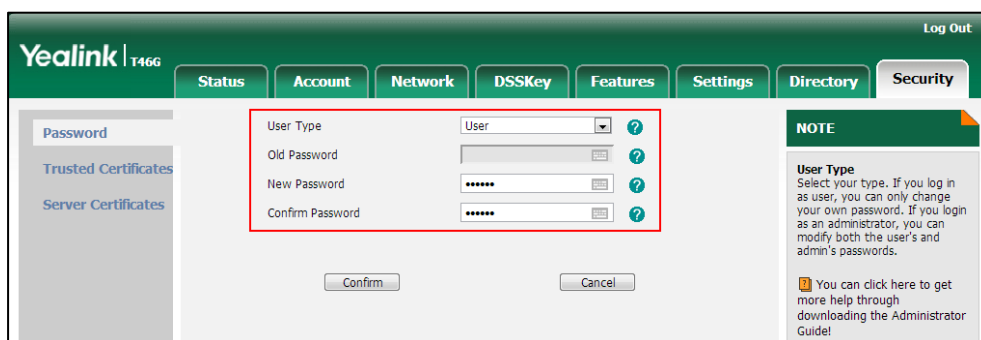
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
security.user_password	String within 32 characters	user
<p>Description: Configures the password of the user for web server access. The IP phone uses "user" as the default user password. The valid value format is username:new password.</p> <p>Example: security.user_password = user:password123 means setting the password of user (current user name is "user") to password123.</p> <p>Note: IP phones support ASCII characters 32-126(0x20-0x7E) in passwords. You can set the password to be empty via web user interface only.</p> <p>Web User Interface: Security->Password</p> <p>Phone User Interface: None</p>		

To change the user password via web user interface:

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

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



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

Note If logging into the web user interface of the IP phone with the user credential, the user needs to enter the current 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. Parameter: security.user_password
Local	Web User Interface	Change the administrator password. Navigate to: http://<phoneIPAddress>/servlet? p=security&q=load
	Phone User Interface	Change the administrator password.

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
security.user_password	String within 32 characters	admin

Description:
 Configures the password of the administrator for web server access.
 The IP phone uses “admin” as the default administrator password.

Example:
 security.user_password = admin:password123 means setting the password of administrator (current user name is “admin”) to password123.

Note: IP phones support ASCII characters 32-126(0x20-0x7E) in passwords. You can set the password to be empty via web user interface only.

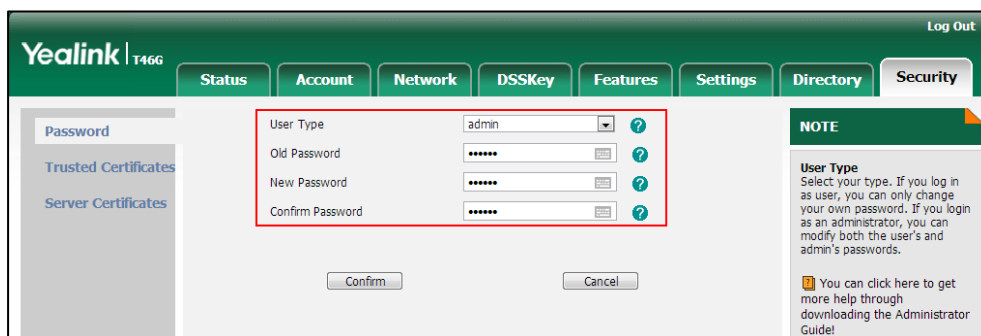
Web User Interface:
 Security->Password

Phone User Interface:
 Menu->Advanced (default password: admin) ->Set 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 a new administrator password in the **New Password** and **Confirm Password** fields.

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



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

To change the administrator password via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Set Password**.
2. Enter the current administrator password in the **Current PWD** field.
3. Enter a new administrator password in the **New PWD** field and **Confirm PWD** field.

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

Phone Lock

Phone lock is used to lock the IP phone to prevent it from unauthorized use. Phone lock is not applicable to SIP-T48G IP phones. Once the IP phone is locked, a user must enter the password to unlock it. IP phones offer three types of phone lock: Menu Key, Function Keys and All Keys. The IP phone will not be locked immediately after the IP phone lock type is configured. One of the following steps is also needed:

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

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

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the IP phone lock type. Parameters: phone_setting.phone_lock.enable phone_setting.phone_lock.lock_key_type Change the unlock password. Parameter: phone_setting.phone_lock.unlock_pin Configure the IP phone to automatically lock the phone after a time interval. Parameter: phone_setting.phone_lock.lock_time_out Assign a phone lock key. Parameter: linekey.X.type/programablekey.X.type
Local	Web User Interface	Configure the phone lock type. Change the unlock PIN. Configure the IP phone to automatically lock the phone after a time interval. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-phonelock&q=load">http://<phoneIPAddress>/servlet?p=features-phonelock&q=load Assign a phone lock key.

		<p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p>
	Phone User Interface	<p>Configure the phone lock type. Change the unlock PIN. Assign a phone lock key.</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.phone_lock.enable	0 or 1	0
<p>Description: Enables or disables phone lock feature. 0-Disabled 1-Enabled Note: It is not applicable to SIP-T48G IP phones. Web User Interface: Features->Phone Lock->Phone Lock Enable Phone User Interface: Menu->Advanced (default password: admin) ->Phone Settings->Phone Lock->Lock Enable</p>		
phone_setting.phone_lock.lock_key_type	0, 1 or 2	0
<p>Description: Configures the type of phone lock. Menu Key: The Menu soft key is locked. Function Keys: MESSAGE, REDIAL, HOLD, MUTE, TRANSFER, OK, X, navigation keys, soft keys and line keys are locked. (For SIP-T42G/T41P, HOLD key and TRANSFER key do not exist). All Keys: All keys are locked, except the Volume, Headset, Speakerphone and digit keys. 0-All Keys 1-Function Keys 2-Menu Key Note: It is not applicable to SIP-T48G IP phones. Web User Interface:</p>		

Parameters	Permitted Values	Default
Features->Phone Lock->Phone Lock Type Phone User Interface: Menu->Advanced (default password: admin) ->Phone Settings->Phone Lock->Lock Type		
phone_setting.phone_lock.unlock_pin	characters within 15 digits	123
Description: Configures the password for unlocking the phone. Note: It is not applicable to SIP-T48G IP phones. Web User Interface: Features->Phone Lock->Phone Unlock PIN(0~15 Digit) Phone User Interface: Menu->Basic->Change PIN		
phone_setting.phone_lock.lock_time_out	Integer from 0 to 3600	0
Description: Configures the interval (in seconds) to automatically lock the phone. The default value is 0 (the phone is locked only by long pressing the pound key or pressing the phone lock key). Note: It works only if the type of phone lock is preset. It is not applicable to SIP-T48G IP phones. Web User Interface: Features->Phone Lock->Phone Lock Time Out(0~3600s) Phone User Interface: None		

Phone Lock Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

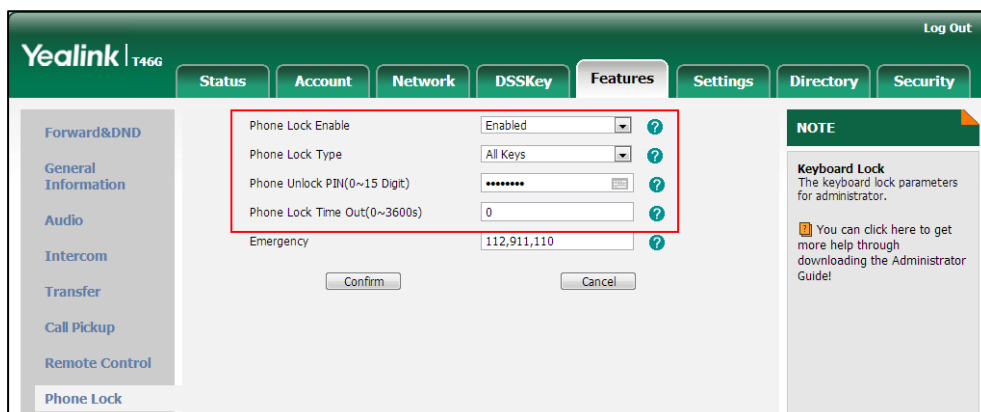
Parameter	Permitted Values	Default
linekey.X.type/programablekey.X.type	50	Refer to the following content

Parameter	Permitted Values	Default
<p>Description:</p> <p>Configures a DSS key as a phone lock key on the IP phone.</p> <p>The digit 50 stands for the key type Phone Lock.</p> <p>For line keys:</p> <p>X ranges from 1 to 27 (for SIP-T46G).</p> <p>X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys:</p> <p>X=1-10, 12-14 (for SIP-T46G)</p> <p>X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example:</p> <p>linekey.2.type = 50</p> <p>Default:</p> <p>For line keys:</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIP-T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p>		

Parameter	Permitted Values	Default
<p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>Note: It is not applicable to SIP-T48G IP phones.</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X/Programable Key->Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Type</p>		

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 **Phone Lock Enable**.
3. Select the desired type from the pull-down list of **Phone Lock Type**.
4. Enter unlock password (numeric characters) in the **Phone Unlock PIN (0~15 Digit)** field.
5. Enter the desired time in the **Phone Lock Time Out (0~3600s)** field.



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

To configure a phone lock key via web user interface:

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

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	4607	Line 1	
Line Key2	Phone Lock			N/A	
Line Key3	Line	Default		Line 3	
Line Key4	Line	Default		Line 4	
Line Key5	Line	Default		Line 5	
Line Key6	Line	Default		Line 6	
Line Key7	Line	Default		Line 7	
Line Key8	Line	Default		Line 8	
Line Key9	Line	Default		Line 9	

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

To configure phone lock type via phone user interface:

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

To change the unlock PIN via phone user interface:

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

To configure a phone lock key via phone user interface:

- Press **Menu->Features->DSS 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 **Phone Lock** from the **Key Type** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.

6. 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 the IP phone. 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 IP phones 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 methods for configuring time and date:

Option	Methods of Configuration
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

Option	Methods of Configuration
Daylight Saving Time	Configuration Files Web User Interface

Procedure

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

Configuration File	<MAC>.cfg	<p>Configure NTP by DHCP priority feature and DHCP time features.</p> <p>Parameters: local_time.manual_ntp_srv_prior local_time.dhcp_time</p> <p>Configure the NTP server, time zone and DST.</p> <p>Parameters: local_time.ntp_server1 local_time.ntp_server2 local_time.interval local_time.time_zone local_time.time_zone_name local_time.summer_time local_time.dst_time_type local_time.start_time local_time.end_time local_time.offset_time</p> <p>Configure the time and date manually.</p> <p>Parameter: local_time.manual_time_enable</p> <p>Configure the time and date formats.</p> <p>Parameters: local_time.time_format local_time.date_format</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>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
local_time.manual_ntp_srv_prior	0 or 1	0
<p>Description: Enables or disables the IP phone to use manually configured NTP server preferentially.</p> <p>0-High (use the NTP server obtained by DHCP preferentially) 1-Low (use the NTP server configured manually preferentially)</p> <p>Web User Interface: Settings->Time & Date->NTP By DHCP Priority</p> <p>Phone User Interface: None</p>		
local_time.dhcp_time	0 or 1	0
<p>Description: Enables or disables the IP phone to update time with the offset time obtained from the DHCP server.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It is only available to offset from GMT 0.</p> <p>Web User Interface: Settings->Time & Date->DHCP Time</p>		

Parameters	Permitted Values	Default
Phone User Interface: Menu->Basic->Time & Date->DHCP Time		
<code>local_time.ntp_server1</code>	IP Address or Domain Name	cn.pool.ntp.org
Description: Configures the IP address or the domain name of the NTP server 1. Example: <code>local_time.ntp_server1 = 192.168.0.5</code> Web User Interface: Settings->Time & Date->Primary Server Phone User Interface: Menu->Basic->Time & Date->General->SNTP Setting->NTP Server 1.		
<code>local_time.ntp_server2</code>	IP Address or Domain Name	cn.pool.ntp.org
Description: Configures the IP address or the domain name of the NTP server 2. If the NTP server 1 is not configured or cannot be accessed, the IP phone will request the time and date from the NTP server 2. Example: <code>local_time.ntp_server2 = 192.168.0.6</code> Web User Interface: Settings->Time & Date->Secondary Server Phone User Interface: Menu->Basic->Time & Date->General->SNTP Setting->NTP Server 2.		
<code>local_time.interval</code>	Integer from 15 to 86400	1000
Description: Configures the interval (in seconds) to update time and date from the NTP server. Example: <code>local_time.interval = 1000</code> Web User Interface: Settings->Time & Date->Synchronism (15~86400s) Phone User Interface: None		

Parameters	Permitted Values	Default
local_time.time_zone	-11 to +14	+8
<p>Description: Configures the time zone. For more available time zones, refer to Appendix B: Time Zones on page 511.</p> <p>Example: local_time.time_zone = +8</p> <p>Web User Interface: Settings->Time & Date->Time Zone</p> <p>Phone User Interface: Menu->Basic->Time & Date->General->SNTP Setting->Time Zone</p>		
local_time.time_zone_name	String within 32 characters	China(Beijing)
<p>Description: Configures the time zone name. The available time zone names depend on the time zone configured by the parameter "local_time.time_zone". For more information on the available time zone names for each time zone, refer to Appendix B: Time Zones on page 511.</p> <p>Note: It works only if the value of the parameter "local_time.summer_time" is set to 2 (Automatic).</p> <p>Example: local_time.time_zone_name = China(Beijing)</p> <p>Web User Interface: Settings->Time & Date->Time Zone</p> <p>Phone User Interface: Menu->Basic->Time & Date->General->SNTP Setting->Time Zone</p>		
local_time.summer_time	0, 1 or 2	2
<p>Description: Configures Daylight Saving Time (DST) feature.</p> <p>0-Disabled 1-Enabled 2-Automatic</p> <p>Web User Interface: Settings->Time & Date->Daylight Saving Time</p>		

Parameters	Permitted Values	Default
Phone User Interface: Menu->Basic->Time & Date->General->SNTP Setting->Daylight Saving		
local_time.dst_time_type	0 or 1	0
<p>Description: Configures the DST time type.</p> <p>0-By Date 1-By Week</p> <p>Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p>Web User Interface: Settings->Time & Date->Fixed Type</p> <p>Phone User Interface: None</p>		
local_time.start_time	Time	1/1/0
<p>Description: Configures the start time of the DST.</p> <p>Value formats are:</p> <ul style="list-style-type: none"> Month/Day/Hour (for By Date) Month/ Day of Week Last in Month/ Day of Week/ Hour of Day (for By Week) <p>If "local_time.dst_time_type" is set to 0 (By Date), use the mapping: Month: 1=Jan, 2=Feb,..., 12=Dec Day:1=the first day in a month,..., 31= the last day in a month Hour:0=0am, 1=1am,..., 23=23pm</p> <p>If "local_time.dst_time_type" is set to 1 (By Week), use the mapping: Month: 1=Jan, 2=Feb,..., 12=Dec Day of Week Last in 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=0am, 1=1am,..., 11=11pm</p> <p>Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p>Web User Interface: For DST By Date:</p>		

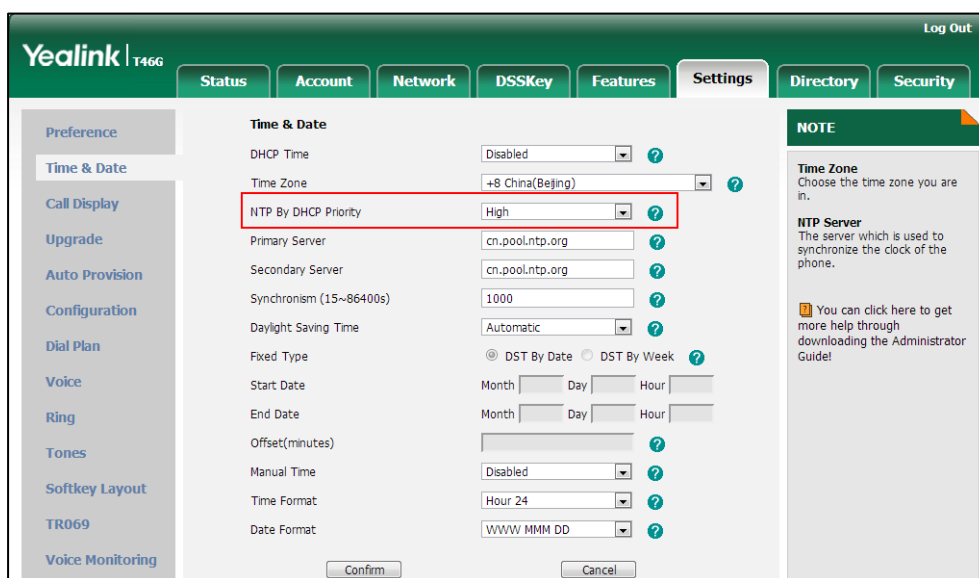
Parameters	Permitted Values	Default
<p>Settings->Time & Date->Start Date</p> <p>For DST By Week:</p> <p>Settings->Time & Date->DST Start Month/DST Start Day of Week/DST Start Day of Week Last in Month/ Start Hour of Day</p> <p>Phone User Interface:</p> <p>None</p>		
local_time.end_time	Time	12/31/23
<p>Description:</p> <p>Configures the end time of the DST.</p> <p>Value formats are:</p> <ul style="list-style-type: none"> • Month/Day/Hour (for By Date) • Month/ Day of Week Last in Month/ Day of Week/ Hour of Day (for By Week) <p>If "local_time.dst_time_type" is set to 0 (By Date), use the mapping:</p> <p>Month: 1=Jan, 2=Feb,..., 12=Dec</p> <p>Day:1=the first day in a month,..., 31= the last day in a month</p> <p>Hour:0=0am, 1=1am,..., 23=11pm</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>Day of Week Last in 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=0am, 1=1am,..., 23=11pm</p> <p>Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p>Web User Interface:</p> <p>For DST By Date:</p> <p>Settings->Time & Date->End Date</p> <p>For DST By Week:</p> <p>Settings ->Time & Date->DST Stop Month/DST Stop Day of Week/DST Stop Day of Week Last in Month/Stop Hour of Day</p>		
local_time.offset_time	Integer from -300 to 300	Blank
<p>Description:</p> <p>Configures the offset time (in minutes) of DST.</p>		

Parameters	Permitted Values	Default
<p>Note: It works only if the value of the parameter "local_time.summer_time" is set to 1 (Enabled).</p> <p>Web User Interface: Settings->Time & Date->Offset(minutes)</p> <p>Phone User Interface: None</p>		
local_time.manual_time_enable	0 or 1	0
<p>Description: Configures the IP phone to obtain time from the NTP server or manual settings.</p> <p>0-NTP 1-Manual</p> <p>Web User Interface: Settings->Time & Date->Manual Time</p> <p>Phone User Interface: Menu->Basic->Time & Date->General</p>		
local_time.time_format	0 or 1	1
<p>Description: Configures the time format.</p> <p>0-12 Hour 1-24 Hour</p> <p>If it is set to 0 (12 Hour), the time will be displayed in 12-hour format with AM or PM specified.</p> <p>If it is set to 1 (24 Hour), the time will be displayed in 24-hour format (eg., 2:00 PM displays as 14:00).</p> <p>Web User Interface: Settings->Time & Date->Time Format</p> <p>Phone User Interface: Menu->Basic->Time & Date->Time & Date Format->Time Format</p>		
local_time.date_format	0, 1, 2, 3, 4, 5 or 6	0

Parameters	Permitted Values	Default
<p>Description: Configures the date format.</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>Web User Interface: Settings->Time & Date->Date Format</p> <p>Phone User Interface: Menu->Basic->Time & Date->Time & Date Format->Date Format</p>		

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

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



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

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

1. Click on **Settings->Time & Date**.
2. Select **Disabled** from the pull-down list of **Manual Time**.
3. Select the desired time zone from the pull-down list of **Time Zone**.

4. Enter the domain names or IP addresses in the **Primary Server** and **Secondary Server** fields respectively.
5. Enter the desired time interval in the **Synchronism (15~86400s)** field.
6. 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.

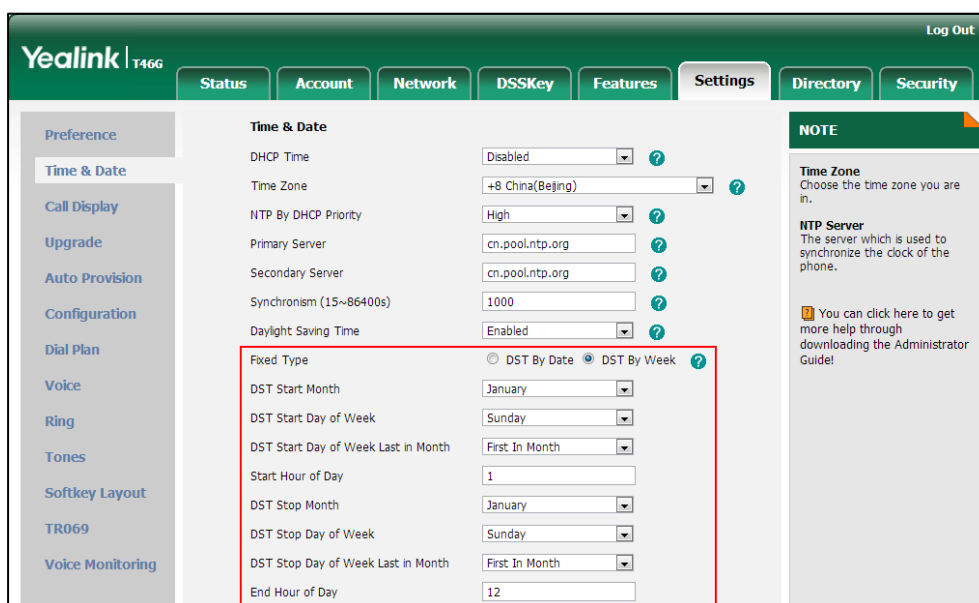
The screenshot shows the 'Time & Date' configuration page in the Yealink T466 web interface. The 'Fixed Type' section is highlighted with a red box, showing 'DST By Date' selected. The 'Start Date' field is set to Month 1, Day 1, Hour 1, and the 'End Date' field is set to Month 12, Day 22, Hour 3. Other fields include DHCP Time (Disabled), Time Zone (+8 China(Beijing)), NTP By DHCP Priority (High), Primary Server (cn.pool.ntp.org), Secondary Server (cn.pool.ntp.org), Synchronism (15~86400s) (1000), Daylight Saving Time (Enabled), Offset (minutes), Manual Time (Disabled), Time Format (Hour 24), and Date Format (WWW MMM DD). A 'NOTE' section on the right provides information about Time Zone and NTP Server.

- 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.

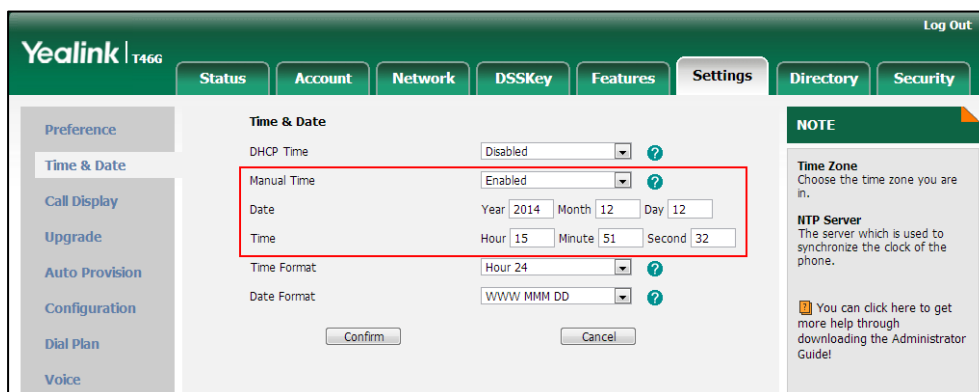
Enter the desired time in the **End Hour of Day** field.



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

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

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



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

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

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

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

The screenshot shows the 'Time & Date' configuration page in the Yealink T466 web interface. The 'Date Format' field is highlighted with a red box, showing 'WWW MMM DD' selected. Other fields include Time Zone (+8 China(Beijing)), NTP By DHCP Priority (High), Primary and Secondary Servers (cn.pool.ntp.org), Synchronism (15~86400s) (1000), Daylight Saving Time (Enabled), Fixed Type (DST By Date), Start and End Dates, Offset (minutes), and Manual Time (Disabled). The 'Time Format' field is also visible, showing 'Hour 24' selected.

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

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

1. Press **Menu->Basic->Time & Date->General->SNTP Setting**.
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 Server 1** and **NTP Server 2** fields respectively.
4. Press **←** or **→** or the **Switch** soft key to select **Automatic** from the **Daylight Saving** field.
5. Press the **Save** soft key to accept the change.

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

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

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

1. Press **Menu->Basic->Time & Date->Time & Date Format**.
2. Press **←** or **→**, or the **Switch** soft key to select the desired date format from the **Date Format** field.
3. Press **←** or **→**, or the **Switch** soft key to select the desired time format (12 Hour or 24 Hour) from the **Time 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 the languages supported by the phone user interface and the web user interface respectively.

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

Loading Language Packs

Languages available for selection depend on language packs currently loaded to the IP phone. You can customize the translation of the existing language for the phone user interface or web user interface. You can also make new languages available for use on the phone user interface and web 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 the associated language packs for the phone user interface:

Available Language	Associated Language Pack for SIP-T48G/T46G/T42G/T41P
English	000.GUI.English.lang
Chinese Simplified	001.GUI.Chinese_S.lang
Chinese Traditional	002.GUI.Chinese_T.lang
French	003.GUI.French.lang
German	004.GUI.German.lang
Italian	005.GUI.Italian.lang

Available Language	Associated Language Pack for SIPT48G/T46G/T42G/T41P
Polish	006.GUI.Polish.lang
Portuguese	007.GUI.Portuguese.lang
Spanish	008.GUI.Spanish.lang
Turkish	009.GUI.Turkish.lang
Russian	010.GUI.Russian.lang

When adding a new language pack for the phone user interface, the language pack must be formatted as “X.GUI.name.lang” (X starts from 011, “name” is replaced with the language name). If the language name is the same as the existing one, the existing language pack will be overridden by the new uploaded one. We recommend that the filename of the new language pack should not be the same as the existing one.

To customize a language file:

1. Open the desired language template file (e.g., 000.GUI.English.lang) using an ASCII editor.
2. Modify the characters within the double quotation marks on the right of the equal sign.

Don't modify the translation item on the left of the equal sign.

The following shows a portion of the language pack "000.GUI.English.lang" for the phone user interface.

```
[Lang]
Name=English
FONT=Tahoma

[Translate]

" s"=" s"
"(Empty)"="(Empty)"
"10min"="10min"
"12 Hour"="12 Hour"
"120s"="120s"
"15s"="15s"
"1800s"="1800s"
"1min"="1min"
"24 Hour"="24 Hour"
"2min"="2min"
"300s"="300s"
"30min"="30min"
"30s"="30s"
"5min"="5min"
"600s"="600s"
"60s"="60s"
"802.1x Mode"="802.1x Mode"
"802.1x"="802.1x"
"ACD Login"="ACD Login"
"ACD State"="ACD State"
"ACD Trace"="Trace"
"ACD"="ACD"
"ALERT"="ALERT"
```

The following table lists available languages, associated language packs and note language packs for the web user interface:

Available Language	Associated Language Pack	Associated Note Language Pack
English	1.English.js	1.English_note.xml
Chinese_S	2.Chinese_S.js	2.Chinese_S_note.xml
Chinese_T	3.Chinese_T.js	3.Chinese_T_note.xml
French	4.French.js	4.French_note.xml
German	5.German.js	5.German_note.xml
Italian	6.Italian.js	6.Italian_note.xml
Polish	7.Polish.js	7.Polish_note.xml
Portuguese	8.Portuguese.js	8.Portuguese_note.xml
Spanish	9.Spanish.js	9.Spanish_note.xml
Turkish	10.Turkish.js	10.Turkish_note.xml
Russian	11.Russian.js	11.Russian_note.xml


When adding a new language pack for the web user interface, the language pack must be formatted as “Y.name.js” (Y starts from 12, “name” is replaced with the language name). If the language name is the same as the existing one, the existing language file will be overridden by the new uploaded one. We recommend that the name of the new language file should not be the same as the existing languages.

To customize a language file:

1. Open the desired language template file (e.g., 1.English.js) using an ASCII editor.
2. Modify the characters within the double quotation marks on the right of the colon.
Don't modify the translation item on the left of the colon.

The following shows a portion of the language pack “1.English.js” for the web user interface:

```
var _objTrans =
{
  " Call Number Filter": "Call Number Filter",
  " Distinctive Ring Tones": "Distinctive Ring Tones",
  " Do you want to reboot ?": "Do you want to reboot?",
  "(800*480)": "(800*480)",
  "0": "0",
  "1": "1",
  "10min": "10min",
  "1min": "1min",
  "2": "2",
  "2min": "2min",
  "3": "3",
  "30min": "30min",
  "4": "4",
  "404 (Not found)": "404 (Not Found)",
  "480 (Temporarily not available)": "480 (Temporarily Not Available)",
  "486 (Busy here)": "486 (Busy Here)",
  "5": "5",
  "5min": "5min",
  "6": "6",
  "603 (Decline)": "603 (Decline)",
  "ACD Auto Available Timer(0~120s)": "ACD Auto Available Timer(0~120s)",
  "ACD Auto Available": "ACD Auto Available",
  "ACD Trace": "ACD Trace",
  "ACD": "ACD",
  "ACD": "ACD",
  "ACD_SubscripPeriod(120~3600s)": "ACD Subscrip Period(120~3600s)",
  "ACS Password": "ACS Password",
```

You can also customize the translation of the note language pack. The note information is integrated in the icon  of the web user interface. The note language pack must be formatted as “Y.name_note.xml” (“Y” and “name” are associated with web language pack).

To customize a note language file:

1. Open the desired note language template file (e.g., 1.English_note.xml) using an ASCII editor.
2. Modify the text of the note field.
Don't modify the name of the note field.

The following shows a portion of the note language pack "1.English_note.xml" for the web user interface:

```
<?xml version="1.0" encoding="utf-8"?>
<notedata>
<status>
  <note name = "version">
    Displays current firmware version and hardware version of the device
  </note>
  <note name = "network">
    Shows details of the phone network configuration
  </note>
  <note name = "network-ipv4">
    Shows details of the phone network configuration
  </note>
  <note name = "network-ipv6">
    Shows details of the phone network configuration
  </note>
  <note name = "network-common">
    Shows details of the phone network configuration
  </note>
  <note name = "AccountStatus">
    According to current state of each account
  </note>
  <note name = "Ext">
    Shows software version and hardware version details of the Expansion LCD Modules
  </note>
</status>
```

Note If the characters in the custom language pack are not supported by the phone, the IP phone will display "?" instead.

The total file sizes of the custom language files must be within 20M (for SIP-T48G/T46G) or 100k (for SIP-T42G/T41P).

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Specify the access URL of the phone user interface language pack.</p> <p>Parameter: gui_lang.url</p> <p>Specify the access URL of the web user interface language pack.</p> <p>Parameter: wui_lang.url</p> <p>Specify the access URL of the note language pack of the web user interface.</p> <p>Parameter: wui_lang_note.url</p> <p>Delete customized language packs of the phone user interface.</p>
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		<p>Parameter: gui_lang.delete</p> <p>Delete customized language packs and note language packs of the web user interface.</p> <p>Parameter: wui_lang.delete</p>
--	--	--

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
gui_lang.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the language pack for the phone user interface.</p> <p>Example: The following example uses HTTP to download the language pack "000.GUI.English.lang" from the provisioning server 192.168.10.25 to the phone user interface. gui_lang.url = http://192.168.10.25/000.GUI.English.lang If you want to download multiple language packs to the SIP-T42G phone simultaneously, you can configure as following: gui_lang.url = http://192.168.10.25/000.GUI.English.lang gui_lang.url = http://192.168.10.25/010.GUI.Russian.lang</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
gui_lang.delete	http://localhost/all or http://localhost/Y.GUI.name.lang	Blank
<p>Description: Deletes customized language packs of the phone user interface.</p> <p>Example: Delete all customized language packs of the phone user interface. gui_lang.delete = http://localhost/all Delete a customized language pack of the phone user interface (e.g.,</p>		

Parameter	Permitted Values	Default
<p>010.GUI.Russian.lang).</p> <p>gui_lang.delete = http://localhost/010.GUI.Russian.lang</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
wui_lang.url	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the access URL of the language pack for the web user interface.</p> <p>Example:</p> <p>The following example uses HTTP to download the language pack "1.English.js" from the provisioning server 192.168.10.25 to the web user interface.</p> <p>wui_lang.url = http://192.168.10.25/1.English.js</p> <p>If you want to download multiple language packs to the phone simultaneously, you can configure as following:</p> <p>wui_lang.url = http://192.168.10.25/1.English.js</p> <p>wui_lang.url = http://192.168.10.25/11.Russian.js</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
wui_lang_note.url	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the access URL of the language pack for web note.</p> <p>Example:</p> <p>The following example uses HTTP to download the language pack "1.English_note.xml" from the provisioning server 192.168.10.25 to the web user interface.</p> <p>wui_lang_note.url = http://192.168.10.25/1.English_note.xml</p> <p>If you want to download multiple language packs to the phone simultaneously, you can configure as following:</p> <p>wui_lang.url = http://192.168.10.25/1.English_note.xml</p> <p>wui_lang.url = http://192.168.10.25/11.Russian_note.xml</p>		

Parameter	Permitted Values	Default
Web User Interface: None Phone User Interface: None		
wui_lang.delete	http://localhost/all or http://localhost/Y.name.js	Blank
Description: Deletes customized language packs and note language packs of the web user interface. Example: Delete all customized language packs and note language packs of the web user interface. web_lang.delete = http://localhost/all Delete a customized language pack (e.g., 11.Russian.js) of the web user interface. gui_lang.delete = http://localhost/11.Russian.js The corresponding note language pack (e.g., 11.Russian_note.xml) will also be deleted. Web User Interface: None Phone User Interface: None		

Specifying the Language to Use

The default language used on the phone user interface is English. You can specify the languages for the phone user interface and web user interface respectively.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Specify the languages for the phone user interface and the web user interface. Parameters: lang.gui
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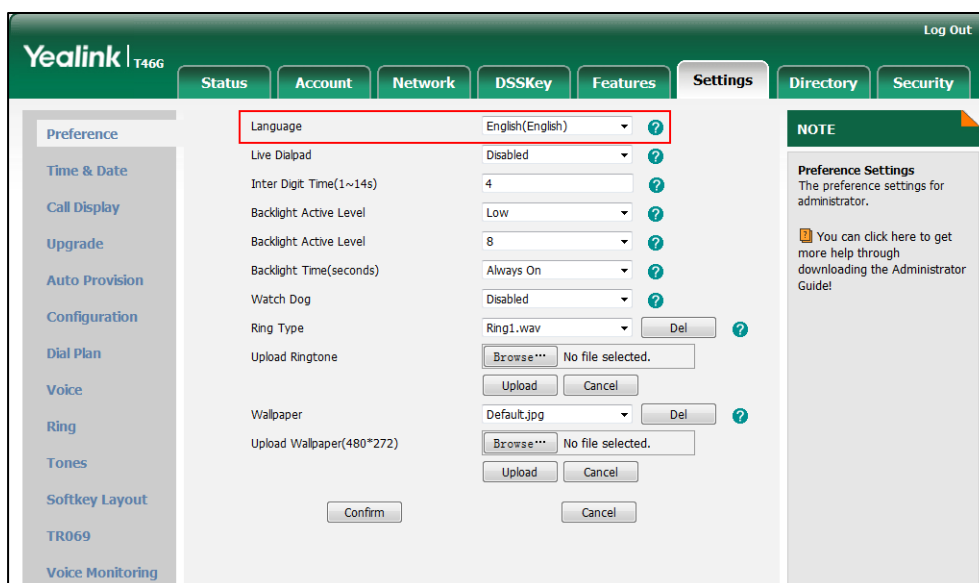
		lang.wui
Local	Web User Interface	Specify the language for the web user interface. Navigate to: http://<phoneIPAddress>/servlet?p=settings-preference&q=load
	Phone User Interface	Specify the language for the phone user interface.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
lang.gui	Refer to the following content	English
<p>Description: Configures the language used on the phone user interface.</p> <p>Permitted Values: English, Chinese_S, Chinese_T, French, German, Turkish, Italian, Polish, Spanish, Portuguese, Russian or the custom language name.</p> <p>Example: lang.gui = English</p> <p>Web User Interface: None</p> <p>Phone User Interface: Menu->Basic->Language</p>		
lang.wui	Refer to the following content	Blank
<p>Description: Configures the language used on the web user interface.</p> <p>Example: lang.wui = French</p> <p>Permitted Values: English, Chinese_S, Chinese_T, French, German, Italian, Polish, Spanish, Turkish, Russian, Portuguese or the custom language name.</p> <p>Web User Interface: Settings->Preference->Language</p> <p>Phone User Interface: None</p>		



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.

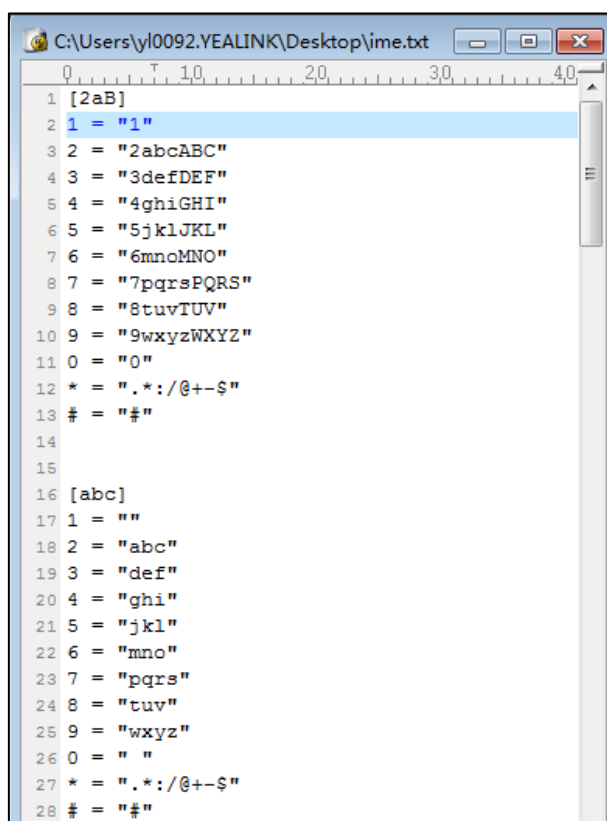
To specify the language for the phone user interface via phone user interface:

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

Input Method Customization

Input method customization allows users to customize the existing input method on IP phones. You can first customize the Yealink-supplied input method file "ime.txt", and then download it to the IP phone. IP phones support 5 input methods: 2aB, abc, Abc, 123, ABC.

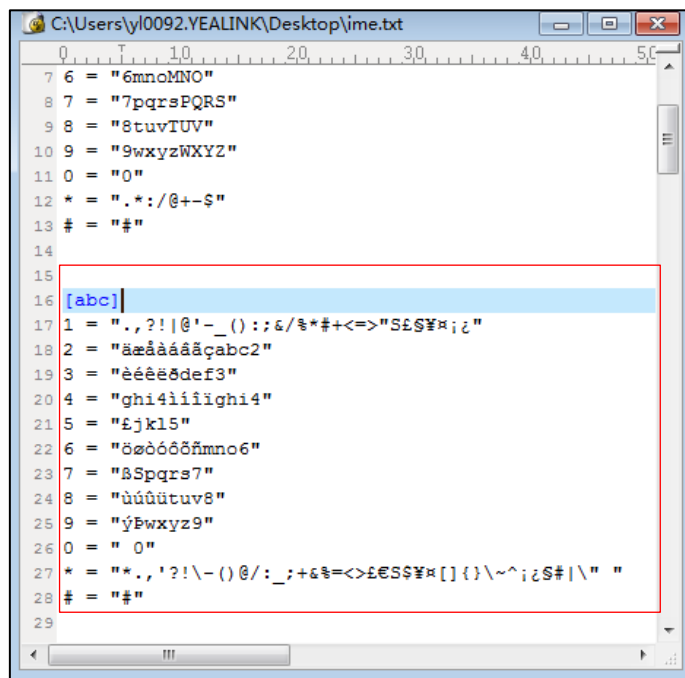
The following shows a portion of the input method file "ime.txt":



The image shows a screenshot of a text editor window titled "C:\Users\y\0092.YEALINK\Desktop\ime.txt". The window contains a list of lines, each representing a mapping from a digit to a string of characters. The lines are numbered 1 through 28. The first section, starting at line 1, is titled "[2aB]" and lists mappings for digits 1 through 9, 0, *, and #. The second section, starting at line 16, is titled "[abc]" and lists mappings for digits 1 through 9, 0, *, and #. The mappings are as follows:

```
1 [2aB]
2 1 = "1"
3 2 = "2abcABC"
4 3 = "3defDEF"
5 4 = "4ghiGHI"
6 5 = "5jklJKL"
7 6 = "6mnoMNO"
8 7 = "7pqrsPQRS"
9 8 = "8tuvTUV"
10 9 = "9wxyzWXYZ"
11 0 = "0"
12 * = ".*:/@+-$"
13 # = "#"
14
15
16 [abc]
17 1 = ""
18 2 = "abc"
19 3 = "def"
20 4 = "ghi"
21 5 = "jkl"
22 6 = "mno"
23 7 = "pqrs"
24 8 = "tuv"
25 9 = "wxyz"
26 0 = " "
27 * = ".*:/@+-$"
28 # = "#"
```


You can add new characters or adjust the character order of the existing input method. The following show an example of adding the Russian characters for the input method "abc".



Note When adding new characters for the existing input method, ensure that the added characters are supported by IP phones. The IP phones can only recognize the input method files uploaded using Unicode encoding. Do not rename the input mode filename.

In addition to customizing the input method file, you can also specify the default input method for the IP phone when editing or searching for contacts.

Procedure

Specify the access URL of the custom input method file and the default input methods using the configuration files.

<p>Configuration File</p>	<p><y000000000xx>.cfg</p>	<p>Specify the access URL of the custom input method file.</p> <p>Parameter: gui_input_method.url</p> <p>Specify the default input method when editing contacts.</p> <p>Parameter: directory.edit_default_input_method</p>
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		Specify the default input method when searching for contacts. Parameter: directory.search_default_input_method
--	--	---

Details of Configuration Parameters:

Parameters	Permitted Values	Default
gui_input_method.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom input method file.</p> <p>Example: The following example uses HTTP to download the custom input method file (ime.txt) from the provisioning server 192.168.10.25. gui_input_method.url = http://192.168.10.25/ime.txt</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
directory.edit_default_input_method	Abc, 2aB, 123, abc or ABC	Abc
<p>Description: Specify the default input method for editing the contact information.</p> <p>Example: directory.edit_default_input_method = abc</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
directory.search_default_input_method	Abc, 2aB, 123, abc or ABC	Abc

Parameters	Permitted Values	Default
<p>Description: Specify the default input method for searching the contacts.</p> <p>Example: directory.search_default_input_method = abc</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

Logo Customization

Logo customization allows unifying the IP phone appearance or displaying a custom image on the idle screen such as a company logo, instead of the default system logo. The logo file format must be *.dob, and the resolution of SIP-T42G/T41P IP phones is 192*64 graphic. Logo is not applicable to SIP-T48G and SIP-T46G IP phones. These two IP phone models use wallpaper instead.

Note

Before uploading your custom logo to IP phones, ensure the logo file is correctly formatted. For more information on customizing a logo file, refer to *Yealink SIP-T2_Series_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	<p>Configure the logo shown on the idle screen and specify the access URL of the custom logo file.</p> <p>Parameters: phone_setting.lcd_logo.mode lcd_logo.url</p>
Local	Web User Interface	<p>Configure the logo shown on the idle screen.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load</p>

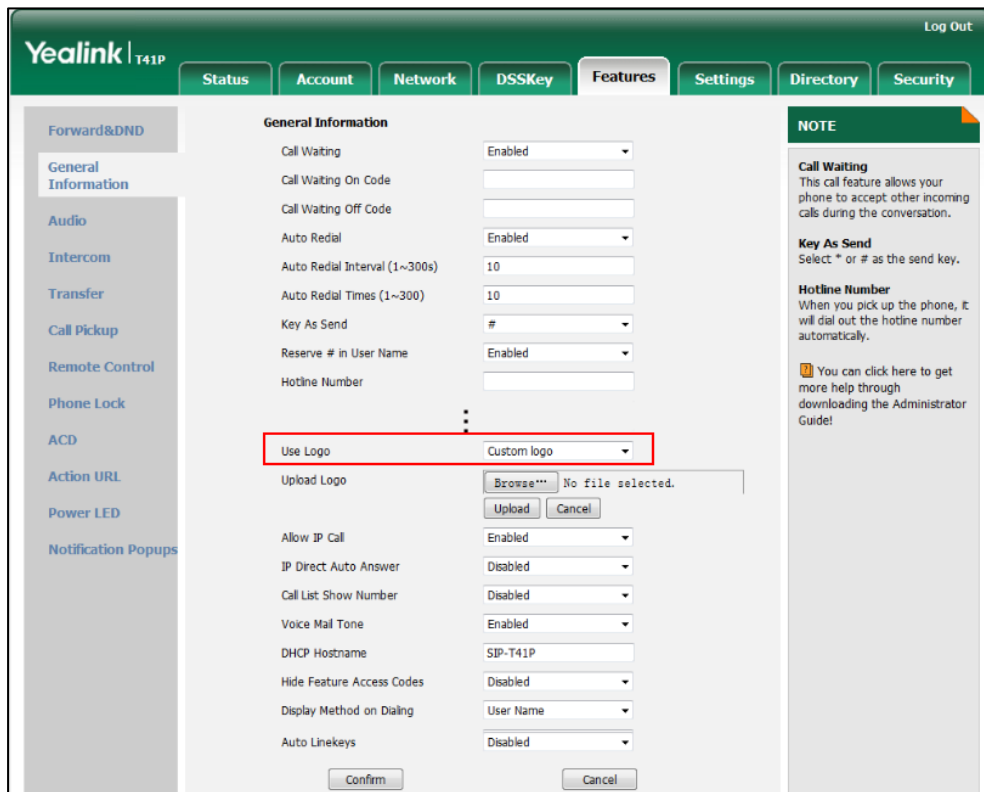
Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.lcd_logo.mode	0, 1 or 2	0
<p>Description: Configures the logo mode of the LCD screen.</p> <p>0-Disabled 1-System logo 2-Custom logo</p> <p>If it is set to 0 (Disabled), the IP phone is not allowed to display a logo. If it is set to 1 (System logo), the LCD screen will display the system logo. If it is set to 2 (Custom logo), the LCD screen will display the custom logo (you need to upload a custom logo file to the IP phone).</p> <p>Note: It is not applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: Features->General Information->Use Logo</p> <p>Phone User Interface: None</p>		
lcd_logo.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom logo file.</p> <p>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</p> <p>Note: It is not applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: Features->General Information->Upload Logo</p> <p>Phone User Interface: None</p>		

To configure a custom logo via web user interface:

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.

The custom logo screen and the idle screen 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. In addition to specifying which soft keys to display, you can determine their display order. It can be configured based on call states.

You can configure the softkey layout using the softkey layout templates for different call states. For more information on how to configure a softkey layout template, refer to [Softkey Layout Template](#) on page 484.

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

Call State	Default Soft Key	Optional Soft Key
CallFailed	NewCall Empty Empty Empty	Empty Switch Cancel

Call State		Default Soft Key	Optional Soft Key
CallIn		Answer Forward Silence Reject	Empty Switch
Connecting	Connecting	Empty Empty Empty Cancel	Empty Switch
	SemiAttendTrans	Transfer Empty Empty Cancel	Empty Switch
Dialing (not applicable to SIP-T48G)		Send IME Delete Cancel	Empty History Switch Line Directory GPickup DPickup Retrieve
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

Call State		Default Soft Key	Optional Soft Key
			Reject PriHold Park GPark
	Hold	Transfer Resume NewCall Cancel	Empty Switch Answer Reject
	Held	Empty Empty Empty Cancel	Empty Switch Answer Reject NewCall
	PreTrans (not applicable to SIP-T48G)	Transfer IME Delete Cancel	Empty Directory Switch Send
	Conferenced	Empty Hold Split Cancel	Empty Switch Answer Reject Mute Manager

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. Parameters: phone_setting.custom_softkey_enable custom_softkey_call_failed.url custom_softkey_call_in.url custom_softkey_connecting.url custom_softkey_dialing.url
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		<p>custom_softkey_ring_back.url</p> <p>custom_softkey_talking.url</p>
Local	Web User Interface	<p>Configure the softkey layout.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet ?p=settings-softkey&q=load</p>


Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.custom_softkey_enable	0 or 1	0
<p>Description:</p> <p>Enables or disables custom soft keys layout feature.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>Web User Interface:</p> <p>Settings->Softkey Layout->Custom Softkey</p> <p>Phone User Interface:</p> <p>None</p>		
custom_softkey_call_failed.url	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Call Failed state.</p> <p>Example:</p> <p>The following example uses HTTP to download the CallFailed state file from the "XMLfiles" directory on provisioning server 10.2.8.16 using 8080 port.</p> <p>custom_softkey_call_failed.url = http://10.2.8.16:8080/XMLfiles/CallFailed.xml</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
custom_softkey_call_in.url	URL within 511 characters	Blank




Parameters	Permitted Values	Default
<p>Description: Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Call In state.</p> <p>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</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
custom_softkey_connecting.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Connecting state.</p> <p>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</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
custom_softkey_dialing.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Dialing state.</p> <p>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</p> <p>Web User Interface:</p>		

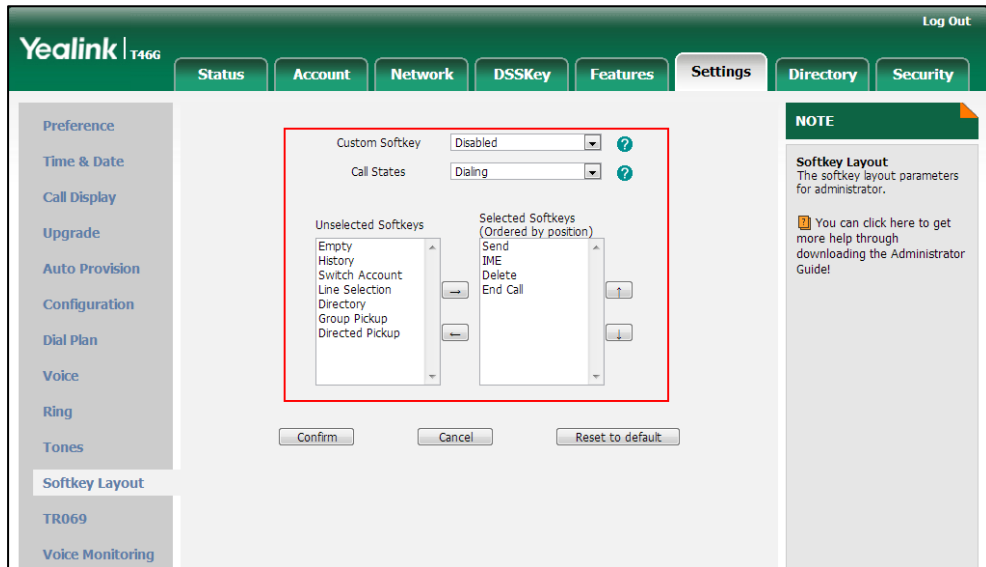
Parameters	Permitted Values	Default
None Phone User Interface: None		
custom_softkey_ring_back.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom file for the soft key presented on the LCD screen when in the RingBack state.</p> <p>Example: The following example 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</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
custom_softkey_talking.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom file for the soft key presented on the LCD screen when in the Talking state.</p> <p>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</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

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 click  .

The selected soft key appears in the **Selected Softkeys** column. If more than four soft keys are selected, a **More** soft key appears on the LCD screen, and the selected soft keys are displayed in two pages.

5. Repeat the step 4 to add more soft keys to the **Selected Softkeys** column.
6. Click  to remove the soft key from the **Selected Softkeys** column.
7. Click  or  to adjust the display order of the soft key.



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

Key as Send

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

Procedure

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

Configuration File	<y0000000000xx>.cfg	<p>Configure a send key.</p> <p>Parameter: features.key_as_send</p> <p>Configure a send sound.</p> <p>Parameter: features.send_key_tone</p> <p>Configure a key tone.</p> <p>Parameter: features.key_tone</p>
---------------------------	---------------------	---

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

Details of Configuration Parameters:

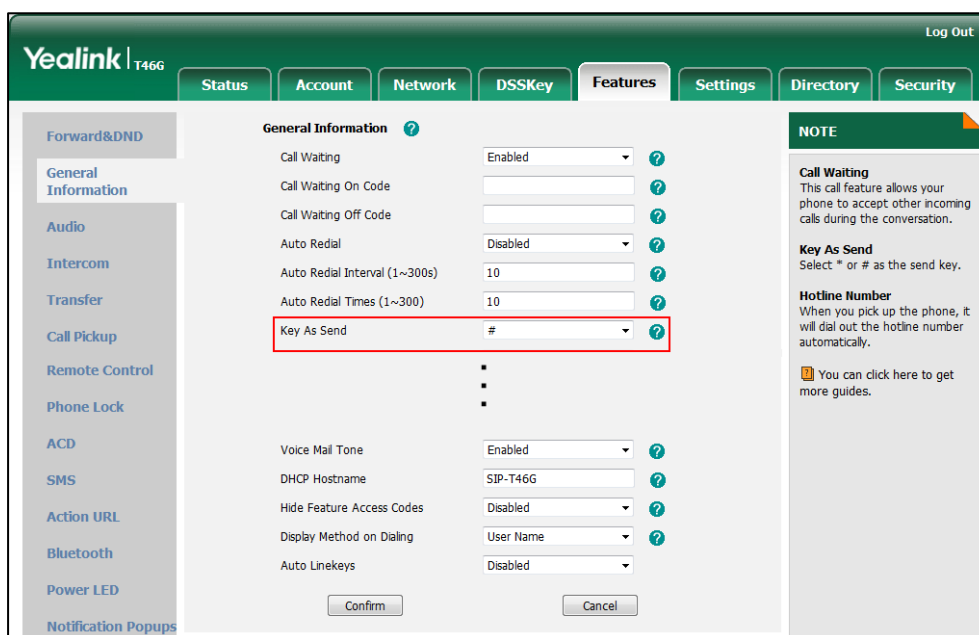
Parameters	Permitted Values	Default
features.key_as_send	0, 1 or 2	1
<p>Description: Configures the "#" or "*" key as the send key.</p> <p>0-Disabled 1-# key 2-* key</p> <p>If it is set to 0 (Disabled), neither "#" nor "*" can be used as a send key. If it is set to 1 (# key), the pound key is used as the send key. If it is set to 2 (* key), the asterisk key is used as the send key.</p> <p>Note: The old parameter "features.pound_key.mode" is also applicable to IP phones.</p> <p>Web User Interface: Features->General Information->Key As Send</p> <p>Phone User Interface: Menu->Features->Others->General->Key As Send</p>		
features.key_tone	0 or 1	1
<p>Description: Enables or disables the IP phone to play a tone when a user presses a key on your phone phone.</p> <p>0-Disabled 1-Enabled</p>		

Parameters	Permitted Values	Default
<p>If it is set to 1 (Enabled), the IP phone will play a tone when a user presses a key on your phone phone.</p> <p>Web User Interface: Features->Audio->Key Tone</p> <p>Phone User Interface: Menu->Basic->Sound->Key Tone</p>		
features.send_key_tone	0 or 1	1
<p>Description: Enables or disables the IP phone to play a tone when a user presses a send key.</p> <p>0-Disabled 1-Enabled</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 value of the parameter "features.key_tone" is set to 1 (Enabled).</p> <p>Web User Interface: Features->Audio->Send Sound</p> <p>Phone User Interface: None</p>		

To configure a send key via web user interface:

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

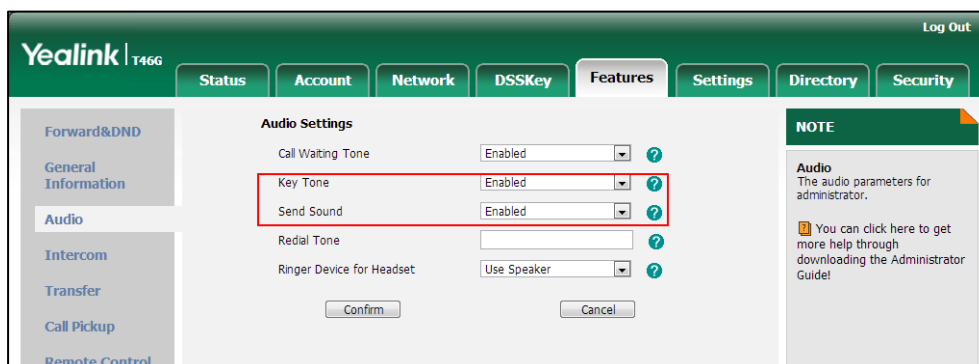
2. Select the desired value from the pull-down list of **Key As Send**.



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

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

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





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

To configure a send key via phone user interface:

1. Press **Menu->Features->Others->General**.
2. Press **←** or **→**, or the **Switch** soft key to select **#** or ***** from the **Key as Send** field, or select **Disabled** to disable this feature.
3. Press the **Save** soft key to accept the change.

To configure a key tone via web user interface:

1. Press **Menu->Basic->Sound->Key Tone**.
2. Press  or , or the **Switch** soft key to select the desired type from the **Key Tone** field.
3. Press the **Save** soft key to accept the change.

Note

Send tone works only if key tone is enabled.
Key tone is enabled by default.

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”.

[]	<p>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".</p>
()	<p>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.</p>
\$	<p>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".</p>

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 482.

Procedure

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

Configuration File	<y0000000000xx>.cfg	<p>Create the replace rule for the IP phone.</p> <p>Parameters: dialplan.replace.prefix.X dialplan.replace.replace.X dialplan.replace.line_id.X</p> <p>Configure the access URL of the replace rule template.</p> <p>Parameter: dialplan_replace_rule.url</p>
Local	Web User Interface	<p>Create the replace rule for the IP phone.</p> <p>Navigate to: http://<phoneIPAddress>/servlet ?p=settings-dialplan&q=load</p>

Details of Configuration Parameters:

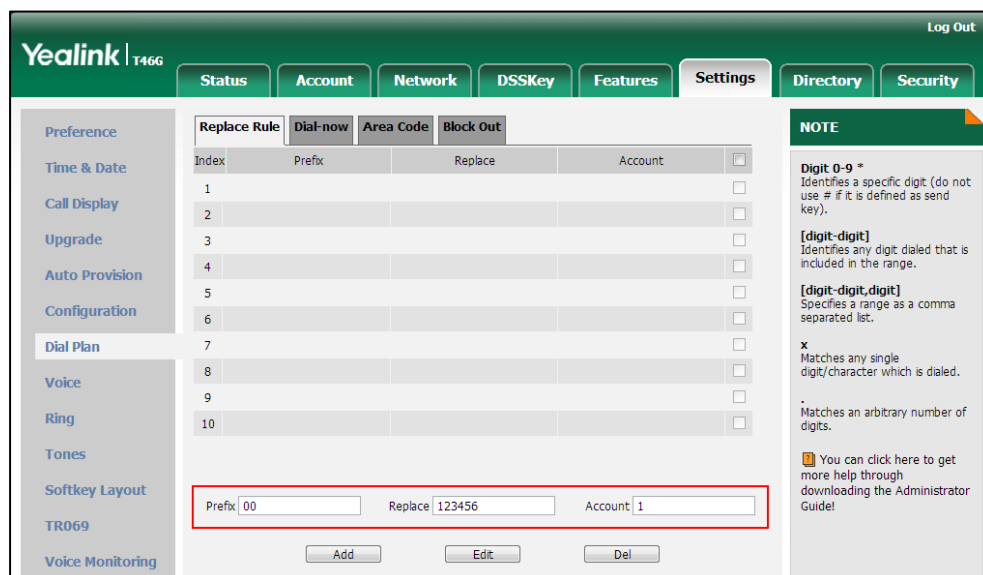
Parameters	Permitted Values	Default
dialplan.replace.prefix.X (X ranges from 1 to 100)	String within 32 characters	Blank
<p>Description: Configures the entered number to be replaced.</p> <p>Example: dialplan.replace.prefix.1 = 00</p> <p>Web User Interface: Settings->Dial Plan->Replace Rule->Prefix</p> <p>Phone User Interface: None</p>		
dialplan.replace.replace.X (X ranges from 1 to 100)	String within 32 characters	Blank
<p>Description: Configures the alternate number to replace the entered number.</p> <p>Example: dialplan.replace.replace.1 = 123456</p> <p>Web User Interface: Settings->Dial Plan->Replace Rule->Replace</p> <p>Phone User Interface: None</p>		
dialplan.replace.line_id.X (X ranges from 1 to 100)	Integer	Blank (for all lines)
<p>Description: Configures the desired line to apply the replace rule. The digit 0 stands for all lines. If it is left blank, the replace rule will apply to all lines on the IP phone.</p> <p>0 to 16 (for SIP-T48G/T46G) 0 to 12 (for SIP-T42G) 0 to 6 (for SIP-T41P)</p> <p>Example: dialplan.replace.line_id.1 = 1</p> <p>Note: Multiple line IDs are separated by commas.</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
Settings->Dial Plan->Replace Rule->Account		
Phone User Interface: None		
dialplan_replace_rule.url	URL within 511 characters	Blank
Description: Configures the access URL of the replace rule template file. Example: dialplan_replace_rule.url = http://192.168.10.25/dialplan.xml Web User Interface: None		

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 the 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 the numbers entered by the user. When entered numbers match the predefined dial-now rule, IP phones 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 483.

Delay Time for Dial-now Rule

IP phones 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	<y000000000xx>.cfg	<p>Create the dial-now rule for the IP phone.</p> <p>Parameters:</p> <p>dialplan.dialnow.rule.X dialplan.dialnow.line_id.X</p> <p>Configure the delay time for the dial-now rule and the access URL of the dial-now template.</p> <p>Parameters:</p> <p>phone_setting.dialnow_delay dialplan_dialnow.url</p>
Local	Web User Interface	<p>Create the dial-now rule for the IP phone.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=settings-dialnow&q=load</p> <p>Configure the delay time for the dial-now rule.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>

Details of Configuration Parameters:

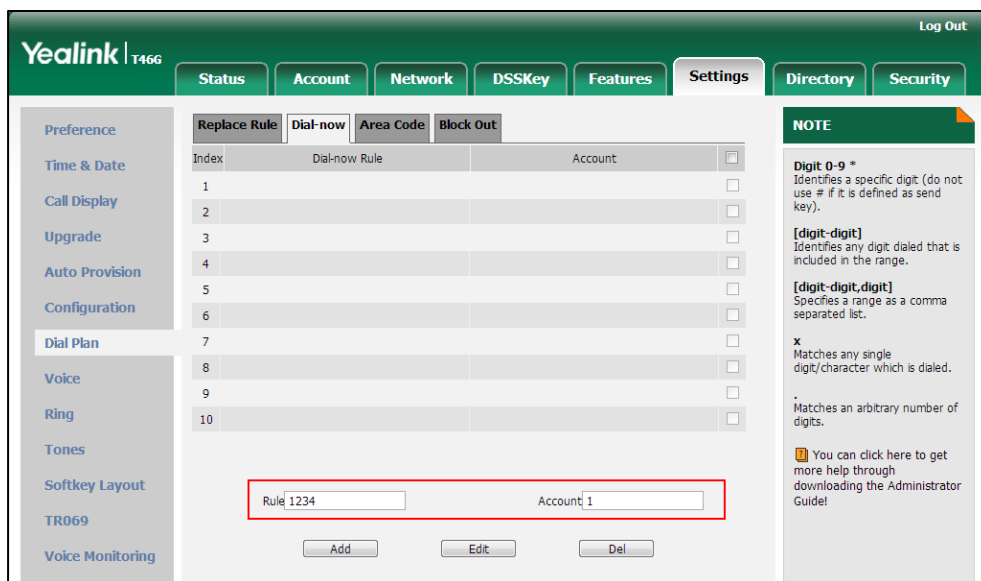
Parameters	Permitted Values	Default
dialplan.dialnow.rule.X (X ranges from 1 to 100)	String within 511 characters	Blank
<p>Description: Configures the dial-now rule (the string used to match the numbers entered by the user). When entered numbers match the predefined dial-now rule, the IP phone will automatically dial out the numbers without pressing the send key.</p> <p>Example: dialplan.dialnow.rule.1 = 1234</p> <p>Web User Interface: Settings->Dial Plan->Dial-now->Rule</p> <p>Phone User Interface: None</p>		
dialplan.dialnow.line_id.X (X ranges from 1 to 100)	Integer	Blank (for all lines)
<p>Description: Configures the desired line to apply the dial-now rule. The digit 0 stands for all lines. If it is left blank, the dial-now rule will apply to all lines on the IP phone. 0 to 16 (for SIP-T48G/T46G) 0 to 12 (for SIP-T42G) 0 to 6 (for SIP-T41P)</p> <p>Example: dialplan.dialnow.line_id.1 = 1</p> <p>Note: Multiple line IDs are separated by commas.</p> <p>Web User Interface: Settings->Dial Plan->Dial-now->Account</p> <p>Phone User Interface: None</p>		
phone_setting.dialnow_delay	Integer from 1 to 14	1

Parameters	Permitted Values	Default
<p>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.</p> <p>Web User Interface: Features->General Information->Time-Out for Dial-Now Rule</p> <p>Phone User Interface: None</p>		
dialplan_dialnow.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the dial-now rule template file.</p> <p>Example: dialplan_dialnow.url = http://192.168.10.25/dialnow.xml</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

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.

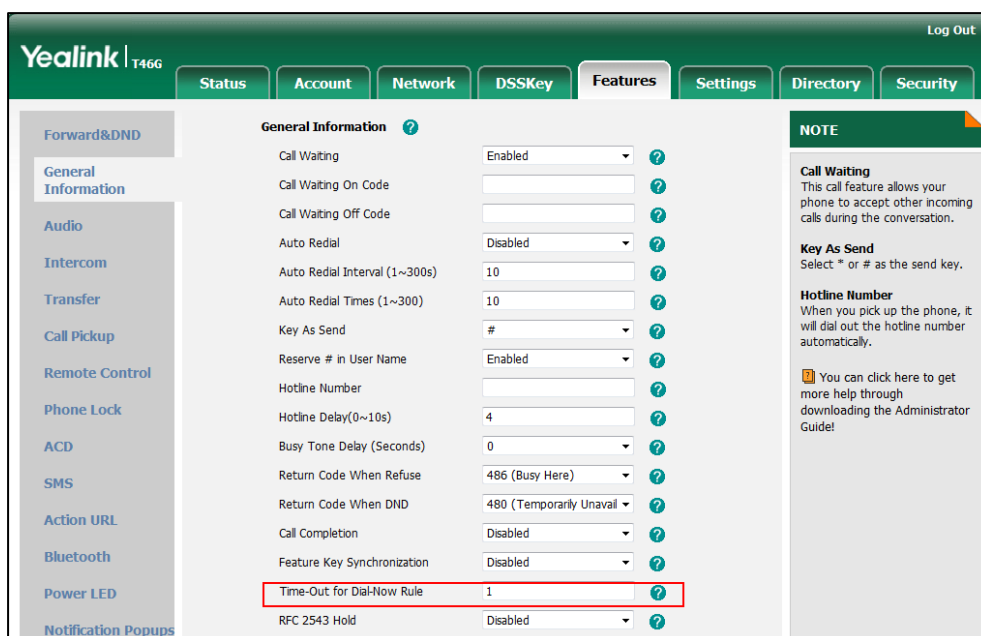
If you leave the field blank or enter 0, the dial-now rule will apply to all accounts on the IP phone.



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



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

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). They usually indicate geographical areas in one country. When the 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 the entered numbers. Parameters: dialplan.area_code.code dialplan.area_code.min_len dialplan.area_code.max_len dialplan.area_code.line_id
Local	Web User Interface	Create the area code rule and specify the maximum and minimum lengths of entered numbers. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-areacode&q=load">http://<phoneIPAddress>/servlet?p=settings-areacode&q=load

Details of Configuration Parameters:

Parameters	Permitted Values	Default
dialplan.area_code.code	String within 16 characters	Blank
<p>Description: Configures the area code to be added before the entered numbers when dialing out.</p> <p>Note: The length of the entered number must be between the minimum length configured by the parameter "dialplan.area_code.min_len" and the maximum length configured by the parameter "dialplan.area_code.max_len".</p> <p>Example: dialplan.area_code.code = 010</p>		

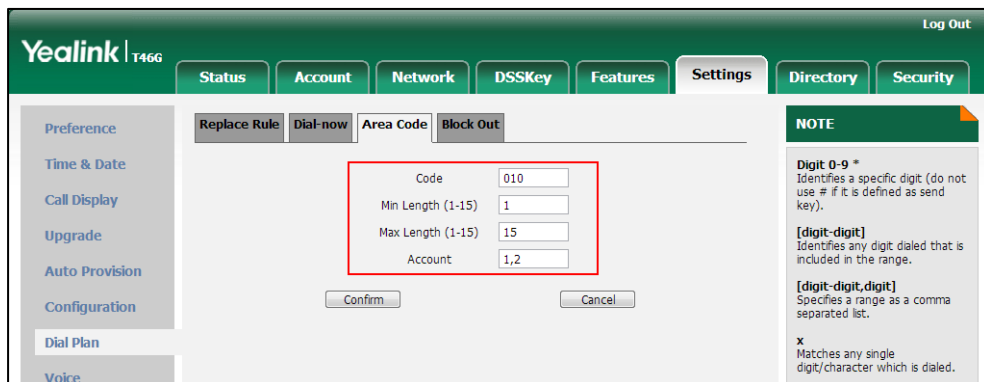
Parameters	Permitted Values	Default
<p>Web User Interface: Settings->Dial Plan->Area Code->Code</p> <p>Phone User Interface: None</p>		
dialplan.area_code.min_len	Integer from 1 to 15	1
<p>Description: Configures the minimum length of the entered numbers.</p> <p>Web User Interface: Settings->Dial Plan->Area Code->Min Length (1-15)</p> <p>Phone User Interface: None</p>		
dialplan.area_code.max_len	Integer from 1 to 15	15
<p>Description: Configures the maximum length of the entered numbers.</p> <p>Note: The value must be larger than the minimum length.</p> <p>Web User Interface: Settings->Dial Plan->Area Code->Max Length (1-15)</p> <p>Phone User Interface: None</p>		
dialplan.area_code.line_id	Integer	Blank (for all lines)
<p>Description: Configures the desired line to apply the area code rule. The digit 0 stands for all lines. If it is left blank, the area code rule will apply to all lines on the IP phone. 0 to 16 (for SIP-T48G/T46G) 0 to 12 (for SIP-T42G) 0 to 6 (for SIP-T41P)</p> <p>Example: dialplan.area_code.line_id = 1,2</p> <p>Note: Multiple line IDs are separated by commas.</p> <p>Web User Interface: Settings->Dial Plan->Area Code->Account</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		

To configure an area code rule via web user interface:

1. Click on **Settings->Dial Plan->Area Code**.
2. Enter 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 the field blank or enter 0, the area code rule will apply to all accounts on the IP phone.



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

Block Out

Block out rule prevents users from dialing out specific numbers. When the 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. Parameters: dialplan.block_out.number.X dialplan.block_out.line_id.X
Local	Web User Interface	Create the block out rule for the desired line. Navigate to: http://<phoneIPAddress>/servlet

		?p=settings-blackout&q=load
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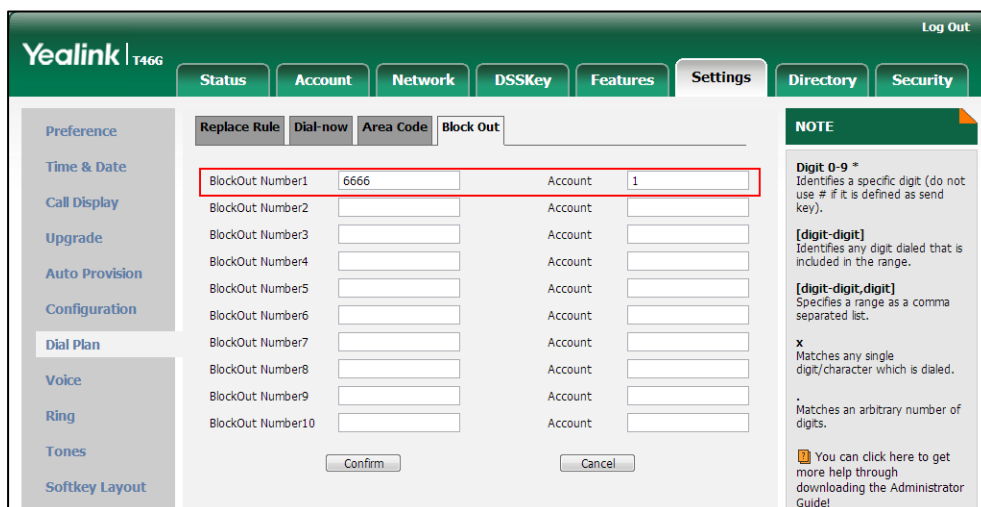
Details of Configuration Parameters:

Parameters	Permitted Values	Default
dialplan.block_out.number.X (X ranges from 1 to 10)	String within 32 characters	Blank
<p>Description: Configures the block out numbers.</p> <p>Example: dialplan.block_out.number.1 = 6666</p> <p>Web User Interface: Settings->Dial Plan->Block Out->BlockOut NumberX</p> <p>Phone User Interface: None</p>		
dialplan.block_out.line_id.X (X ranges from 1 to 10)	Integer	Blank (for all lines)
<p>Description: Configures the desired line to apply the block out rule. The digit 0 stands for all lines. If it is left blank, the block out rule will apply to all lines on the IP phone. 0 to 16 (for SIP-T48G/T46G) 0 to 12 (for SIP-T42G) 0 to 6 (for SIP-T41P)</p> <p>Example: dialplan.block_out.line_id.1 = 1</p> <p>Note: Multiple line IDs are separated by commas.</p> <p>Web User Interface: Settings->Dial Plan->Block Out->Account</p> <p>Phone User Interface: None</p>		

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 the 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.

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.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the hotline number.</p> <p>Parameter: features.hotline_number</p> <p>Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number.</p> <p>Parameter: features.hotline_delay</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the hotline number.</p> <p>Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number.</p> <p>Navigate to: http://<phoneIPAddress>/servlet</p>

		?p=features-general&q=load
	Phone User Interface	Configure the hotline number. Specify the time (in seconds) the IP phone waits to automatically dial out the hotline number.

Details of Configuration Parameters:

Parameter	Permitted Values	Default
features.hotline_number	String within 32 characters	Blank
<p>Description: Configures the hotline 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.</p> <p>Example: features.hotline_number = 3601</p> <p>Web User Interface: Features->General Information->Hotline Number</p> <p>Phone User Interface: Menu->Features->Others->Hot Line->Number</p>		
features.hotline_delay	Integer from 0 to 10	4
<p>Description: Configures the waiting time (in seconds) for the IP phone to automatically dial out the hotline number. If it is set to 0 (0s), the IP phone will immediately dial out the preconfigured hotline number when you lift the handset, press the speakerphone key or press the line key. If it is set to a value greater than 0, the IP phone will wait the designated seconds before dialing out the predefined hotline number when you lift the handset, press the speakerphone key or press the line key.</p> <p>Web User Interface: Features->General Information->Hotline Delay (0~10s)</p> <p>Phone User Interface: Menu->Features->Others->Hot Line->HotLine Delay</p>		

To configure hotline via web user interface:

1. Click on **Features->General Information**.
2. Enter the hotline number in the **Hotline Number** field.

- Enter the delay time in the **Hotline Delay (0~10s)** field.

The screenshot shows the Yealink T46G web interface with the 'Features' tab selected. The 'General Information' section is expanded, and the 'Hotline Number' and 'Hotline Delay(0~10s)' fields are highlighted with a red box. The 'Hotline Number' is set to 3601 and the 'Hotline Delay' is set to 4. Other settings include Call Waiting (Enabled), Auto Redial (Disabled), and various return codes.

Feature	Value
Call Waiting	Enabled
Call Waiting On Code	
Call Waiting Off Code	
Auto Redial	Disabled
Auto Redial Interval (1~300s)	10
Auto Redial Times (1~300)	10
Key As Send	#
Reserve # in User Name	Enabled
Hotline Number	3601
Hotline Delay(0~10s)	4
Busy Tone Delay (Seconds)	0
Return Code When Refuse	486 (Busy Here)
Return Code When DND	480 (Temporarily Unava)
Call Completion	Disabled
Feature Key Synchronization	Disabled
Time-Out for Dial-Now Rule	1

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

To configure hotline via phone user interface:

- Press **Menu->Features->Others->Hot Line**.
- Enter the hotline number in the **Number** field.
- Enter the delay time in the **HotLine Delay** field.
- Press the **Save** soft key to accept the change.

Off Hook Hot Line Dialing

For security reasons, IP phones support off hook hot line dialing feature, which allows the phone to first dial out the pre-configured number when the user presses the speakerphone key or desired line key, dials out a call or off hook the phone using the account with this feature enabled. The SIP server may then prompt the user to enter an activation code for call service. Only if the user enters a valid activation code, the IP phone will be permitted to place the call.

Off hook hot line dialing feature is configurable on a per-line basis and depends on support from a SIP server.

Note

Off hook hot line dialing feature limits the call-out permission of this account and disables the hotline feature. For example, when the phone goes off hook using the account with this feature enabled, the configured hotline number will not be dialed out automatically.

The server actions may vary from different servers.

This feature is also applicable to the IP call and intercom call.

Procedure

Off hook hot line dialing can be configured using the configuration files.

Configuration File	<y0000000000xx>.cfg	<p>Configure off hook hot line dialing feature.</p> <p>Parameter: account.X.auto_dial_enable Specify the number that the phone first dials out.</p> <p>Parameter: account.X.auto_dial_num</p>
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Details of Configuration Parameters:

Parameter	Permitted Values	Default
account.X.auto_dial_enable	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to first dial out a pre-configured number when a user presses the speakerphone key or desired line key, dials out a call or off hook the phone using account X.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>If it is set to 1(Enabled), the phone will first dial out the pre-configured number (configured by the parameter "account.X.auto_dial_num") when a user presses the speakerphone key or desired line key, dials out a call or off hook the phone using account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
account.X.auto_dial_num	String within 32 characters	Blank
<p>Description:</p> <p>Configures the number that the IP phone first dials out when a user presses the speakerphone key or desired line key, dials out a call or off hook the phone using</p>		

Parameter	Permitted Values	Default
account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Note: It works only if the value of the parameter "account.X.auto_dial_enable" is set to 1 (Enabled). Web User Interface: None Phone User Interface: None		

Directory

Directory provides easy access to frequently used lists. The lists can be Local Directory, History, Remote Phone Book and LDAP. The desired list(s) can be added to Directory using a directory file. For more information on the directory file, refer to [Directory Template](#) on page 485.

Procedure

Directory can be configured using the configuration files or locally.


Configuration File	<y0000000000xx>.cfg	Specify the access URL of the Directory file. Parameter: directory_setting.url
Local	Web User Interface	Configure the Directory. Navigate to: http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

Details of the Configuration Parameter:




Parameter	Permitted Values	Default
directory_setting.url	URL within 511 characters	Blank

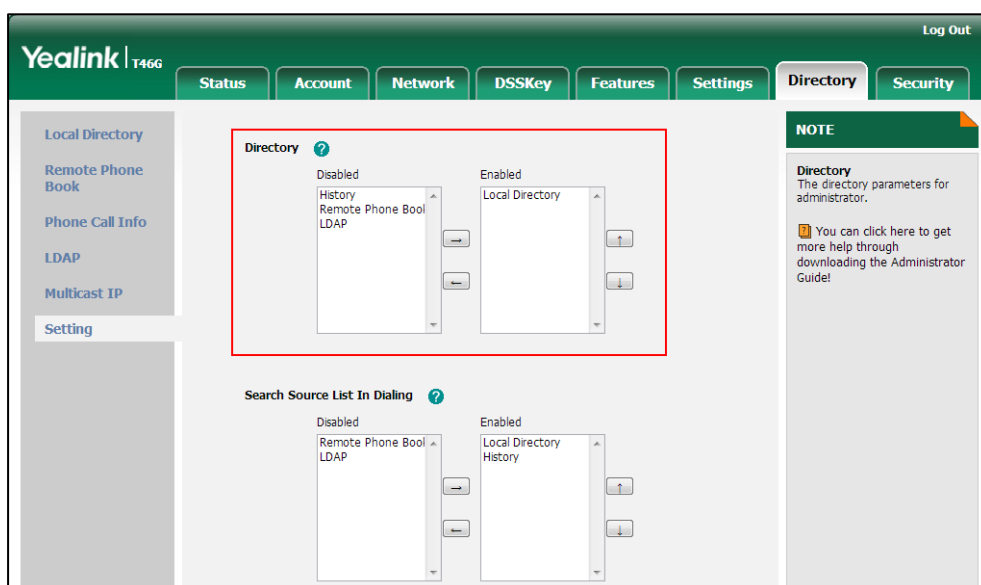
Parameter	Permitted Values	Default
<p>Description: Configures the access URL of the directory template.</p> <p>Example: directory_setting.url = http://192.168.1.20/favorite_setting.xml</p> <p>Web User Interface: Directory->Setting->Directory</p> <p>Phone User Interface: None</p>		

To configure the directory via web user interface:

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

The selected list appears in the **Enabled** column.

3. Repeat the step 2 to add more lists to the **Enabled** column.
4. To remove a list from the **Enabled** column, select the desired list and then click .
5. To adjust the display order of enabled lists, select the desired list and then click  or .



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

The IP phone LCD screen will display the enabled list(s) in the adjusted order.

Search Source List in Dialing

Search source list in dialing allows the IP phone to automatically search entries from the search source list based on the entered string, and display results on the pre-dialing screen. The search source list can be Local Directory, History, Remote Phone Book and LDAP. You can configure the search source list in dialing using a super search file. For more information on the super search template, refer to [Super Search Template](#) on page 486.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Specify the access URL of the super search file. Parameter: super_search.url
Local	Web User Interface	Configure the search source list in dialing. Navigate to: http://<phoneIPAddress>/servlet?p=contacts-favorite&q=load

Details of the Configuration Parameter:




Parameter	Permitted Values	Default
super_search.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the super search template.</p> <p>Web User Interface: Directory->Setting->Search Source List In Dialing</p> <p>Phone User Interface: None</p>		

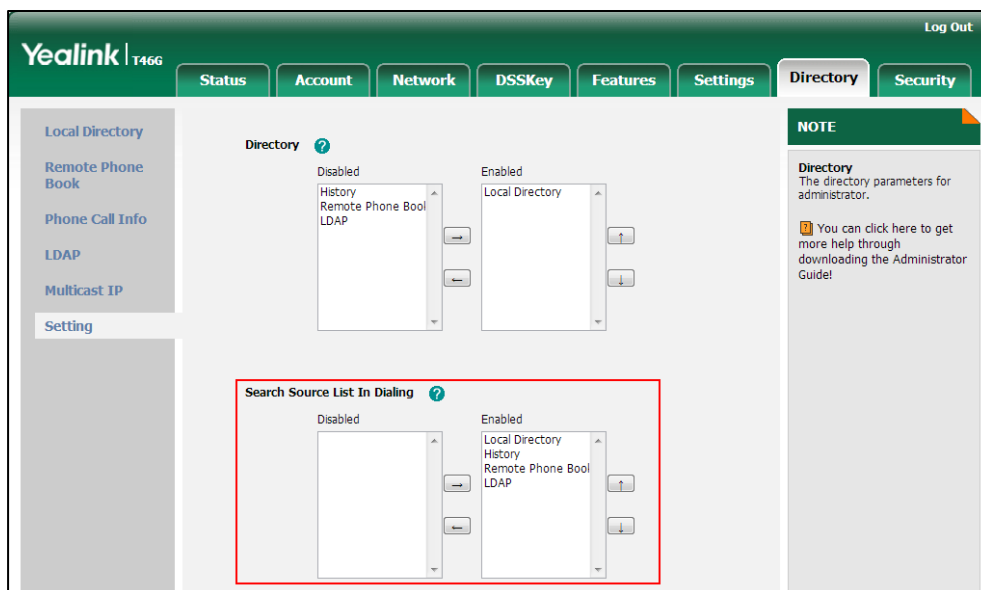
To configure search source list in dialing via web user interface:

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

The selected list appears in the **Enabled** column.

3. Repeat step 2 to add more lists to the **Enabled** column.

4. To remove a list from the **Enabled** column, select the desired list and then click  .
5. To adjust the display order of the enabled list, select the desired list, and click  or  .



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

The pre-dialing screen displays the search results in the adjusted order.

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: Missed calls, Placed calls, Received calls and Forwarded calls. Each call log list supports up to 100 entries. To store call information, you must enable the save call log feature in advance.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the call log. Parameter: features.save_call_history
Local	Web User Interface	Configure the call log. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure the call log.

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.save_call_history	0 or 1	1

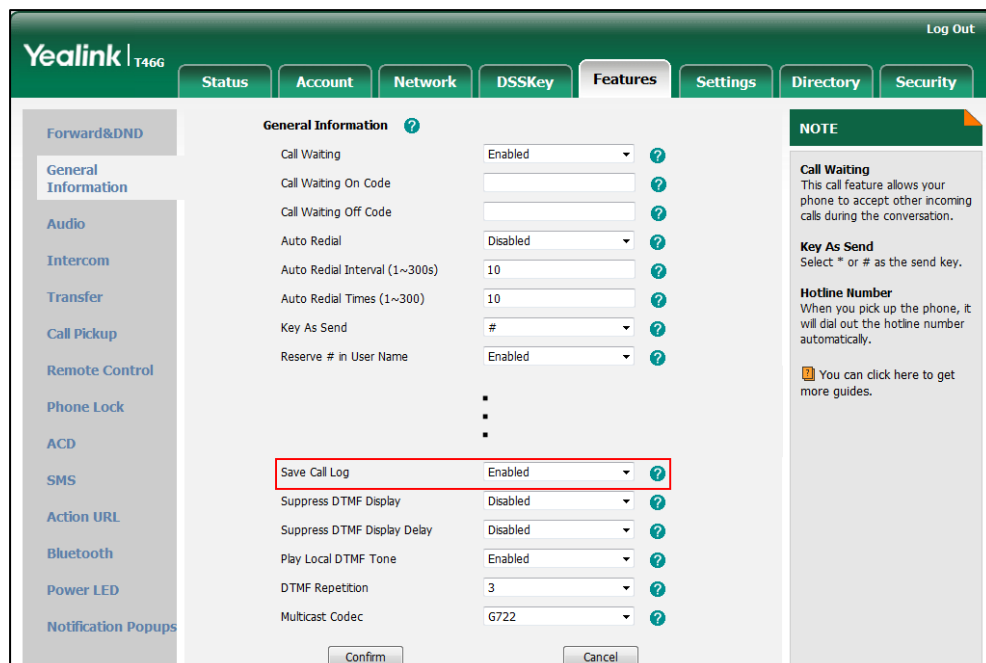
Description:
Enables or disables the IP phone to save call log.
0-Disabled
1-Enabled
If it is set to 0 (Disabled), the IP phone cannot log the placed calls, received calls, missed calls and the forwarded calls in the call log lists.

Web User Interface:
Features->General Information->Save Call Log

Phone User Interface:
Menu->Features->Others->General->History Record

To configure the call log 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 the call log via phone user interface:

1. Press **Menu->Features->Others->General**.

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 IP phones to display the number of the missed calls with an indicator icon on the idle screen, and to log the 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 the missed call log feature. Parameter: account.X.missed_callog
Local	Web User Interface	Configure the missed call log feature. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0

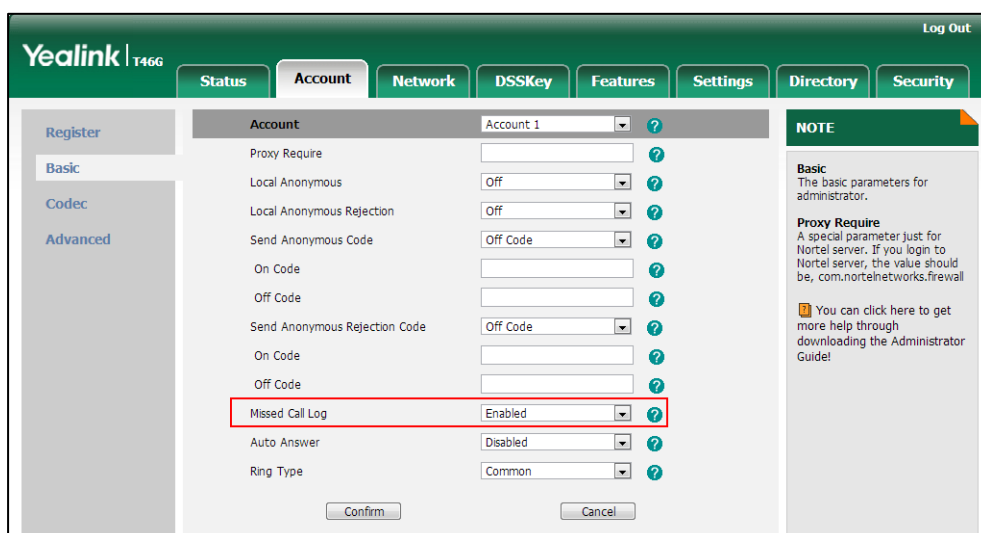
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.missed_callog	0 or 1	1
<p>Description: Enables or disables the IP phone to record missed calls for account X.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 0 (Disabled), there is no indicator displaying on the LCD screen, the IP phone does not log the missed call in the Missed Calls list.</p> <p>If it is set to 1 (Enabled), a prompt message "<number> New Missed Call(s)" along with an indicator icon is displayed on the IP phone idle screen when the IP phone misses calls.</p>		

Parameter	Permitted Values	Default
X ranges from 1 to 16 (for SIP-T48G/T46G).		
X ranges from 1 to 12 (for SIP-T42G).		
X ranges from 1 to 6 (for SIP-T41P).		
Web User Interface:		
Account->Basic->Missed Call Log		
Phone User Interface:		
None		

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

The IP phone maintains a local directory. The local directory can store up to 1000 contacts and 48 groups (including the default groups: Company, Family and Friend). 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. Yealink IP phones support both *.xml and *.csv format contact files. For more information on how to customize a contact file (*.xml), refer to [Local Contact File](#) on page 488.

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 (*.xml). Parameter: local_contact.data.url
Local	Web User Interface	Add a new group and a contact to the local directory. To import or export contact file. Navigate to: http://<phoneIPAddress>/servlet?p=contactsbasic&q=load&num=1&group=
	Phone User Interface	Add a new group and a contact to the local directory.

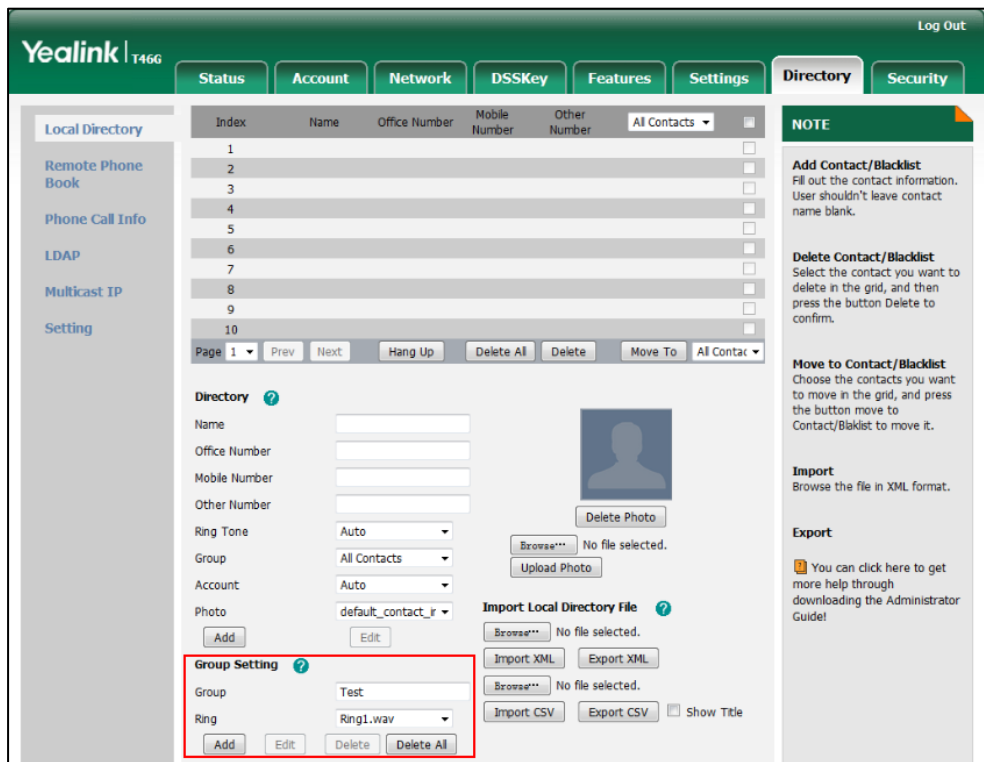
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
local_contact.data.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the local contact file (*.xml).</p> <p>Example: local_contact.data.url = http://192.168.10.25/contact.xml</p> <p>Web User Interface: Directory->Local Directory->Import Local Directory File</p> <p>Phone User Interface: None</p>		

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

1. Click on **Directory->Local Directory**.
2. In the **Group Setting** block, enter the new group name in the **Group** field.

3. Select the desired group ring tone from the pull-down list of **Ring**.

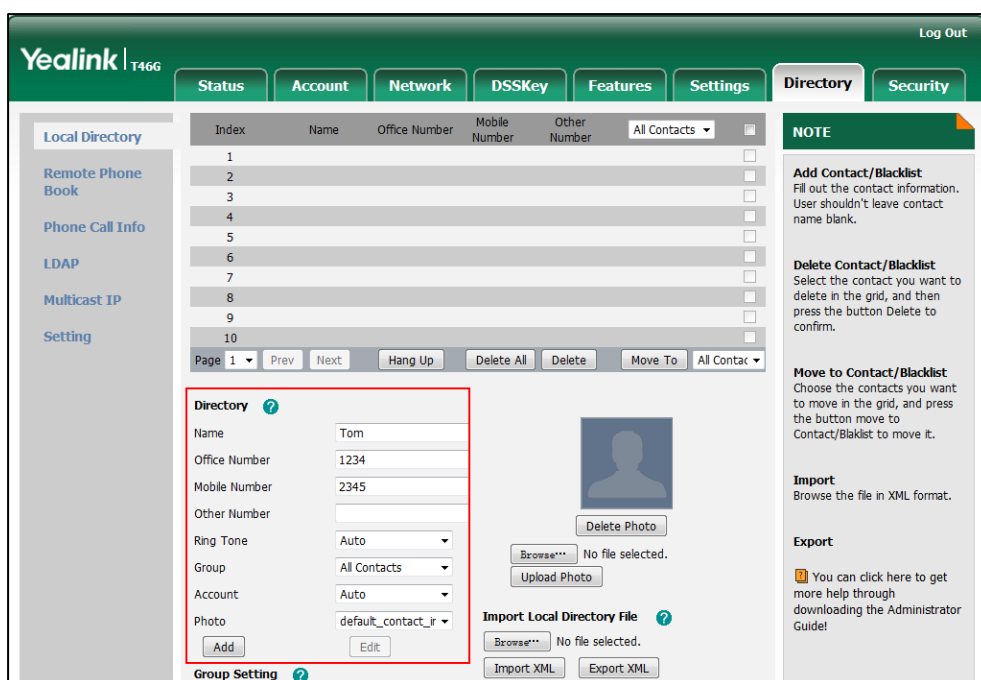


4. Click **Add** to add the group.

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

1. Click on **Directory->Local Directory**.
2. 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**.
6. Select the desired photo from the pull-down list of **Photo**.

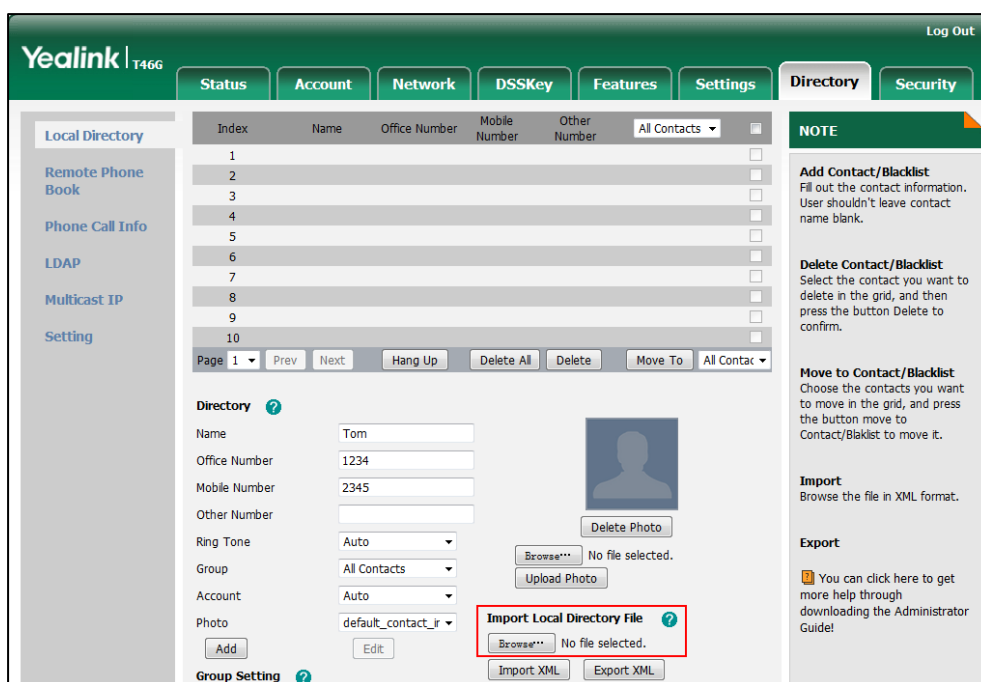
It is not applicable to SIP-T42G and SIP-T41P IP phones.



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

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

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



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

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

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

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

1. Click on **Directory->Local Directory**.
2. Click **Browse** to locate a contact list file (the file format must be *.csv) from your local system.
3. (Optional.) Check the **Show Title** checkbox.

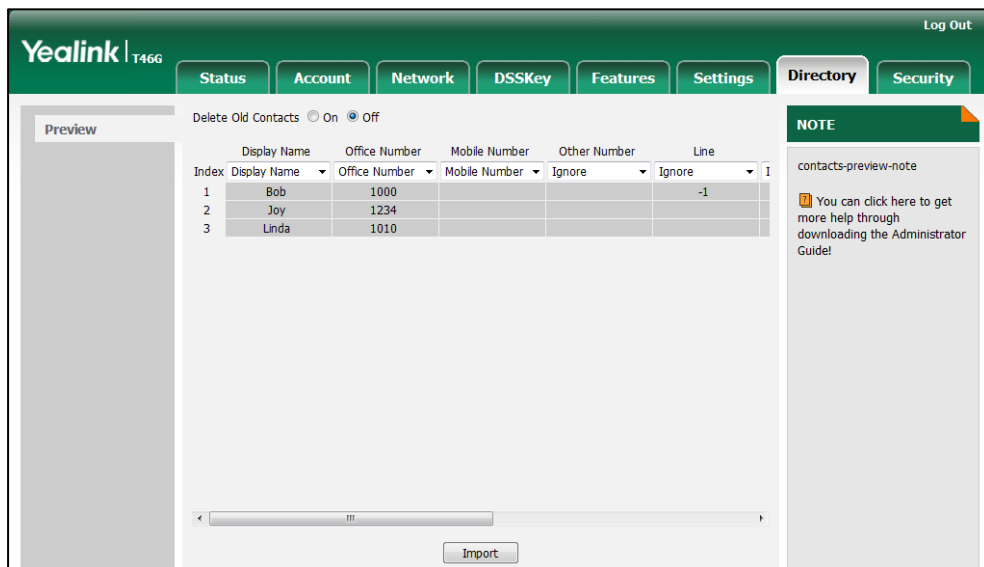
It will prevent importing the title of the contact information which is located in the first line of the CSV file.

4. Click **Import CSV** to import the contact list.
5. (Optional.) Mark the **On** radio box in the **Delete Old Contacts** field.

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

6. Select the contact information you want to import into the local directory from the pull-down list of **Index**.

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





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

To export a contact list via web user interface:



1. Click on **Directory->Local Directory**.
2. Click **Export XML** (or **Export CSV**).
3. Click **Save** to save the contact list to your local system.

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





1. Press **Menu->Directory->Local Directory**.
2. Press the **Add Group** soft key.
3. Enter the desired group name in the **Name** field.

4. Press  or  to select the desired group ring tone from the **Ring Tones** field.
5. Press the **Save** soft key to accept the change or the **Back** soft key to cancel.

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

1. Press **Menu->Directory->Local Directory**.
2. Select the desired contact group and then press the **Enter** soft key.
3. Press the **Add** soft key.
4. Enter the name and the office, mobile or other numbers in the corresponding fields.
5. Press  or , or the **Switch** soft key to select the desired account from the **Account** field.

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

6. Press  or , or the **Switch** soft key to select the desired ring tone from the **Ring** field.
7. Press  or , or the **Switch** soft key to select the desired photo from the **Photo** field.
8. 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. Parameters: phone_setting.predial_autodial phone_setting.inter_digit_time
Local	Web User Interface	Configure live dialpad. Navigate to: http://<phoneIPAddress>/servlet ?p=settings-preference&q=load

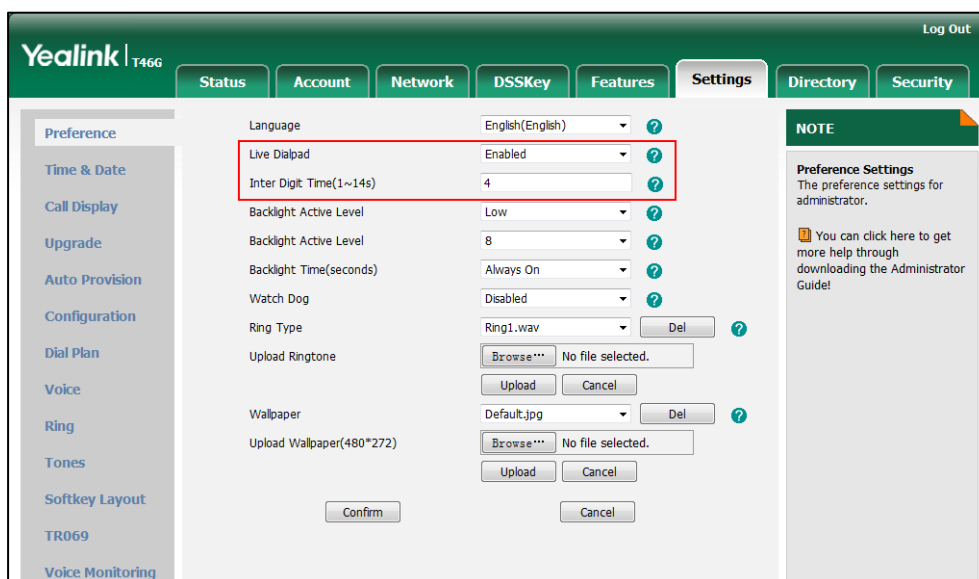
Details of Configuration Parameters:

Parameters	Permitted Values	Default
<code>phone_setting.predial_autodial</code>	0 or 1	0
<p>Description: Enables or disables live dialpad feature.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone will automatically dial out the entered phone number in the pre-dialing screen without pressing a send key.</p> <p>Web User Interface: Settings->Preference->Live Dialpad</p> <p>Phone User Interface: None</p>		
<code>phone_setting.inter_digit_time</code>	Integer from 1 to 14	4
<p>Description: Configures the time (in seconds) for the IP phone to automatically dial out the entered digits without pressing a send key.</p> <p>Note: It works only if the value of the parameter "phone_setting.predial_autodial" is set to 1 (Enabled).</p> <p>Web User Interface: Settings->Preference->Inter Digit Time (1~14s)</p> <p>Phone User Interface: None</p>		

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 incoming 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 IP 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.

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

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure call waiting. Parameters: call_waiting.enable call_waiting.tone call_waiting.on_code call_waiting.off_code
Local	Web User Interface	Configure call waiting. Navigate to: http://<phoneIPAddress>/servlet

		?p=features-general&q=load
	Phone User Interface	Configure call waiting.

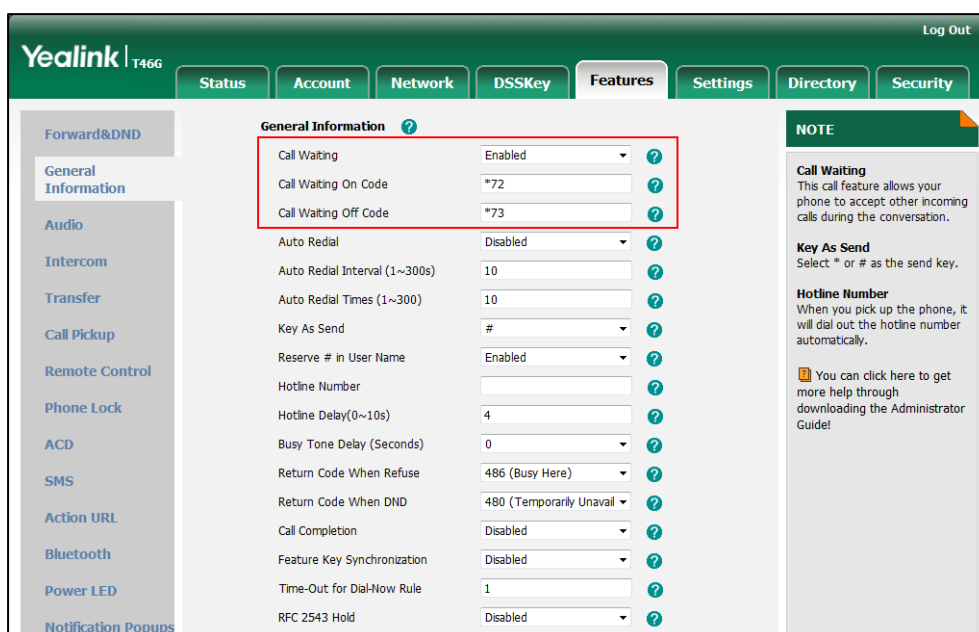
Details of Configuration Parameters:

Parameters	Permitted Values	Default
call_waiting.enable	0 or 1	1
<p>Description: Enables or disables call waiting feature.</p> <p>0-Disabled 1-Enabled</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> <p>Web User Interface: Features->General Information->Call Waiting</p> <p>Phone User Interface: Menu->Features->Call Waiting->Call Waiting</p>		
call_waiting.tone	0 or 1	1
call_waiting.on_code	String within 32 characters	Blank
<p>Description: Configures the call waiting on code to activate the server-side call waiting feature. The IP phone will send the call waiting on code to the server when you activate call waiting feature on the IP phone.</p> <p>Example: call_waiting.on_code = *72</p> <p>Web User Interface: Features->General Information->Call Waiting On Code</p> <p>Phone User Interface: Menu->Features->Call Waiting->On Code</p>		
call_waiting.off_code	String within 32 characters	Blank
Description:		

Parameters	Permitted Values	Default
<p>Configures the call waiting off code to deactivate the server-side call waiting feature. The IP phone will send the call waiting off code to the server when you deactivate call waiting feature on the IP phone.</p> <p>Example:</p> <p>call_waiting.off_code = *73</p> <p>Web User Interface:</p> <p>Features->General Information->Call Waiting Off Code</p> <p>Phone User Interface:</p> <p>Menu->Features->Call Waiting->Off Code</p>		

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

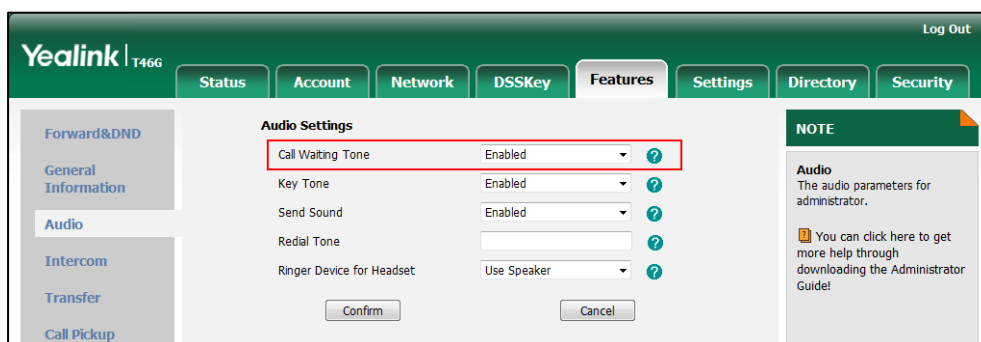


5. Click **Confirm** to accept the change.

To configure the call waiting tone via web user interface:

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

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



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

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

1. Press **Menu->Features->Call Waiting**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Call Waiting** field.
3. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Play Tone** field.
4. (Optional.) Enter the call waiting on code in the **On Code** field.
5. (Optional.) Enter the call waiting off code in the **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.

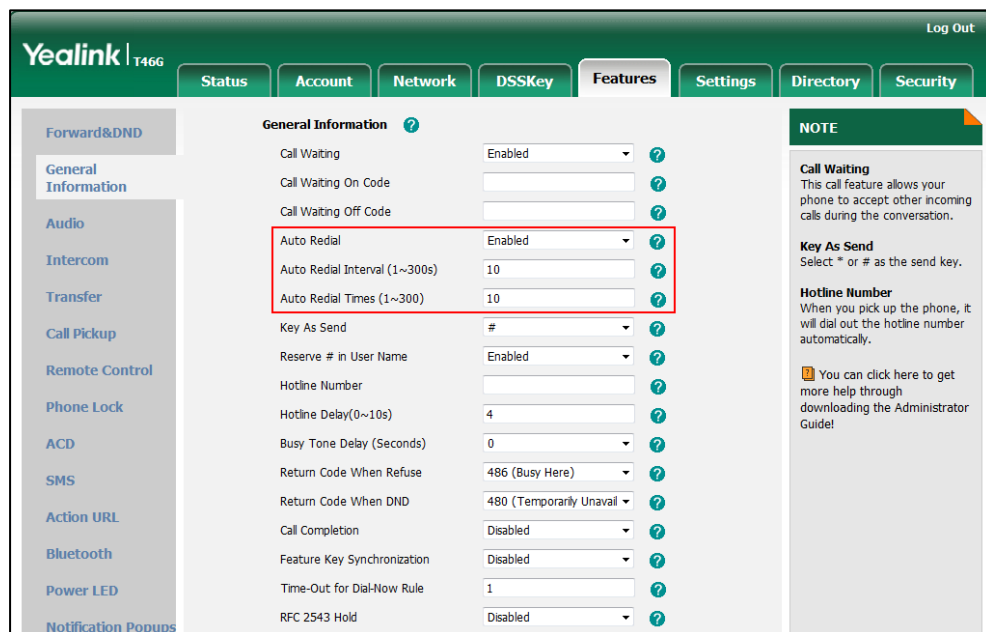
Configuration File	<y0000000000xx>.cfg	Configure auto redial feature. Parameters: auto_redial.enable auto_redial.interval auto_redial.times
Local	Web User Interface	Configure auto redial feature. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure auto redial.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
auto_redial.enable	0 or 1	0
<p>Description: Enables or disables the IP phone to automatically redial the dialed number when the callee is temporarily unavailable.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone will dial the previous dialed out number automatically when the dialed number is temporarily unavailable.</p> <p>Web User Interface: Features->General Information->Auto Redial</p> <p>Phone User Interface: Menu->Features->Others->Auto Redial->Auto Redial</p>		
auto_redial.interval	Integer from 1 to 300	10
<p>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.</p> <p>Web User Interface: Features->General Information->Auto Redial Interval (1~300s)</p> <p>Phone User Interface: Menu->Features->Others->Auto Redial->Redial Interval</p>		
auto_redial.times	Integer from 1 to 300	10
<p>Description: Configures the auto redial times when the callee is temporarily unavailable. The IP phone tries to redial the dialed number as many times as configured till the callee answers the call.</p> <p>Web User Interface: Features->General Information->Auto Redial Times (1~300)</p> <p>Phone User Interface: Menu->Features->Others->Auto Redial->Redial Times</p>		

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 desired time interval (in seconds) 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->Others->Auto Redial**.
2. Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Auto Redial** field.
3. Enter the desired time in the **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. Parameter: account.X.auto_answer
	<y0000000000xx>.cfg	Specify a period of delay time for auto answer. Parameter: features.auto_answer_delay
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. http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure auto answer.

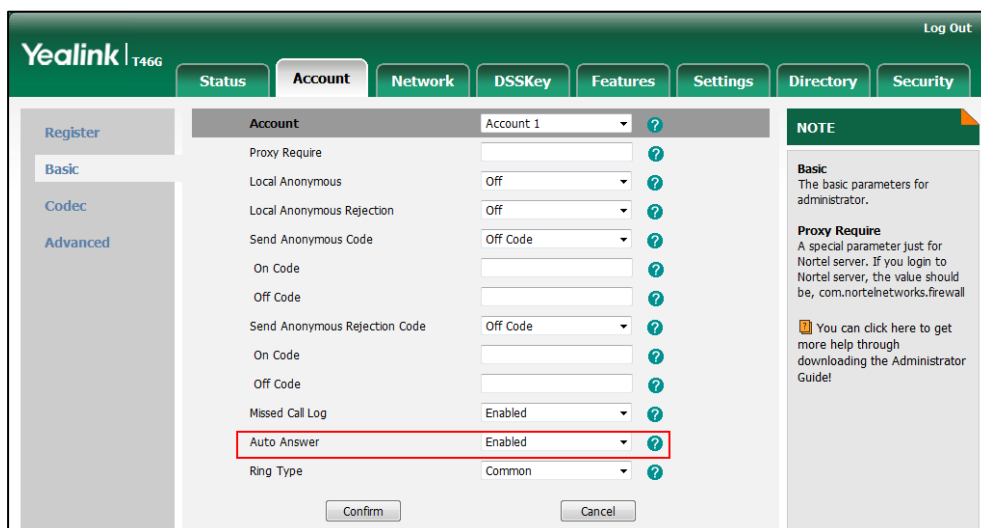
Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.auto_answer	0 or 1	0
<p>Description: Enables or disables auto answer feature for account X.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone can automatically answer an incoming call. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Note: The IP phone cannot automatically answer the incoming call during a call even if auto answer is enabled.</p> <p>Web User Interface: Account->Basic->Auto Answer</p>		

Parameters	Permitted Values	Default
Phone User Interface: Menu->Features ->Auto Answer->Line X->Auto Answer		
features.auto_answer_delay (X ranges from 1 to 16)	Integer from 1 to 4	1
<p>Description: Configures the delay time (in seconds) before the IP phone automatically answers an incoming call.</p> <p>Web User Interface: Features->General Information->Auto-Answer Delay (1~4s)</p> <p>Phone User Interface: None</p>		

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**.

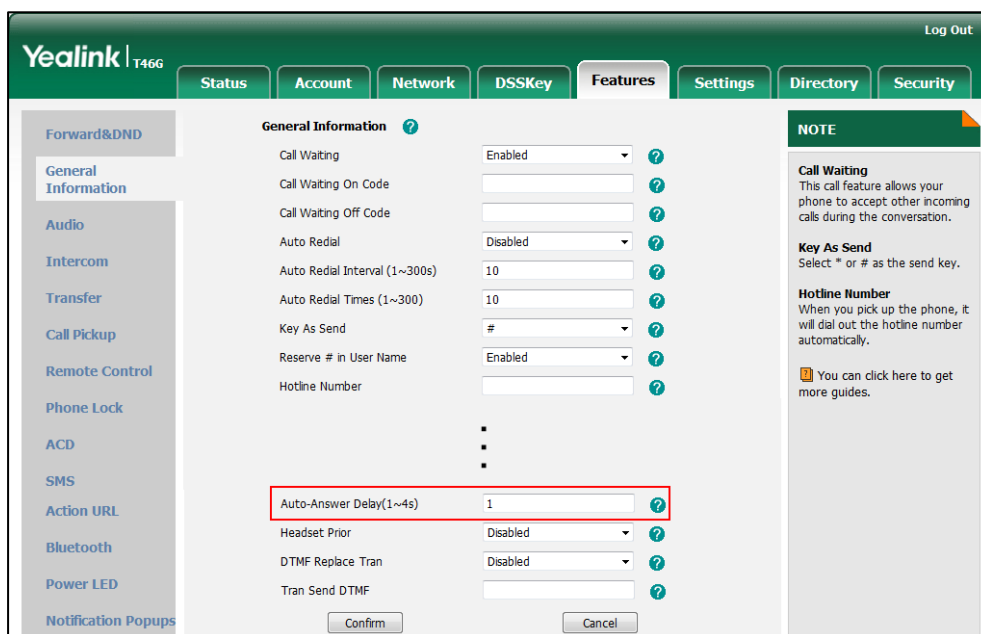


5. Click **Confirm** to accept the change.

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

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

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



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

To configure auto answer via phone user interface:

1. Press **Menu->Features->Auto Answer**.
2. Select the desired line and then press the **Enter** soft key.
3. Press **←** or **→**, or the **Switch** soft key to select the desired value from the **Auto Answer** field.
4. Press the **Save** soft key to accept the change.

Call Completion

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. Parameter: features.call_completion_enable
Local	Web User Interface	Configure call completion. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load
	Phone User Interface	Configure call completion.

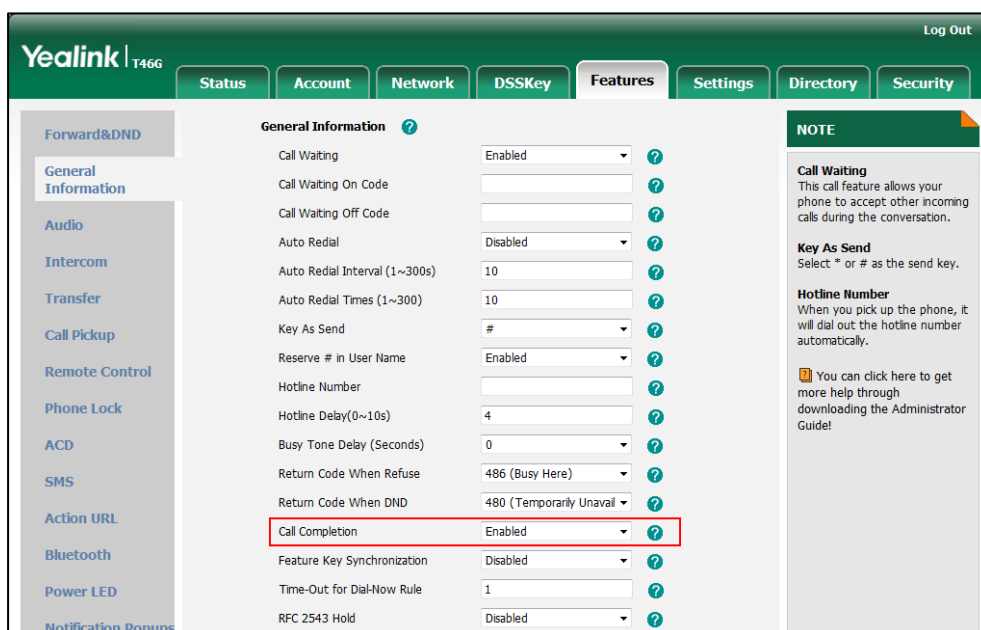
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.call_completion_enable	0 or 1	0
<p>Description:</p> <p>Enables or disables call completion feature. If a user places a call and the callee is temporarily unavailable to answer the call, call completion feature allows notifying the user when the callee becomes available to receive a call.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the caller is notified when the callee becomes available to receive a call.</p> <p>Web User Interface: Features->General Information->Call Completion</p> <p>Phone User Interface: Menu->Features->Others->Call Completion</p>		

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->Others->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-T46G 28.73.0.10
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 call 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. Parameters: account.X.anonymous_call account.X.send_anonymous_code account.X.anonymous_call_oncode account.X.anonymous_call_offcode
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.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.anonymous_call	0 or 1	0
Description: Enables or disables anonymous call feature for account X. 0-Disabled 1-Enabled 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.		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Basic->Local Anonymous</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X ->Local Anonymous</p>		
account.X.send_anonymous_code	0 or 1	0
<p>Description: Configures the IP phone to send anonymous on/off code to activate/deactivate the server-side anonymous call feature for account X.</p> <p>0-Off Code 1-On Code</p> <p>If it is set to 0 (Off Code), the IP phone will send off code to deactivate the server-side anonymous call feature when you enable or disable local anonymous feature on the phone.</p> <p>If it is set to 1 (On Code), the IP phone will send on code to activate the server-side anonymous call feature when you enable or disable local anonymous feature on the phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Basic->Send Anonymous Code</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X->Send Anonymous Code</p>		
account.X.anonymous_call_oncode	String within 32 characters	Blank
<p>Description: Configures the anonymous call on code to activate the server-side anonymous call feature for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p>		

Parameters	Permitted Values	Default
<p>Example: account.1.anonymous_call_oncode = *72</p> <p>Note: It works only if the value of the parameter "account.X.send_anonymous_code" is set to 1 (On Code).</p> <p>Web User Interface: Account->Basic->Send Anonymous Code->On Code</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X ->On Code</p>		
account.X.anonymous_call_offcode	String within 32 characters	Blank
<p>Description: Configures the anonymous call off code to deactivate the server-side anonymous call feature for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.anonymous_call_offcode = *73</p> <p>Note: It works only if the value of the parameter "account.X.send_anonymous_code" is set to 0 (Off Code).</p> <p>Web User Interface: Account->Basic->Send Anonymous Code->Off Code</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X ->Off Code</p>		

To configure the 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. (Optional.) 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.

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

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected, and the 'Account 1' configuration page is displayed. The 'Local Anonymous' field is set to 'On' and the 'Send Anonymous Code' field is set to 'Off Code'. Both fields are highlighted with a red box. The 'On Code' field contains '*72' and the 'Off Code' field contains '*73'. Other fields include 'Proxy Require', 'Local Anonymous Rejection', 'Send Anonymous Rejection Code', 'On Code', 'Off Code', 'Missed Call Log', 'Auto Answer', and 'Ring Type'. A 'NOTE' section on the right provides information about the 'Proxy Require' parameter.

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

To configure the anonymous call via phone user interface:

- Press **Menu->Features->Anonymous**.
- Select the desired line and then press **Enter** soft key.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Local Anonymous** field.
- (Optional.) Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Send Anonymous Code** field.
- (Optional.) Enter the anonymous call on code in the **On Code** field.
- (Optional.) Enter the anonymous call off code in the **Off Code** field.
- Press the **Save** soft key to 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 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. Send Anonymous Rejection Code feature allows IP phones to send anonymous call rejection on/off code to the server.

Procedure

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

Configuration File	<MAC>.cfg	Configure anonymous call rejection. Parameters: account.X.reject_anonymous_call account.X.send_anonymous_rejection_code account.X.anonymous_reject_oncode account.X.anonymous_reject_offcode
Local	Web User Interface	Configure anonymous call rejection. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-basic&q=load&acc=0
	Phone User Interface	Configure anonymous call rejection.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.reject_anonymous_call	0 or 1	0
<p>Description: Enables or disables anonymous call rejection feature for account X.</p> <p>0-Disabled 1-Enabled</p> <p>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".</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Basic->Local Anonymous Rejection</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X->Local Anonymous Rejection</p>		
account.X.send_anonymous_rejection_code	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Configures the IP phone to send anonymous rejection on/off code to activate/deactivate the server-side anonymous rejection feature for account X.</p> <p>0-Off Code 1-On Code</p> <p>If it is set to 0 (Off Code), the IP phone will send reject off code to deactivate the server-side anonymous rejection feature when you enable or disable anonymous rejection feature on the phone.</p> <p>If it is set to 1 (On Code), the IP phone will send reject on code to activate the server-side anonymous rejection feature when you enable or disable anonymous rejection feature on the phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Basic->Send Anonymous Rejection Code</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X->Send Anonymous Rejection Code</p>		
<p>account.X.anonymous_reject_oncode</p>	<p>String within 32 characters</p>	<p>Blank</p>
<p>Description: Configures the anonymous call rejection on code to activate the server-side anonymous call rejection feature for account X. The IP phone will send the anonymous call rejection on code to the server when you activate anonymous call rejection feature for account X on the IP phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.anonymous_reject_oncode = *74</p> <p>Web User Interface: Account->Basic->Send Anonymous Rejection Code->On Code</p> <p>Phone User Interface: Menu->Features->Anonymous->Line X->Send Anonymous Rejection Code->On Code</p>		

Parameters	Permitted Values	Default
account.X.anonymous_reject_offcode	String within 32 characters	Blank
<p>Description:</p> <p>Configures the anonymous call rejection off code to deactivate the server-side anonymous call rejection feature for account X. The IP phone will send the anonymous call rejection off code to the server when you deactivate anonymous call rejection feature for account X on the IP phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.anonymous_reject_offcode = *75</p> <p>Web User Interface:</p> <p>Account->Basic->Send Anonymous Rejection Code->Off Code</p> <p>Phone User Interface:</p> <p>Menu->Features->Anonymous->Line X->Send Anonymous Rejection Code->Off Code</p>		

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.) Select the desired value from the pull-down list of **Send Anonymous Rejection Code**.
6. (Optional.) Enter the anonymous call rejection on code in the **On Code** field.

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

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected, and the settings for 'Account 1' are displayed. The 'Local Anonymous Rejection' dropdown is set to 'On', and the 'Send Anonymous Rejection Code' dropdown is set to 'Off Code'. The 'On Code' field contains '*74' and the 'Off Code' field contains '*75'. A red box highlights the 'Local Anonymous Rejection' and 'Send Anonymous Rejection Code' sections. A 'NOTE' sidebar on the right provides information about the Proxy Require parameter.

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

To configure anonymous call rejection via phone user interface:

- Press **Menu->Features->Anonymous**.
- Select the desired line and then press **Enter** soft key.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Local Anonymous Rejection** field.
- (Optional.) Press **◀** or **▶** to select the desired value from the **Send Anonymous Rejection Code** field.
- (Optional.) Enter the anonymous call rejection on code in the **On Code** field.
- (Optional.) Enter the anonymous call rejection off code in the **Off Code** field.
- Press the **Save** soft key to accept the change.

Do Not Disturb

Do Not Disturb (DND) allows IP phones to ignore incoming calls. DND can be configured on a phone or per-line basis depending on the DND mode. Two DND modes:

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

A user can activate or deactivate DND using the DND soft key or DND key. The DND configurations on IP phones may be overridden by the server settings. The server-side DND feature disables the local DND and call forward settings. If the server-side DND feature is enabled on any of the IP phone's registrations, the other registrations are not affected. For more information on call forward, refer to [Call Forward](#) on page 188.

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 IP phones. The caller's LCD screen displays the received return code.

Procedure

DND can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	<p>Configure DND in the custom mode.</p> <p>Parameters: account.X.dnd.enable account.X.dnd.on_code account.X.dnd.off_code</p>
	<y0000000000xx>.cfg	<p>Assign a DND key.</p> <p>Parameters: linekey.X.type/ programablekey.X.type</p> <p>Configure the DND mode.</p> <p>Parameter: features.dnd_mode</p> <p>Configure DND in the phone mode.</p> <p>Parameters: features.dnd.enable features.dnd.on_code features.dnd.off_code</p> <p>Specify return code and reason of the SIP response message.</p> <p>Parameter: features.dnd_refuse_code</p>
Local	Web User Interface	<p>Assign a DND key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p> <p>Configure DND.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?">http://<phoneIPAddress>/servlet?</p>

		<p>p=features-forward&q=load</p> <p>Specify return code and reason of the SIP response message.</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>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.dnd.enable	0 or 1	0
<p>Description:</p> <p>Enables or disables DND feature for account X when the DND mode is configured as Custom.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone will reject incoming calls on account X.</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G).</p> <p>X ranges from 1 to 12 (for SIPT42G).</p> <p>X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface:</p> <p>Features->Forward& DND->DND ->DND Status</p> <p>Phone User Interface:</p> <p>Menu->Features->DND->Account->DND Enable.</p>		
account.X.dnd.on_code	String within 32 characters	Blank
<p>Description:</p> <p>Configures the DND on code to activate the server-side DND feature for account X when the DND mode is configured as Custom. The IP phone will send the DND on code to the server when you activate DND feature for account X on the IP phone.</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G).</p> <p>X ranges from 1 to 12 (for SIPT42G).</p> <p>X ranges from 1 to 6 (for SIPT41P).</p> <p>Example:</p> <p>account.1.dnd.on_code = *73</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: Features->Forward& DND->DND->DND On Code</p> <p>Phone User Interface: Menu->Features->DND->Account->On Code</p>		
account.X.dnd.off_code	String within 32 characters	Blank
<p>Description: Configures the DND off code to deactivate the server-side DND feature for account X when the DND mode is configured as Custom. The IP phone will send the DND off code to the server when you deactivate DND feature for account X on the IP phone. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.dnd.off_code = *74</p> <p>Web User Interface: Features->Forward& DND->DND ->DND Off Code</p> <p>Phone User Interface: Menu->Features->DND->Account->Off Code</p>		
features.dnd_mode	0 or 1	0
<p>Description: Configures the DND mode for the IP phone. 0-Phone 1-Custom If it is set to 0 (Phone), DND feature is effective for the IP phone. If it is set to 1 (Custom), you can configure DND feature for each account.</p> <p>Web User Interface: Features->Forward& DND->DND->Mode</p> <p>Phone User Interface: None</p>		
features.dnd.enable	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables DND feature when the DND mode is configured as Phone. 0-Disabled 1-Enabled If it is set to 1 (Enabled), the IP phone will reject incoming calls on all accounts.</p> <p>Web User Interface: Features->Forward& DND->DND->DND Status</p> <p>Phone User Interface: Menu->Features->DND->DND Enable</p>		
features.dnd.on_code	String within 32 characters	Blank
<p>Description: Configures the DND on code to activate the server-side DND feature when the DND mode is configured as Phone. The IP phone will send the DND on code to the server when you activate DND feature on the IP phone.</p> <p>Example: features.dnd.on_code = *71</p> <p>Web User Interface: Features->Forward& DND->DND->On Code</p> <p>Phone User Interface: Menu->Features->DND->On Code</p>		
features.dnd.off_code	String within 32 characters	Blank
<p>Description: Configures the DND off code to deactivate the server-side DND feature when the DND mode is configured as Phone. The IP phone will send the DND off code to the server when you deactivate DND feature on the IP phone.</p> <p>Example: features.dnd.off_code = *72</p> <p>Web User Interface: Features->Forward& DND->DND->DND Off Code</p> <p>Phone User Interface: Menu->Features->DND->Off Code</p>		
features.dnd_refuse_code	404, 480 or 486	480

Parameters	Permitted Values	Default
<p>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.</p> <p>404-No Found 480-Temporarily Unavailable 486-Busy Here</p> <p>If it is set to 486 (Busy Here), the caller's LCD screen will display the reason "Busy Here" when the callee enables DND.</p> <p>Web User Interface: Features->General Information->Return Code When DND</p> <p>Phone User Interface: None</p>		

DND Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

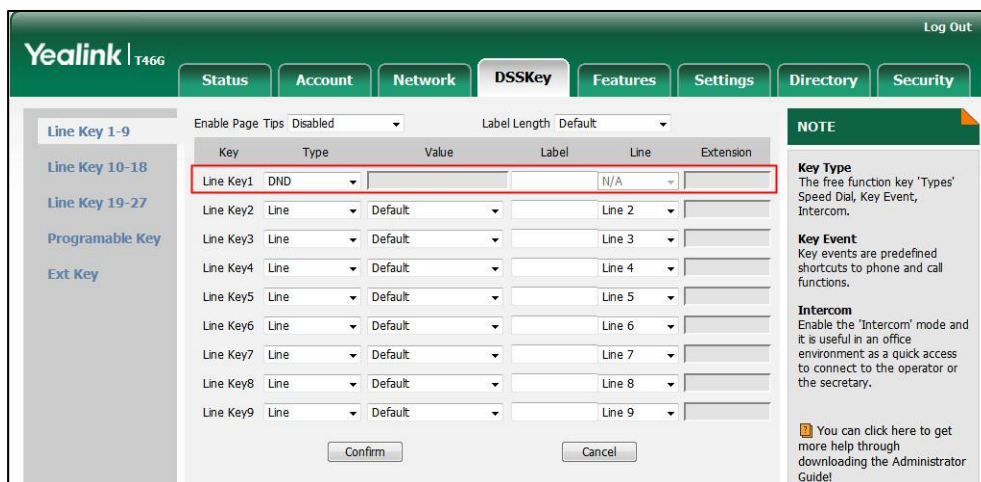
Parameter	Permitted Values	Default
linekey.X.type/programablekey.X.type	5	Refer to the following content
<p>Description: Configures a DSS key as a DND key on the IP phone. The digit 5 stands for the key type DND.</p> <p>For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: linekey.2.type = 5</p> <p>Default:</p>		

Parameter	Permitted Values	Default
<p>For line keys:</p> <p>For SIP-T48G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programmable keys:</p> <p>For SIP-T48G/T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p>		

Parameter	Permitted Values	Default
When X=7, the default value is 51 (Switch Account Up).		
When X=8, the default value is 52 (Switch Account Down).		
When X=9, the default value is 33 (Status).		
When X=10, the default value is 0 (NA).		
When X=13, the default value is 0 (NA).		
Web User Interface:		
DSSKey->Line Key X / Programmable Key->Type		
Phone User Interface:		
Menu->Features->DSS Keys->Line Key X->Type		

To configure a DND key via web user interface:

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

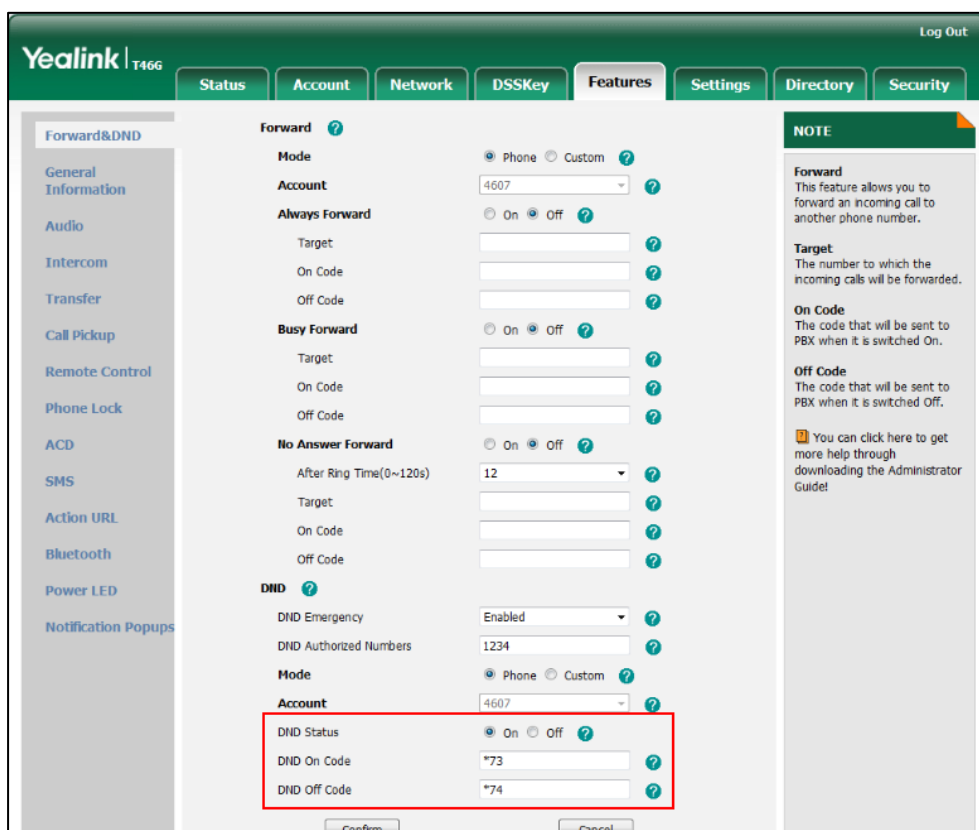


3. Click **Confirm** to accept the change.

To configure the DND feature via web user interface:

1. Click on **Features->Forward & DND**.
2. In the **DND** 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 **DND Status** field.
 - 2) (Optional.) Enter the DND on code in the **DND On Code** field.

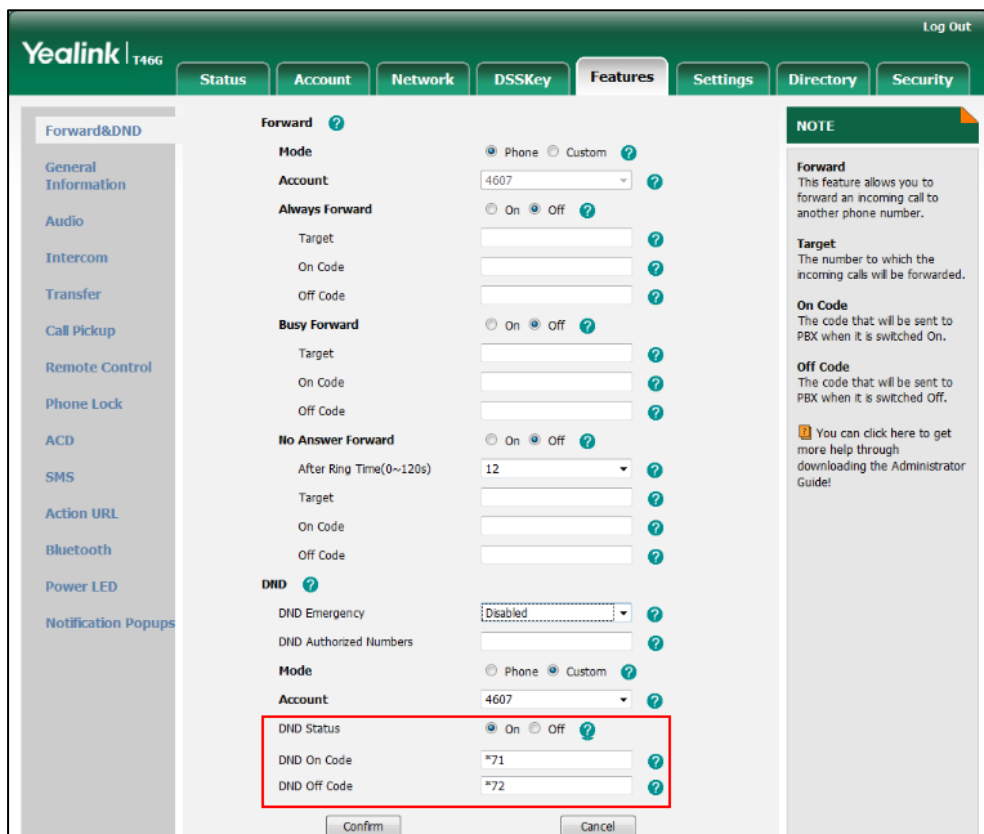
3) (Optional.) Enter the DND off code in the **DND Off Code** field.



b) If you mark the **Custom** radio box:

- 1) Select the desired account from the pull-down list of **Account**.
- 2) Mark the desired value in the **DND Status** field.
- 3) (Optional.) Enter the DND on code in the **DND On Code** field.

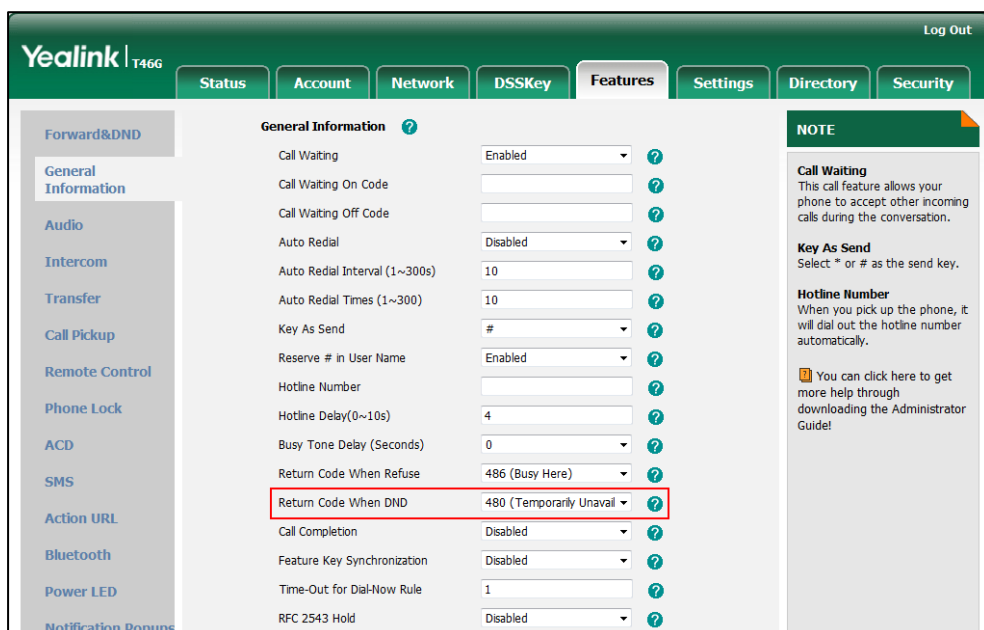
4) (Optional.) Enter the DND off code in the **DND Off Code** field.



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





To specify the return code 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**.
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. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. 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 the accounts registered on the IP phone.
2. Press  or  to select the desired account and press the **Enter** soft key.
3. Press  or  to select **On** to activate DND.
4. Press the **Save** soft key to accept the change.

To configure DND in the custom mode for all accounts 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 the accounts registered on the IP phone.
2. Press the **All On** soft key to activate DND for all accounts.
3. 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 the busy tone delay feature. Parameter: features.busy_tone_delay
Local	Web User Interface	Configure the busy tone delay feature.

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

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.busy_tone_delay	0, 3 or 5	0

Description:
 Configures the duration time (in seconds) for the busy tone.
 When one party releases the call, a busy tone is audible to the other party indicating that the call connection breaks.

0-0s
3-3s
5-5s

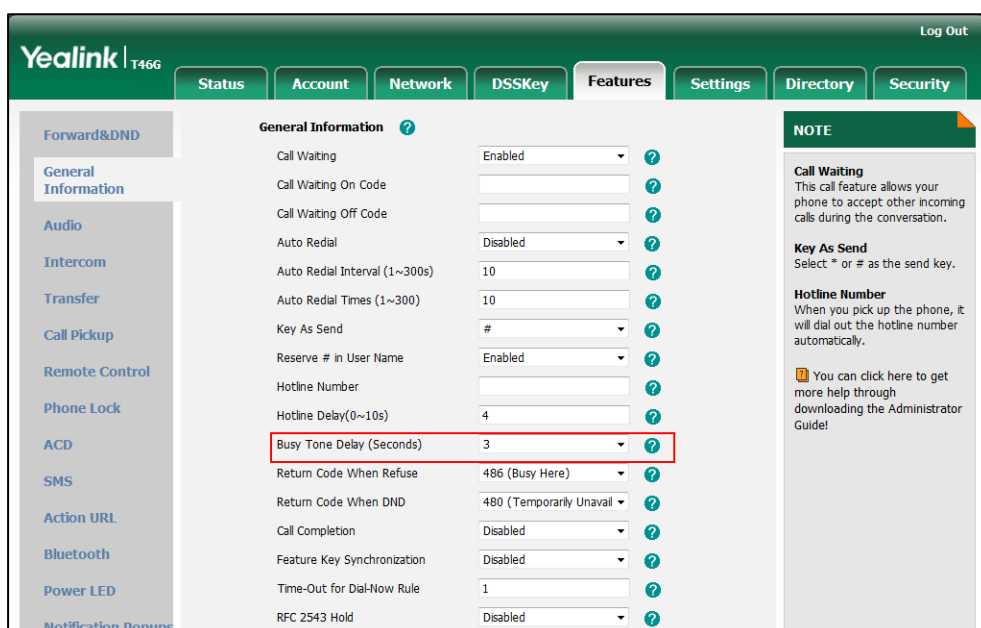
If it is set to 3 (3s), a busy tone is audible for 3 seconds on the IP phone.

Web User Interface:
 Features->General Information->Busy Tone Delay (Seconds)

Phone User Interface:
 None

To configure busy tone delay via web user interface:

1. Click on **Features->General Information**.
2. 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 call rejection. The caller's LCD screen displays the reason according to the return code received. Available return codes and reasons are:

- 404 (Not Found)
- 480 (Temporarily Unavailable)
- 486 (Busy Here)

Procedure

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

Configuration File	<y0000000000xx>.cfg	Configure the return code when refusing a call. Parameter: features.normal_refuse_code
Local	Web User Interface	Configure the return code when refusing a call. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

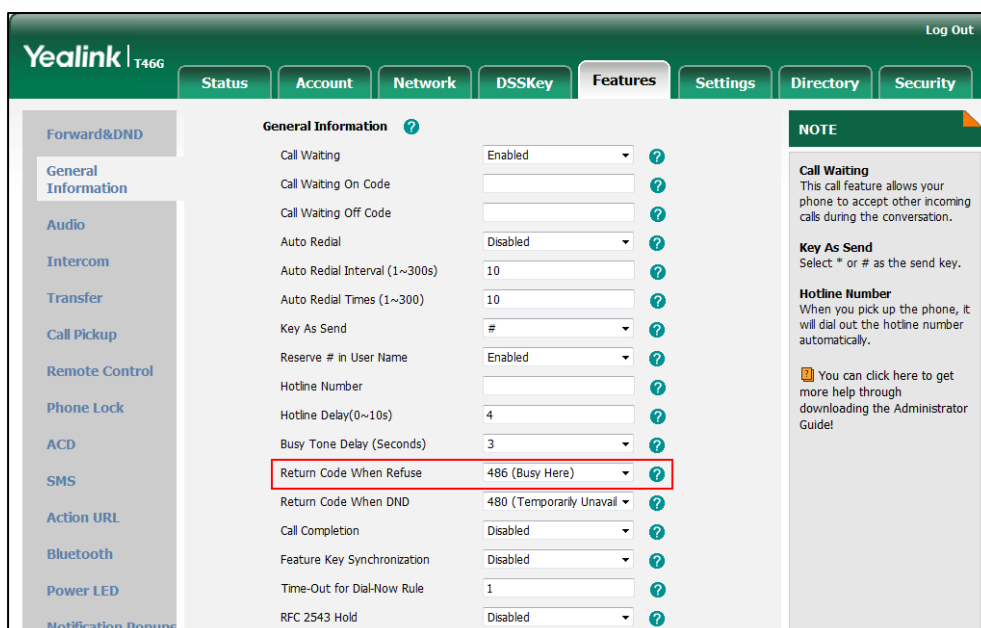
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.normal_refuse_code	404, 480 or 486	486
<p>Description:</p> <p>Configures a return code and reason of SIP response messages when the IP phone rejects an incoming call. A specific reason is displayed on the caller's phone LCD screen.</p> <p>404-No Found</p> <p>480-Temporarily Unavailable</p> <p>486-Busy Here</p> <p>If it is set to 486 (Busy Here), the caller's phone LCD screen will display the message "Busy Here" when the callee rejects the incoming call.</p> <p>Web User Interface:</p>		

Parameter	Permitted Values	Default
Features->General Information->Return Code When Refuse		
Phone User Interface:		
None		

To specify the return code 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. Parameter: phone_setting.is_deal180
Local	Web User Interface	Configure 180 ring workaround. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

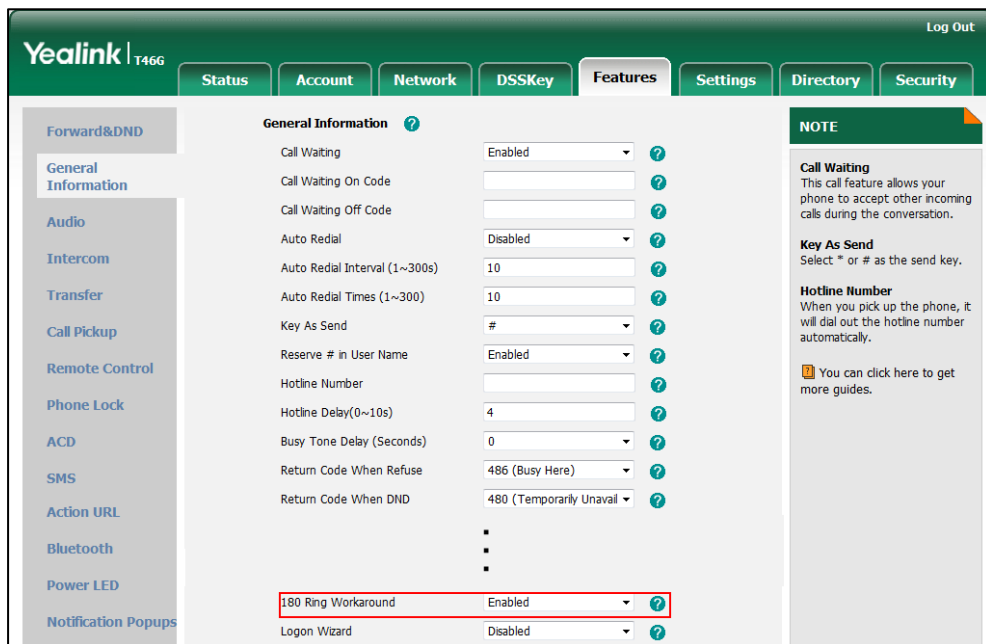
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
phone_setting.is_deal180	0 or 1	1
<p>Description: Enables or disables the IP phone to deal with the 180 SIP message received after the 183 SIP message.</p> <p>0-Disabled 1-Enabled</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> <p>Web User Interface: Features->General Information->180 Ring Workaround</p> <p>Phone User Interface: None</p>		

To configure 180 ring workaround via web user interface:

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

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



- 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. Parameter: sip.use_out_bound_in_dialog
Local	Web User Interface	Specify whether to use outbound proxy in a dialog. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
sip.use_out_bound_in_dialog	0 or 1	1

Description:
 Enables or disables the IP phone to keep sending SIP requests to the outbound proxy server in a dialog.

0-Disabled
1-Enabled

If it is set to 1 (Enabled), all the SIP request messages from the IP phone will be forced to send to the outbound proxy server in a dialog.

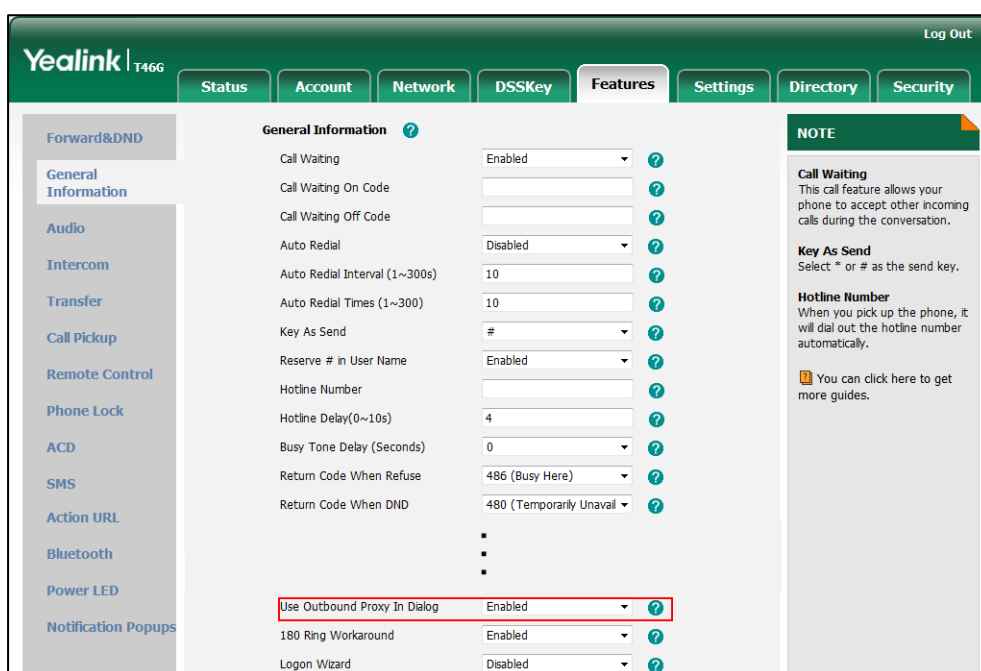
Note: If you change this parameter, the IP phone will reboot to make the change take effect.

Web User Interface:
 Features->General Information->Use Outbound Proxy In Dialog

Phone User Interface:
 None

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

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



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

SIP Session Timer

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 retransmitting time of any SIP request message. The re-transmitting and doubling of T1 will continue until the retransmitting time reaches the T2 value. Timer T4 represents the time the network will take to clear messages between the SIP client and server. 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. Parameters: account.X.advanced.timer_t1 account.X.advanced.timer_t2 account.X.advanced.timer_t4
Local	Web User Interface	Configure SIP session timer. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.advanced.timer_t1	Float from 0.5 to10	0.5
<p>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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->SIP Session Timer T1 (0.5~10s)</p> <p>Phone User Interface:</p>		

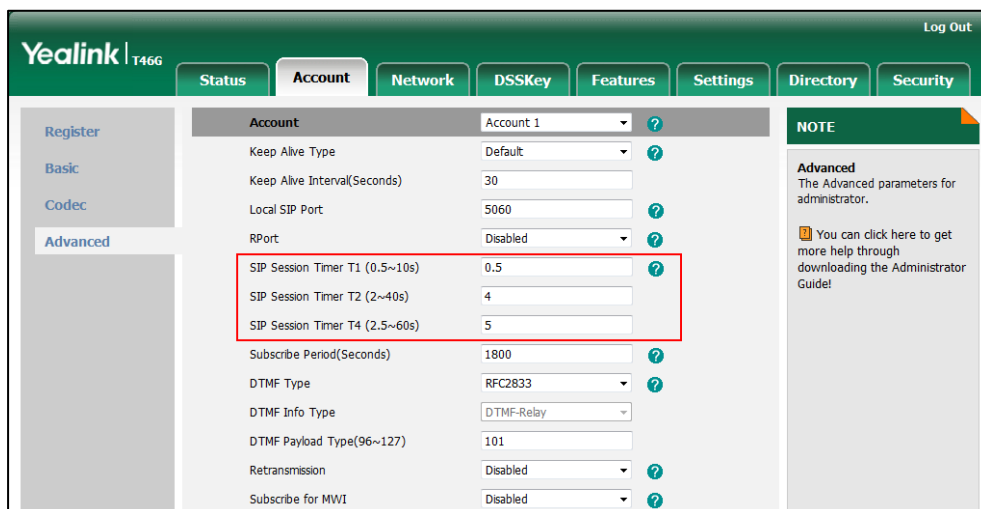
Parameters	Permitted Values	Default
None		
account.X.advanced.timer_t2	Float from 2 to 40	4
<p>Description: Configures the session timer T2 (in seconds) for account X. T2 represents the maximum retransmit interval for non-INVITE requests and INVITE responses. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->SIP Session Timer T2 (2~40s)</p> <p>Phone User Interface: None</p>		
account.X.advanced.timer_t4	Float from 2.5 to 60	5
<p>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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->SIP Session Timer T4 (2.5~60s)</p> <p>Phone User Interface: None</p>		

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.



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 phones send an INVITE request with HOLD SDP to request remote parties to stop sending media and to inform them that they are being held. IP phones support two call hold methods, one is RFC 3264, which sets the “a” (media attribute) in the SDP to sendonly, recvonly or inactive (e.g., a=sendonly). The other is RFC 2543, which sets the “c” (connection addresses for the media streams) in the SDP to zero (e.g., c=0.0.0.0). Call hold tone allows IP phones to play a warning tone at regular intervals when there is a call on hold. The warning tone is played through the speakerphone.

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

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the call hold tone and call hold tone delay.</p> <p>Parameters:</p> <p>features.play_hold_tone.enable</p>
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		<p>features.play_hold_tone.delay</p> <p>Specify whether RFC 2543 (c=0.0.0.0) outgoing hold signaling is used.</p> <p>Parameters:</p> <p>sip.rfc2543_hold</p>
	<MAC>.cfg	<p>Configure MoH on a per-line basis.</p> <p>Parameter:</p> <p>account.X.music_server_uri</p>
Local	Web User Interface	<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> <p>Configure MoH on a per-line basis.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>

Details of Configuration Parameters:

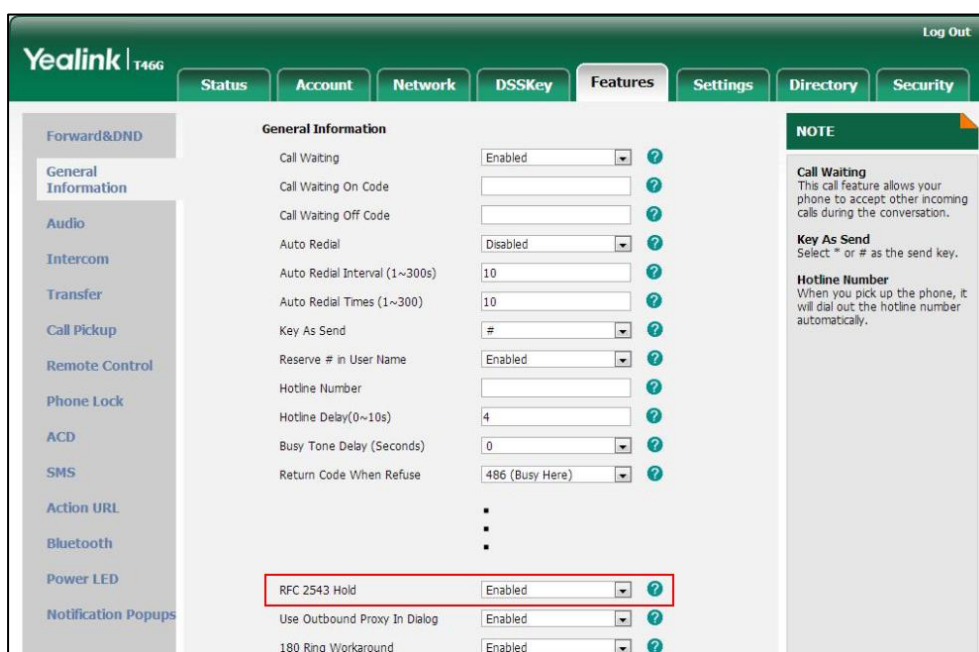
Parameters	Permitted Values	Default
features.play_hold_tone.enable	0 or 1	1
<p>Description:</p> <p>Enables or disables the IP phone to play a tone when there is a call on hold.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>Web User Interface:</p> <p>Features->General Information->Play Hold Tone</p> <p>Phone User Interface:</p> <p>None</p>		

Parameters	Permitted Values	Default
features.play_hold_tone.delay	Integer from 3 to 3600	30
<p>Description: Configures the interval (in seconds) at which the IP phone plays a hold tone. If it is set to 30 (30s), the IP phone will play a hold tone every 30 seconds when there is a hold call on the IP phone.</p> <p>Note: It works only if the value of the parameter “features.play_hold_tone.enable” is set to 1 (Enabled).</p> <p>Web User Interface: Features->General Information->Play Hold Tone Delay</p> <p>Phone User Interface: None</p>		
sip.rfc2543_hold	0 or 1	0
<p>Description: Enables or disables the IP phone to use RFC 2543 (c=0.0.0.0) outgoing hold signaling.</p> <p>0-Disabled 1-Enabled</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> <p>Web User Interface: Features->General Information->RFC 2543 Hold</p> <p>Phone User Interface: None</p>		
account.X.music_server_uri	SIP URI within 256 characters	Blank

Parameters	Permitted Values	Default
<p>Description:</p> <p>Configures the address of the Music On Hold server for account X. 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.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.music_server_uri = sip:moh@sip.com</p> <p>Note: The DNS query in this parameter only supports A query.</p> <p>Web User Interface:</p> <p>Account->Advanced->Music Server URI</p> <p>Phone User Interface:</p> <p>None</p>		

To configure call hold method via web user interface:

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

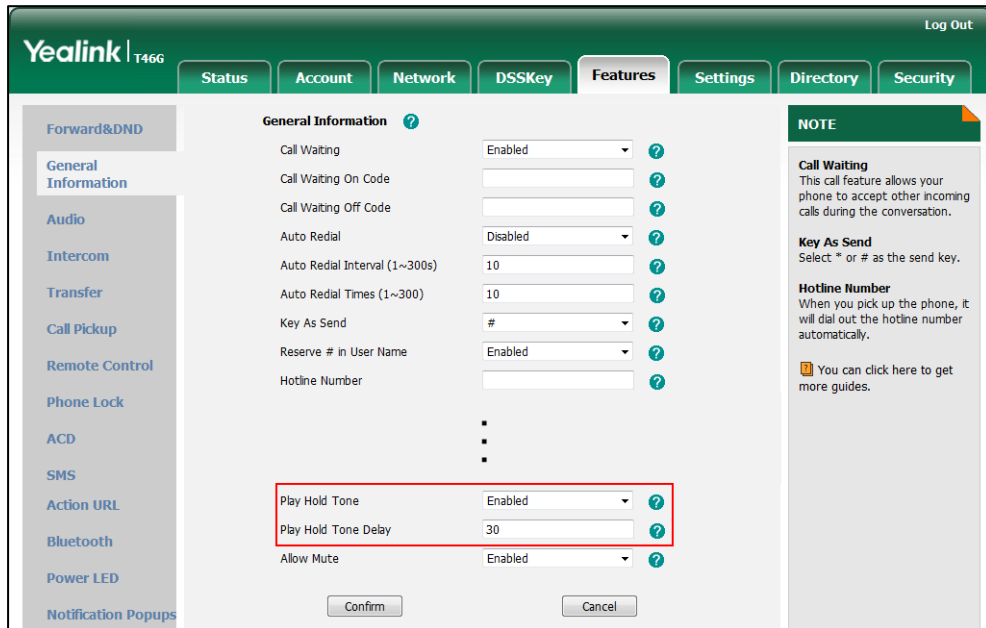


3. Click **Confirm** to accept the change.

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

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

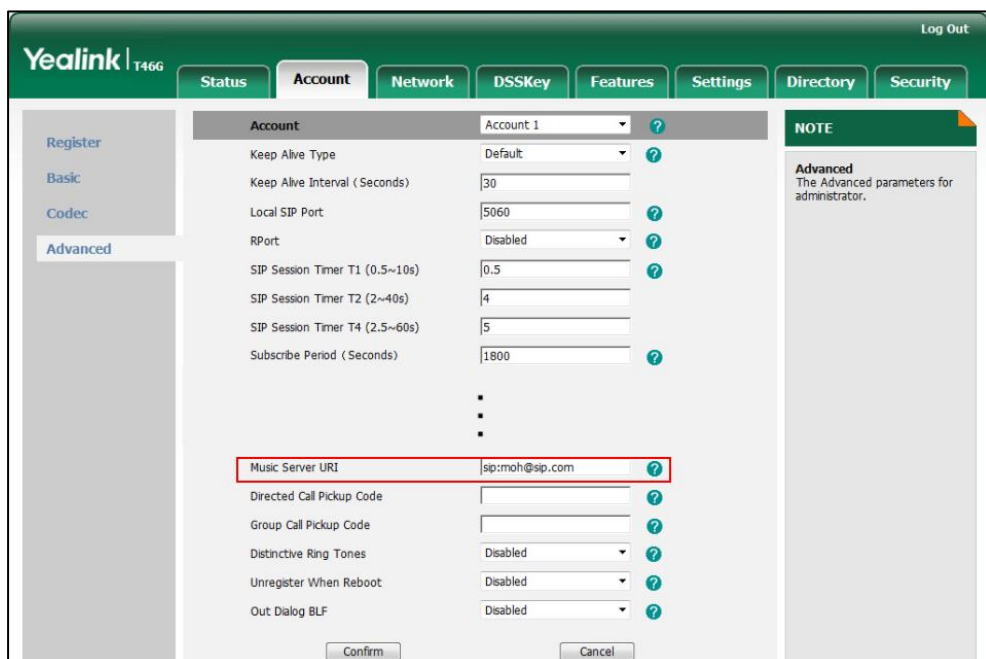
2. Select the desired value from the pull-down list of **Play Hold Tone**.
3. Enter the desired time in the **Play Hold Tone Delay** field.



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

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



5. 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 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. Parameters: account.X.session_timer.enable account.X.session_timer.expires account.X.session_timer.refresher
Local	Web User Interface	Configure session timer. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc= 0

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.session_timer.enable	0 or 1	0
<p>Description: Enables or disables the session timer for account X.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), IP phone will send periodic re-INVITE requests to refresh the session during a call.</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
Account->Advanced->Session Timer Phone User Interface: None		
account.X.session_timer.expires	Integer from 30 to 7200	1800
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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Example: account.1.session_timer.expires = 1800 Web User Interface: Account->Advanced->Session Expires(30~7200s) Phone User Interface: None		
account.X.session_timer.refresher	0 or 1	0
Description: Configures the session timer refresher for account X. 0-UAC 1-UAS 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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Web User Interface: Account->Advanced->Session Refresher Phone User Interface: None		

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**.

The screenshot shows the Yealink web interface for configuring an account. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'Session Timer' section is highlighted with a red box. The configuration for this section is as follows:

Parameter	Value
Session Timer	Enabled
Session Expires(30~7200s)	1800
Session Refresher	UAC

Other visible settings include: Keep Alive Type: Default, Keep Alive Interval (Seconds): 30, Local SIP Port: 5062, RPort: Disabled, SIP Session Timer T1 (0.5~10s): 0.5, Send user=phone: Disabled, Directed Call Pickup Code: (empty), Group Call Pickup Code: (empty), Distinctive Ring Tones: Disabled, Unregister When Reboot: Disabled, and Out Dialog BLF: Disabled. There are 'Confirm' and 'Cancel' buttons at the bottom of the form.

7. 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 calls immediately.
- **Busy Forward** -- Forward the incoming call when the IP phone or the specified account is busy.
- **No Answer Forward** -- Forward the incoming call after a period of ring time.

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

- **Phone** (default): Call forward feature is effective for all accounts on the IP phone.

- **Custom:** Call forward feature can be configured for each or all accounts.

The server-side call forward settings disable the local call forward settings. If the server-side call forward feature is enabled on any of the IP phone's registrations, the other registrations are not affected. DND activated on the IP phone disables the local no answer forward settings.

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

IP phones support the redirected call information sent by the SIP server with Diversion header, per draft-levy-sip-diversion-08, or History-info header, per RFC 4244. The Diversion/History-info header is used to inform the IP phone of a call's history. For example, when a phone has been set to enable call forward, the Diversion/History-info header allows the receiving phone to indicate who the call was from, and from which phone number it was forwarded.

Forward International

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

Procedure

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

<p>Configuration File</p>	<p><MAC>.cfg</p>	<p>Configure call forward in custom mode.</p> <p>Parameters:</p> <p>account.X.always_fwd.enable account.X.always_fwd.target account.X.always_fwd.on_code account.X.always_fwd.off_code account.X.busy_fwd.enable account.X.busy_fwd.target account.X.busy_fwd.on_code account.X.busy_fwd.off_code account.X.timeout_fwd.enable account.X.timeout_fwd.target account.X.timeout_fwd.timeout account.X.timeout_fwd.on_code account.X.timeout_fwd.off_code</p>
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	<p><y0000000000xx>.cf g</p>	<p>Configure the call forward mode.</p> <p>Parameter:</p> <p>features.fwd_mode</p> <p>Configure call forward in phone mode.</p> <p>Parameters:</p> <p>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 features.fwd_diversion_enable</p> <p>Configure forward international.</p> <p>Parameter:</p> <p>forward.international.enable</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure call forward.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-forward&q=load</p> <p>Configure forward international.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
	<p>Phone User Interface</p>	<p>Configure call forward.</p> <p>Configure forward international.</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.always_fwd.enable	0 or 1	0
<p>Description: Enables or disables always forward feature for account X when the call forward mode is configured as Custom.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls to the account X are forwarded to the destination number immediately.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Features->Forward& DND->Forward->Always Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Always Forward ->Always Forward</p>		
account.X.always_fwd.target	String within 32 characters	Blank
<p>Description: Configures the destination number of the always forward for account X when the call forward mode is configured as Custom.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.always_fwd.target = 3601</p> <p>Web User Interface: Features->Forward& DND->Forward->Always Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Always Forward ->Forward To</p>		
account.X.always_fwd.on_code	String within 32 characters	Blank
<p>Description: Configures the always forward on code to activate the server-side always forward</p>		

Parameters	Permitted Values	Default
<p>feature for account X when the call forward mode is configured as Custom. The IP phone will send the always forward on code and the pre-configured destination number to the server when you activate always forward feature for account X on the IP phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.always_fwd.on_code = *72</p> <p>Web User Interface:</p> <p>Features->Forward& DND->Forward->Always Forward->On Code</p> <p>Phone User Interface:</p> <p>Menu->Features->Call Forward->Account ->Always Forward ->On Code</p>		
account.X.always_fwd.off_code	String within 32 characters	Blank
<p>Description:</p> <p>Configures the always forward off code to deactivate the server-side always forward feature for account X when the call forward mode is configured as Custom. The IP phone will send the always forward off code to the server when you deactivate always forward feature for account X on the IP phone.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.busy_fwd.off_code = *73</p> <p>Web User Interface:</p> <p>Features->Forward& DND->Forward->Always Forward ->Off Code</p> <p>Phone User Interface:</p> <p>Menu->Features->Call Forward->Account ->Always Forward ->Off Code</p>		
account.X.busy_fwd.enable	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables busy forward feature for account X when the call forward mode is configured as Custom.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls to the account X are forwarded to the destination number when the callee is busy.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Features->Forward& DND->Forward->Busy Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Busy Forward->Busy Forward</p>		
account.X.busy_fwd.target	String within 32 characters	Blank
<p>Description: Configures the destination number of the busy forward for account X when the call forward mode is configured as Custom.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.busy_fwd.target = 3602</p> <p>Web User Interface: Features->Forward& DND->Forward->Busy Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Busy Forward ->Forward To</p>		
account.X.busy_fwd.on_code	String within 32 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the busy forward on code to activate the server-side busy forward feature for account X when the call forward mode is configured as Custom. The IP phone will send the busy forward on code and the pre-configured destination number to the server when you activate busy forward feature for account X on the IP phone.</p> <p>Example: account.1.busy_fwd.on_code = *74</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward ->On Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Busy Forward ->On Code</p>		
account.X.busy_fwd.off_code	String within 32 characters	Blank
<p>Description: Configures the busy forward off code to deactivate the server-side busy forward feature for account X when the call forward mode is configured as Custom. The IP phone will send the busy forward off code to the server when you deactivate busy forward feature for account X on the IP phone. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.busy_fwd.off_code = *75</p> <p>Web User Interface: Features->Forward& DND->Forward ->No Answer Forward ->Off Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->Busy Forward ->Off Code</p>		
account.X.timeout_fwd.enable	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables no answer forward feature for account X when the call forward mode is configured as Custom.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls to the account X are forwarded to the destination number after a period of ring time.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->No Answer Forward->No Answer Forward</p>		
<p>account.X.timeout_fwd.target</p>	<p>String within 32 characters</p>	<p>Blank</p>
<p>Description: Configures the destination number of the no answer forward for account X when the call forward mode is configured as Custom.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.timeout_fwd.target = 3603</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->No Answer Forward ->Forward To</p>		
<p>account.X.timeout_fwd.timeout</p>	<p>Integer from 0 to 20</p>	<p>2</p>

Parameters	Permitted Values	Default
<p>Description: Configures ring times (N) to wait before forwarding incoming calls for account X when the call forward mode is configured as Custom. Incoming calls will be forwarded when not answered after N*6 seconds. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward->After Ring Time</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->No Answer Forward ->After Ring Time</p>		
account.X.timeout_fwd.on_code	String within 32 characters	Blank
<p>Description: Configures the no answer forward on code to activate the server-side no answer forward feature for account X when the call forward mode is configured as Custom. The IP phone will send the no answer forward on code and the pre-configured destination number to the server when you activate no answer forward feature for account X on the IP phone. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.timeout_fwd.on_code = *76</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward ->On Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->No Answer Forward ->On Code</p>		
account.X.timeout_fwd.off_code	String within 32 characters	Blank
<p>Description: Configures the no answer forward off code to deactivate the server-side no answer forward feature for account X when the call forward mode is configured as Custom. The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature for account X on the IP phone.</p>		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.timeout_fwd.off_code = *77</p> <p>Web User Interface: Features->Forward& DND->Forward->No Answer Forward ->Off Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Account ->No Answer Forward ->Off Code</p>		
features.fwd_mode	0 or 1	0
<p>Description: Configures the call forward mode for the IP phone.</p> <p>0-Phone 1-Custom</p> <p>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.</p> <p>Web User Interface: Features->Forward&DND->Forward->Mode</p> <p>Phone User Interface: None</p>		
forward.always.enable	0 or 1	0
<p>Description: Enables or disables always forward feature.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls are forwarded to the destination number immediately.</p> <p>Web User Interface: Features->Forward &DND->Forward->Always Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->Always Forward->Always Forward</p>		
forward.always.target	String within 32 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the destination number the IP phone forwards all incoming calls to.</p> <p>Web User Interface: Features->Forward &DND->Always Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->Always Forward->Forward To</p>		
forward.always.on_code	String within 32 characters	Blank
<p>Description: Configures the always forward on code to activate the server-side always forward feature. The IP phone will send the always forward on code and the pre-configured destination number to the server when you activate always forward feature on the IP phone.</p> <p>Example: forward.always.on_code = *72</p> <p>Web User Interface: Features->Forward &DND->Forward->Always Forward->On Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Always Forward->On Code</p>		
forward.always.off_code	String within 32 characters	Blank
<p>Description: Configures the always forward off code to deactivate the server-side always forward feature. The IP phone will send the always forward off code to the server when you deactivate always forward feature on the IP phone.</p> <p>Example: forward.always.off_code = *73</p> <p>Web User Interface: Features->Forward &DND->Forward->Always Forward->Off Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Always Forward->Off Code</p>		
forward.busy.enable	0 or 1	0
<p>Description: Enables or disables busy forward feature.</p>		

Parameters	Permitted Values	Default
<p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls are forwarded to the destination number when the callee is busy.</p> <p>Web User Interface: Features->Forward &DND->Forward->Busy Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->Busy Forward->Busy Forward</p>		
forward.busy.target	String within 32 characters	Blank
<p>Description: Configures the destination number the IP phone forwards incoming calls to when busy.</p> <p>Example: forward.busy.target = 3602</p> <p>Web User Interface: Features->Forward &DND->Forward->Busy Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->Busy Forward->Forward To</p>		
forward.busy.on_code	String within 32 characters	Blank
<p>Description: Configures the busy forward on code to activate the server-side busy forward feature. The IP phone will send the busy forward on code and the pre-configured destination number to the server when you activate busy forward feature on the IP phone.</p> <p>Example: forward.busy.on_code = *74</p> <p>Web User Interface: Features->Forward &DND->Forward->Busy Forward->On Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Busy Forward->On Code</p>		
forward.busy.off_code	String within 32 characters	Blank
<p>Description:</p>		

Parameters	Permitted Values	Default
<p>Configures the busy forward off code to deactivate the server-side busy forward feature. The IP phone will send the busy forward off code to the server when you deactivate busy forward feature on the IP phone.</p> <p>Example: forward.busy.off_code = *75</p> <p>Web User Interface: Features->Forward &DND->Forward->Busy Forward->Off Code</p> <p>Phone User Interface: Menu->Features->Call Forward->Busy Forward->Off Code</p>		
forward.no_answer.enable	0 or 1	0
<p>Description: Enables or disables no answer forward feature.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), incoming calls are forwarded to the destination number after a period of ring time.</p> <p>Web User Interface: Features->Forward &DND->Forward->No Answer Forward->On/Off</p> <p>Phone User Interface: Menu->Features->Call Forward->No Answer Forward->No Answer Forward</p>		
forward.no_answer.target	String within 32 characters	Blank
<p>Description: Configures the destination number the IP phone forwards incoming calls to after a period of ring time.</p> <p>Example: forward.no_answer.target = 3603</p> <p>Web User Interface: Features->Forward &DND->Forward->No Answer Forward->Target</p> <p>Phone User Interface: Menu->Features->Call Forward->No Answer Forward->Forward To</p>		
forward.no_answer.timeout	Integer from 0 to 20	2

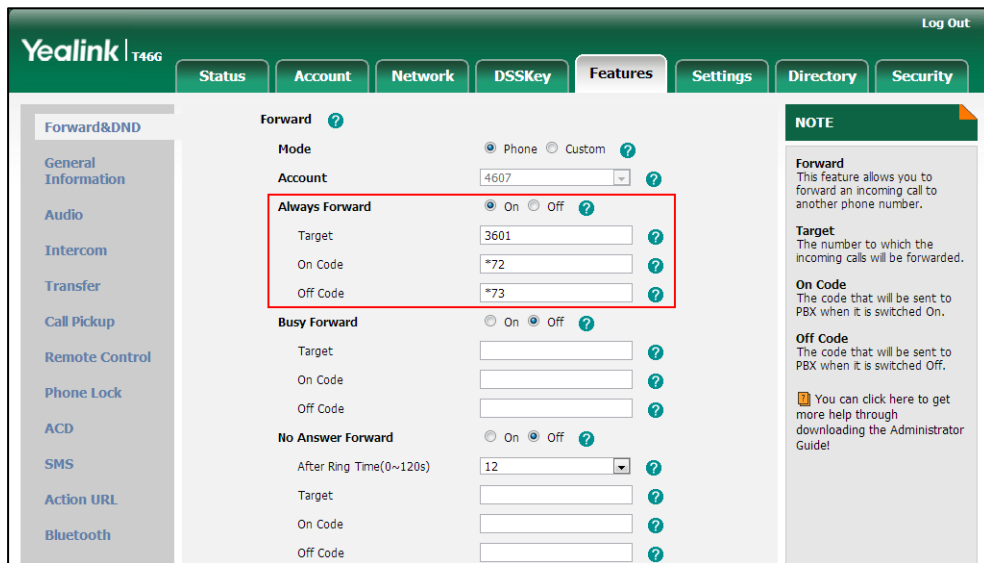
Parameters	Permitted Values	Default
<p>Description: Configures ring times (N) to wait before forwarding incoming calls. Incoming calls will be forwarded when not answered after N*6 seconds.</p> <p>Web User Interface: Features->Forward &DND->Forward->No Answer Forward->After Ring Time (0~120s)</p> <p>Phone User Interface: Menu->Features->Call Forward->No Answer Forward->After Ring Time</p>		
forward.no_answer.on_code	String within 32 characters	Blank
<p>Description: Configures the no answer forward on code to activate the server-side no answer forward feature. The IP phone will send the no answer forward on code and the pre-configured destination number to the server when you activate no answer forward feature on the IP phone.</p> <p>Example: forward.no_answer.on_code = *76</p> <p>Web User Interface: Features->Forward &DND->Forward->No Answer Forward->On Code</p> <p>Phone User Interface: Menu->Features->Call Forward->No Answer Forward->On Code</p>		
forward.no_answer.off_code	String within 32 characters	Blank
<p>Description: Configures the no answer forward off code to deactivate the server-side no answer forward feature. The IP phone will send the no answer forward off code to the server when you deactivate no answer forward feature on the IP phone.</p> <p>Example: forward.no_answer.off_code = *77</p> <p>Web User Interface: Features->Forward &DND->Forward->No Answer Forward->Off Code</p> <p>Phone User Interface: Menu->Features->Call Forward->No Answer Forward->Off Code</p>		
features.fwd_diversion_enable	0 or 1	1

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to present the diversion information when an incoming call is forwarded to your IP phone.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->General Information->Diversion/History-Info</p> <p>Phone User Interface: None</p>		
forward.international.enable	0 or 1	1
<p>Description: Enables or disables the IP phone to forward incoming calls to international numbers (the prefix is 00).</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->General Information->Fwd International</p> <p>Phone User Interface: None</p>		

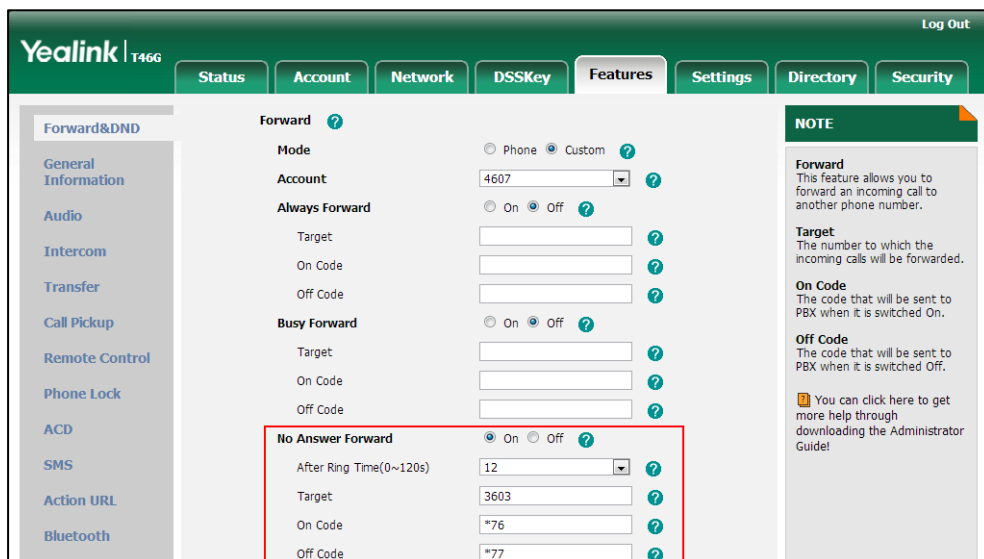
To configure call forward via web user interface:

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 Forward/Busy Forward/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).



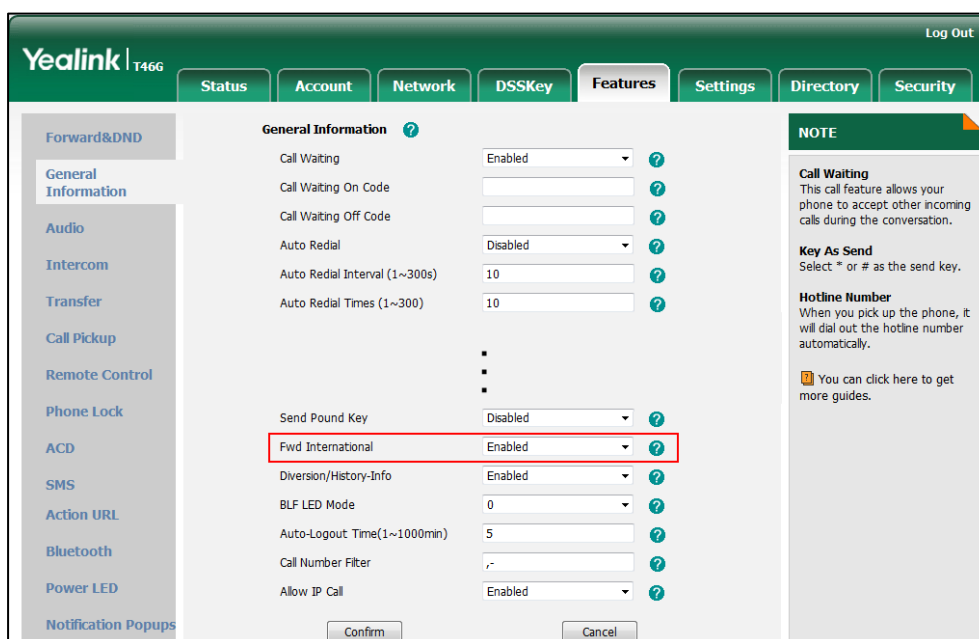
- b) If you mark the **Custom** radio box:
 - 1) Select the desired account from the pull-down list of **Account**.
 - 2) Mark the desired radio box in the **Always Forward/Busy Forward/No Answer Forward** field.
 - 3) Enter the destination number you want to forward in the **Target** field.
 - 4) (Optional.) 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 no answer forward).



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

To configure the forward international feature via web user interface:





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









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

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



1. Press **Menu->Features->Call Forward**.
2. Press **▲** or **▼** to select the desired forwarding type, and then press the **Enter** soft key.
3. Depending on your selection:
 - a) If you select **Always Forward**:
 - 1) Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Always Forward** 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.
 - b) If you select **Busy Forward**:
 - 1) Press **◀** or **▶**, or the **Switch** soft key to select the desired value from the **Busy Forward** 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.



- c) If you select **No Answer Forward**:
- 1) Press  or  , or the **Switch** soft key to select the desired value from the **No Answer Forward** 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 Forward** 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 Forward** 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 Forward** 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 Forward** 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 Forward** 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 Forward** 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.

To configure forward international via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**FWD International**.
2. Press  or  , or the **Switch** soft key to select the desired type from the **FWD International** field.
3. 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 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 determines whether to display the prompt "1 New Missed Call(s)" ("n" indicates the number of the missed calls) on the destination party's LCD screen.

Procedure

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

<p>Configuration File</p>	<p><y000000000xx>.cfg</p>	<p>Specify whether to complete the transfer through on-hook.</p> <p>Parameters:</p> <p>transfer.blind_tran_on_hook_enable transfer.on_hook_trans_enable</p> <p>Configure the semi-attended transfer feature.</p> <p>Parameter:</p> <p>transfer.semi_attend_tran_enable</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Specify whether to complete the transfer through on-hook.</p> <p>Configure the semi-attended transfer feature.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?parameters=features-transfer&q=load</p>

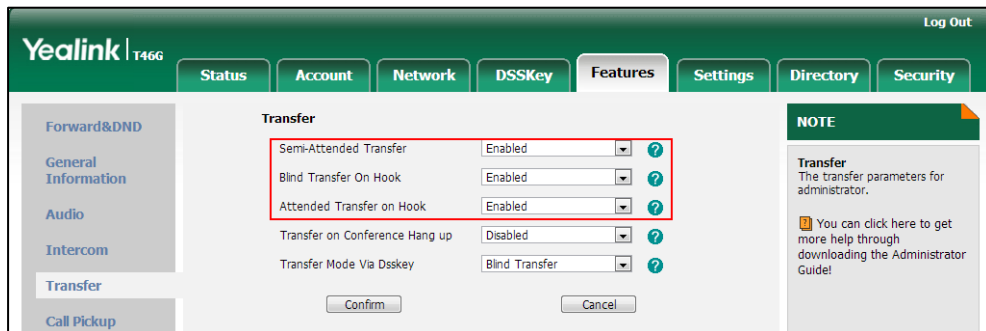
Details of Configuration Parameters:

Parameters	Permitted Values	Default
transfer.blind_tran_on_hook_enable	0 or 1	1
<p>Description: Enables or disables the IP phone to complete the blind transfer through on-hook besides pressing the Transfer/Tran soft key or TRAN/TRANSFER key.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Transfer ->Blind Transfer On Hook</p> <p>Phone User Interface: None</p>		
transfer.on_hook_trans_enable	0 or 1	1
<p>Description: Enables or disables the IP phone to complete the semi-attended/attended transfer through on-hook besides pressing the Transfer/Tran soft key or TRAN/TRANSFER key.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Transfer ->Attended Transfer On Hook</p>		
transfer.semi_attend_tran_enable	0 or 1	1
<p>Description: Enables or disables the transferee party's phone to prompt a missed call on the LCD screen before displaying the caller ID when performing a semi-attended transfer.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Transfer ->Semi-Attended Transfer</p> <p>Phone User Interface: None</p>		

To configure call transfer via web user interface:

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

- Select the desired values from the pull-down lists of **Semi-Attend Transfer**, **Blind Transfer On Hook** and **Semi Attend Transfer On Hook**.



- 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. Parameters: account.X.conf_type account.X.conf_uri
Local	Web User Interface	Configure network conference. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc=0

Details of Configuration Parameters:

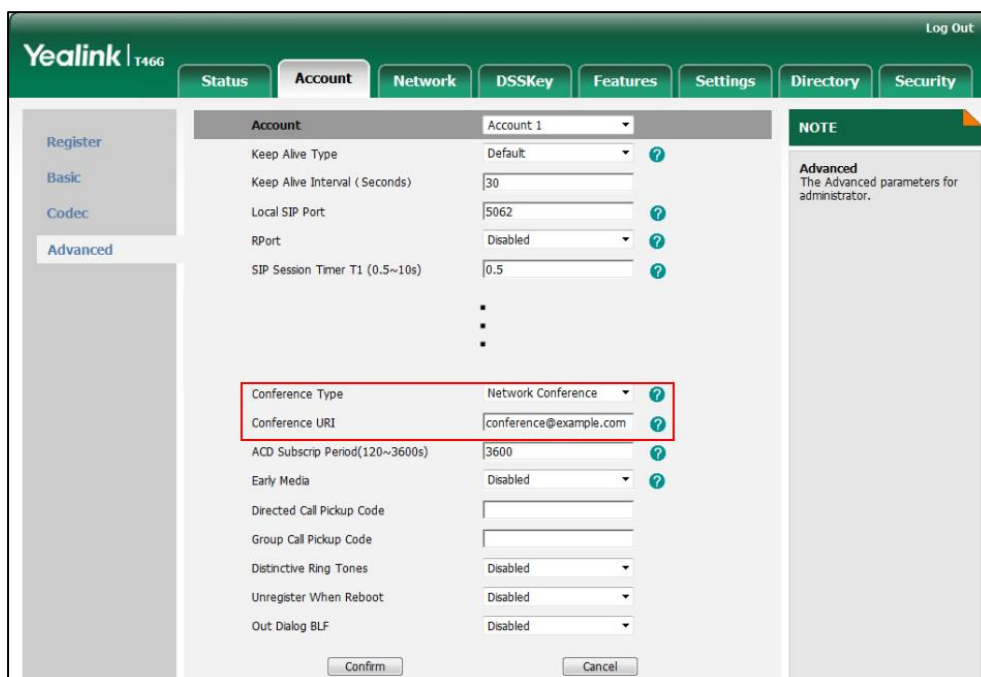
Parameters	Permitted Values	Default
account.X.conf_type	0 or 2	0

Parameters	Permitted Values	Default
<p>Description: Configures the network conference type for account X. 0-Local Conference 2-Network Conference 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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Web User Interface: Account->Advanced->Conference Type Phone User Interface: None</p>		
account.X.conf_uri	SIP URI within 511 characters	Blank
<p>Description: Configures the network conference URI for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Example: account.1.conf_uri = conference@example.com Note: It works only if the value of the parameter "account.X.conf_type" is set to 2 (Network Conference). Web User Interface: Account->Advanced->Conference URI Phone User Interface: None</p>		

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.

Feature Key Synchronization

Feature key synchronization provides the capability to synchronize the status of the following features between the IP phone and the server:

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

Note Feature key synchronization is applicable to IP phones running firmware version 73 or later in the neutral version.

Procedure

Feature key synchronization can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure feature key synchronization. Parameters: bw.feature_key_sync
Local	Web User Interface	Configure feature key synchronization.

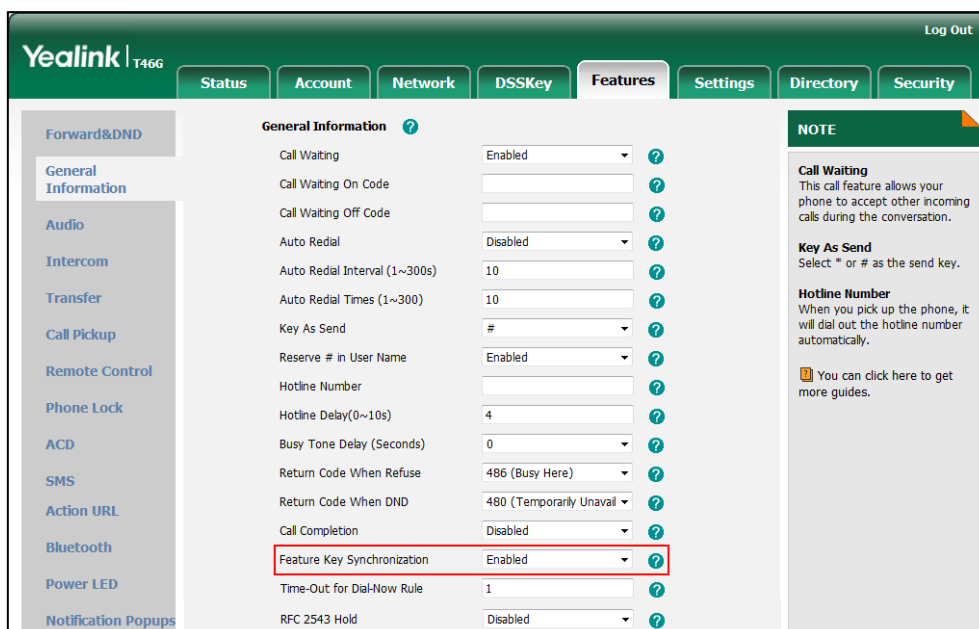
		<p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet ?p=features-general&q=load</p>
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Details of Configuration Parameter:

Parameters	Permitted Values	Default
bw.feature_key_sync	0 or 1	0
<p>Description:</p> <p>Enables or disables feature key synchronization.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface:</p> <p>Features->General Information ->Feature Key Synchronization</p> <p>Phone User Interface:</p> <p>None</p>		

To configure feature key synchronization via web user interface:

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



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

Procedure

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

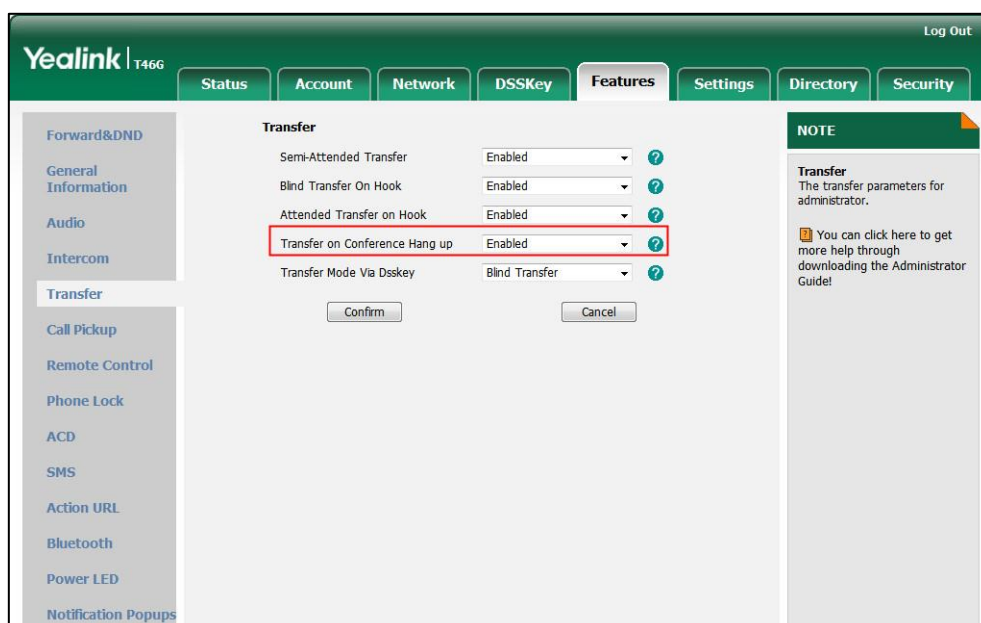
Configuration File	<y0000000000xx>.cfg	Configure transfer on conference hang up. Parameter: transfer.tran_others_after_conf_enable
Local	Web User Interface	Configure transfer on conference hang up. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-transfer&q=load">http://<phoneIPAddress>/servlet?p=features-transfer&q=load

Details of the Configuration Parameter:

Parameter & Description	Permitted Values	Default
transfer.tran_others_after_conf_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to transfer the local conference call to the two parties after the conference initiator drops the local conference call.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the other two parties remain connected when the conference initiator drops the conference call.</p> <p>Note: It is only applicable to the local conference.</p> <p>Web User Interface: Features->Transfer ->Transfer on Conference Hang up</p> <p>Phone User Interface: None</p>		

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. 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 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.

Configuration File	<MAC>.cfg	Configure the directed call pickup code on a per-line basis. Parameter: account.X.direct_pickup_code
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	<p><y000000000xx>.cfg</p>	<p>Configure directed call pickup features on a phone basis.</p> <p>Parameters:</p> <p>features.pickup.direct_pickup_enable</p> <p>features.pickup.direct_pickup_code</p> <p>Assign a directed call pickup key.</p> <p>Parameters:</p> <p>linekey.X.type/ programablekey.X.type linekey.X.line/ programablekey.X.line linekey.X.value/ programablekey.X.value</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Assign a directed call pickup key.</p> <p>Navigate to:</p> <p><a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p> <p>Configure the directed call pickup feature on a phone basis.</p> <p>Navigate to:</p> <p><a href="http://<phoneIPAddress>/servlet?p=features-callpickup&q=load">http://<phoneIPAddress>/servlet?p=features-callpickup&q=load</p> <p>Configure the directed call pickup code on a per-line basis.</p> <p>Navigate to:</p> <p><a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>
	<p>Phone User Interface</p>	<p>Assign a directed call pickup key.</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.direct_pickup_code	String within 32 characters	Blank
<p>Description : Configures the directed call pickup code for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.direct_pickup_code = *97</p> <p>Note: The directed call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.</p> <p>Web User Interface: Account->Advanced->Directed Call Pickup Code</p> <p>Phone User Interface: None</p>		
features.pickup.direct_pickup_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to display the DPickup soft key when the IP phone is in the pre-dialing screen.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Call Pickup->Directed Call Pickup</p> <p>Phone User Interface: None</p>		
features.pickup.direct_pickup_code	String within 32 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the directed call pickup code on a phone basis.</p> <p>Example: features.pickup.direct_pickup_code = *97</p> <p>Note: The directed call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.</p> <p>Web User Interface: Features->Call Pickup->Directed Call Pickup Code</p> <p>Phone User Interface: None</p>		

Directed Call Pickup Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameters	Permitted Values	Default
linekey.X.type/programablekey.X.type	9	Refer to the following content
<p>Description: Configures a DSS key as a directed call pickup key on the IP phone. The digit 9 stands for the key type Directed Pickup. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: linekey.2.type = 9</p> <p>Default: For line keys: For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29</p>		

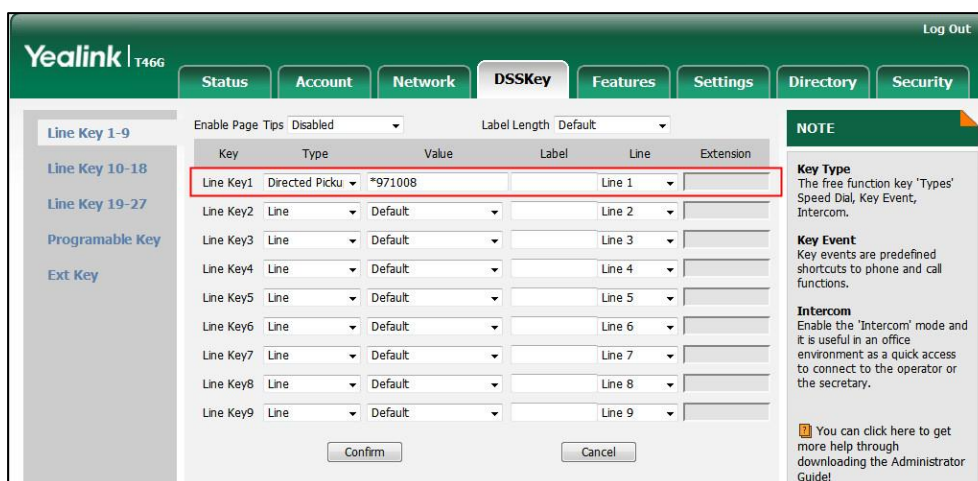
Parameters	Permitted Values	Default
<p>is 0.</p> <p>For SIPT46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIPT42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIPT41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIPT48G/T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIPT42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p>		

Parameters	Permitted Values	Default
<p>When X=10, the default value is 0 (NA). When X=13, the default value is 0 (NA). Web User Interface: DSSKey->Line Key X/ Programable Key ->Type Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		
<p>linekey.X.line/programablekey.X.line</p>	<p>Integer from 1 to 16</p>	<p>1-6 correspond to the lines 1-6</p>
<p>Description: Configures the desired line to apply the directed call pickup key. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P) The valid values are: 1 to 16 (for SIP-T48G/T46G) 1 to 12 (for SIP-T42G) 1 to 6 (for SIP-T41P) 1-Line 1 2-Line 2 ... 16-Line 16 Web User Interface: DSSKey->Line Key X/ Programable Key ->Line Phone User Interface: Menu->Features->DSS Keys->Line Key X->Account ID</p>		
<p>linekey.X.value/ programablekey.X.value</p>	<p>String within 99 characters</p>	<p>Blank</p>
<p>Description: Configures the directed call pickup feature code followed by the number of monitored extension.</p>		

Parameters	Permitted Values	Default
<p>For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Web User Interface: DSSKey->Line Key X/Programable Key ->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		

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

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

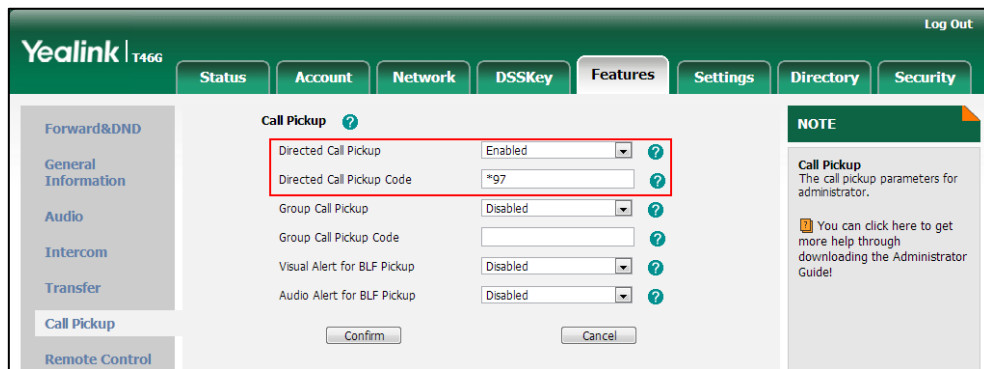


5. Click **Confirm** to accept the change.

To configure the 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**.

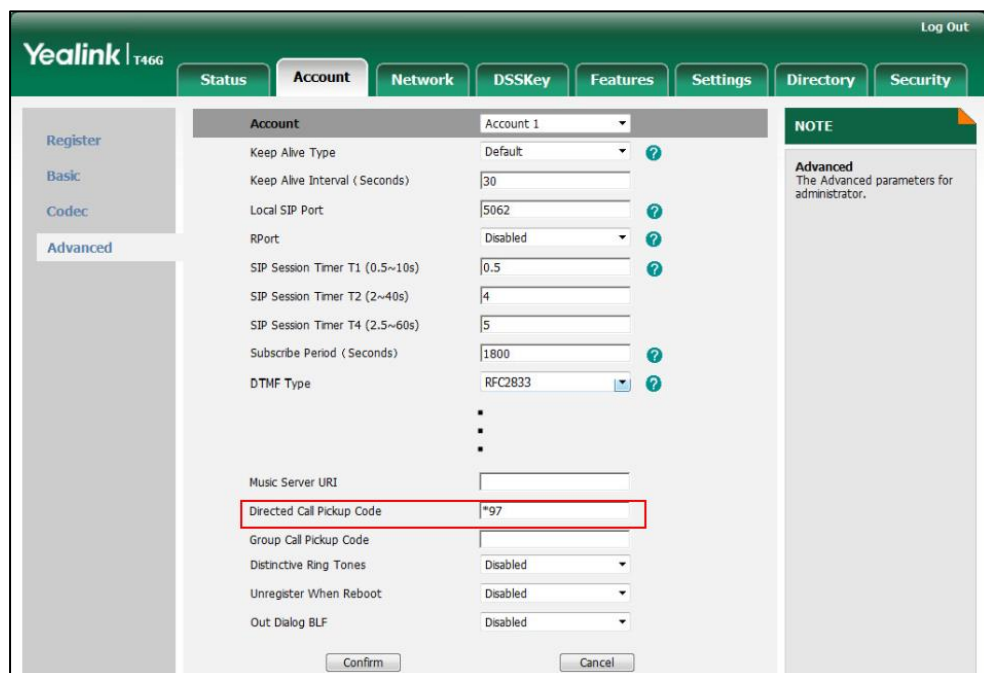
3. Enter the directed call pickup code in the **Directed Call Pickup Code** field.



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**.
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 **DPickup** from the **Key Type** field.

5. Press  or  , or the **Switch** soft key to select the desired line from the **Account ID** field.
6. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
7. Enter the directed call pickup code followed by the specific extension in the **Value** field.
8. 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. 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 per-line basis.

Procedure

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

Configuration File	<MAC>.cfg	Configures the group call pickup code on a per-line basis. Parameter: account.X.group_pickup_code
	<y0000000000xx>.cfg	Configures the group call pickup features on a phone basis. Parameters: features.pickup.group_pickup_enable features.pickup.group_pickup_code Assign a group call pickup key. Parameters: linekey.X.type/ programablekey.X.type linekey.X.line/ programablekey.X.line linekey.X.value/ programablekey.X.value
Local	Web User Interface	Assign a group call pickup key. Navigate to: <a href="http://<phoneIPAddress>/servl">http://<phoneIPAddress>/servl

		<p>et?p=dsskey&model=1&q=load&linepage=1</p> <p>Configure the group call pickup feature on a phone basis.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-callpickup&q=load</p> <p>Configure the group call pickup code on a per-line basis.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>
	Phone User Interface	Assign a group call pickup key.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.pickup.group_pickup_enable	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to display the GPickup soft key when the IP phone is in the pre-dialing screen.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>Web User Interface:</p> <p>Features->Call Pickup->Group Call Pickup</p> <p>Phone User Interface:</p> <p>None</p>		
account.X.group_pickup_code	String within 32 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the group pickup code for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.group_pickup_code = *98</p> <p>Note: The group call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.</p> <p>Web User Interface: Account->Advanced->Group Call Pickup Code</p> <p>Phone User Interface: None</p>		
features.pickup.group_pickup_code	String within 32 characters	Blank
<p>Description: Configures the group call pickup code on a phone basis.</p> <p>Example: features.pickup.group_pickup_code = *98</p> <p>Note: The group call pickup code configured on a per-line basis takes precedence over that configured on a phone basis.</p> <p>Web User Interface: Features->Call Pickup->Group Call Pickup Code</p> <p>Phone User Interface: None</p>		

Group Call Pickup Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameters	Permitted Values	Default
linekey.X.type/ programablekey.X.type	23	Refer to the following content

Parameters	Permitted Values	Default
<p>Description:</p> <p>Configures a DSS key as a group call pickup key on the IP phone.</p> <p>The digit 23 stands for the key type Group Pickup.</p> <p>For line keys:</p> <p>X ranges from 1 to 29 (for SIP-T48G).</p> <p>X ranges from 1 to 27 (for SIP-T46G).</p> <p>X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys:</p> <p>X=1-10, 12-14 (for SIP-T48G/T46G)</p> <p>X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example:</p> <p>linekey.2.type = 23</p> <p>Default:</p> <p>For line keys:</p> <p>For SIP-T48G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIP-T48G/T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p>		

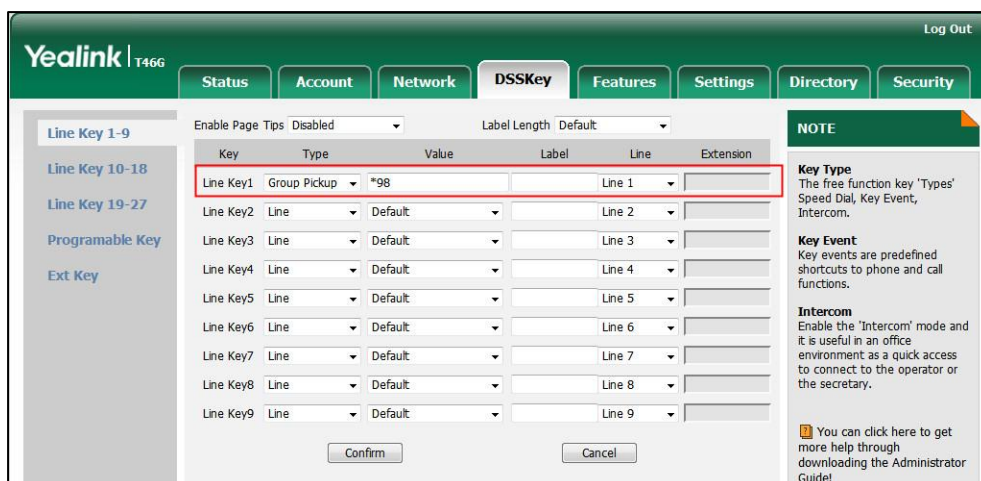
Parameters	Permitted Values	Default
<p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X/Programable Key->Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.line/programablekey.X.line	Integer from 1 to 6	1-6 correspond to the lines 1-6.
<p>Description:</p> <p>Configures the desired line to apply the group call pickup key.</p> <p>For line keys:</p> <p>X ranges from 1 to 29 (for SIP-T48G).</p> <p>X ranges from 1 to 27 (for SIP-T46G).</p> <p>X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys:</p> <p>X=1-10, 12-14 (for SIP-T48G/T46G)</p> <p>X=1-10, 13 (for SIP-T42G/T41P)</p> <p>The valid values are:</p> <p>1 to 16 (for SIP-T48G/T46G)</p>		

Parameters	Permitted Values	Default
1 to 12 (for SIP-T42G) 1 to 6 (for SIP-T41P) 1-Line 1 2-Line 2 ... 16-Line 16 Web User Interface: DSSKey->Line Key X/ Programable Key->Line Phone User Interface: Menu->Features->DSS Keys->Line Key X->Account ID		
linekey.X.value/ programablekey.X.value	String within 99 characters	blank
Description: Configures the group call pickup feature code. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P) Example: linekey.1.value = *98 Web User Interface: DSSKey->Line Key X/ Programable Key->Value Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value		

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

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

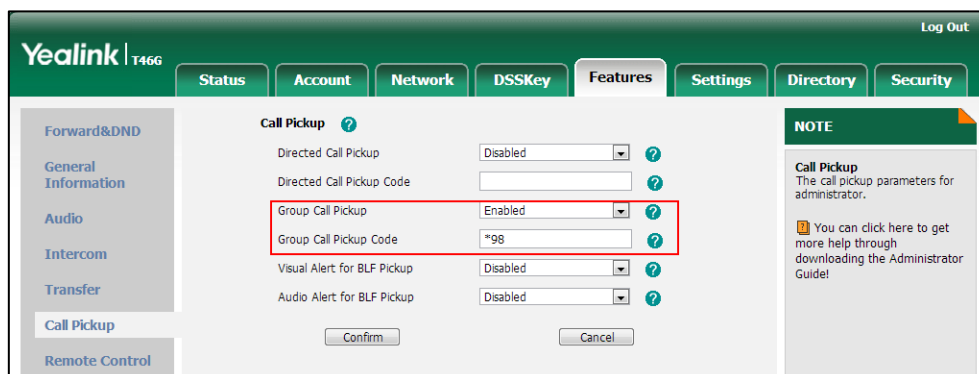
4. Select the desired line from the pull-down list of **Line**.



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

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

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



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

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

1. Click on **Account**.
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 T46G web interface with the 'Account' tab selected. The 'Account' configuration page is displayed for 'Account 1'. The 'Group Call Pickup Code' field is highlighted with a red box and contains the value '*98'. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5062), '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), and 'DTMF Type' (RFC2833). A 'NOTE' section on the right states: 'Advanced: The Advanced parameters for administrator.' At the bottom, there are 'Confirm' and 'Cancel' buttons.

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

To configure a group pickup key via phone user interface:

- Press **Menu->Features->DSS 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 **GPickup** from the **Key Type** field.
- Press **◀** or **▶**, or the **Switch** soft key to select the desired line from the **Account ID** field.
- (Optional.) Enter the string that will appear on the LCD screen in the **Label** 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 can pick up incoming calls via a NOTIFY message with dialog-info event. A user can pick up an incoming call by pressing a DSS key used to monitor a specific extension (such as a BLF key).

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 on the IP phone. Parameter: account.X.dialoginfo_callpickup
Local	Web User Interface	Configure dialog-info call pickup on the IP phone. Navigate to: http://<phoneIPAddress>/servlet? p=account-adv&q=load&acc=0

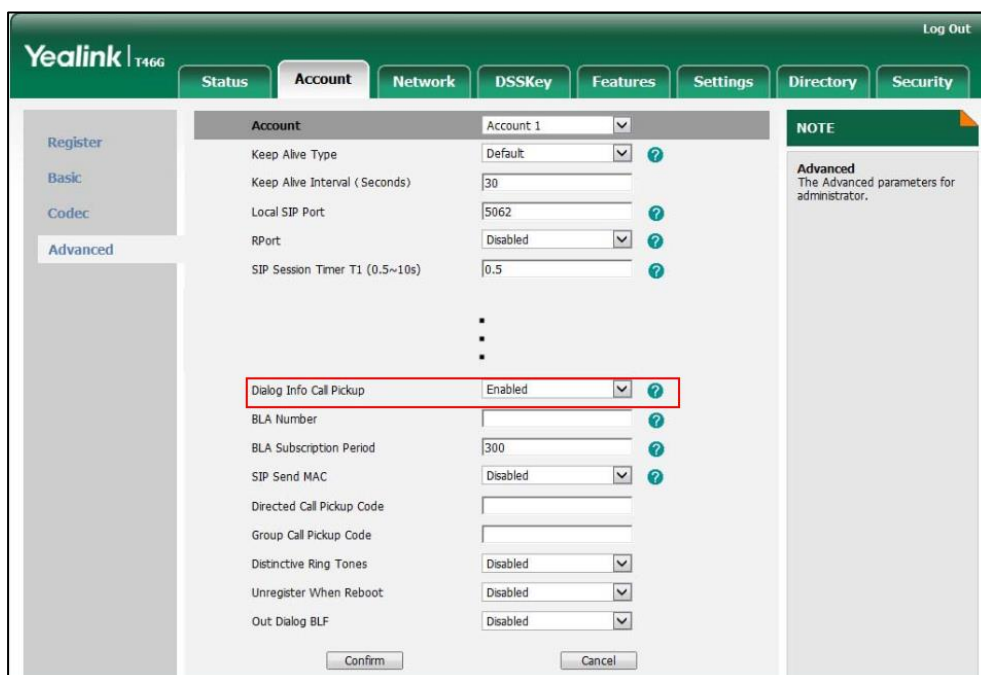
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.dialoginfo_callpickup	0 or 1	0

Parameter	Permitted Values	Default
<p>Description:</p> <p>Enables or disables the IP phone to pick up a call according to the SIP header of dialog-info for account X.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>If it is set to 1 (Enabled), call pickup is implemented through SIP signals.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface:</p> <p>Account->Advanced->Dialog Info Call Pickup</p> <p>Phone User Interface:</p> <p>None</p>		

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



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

ReCall

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

Procedure

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

Configuration File	<y0000000000xx>.cfg	Assign a recall key. Parameter: linekey.X.type/ programablekey.X.type
Local	Web User Interface	Assign a recall key. Navigate to: http://<phoneIPAddress>/servlet ?p=dsskey&model=1&q=load&li nepage=1
	Phone User Interface	Assign a recall key.

Recall Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

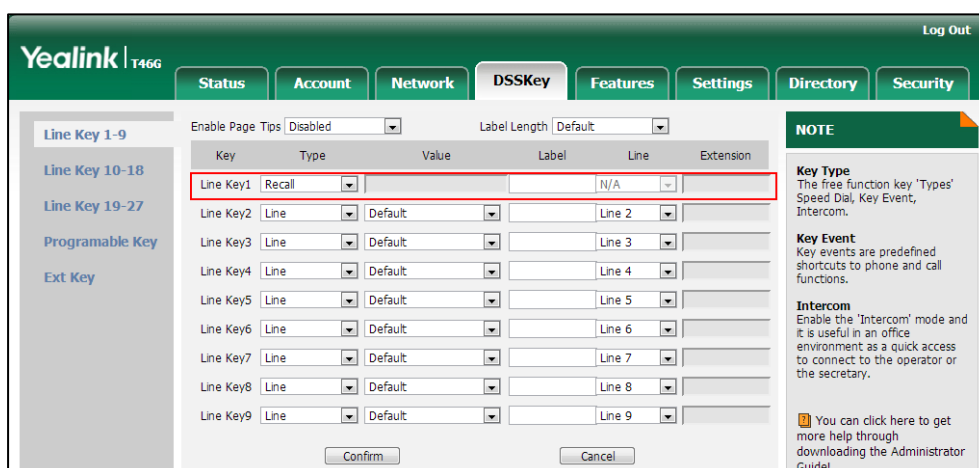
Parameter	Permitted Values	Default
linekey.X.type/programablekey.X.type	7	Refer to the following content
<p>Description: Configures a DSS key as a recall key on the IP phone. The digit 7 stands for the key type ReCall. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: linekey.1.type = 7</p>		

Parameter	Permitted Values	Default
<p>Default:</p> <p>For line keys:</p> <p>For SIP-T48G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIP-T48G/T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p>		

Parameter	Permitted Values	Default
When X=6, the default value is 61 (Directory).		
When X=7, the default value is 51 (Switch Account Up).		
When X=8, the default value is 52 (Switch Account Down).		
When X=9, the default value is 33 (Status).		
When X=10, the default value is 0 (NA).		
When X=13, the default value is 0 (NA).		
Web User Interface:		
DSSKey->Line Key X/ Programable Key->Type		
Phone User Interface:		
Menu->Features->DSS Keys->Line Key X->Type		

To configure a recall key via web user interface:

1. Click on **DSSKey->Line Key X**.
2. In the desired DSS key field, select **ReCall** from the pull-down list of **Type**.



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

To configure a recall key via phone user interface:

1. Press **Menu->Features->DSS 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 **ReCall** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. 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.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Assign a call park key. linekey.X.type
Local	Web User Interface	Assign a call park key. Navigate to: http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1
	Phone User Interface	Assign a call park key.

Call Park Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

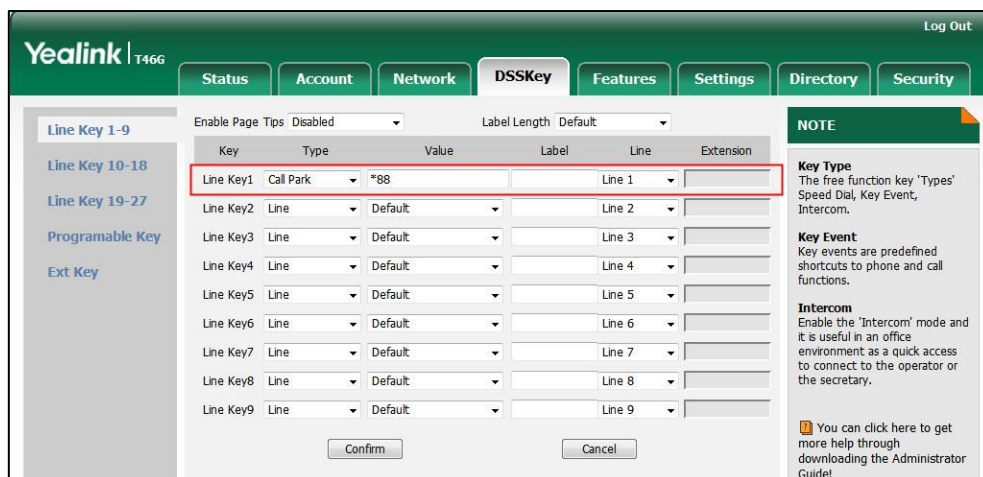
Parameter	Permitted Values	Default
linekey.X.type	10	Refer to the following content
<p>Description: Configures a line key to be a call park key on the IP phone. The digit 10 stands for the key type Call Park. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.type = 10</p> <p>Default: For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29</p>		

Parameter	Permitted Values	Default
<p>is 0.</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X->Line Key X->Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.line	Integer from 1 to 16	1-6 correspond to the lines 1-6
<p>Description:</p> <p>Configures the desired line to apply the call park key.</p> <p>X ranges from 1 to 29 (for SIP-T48G).</p> <p>X ranges from 1 to 27 (for SIP-T46G).</p> <p>X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>The valid values are:</p> <p>1 to 16 (for SIP-T48G/T46G)</p> <p>1 to 12 (for SIP-T42G)</p> <p>1 to 6 (for SIP-T41P)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X->Line Key X->Line</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Account ID</p>		
linekey.X.value	String	Blank

Parameter	Permitted Values	Default
<p>Description: Configures the call park feature code. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.value = *88</p> <p>Web User Interface: DSSKey->Line Key X->Line Key X->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		

To configure a call park key via web user interface:

1. Click on **DSSKey->Line Key X**.
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**.
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. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
7. Enter the desired value (e.g., call park feature code) in the **Value** field.
8. Press the **Save** soft key to accept the change.

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 and stored in the call log.

For more information on calling line identification presentation, refer to *Calling and Connected Line Identification Presentation on Yealink IP Phones*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatelId=142&flag=142>.

Procedure

CLIP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the presentation of the caller identity. Parameter: account.X.cid_source
Local	Web User Interface	Configure the presentation of the caller identity. Navigate to: http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.cid_source	0, 1, 2, 3, 4 or 5	0

Parameter	Permitted Values	Default
<p>Description: Configures the presentation of the caller identity when receiving an incoming call 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 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->Caller ID Source</p> <p>Phone User Interface: None</p>		

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**.

- Select the desired value from the pull-down list of the **Caller ID Source**.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is selected, and the 'Account 1' settings are displayed. The 'Caller ID Source' field is highlighted with a red box and set to 'FROM'. Other settings include Keep Alive Type (Default), Keep Alive Interval (30), Local SIP Port (5060), RPort (Disabled), SIP Session Timer T1 (0.5), SIP Session Timer T2 (4), SIP Session Timer T4 (5), Subscribe Period (1800), DTMF Type (RFC2833), DTMF Info Type (DTMF-Relay), DTMF Payload Type (101), Retransmission (Disabled), Subscribe for MWI (Disabled), MWI Subscription Period (3600), Subscribe MWI To Voice Mail (Disabled), Voice Mail (Enabled), and Voice Mail Display (Enabled). A 'NOTE' section on the right indicates that advanced parameters are for administrators and provides a link to the Administrator Guide.

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

Connected Line Identification Presentation

Connected line identification presentation (COLP) allows IP phones to display the identity of the connected party specified for outgoing calls. IP phones can display the Dialed Digits, or the identity in a SIP header (Remote-Party-ID or P-Asserted-Identity) received, or the identity in the From header carried in the UPDATE message sent by the callee as described in RFC 4916. Connected line identification presentation is also known as Called line identification presentation. In some cases, the remote party will be different from the called line identification presentation due to call diversion.

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

For more information on connected line identification presentation, refer to *Calling and Connected Line Identification Presentation on Yealink IP Phones*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

COLP can be configured only using the configuration files.

Configuration File	<MAC>.cfg	Configure the presentation of the callee identity. Parameter: account.X.cp_source
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Details of the Configuration Parameter:

Parameter	Permitted Values	Default
account.X.cp_source	0, 1 or 2	0
<p>Description:</p> <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 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 16 (for SIPT48G/T46G).</p> <p>X ranges from 1 to 12 (for SIPT42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		

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 4x4 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 the 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 the 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 the conversation partners.

SIP INFO

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

Procedure

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

Configuration File	<MAC>.cfg	<p>Configure the method of transmitting DTMF digit and the payload type.</p> <p>Parameters:</p> <p>account.X.dtmf.type account.X.dtmf.dtmf_payload account.X.dtmf.info_type</p>
	<y0000000000xx>.cfg	<p>Configure the number of times for the IP phone to send the end RTP Event packet.</p> <p>Parameter:</p> <p>features.dtmf.repetition</p> <p>Configure the frequency level of DTMF digits.</p> <p>Parameter:</p> <p>features.dtmf.volume</p>
Local	Web User Interface	<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>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.dtmf.type	0, 1, 2 or 3	1

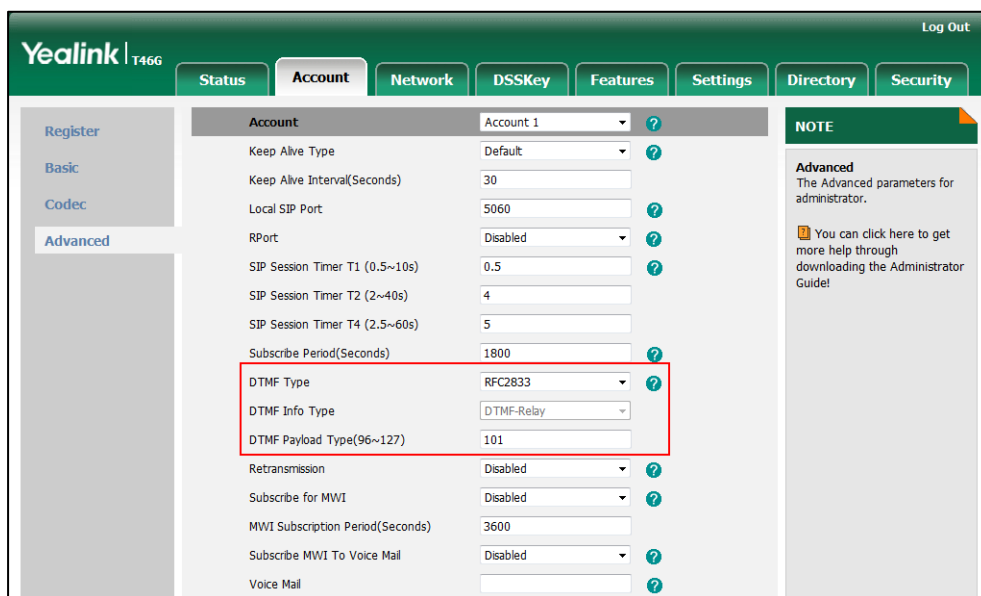
Parameters	Permitted Values	Default
<p>Description: Configures the DTMF type for account X. 0-INBAND 1-RFC 2833 2-SIP INFO 3-RFC2833 + SIP INFO If it is set to 0 (INBAND), DTMF digits are transmitted in the voice band. If it is set to 1 (RFC 2833), DTMF digits are transmitted by RTP Events compliant to RFC 2833. If it is set to 2 (SIP INFO), DTMF digits are transmitted by the SIP INFO messages. If it is set to 3 (RFC2833 + SIP INFO), DTMF digits are transmitted by RTP Events compliant to RFC 2833 and the SIP INFO messages. X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIP-T41P). Web User Interface: Account->Advanced->DTMF Type Phone User Interface: None</p>		
account.X.dtmf.dtmf_payload	Integer from 96 to 127	101
<p>Description: Configures the RFC 2833 payload type for account X. X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIP-T41P). Web User Interface: Account->Advanced->DTMF Payload Type (96~127) Phone User Interface: None</p>		
account.X.dtmf.info_type	1, 2 or 3	0

Parameters	Permitted Values	Default
<p>Description: Configures the DTMF info type when the DTMF type is configured as "SIP INFO", "RFC2833 + SIP INFO" for account X.</p> <p>0-Disabled 1-DTMF-Relay 2-DTMF 3-Telephone-Event</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->DTMF Info Type</p> <p>Phone User Interface: None</p>		
features.dtmf.repetition	1, 2 or 3	3
<p>Description: Configures the repetition times for the IP phone to send the end RTP EVENT packet during an active call.</p> <p>Web User Interface: Features->General Information->DTMF Repetition</p> <p>Phone User Interface: None</p>		
features.dtmf.volume	Integer from -33 to 0	-10
<p>Description: Configures the frequency level of DTMF digits (in db).</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

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**.

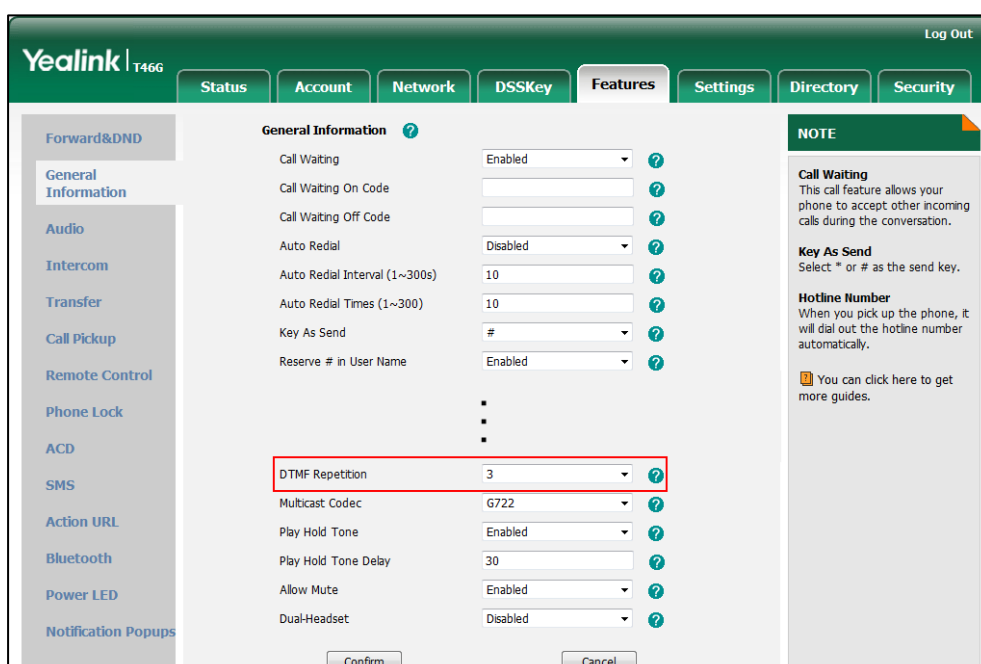
3. Click on **Advanced**.
4. Select the desired value from the pull-down list of **DTMF Type**.
If **SIP INFO** or **RFC2833 + SIP INFO** is selected, select the desired value from the pull-down list of **DTMF Info Type**.
5. Enter the desired value in the **DTMF Payload Type (96~127)** field.



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

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

1. Click on **Features->General Information**.
2. 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. The 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. Parameters: features.dtmf.hide features.dtmf.hide_delay
Local	Web User Interface	Configure suppress DTMF display and suppress DTMF display delay. Navigate to: http://<phoneIPAddress>/servlet?p=features-general&q=load

Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.dtmf.hide	0 or 1	0
<p>Description: Enables or disables the IP phone to suppress the display of DTMF digits during an active call. 0-Disabled 1-Enabled If it is set to 1 (Enabled), the DTMF digits are displayed as asterisks.</p> <p>Web User Interface: Features->General Information->Suppress DTMF Display</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
features.dtmf.hide_delay	0 or 1	0

Description:
 Enables or disables the IP phone to display the DTMF digits for a short period before displaying asterisks during an active call.

0-Disabled
1-Enabled

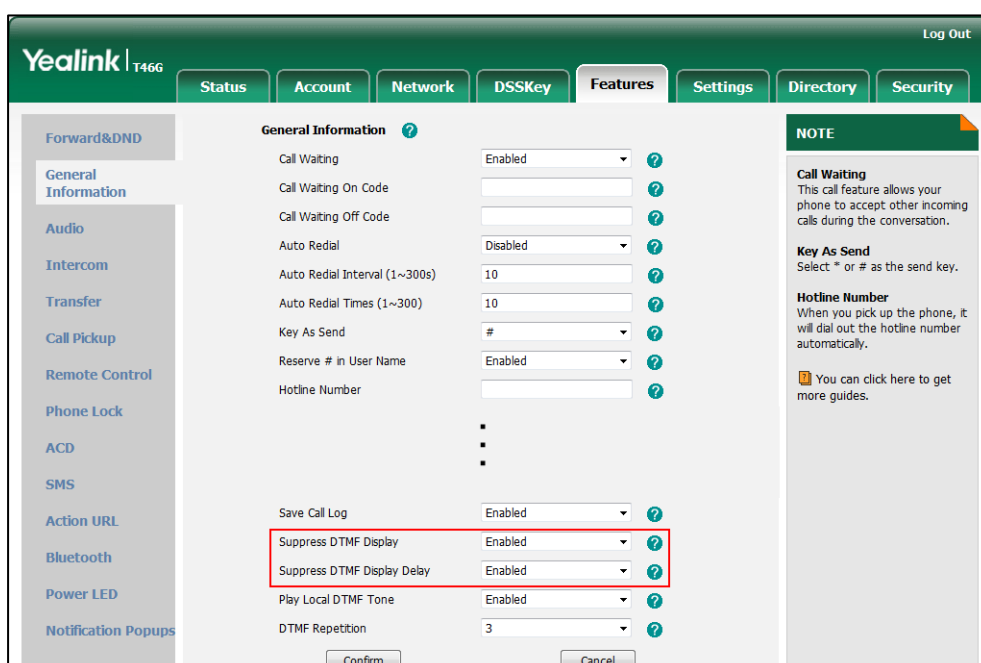
Note: It works only if the value of the parameter "features.dtmf.hide" is set to 1 (Enabled).

Web User Interface:
 Features->General Information->Suppress DTMF Display Delay

Phone User Interface:
 None

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**.



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. Parameters: features.dtmf.replace_tran features.dtmf.transfer
Local	Web User Interface	Configure transfer via DTMF. Navigate to: http://<phoneIPAddress>/servl et?p=features-general&q=loa d

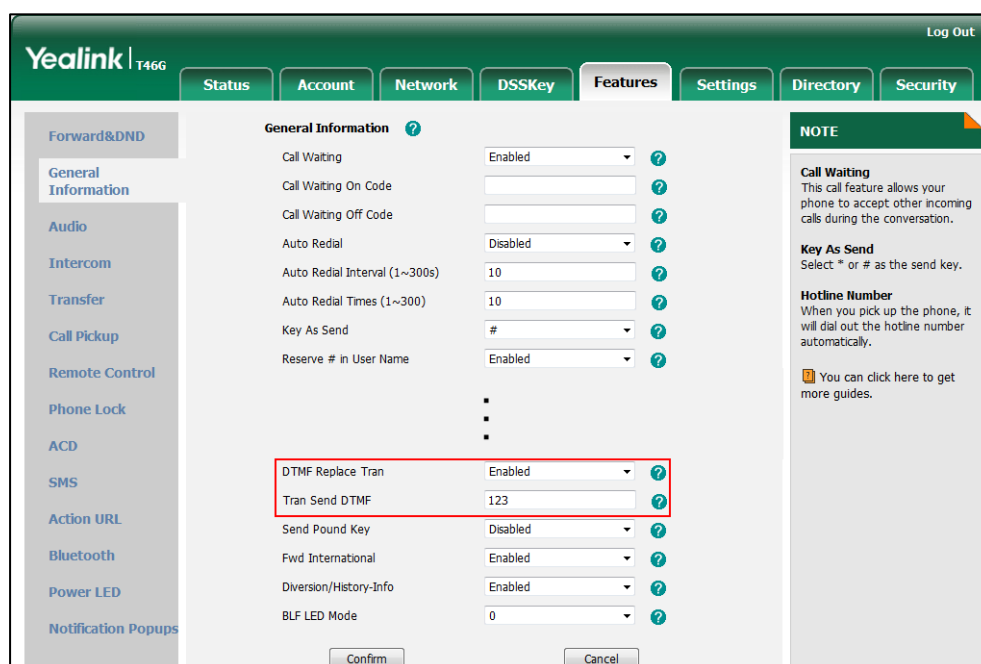
Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.dtmf.replace_tran	0 or 1	0
<p>Description: Enables or disables the IP phone to send DTMF sequences for transfer function when pressing the transfer soft key or the TRAN key.</p> <p>0-Disabled 1-Enabled</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 designated DTMF digits to the server for completing call transfer when pressing the transfer key during a call.</p> <p>Web User Interface: Features->General Information->DTMF Replace Tran</p> <p>Phone User Interface: None</p>		
features.dtmf.transfer	String within 32 characters	Blank

Parameters	Permitted Values	Default
<p>Description:</p> <p>Configures the DTMF digits to be transmitted to perform call transfer. Valid values are: 0-9, *, # and A-D.</p> <p>Example:</p> <p>features.dtmf.transfer = 123</p> <p>Note: It works only if the value of the parameter "features.dtmf.replace_tran" is set to 1 (Enabled).</p> <p>Web User Interface:</p> <p>Features->General Information->Tran Send DTMF</p> <p>Phone User Interface:</p> <p>None</p>		

To configure transfer via DTMF feature 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.

Procedure

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

Configuration File	<y0000000000xx>.cfg	Assign an intercom key. Parameters: linekey.X.type linekey.X.line linekey.X.value
Local	Web User Interface	Assign an intercom key. Navigate to: http://<phoneIPAddress>/servlet ?p=dsskey&model=1&q=load&linepage=1
	Phone User Interface	Assign an intercom key.

Intercom Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

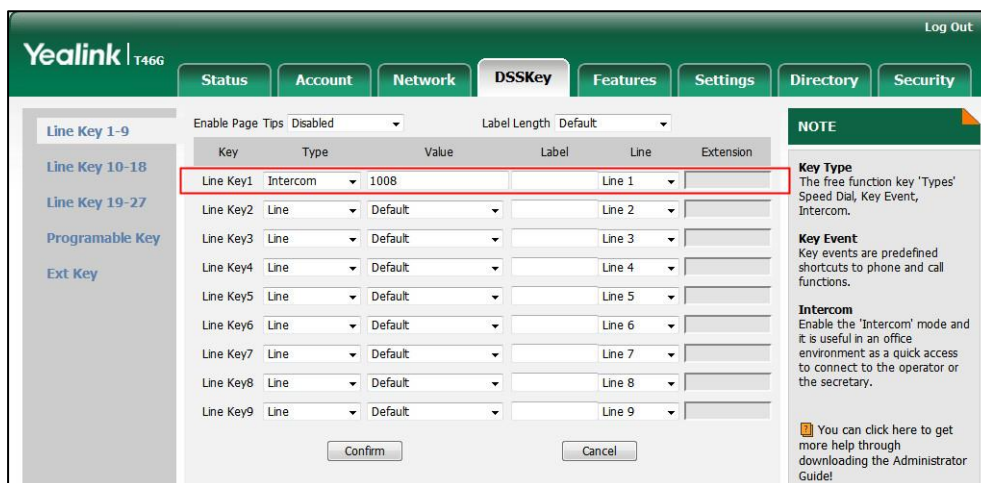
Parameters	Permitted Values	Default
linekey.X.type	14	Refer to the following content
<p>Description: Configures a line key to be an intercom key. The digit 14 stands for the key type Intercom. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.type = 14</p> <p>Default: For SIP-T48G IP phones:</p>		

Parameters	Permitted Values	Default
<p>The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIPT46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIPT42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIPT41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X->Line KeyX->Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.line	Integer from 1 to 16	1-6 correspond to the lines 1-6
<p>Description:</p> <p>Configures the desired line to apply the intercom key.</p> <p>X ranges from 1 to 29 (for SIPT48G).</p> <p>X ranges from 1 to 27 (for SIPT46G).</p> <p>X ranges from 1 to 15 (for SIPT42G/T41P).</p> <p>The valid values are:</p> <p>1 to 16 (for SIPT48G/T46G)</p> <p>1 to 12 (for SIPT42G)</p> <p>1 to 6 (for SIPT41P)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X->Line KeyX->Line</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Account ID</p>		
linekey.X.value	String within 99 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the intercom number. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		

To configure an intercom key via web user interface:

1. Click on **DSSKey->Line Key X**.
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**.
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. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the remote extension number in the **Value** field.
7. 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 the incoming intercom call feature. Parameters: features.intercom.allow features.intercom.mute features.intercom.tone features.intercom.barge
Local	Web User Interface	Configure the incoming intercom call feature. Navigate to: http://<phoneIPAddress>/servlet?p=features-intercom&q=load
	Phone User Interface	Configure the incoming intercom call feature.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.intercom.allow	0 or 1	1

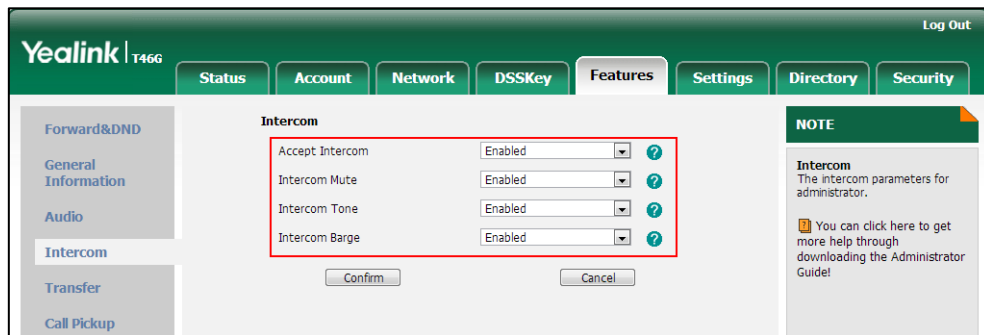
Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to automatically answer an incoming intercom call. 0-Disabled 1-Enabled If it is set to 0 (Disabled), the IP phone will reject incoming intercom calls and sends a busy signal to the caller. If it is set to 1 (Enabled), the IP phone will automatically answer an incoming intercom call.</p> <p>Web User Interface: Features->Intercom ->Accept Intercom</p> <p>Phone User Interface: Menu->Features->Intercom ->Accept Intercom</p>		
features.intercom.mute	0 or 1	0
<p>Description: Enables or disables the IP phone to mute the microphone when answering an intercom call. 0-Disabled 1-Enabled If it is set to 1 (Enabled), the microphone is muted for intercom calls, and then the other party cannot hear you.</p> <p>Note: It works only if the value of the parameter "features.intercom.allow" is set to 1 (Enabled).</p> <p>Web User Interface: Features->Intercom ->Intercom Mute</p> <p>Phone User Interface: Menu->Features->Intercom ->Intercom Mute</p>		
features.intercom.tone	0 or 1	1

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to play a warning tone when receiving an intercom call.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "features.intercom.allow" is set to 1 (Enabled).</p> <p>Web User Interface: Features->Intercom ->Intercom Tone</p> <p>Phone User Interface: Menu->Features->Intercom ->Intercom Tone</p>		
features.intercom.barge	0 or 1	0
<p>Description: Enables or disables the IP phone to automatically answer an incoming intercom call while there is already an active call on the IP phone.</p> <p>0-Disabled 1-Enabled</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> <p>Note: It works only if the value of the parameter "features.intercom.allow" is set to 1 (Enabled).</p> <p>Web User Interface: Features->Intercom->Intercom Barge</p> <p>Phone User Interface: Menu->Features->Intercom->Intercom Barge</p>		

To configure intercom via web user interface:

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

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



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

To configure intercom via phone user interface:

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

Configuring Advanced Features

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

- [Distinctive Ring Tones](#)
- [Tones](#)
- [Remote Phone Book](#)
- [LDAP](#)
- [Busy Lamp Field](#)
- [BLF List](#)
- [Hide Features Access Code](#)
- [Automatic Call Distribution](#)
- [Message Waiting Indicator](#)
- [Multicast Paging](#)
- [Call Recording](#)
- [Hot Desking](#)
- [Action URL](#)
- [Action URI](#)
- [Server Redundancy](#)
- [Static DNS Cache](#)
- [LLDP](#)
- [VLAN](#)
- [VPN](#)
- [Voice Quality Monitoring](#)
- [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 or keyword parameter and maps it to the appropriate ring tone.

Note

If the caller already exists in the local directory, the ring tone assigned to the caller should be preferentially played.

Alert-Info headers in the following four formats:

Alert-Info: 127.0.0.1/Bellcore-drN (or Alert-Info: Bellcore-drN)

Alert-Info: ringtone-N (or Alert-Info: MyMelodyN)

Alert-Info: <URL>

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

- When the Alert-Info header contains the keyword "Bellcore-drN", the IP phone will play the Bellcore-drN (N=1, 2, 3, 4 or 5) ring tone if the parameter "features.alert_info_tone" is set to 1, or play the corresponding local ring tone (RingN.wav) in about ten seconds if the parameter "features.alert_info_tone" is set to 0.

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

Bellcore Tone	Pattern ID	Pattern	Cadence	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
		Silent		145	200	525
		Ringling	Long	630	800	1025
		Silent		2975	4000	4400
Bellcore-dr4	4	Ringling	Short	200	300	525
		Silent		145	200	525
		Ringling	Long	800	1000	1100
		Silent		145	200	525
		Ringling	Short	200	300	525
		Silent		2975	4000	4400
Bellcore-dr5	5	Ringling		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.

- When the Alter-Info header contains the keyword "ringtone-N" or "MyMelodyN", the IP phone will play the corresponding local ring tone (RingN.wav), or play the first local ring tone (Ring1.wav) in about ten seconds if "N" is greater than 10 or less than 1.

Example:

Alert-Info: ringtone-2

Alert-Info: MyMelody2

The following table identifies the corresponding local ring tone:

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

Value of N	Ring Tone
10	Splash.wav
N<1 or N>10	Ring1.wav

- When 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 (or the item called "Distinctive Ring Tones" on the web user interface is Enabled), or play the preconfigured local ring tone in about ten seconds if the parameter "account.X.alert_info_url_enable" is set to 0 or if the IP phone fails to download the remote ring tone.

Example:

Alert-Info: http://192.168.0.12:8080/Custom.wav

- When the Alert-Info header contains an info text, the IP phone will map the text with the internal ringer text preconfigured on the IP phone, and then play the ring tone associated with the internal ringer text. If no internal ringer text maps, the IP phone will play the preconfigured local ring tone in about ten seconds.

Example:

Alert-Info: info=family;x-line-id=0

Auto Answer

If the Alert-Info header contains the following type of strings, the IP phone will answer incoming calls automatically without playing the ring tone:

- Alert-Info: Auto Answer
- Alert-Info: info = alert-autoanswer
- Alert-Info: answer-after = 0 (or Alert-Info: Answer-After = 0)

Note

If the Alert-Info header contains multiple types of keywords, the IP phone will process the keywords in the following order:
 AutoAnswer>URL>"Bellcore-drN/ringtone-N/MyMelodyN">info text.

Procedure

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

Configuration File	<MAC>.cfg	Configure distinctive ring tones feature. Parameter: account.X.alert_info_url_enable
	<y000000000xx>.cfg	Configure the internal ringer text and internal ringer file.

		<p>Parameters:</p> <p>features.alert_info_tone</p> <p>distinctive_ring_tones.alert_info.X.text</p> <p>distinctive_ring_tones.alert_info.X.ringer</p>
Local	Web User Interface	<p>Configure distinctive ring tones feature.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p> <p>Configure the internal ringer text and internal ringer file.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=settings-ring&q=load</p>

Details of Configuration Parameters:

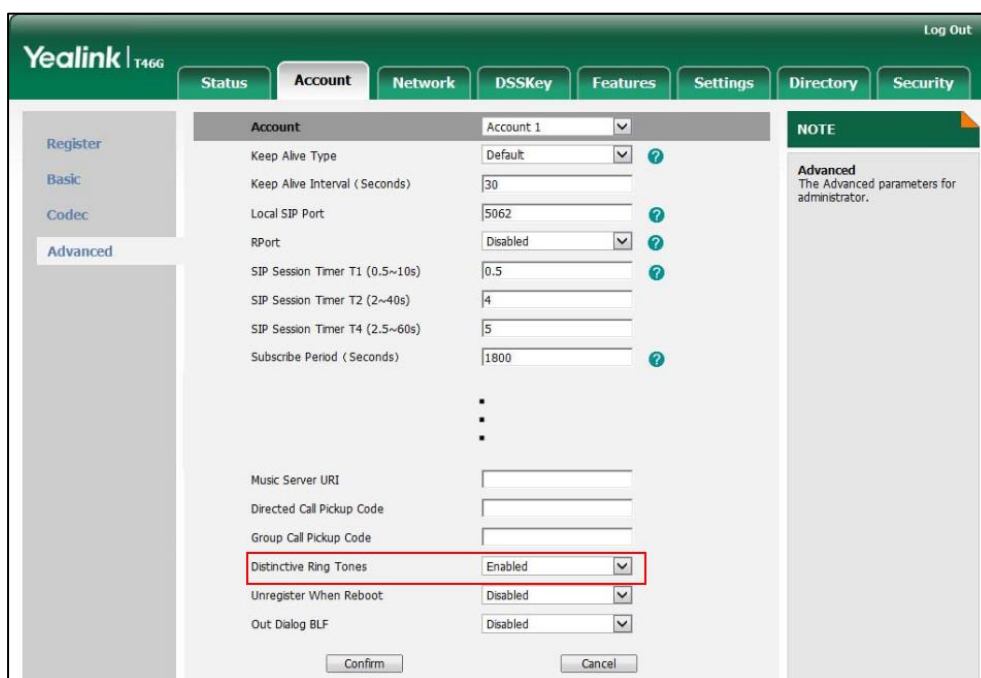
Parameters	Permitted Values	Default
account.X.alert_info_url_enable	0 or 1	1
<p>Description:</p> <p>Enables or disables the IP phone to download the ring tone from the URL contained in the Alert-Info header for account X.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface:</p> <p>Account->Advanced->Distinctive Ring Tones</p> <p>Phone User Interface:</p> <p>None</p>		
features.alert_info_tone	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to map the keywords in the Alert-info header to the specified Bellcore ring tones.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>distinctive_ring_tones.alert_info.X.text (X ranges from 1 to 10)</p>	<p>String within 32 characters</p>	<p>Blank</p>
<p>Description: Configures the internal ringer text to map the keywords contained in the Alert-Info header.</p> <p>Example: distinctive_ring_tones.alert_info.1.text = Family</p> <p>Web User Interface: Settings->Ring->Internal Ringer Text</p> <p>Phone User Interface: None</p>		
<p>distinctive_ring_tones.alert_info.X.ringer (X ranges from 1 to 10)</p>	<p>Integer from 1 to 10</p>	<p>1</p>
<p>Description: Configures the desired ring tones for each text. The value ranges from 1 to 10, the digit stands for the appropriate ring tone.</p> <p>1-Ring1.wav 2-Ring2.wav 3-Ring3.wav 4-Ring4.wav 5-Ring5.wav 6-Ring6.wav 7-Ring7.wav 8-Ring8.wav 9-Silent.wav</p>		

Parameters	Permitted Values	Default
10-Splash.wav		
Web User Interface:		
Settings->Ring->Internal Ringer File		
Phone User Interface:		
None		

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**.

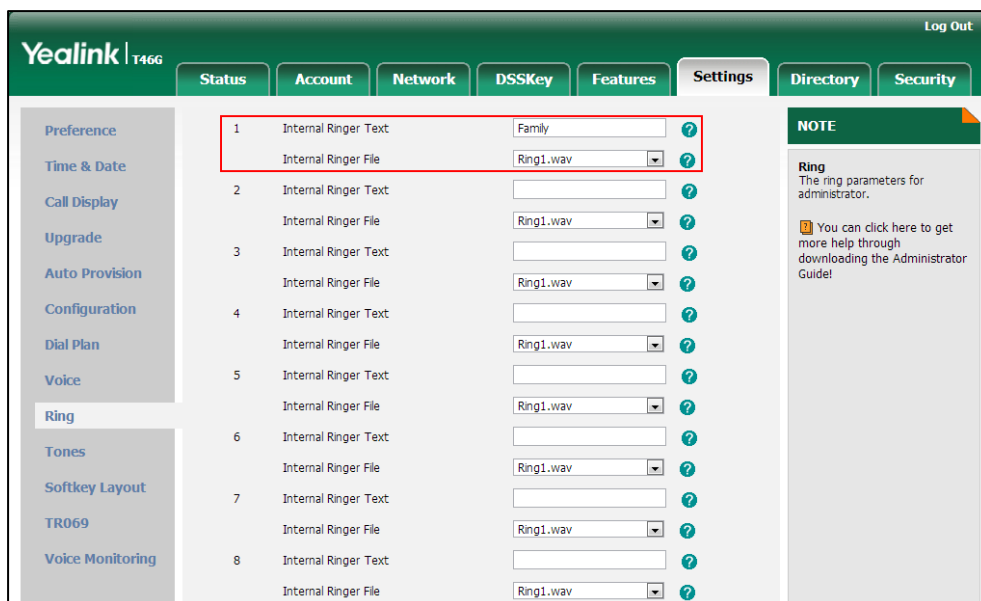


5. Click **Confirm** to accept the change.

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 the IP phone for the following conditions:

Condition	Description
Dial	When in the pre-dialing interface
Ring Back	Ring-back tone
Busy	When the callee is busy
Congestion	When the network is congested
Call Waiting	Call waiting tone
Dial Recall	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-T48G 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
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		<p>phone.</p> <p>Parameters:</p> <p>voice.tone.country</p> <p>voice.tone.dial</p> <p>voice.tone.ring</p> <p>voice.tone.busy</p> <p>voice.tone.congestion</p> <p>voice.tone.callwaiting</p> <p>voice.tone.dialrecall</p> <p>voice.tone.info</p> <p>voice.tone.stutter</p> <p>voice.tone.message</p> <p>voice.tone.autoanswer</p>
Local	Web User Interface	<p>Configure the tones for the IP phone.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=settings-tones&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
voice.tone.country	Refer to the following content	Custom
<p>Description:</p> <p>Configures the country tone for the IP phone.</p> <p>Example:</p> <p>voice.tone.country = Custom</p> <p>Permitted Values:</p> <p>Custom, Australia, Austria, Brazil, Belgium, China, Czech, Denmark, Finland, France, Germany, Great Britain, Greece, Hungary, Lithuania, India, Italy, Japan, Mexico, New Zealand, Netherlands, Norway, Portugal, Spain, Switzerland, Sweden, Russia, United States, Chile, Czech ETSI</p> <p>Web User Interface:</p> <p>Settings->Tones->Select Country</p> <p>Phone User Interface:</p> <p>None</p>		

Parameters	Permitted Values	Default
voice.tone.dial	String	Blank
<p>Description: Customizes the dial tone. tonelist = element[,element] [,element]...</p> <p>Where element = [!]Freq1[+Freq2][+Freq3][+Freq4] /Duration</p> <p>Freq: the frequency of the tone (ranges from 200 to 7000 Hz). If it is set to 0Hz, it means the tone is not played. A tone is comprised of at most four different frequencies.</p> <p>Duration: the duration (in milliseconds) of the dial tone, ranges from 0 to 30000ms. You can configure at most eight different tones for one condition, and separate them by commas. (e.g., 250/200, 0/1000, 200+300/500, 600+700+800+1000/2000). If you want the IP phone to play tones once, add an exclamation mark “!” before tones (e.g., !250/200, 0/1000, 200+300/500, 600+700+800+1000/2000).</p> <p>Note: It works only if the value of the parameter “voice.tone.country” is set to Custom.</p> <p>Web User Interface: Settings->Tones->Dial</p> <p>Phone User Interface: None</p>		
voice.tone.ring	String	Blank
<p>Description: Customizes the ringback tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter “voice.tone.dial”.</p> <p>Note: It works only if the value of the parameter “voice.tone.country” is set to Custom.</p> <p>Web User Interface: Settings->Tones->Ring Back</p> <p>Phone User Interface: None</p>		
voice.tone.busy	String	Blank

Parameters	Permitted Values	Default
<p>Description: Customizes the tone when the callee is busy. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". Note: It works only if the value of the parameter "voice.tone.country" is set to Custom. Web User Interface: Settings->Tones->Busy Phone User Interface: None</p>		
voice.tone.congestion	String	Blank
<p>Description: Customizes the tone when the network is congested. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank. Note: It works only if the value of the parameter "voice.tone.country" is set to Custom. Web User Interface: Settings->Tones->Congestion Phone User Interface: None</p>		
voice.tone.callwaiting	String	Blank
<p>Description: Customizes the call waiting tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank. Note: It works only if the value of the parameter "voice.tone.country" is set to Custom. Web User Interface: Settings->Tones->Call Waiting Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
voice.tone.dialrecall	String	Blank
<p>Description: Customizes the call back tone. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial".</p> <p>Note: It works only if the value of the parameter "voice.tone.country" is set to Custom.</p> <p>Web User Interface: Settings->Tones->Dial Recall</p> <p>Phone User Interface: None</p>		
voice.tone.info	String	Blank
<p>Description: Customizes the info tone. The phone will play the info tone with the special information, for example, the number you are calling is not in service. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank.</p> <p>Note: It works only if the value of the parameter "voice.tone.country" is set to Custom.</p> <p>Web User Interface: Settings->Tones->Info</p> <p>Phone User Interface: None</p>		
voice.tone.stutter	String	Blank
<p>Description: Customizes the tone when the IP phone receives a voice mail. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank.</p> <p>Note: It works only if the value of the parameter "voice.tone.country" is set to</p>		

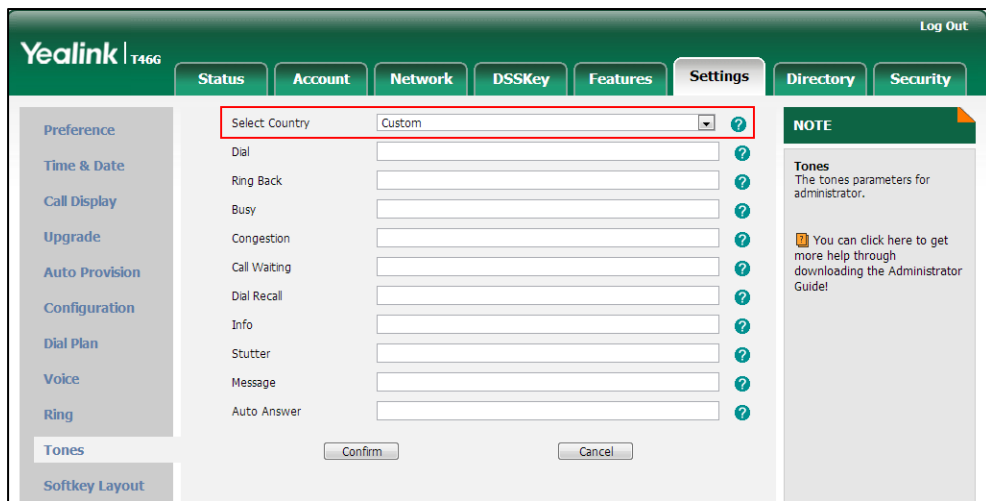
Parameters	Permitted Values	Default
Custom. Web User Interface: Settings->Tones->Stutter Phone User Interface: None		
voice.tone.message	String	Blank
Description: Customizes the tone when the IP phone receives a text message or voice message. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank. Note: It works only if the value of the parameter "voice.tone.country" is set to Custom. Web User Interface: Settings->Tones->Message Phone User Interface: None		
voice.tone.autoanswer	String	Blank
Description: Customizes the warning tone for auto answer. The value format is Freq/Duration. For more information on the value format, refer to the parameter "voice.tone.dial". The default value is blank. Note: It works only if the value of the parameter "voice.tone.country" is set to Custom. Web User Interface: Settings->Tones->Auto Answer Phone User Interface: None		

To configure tones via web user interface:

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

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

phone.



3. 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 entries, and then display the remote phone book entries on the phone user interface. IP phones support up to 5 remote phone books and 5000 entries. Remote phone book is customizable. For more information, refer to [Remote XML Phone Book](#) on page 490.

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

Procedure

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

<p>Configuration File</p>	<pre><y0000000000xx>.cf g</pre>	<p>Specify the access URL and the display name of the remote phone book.</p> <p>Parameters:</p> <ul style="list-style-type: none"> remote_phonebook.data.X.url remote_phonebook.data.X.name remote_phonebook.display_name <p>Specify whether to query the entry name from the remote phone book for outgoing/incoming calls.</p> <p>Parameter:</p> <ul style="list-style-type: none"> features.remote_phonebook.enable
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		<p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p>Parameter: features.remote_phonebook.flash_time</p> <p>Specify whether to refresh the local cache of the remote phone book at a time when accessing the remote phone book.</p> <p>Parameter: features.remote_phonebook.enter_update_enable</p>
Local	Web User Interface	<p>Specify the access URL of the remote phone book.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p> <p>Specify whether to query the entry name from the remote phone book for outgoing/incoming calls.</p> <p>Specify how often the IP phone refreshes the local cache of the remote phone book.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-remote&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
remote_phonebook.data.X.url (X ranges from 1 to 5)	URL within 511 characters	Blank
<p>Description: Configures the access URL of the remote phone book.</p> <p>Example: remote_phonebook.data.1.url = http://192.168.1.20/phonebook.xml</p> <p>Web User Interface:</p>		

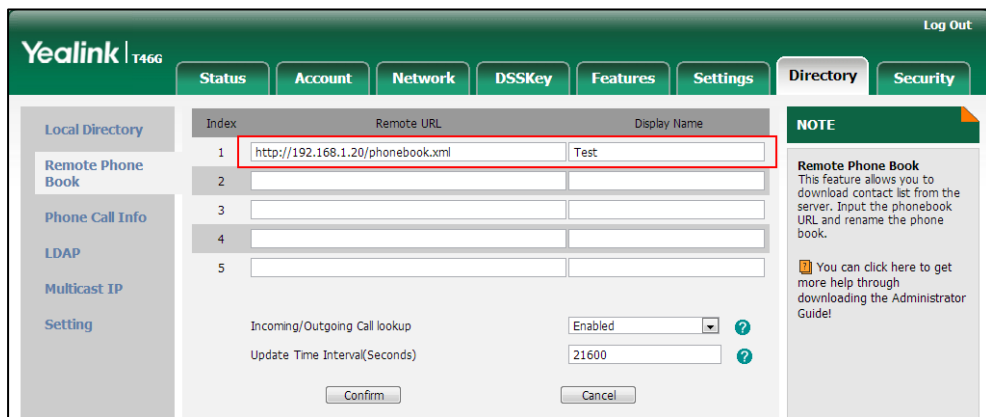
Parameters	Permitted Values	Default
Directory->Remote Phone Book->Remote URL Phone User Interface: None		
remote_phonebook.data.X.name (X ranges from 1 to 5)	String within 99 characters	Blank
Description: Configures the display name of the remote phone book item. Example: remote_phonebook.data.1.name = Test Web User Interface: Directory->Remote Phone Book->Display Name Phone User Interface: None		
remote_phonebook.display_name	String within 99 characters	Blank
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. Example: remote_phonebook.display_name = Remote Phone Book Note: It is not applicable to SIP-T42G/T41P IP phones. Web User Interface: None Phone User Interface: None		
features.remote_phonebook.enable	0 or 1	0
Description: Enables or disables the IP phone to perform a remote phone book search for an incoming or outgoing call and display the matched call on the LCD screen. 0-Disabled 1-Enabled Web User Interface: Directory->Remote Phone Book->Incoming/Outgoing Call lookup Phone User Interface:		

Parameters	Permitted Values	Default
None		
features.remote_phonebook.flash_time	0, Integer from 3600 to 2592000	21600
<p>Description: Configures how often to refresh the local cache of the remote phone book. If it is set to 3600, the IP phone will refresh the local cache of the remote phone book every 3600 seconds.</p> <p>Note: If it is set to 0, the IP phone will not refresh the local cache of the remote phone book.</p> <p>Web User Interface: Directory->Remote Phone Book->Update Time Interval(Seconds)</p> <p>Phone User Interface: None</p>		
features.remote_phonebook.enter_update_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to refresh the local cache of the remote phone book at a time when accessing the remote phone book.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

To specify the 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.

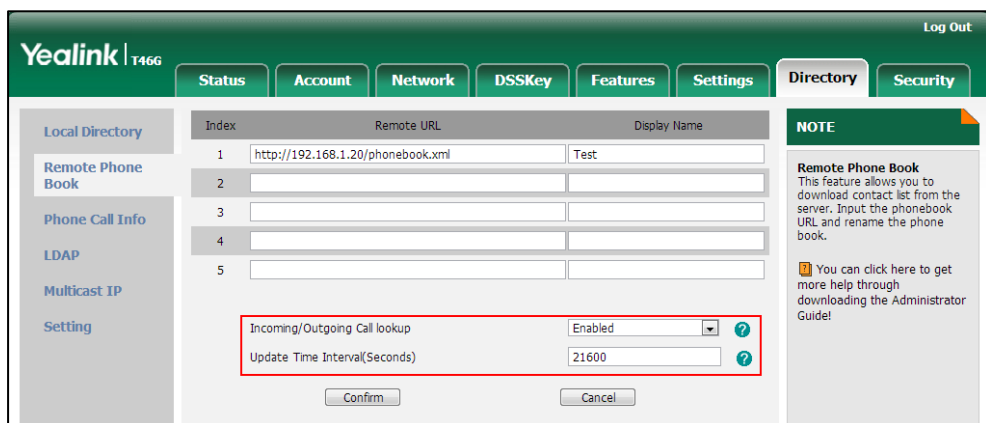
3. Enter the name in the **Display Name** field.



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

To configure the remote phone book via web user interface:

1. Click on **Directory->Remote Phone Book**.
2. Select the desired value from the pull-down list of **Incoming/Outgoing Call lookup**.
3. Enter the desired time in the **Update Time Interval (seconds)** field.



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. The following LDAP servers are supported:

- Microsoft Active Directory
- Sun ONE Directory Server
- Open LDAP Directory Server
- Microsoft Active Directory Application Mode (ADAM)

The biggest 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 local directory. 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" may be used to select the desired entry or group, and return the desired information.

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

You can assign a DSS key to be an LDAP key, and 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

For more information on LDAP, refer to *LDAP Phonebook on Yealink IP Phones*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

LDAP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the LDAP feature. Parameters: ldap.enable ldap.name_filter
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		<p>ldap.number_filter ldap.tls_mode 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.call_out_lookup ldap.ldap_sort</p> <p>Assign an LDAP key.</p> <p>Parameter: linekey.X.type/ programablekey.X.type</p>
Local	Web User Interface	<p>Configure the LDAP feature.</p> <p>Navigate to: http://<phoneIPAddress>/servl et?p=contacts-LDAP&q=load</p> <p>Assign an LDAP key.</p> <p>Navigate to: http://<phoneIPAddress>/servl et?p=dsskey&model=1&q=loa d&linepage=1</p>
	Phone User Interface	Assign an LDAP key.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
ldap.enable	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables LDAP feature on the IP phone. 0-Disabled 1-Enabled</p> <p>Web User Interface: Directory->LDAP->Enable LDAP</p> <p>Phone User Interface: None</p>		
ldap.name_filter	String within 99 characters	Blank
<p>Description: Configures the criteria for searching the LDAP contact name attributes. 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.</p> <p>Example: ldap.name_filter = ((cn=%)(sn=%))</p> <p>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.</p> <p>Web User Interface: Directory->LDAP->LDAP Name Filter</p> <p>Phone User Interface: None</p>		
ldap.number_filter	String within 99 characters	Blank
<p>Description: Configures the criteria for searching the LDAP contact number attributes. 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.</p> <p>Example: ldap.number_filter = ((telephoneNumber=%)(Mobile=%)(ipPhone=%))</p> <p>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.</p> <p>Web User Interface: Directory->LDAP->LDAP Number Filter</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
ldap.tls_mode	Integer from 0 to 2	0
<p>Description: Configures the connection mode between the LDAP server and the IP phone. 0-LDAP—Unencrypted connection between LDAP server and the IP phone. (port 389 is used by default). 1-LDAP TLS Start—TLS/SSL connection between LDAP server and the IP phone (port 389 is used by default). 2-LDAPs—TLS/SSL connection between LDAP server and the IP phone (port 636 is used by default).</p> <p>Web User Interface: Directory->LDAP->LDAP TLS Mode</p> <p>Phone User Interface: None</p>		
ldap.host	String within 99 characters	Blank
<p>Description: Configures the IP address or domain name of the LDAP server.</p> <p>Example: ldap.host = 192.168.1.20</p> <p>Web User Interface: Directory->LDAP->Server Address</p> <p>Phone User Interface: None</p>		
ldap.port	Integer from 1 to 65535	389
<p>Description: Configures the port of the LDAP server.</p> <p>Example: ldap.port = 389</p> <p>Web User Interface: Directory->LDAP->Port</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
ldap.base	String within 99 characters	Blank
<p>Description: Configures the LDAP search base which corresponds to the location of the LDAP phone book from which the LDAP search request begins. The search base narrows the search scope and decreases directory search time.</p> <p>Example: ldap.base = dc=yealink,dc=cn</p> <p>Web User Interface: Directory->LDAP->Base</p> <p>Phone User Interface: None</p>		
ldap.user	String within 99 characters	Blank
<p>Description: Configures the user name used to login the LDAP server. This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the user name to login the LDAP server.</p> <p>Example: ldap.user = cn=manager,dc=yealink,dc=cn</p> <p>Web User Interface: Directory->LDAP->Username</p> <p>Phone User Interface: None</p>		
ldap.password	String within 99 characters	Blank
<p>Description: Configures the password to login the LDAP server. This parameter can be left blank in case the server allows anonymous to login. Otherwise you will need to provide the password to login the LDAP server.</p> <p>Example: ldap.password = secret</p> <p>Web User Interface: Directory->LDAP->Password</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
ldap.max_hits	Integer from 1 to 32000	50
<p>Description: Configures the maximum number of search results to be returned by the LDAP server. If the value of the "Max.Hits" is blank, the LDAP server will return all searched results. Please note that a very large value of the "Max. Hits" will slow down the LDAP search speed, therefore it should be configured according to the available bandwidth.</p> <p>Example: ldap.max_hits = 50</p> <p>Web User Interface: Directory->LDAP->Max. Hits (1~32000)</p> <p>Phone User Interface: None</p>		
ldap.name_attr	String within 99 characters	Blank
<p>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.</p> <p>Example: ldap.name_attr = cn sn</p> <p>Web User Interface: Directory->LDAP->LDAP Name Attributes</p> <p>Phone User Interface: None</p>		
ldap.numb_attr	String within 99 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the number attributes of each record to be returned by the LDAP server. You can configure multiple number attributes separated by spaces.</p> <p>Example: ldap.numb_attr = telephoneNumber</p> <p>Web User Interface: Directory->LDAP->LDAP Number Attributes</p> <p>Phone User Interface: None</p>		
ldap.display_name	String within 99 characters	Blank
<p>Description: Configures the display name of the contact record displayed on the LCD screen. The value must start with “%” symbol.</p> <p>Example: ldap.display_name = %cn The cn of the contact record is displayed on the LCD screen.</p> <p>Web User Interface: Directory->LDAP->LDAP Display Name</p> <p>Phone User Interface: None</p>		
ldap.version	2 or 3	3
<p>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.</p> <p>Web User Interface: Directory->LDAP->Protocol</p> <p>Phone User Interface: None</p>		
ldap.call_in_lookup	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to perform an LDAP search when receiving an incoming call.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Directory->LDAP->LDAP Lookup For Incoming Call</p> <p>Phone User Interface: None</p>		
ldap.call_out_lookup	0 or 1	1
<p>Description: Enables or disables the IP phone to perform an LDAP search when placing a call.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Directory->LDAP->LDAP Lookup For Callout</p> <p>Phone User Interface: None</p>		
ldap.ldap_sort	0 or 1	0
<p>Description: Enables or disables the IP phone to sort the search results in alphabetical order or numerical order.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Directory->LDAP->LDAP Sorting Results</p> <p>Phone User Interface: None</p>		

LDAP Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

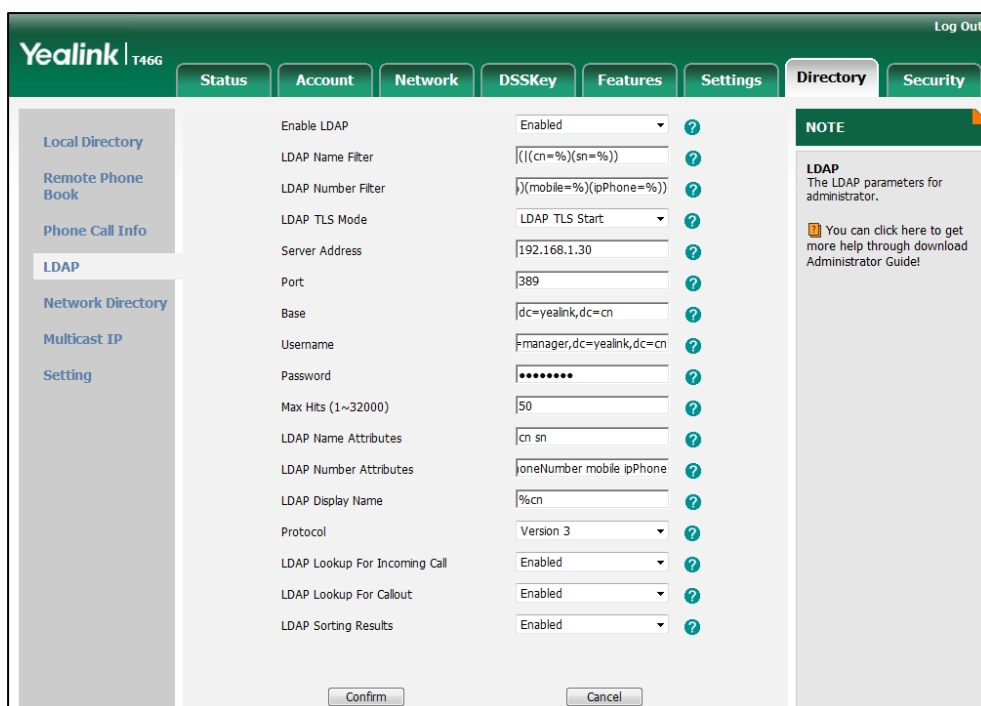
Parameter	Permitted Values	Default
linekey.X.type/programablekey.X.type	38	Refer to the following content
<p>Description: Configures a DSS key as an LDAP key on the IP phone. The digit 38 stands for the key type LDAP. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: linekey.2.type = 38</p> <p>Default: For line keys: For SIPT48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0. For SIPT46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For SIPT42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0. For SIPT41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0. For programable keys: For SIPT48G/T46G IP phones: When X=1, the default value is 28 (History).</p>		

Parameter	Permitted Values	Default
<p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X/ Programmable Key->Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Key X->Type</p>		

To configure LDAP via web user interface:

1. Click on **Directory->LDAP**.
2. Select **Enabled** from the pull-down list of **Enable LDAP**.
3. Enter the values in the corresponding fields.

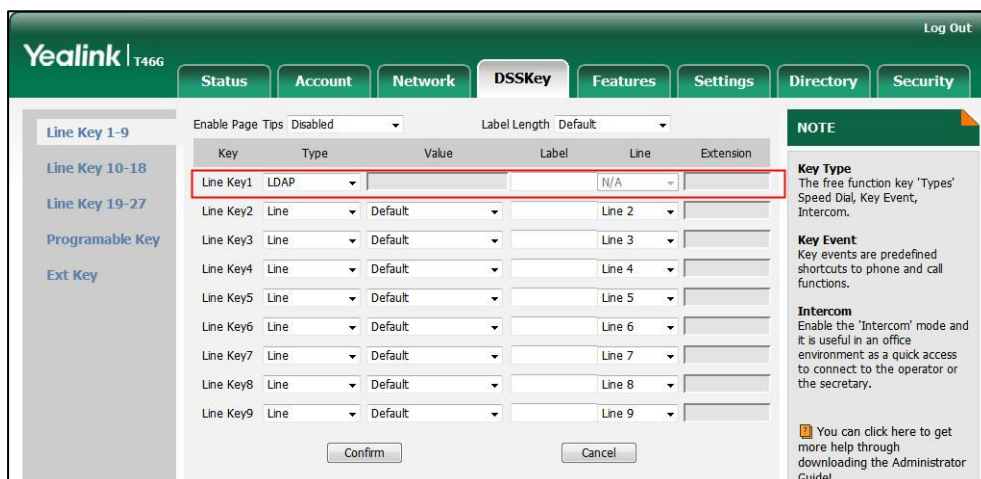
4. Select the desired values from the corresponding pull-down lists.



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

To configure an LDAP key via web user interface:

1. Click on **DSSKey->Line Key X**.
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**.
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. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. 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 IP phone user status (busy or idle). Then when the user places a call, a busy indicator on the supervisor's phone indicates that the user's phone is in use.

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

Visual Alert and Audio Alert for BLF Pickup

Visual alert 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 and 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 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 214.

BLF LED Mode

BLF LED Mode provides four kinds of definition for the BLF key LED status. As there is no hard line key on SIP-T48G IP phones, BLF LED mode is only applicable to SIP-T46G/T42G/T41P IP phones. BLF LED mode configuration is also applicable to the expansion module EXP40 connected to SIP-T48G/T46G IP phones. 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.

BLF LED mode feature is also applicable to BLF list key. For more information on BLF List key, refer to [BLF List](#) on page 297.

Line Key/Expansion Module Key LED (configured as a BLF key or a BLF List 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.

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

Line Key/Expansion Module Key LED (configured as a BLF key or a BLF List 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. The monitored user's conversation is placed on hold (This LED status requires server support).
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.

Line Key/Expansion Module Key LED (configured as a BLF key a BLF List 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. The monitored user's conversation is placed on hold (This LED status requires server support).
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.

Line Key/Expansion Module Key LED (configured as a BLF key a BLF List key and BLF LED Mode is set to 3)

LED Status	Description
Fast flashing green (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 (This LED status requires server support).

LED Status	Description
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.

Note BLF LED Mode feature is only applicable to IP phones running firmware version 72 or

Procedure

BLF can be configured using the configuration files or locally.

Configuration File	y000000000xx.cfg	<p>Specify whether to use visual alert and audio alert for BLF pickup.</p> <p>Parameters:</p> <p>features.pickup.blf_visual_enable features.pickup.blf_audio_enable</p> <p>Assign a BLF key.</p> <p>Parameters:</p> <p>linekey.X.type linekey.X.line linekey.X.value linekey.X.pickup_value</p> <p>Configure BLF LED mode.</p> <p>Parameter:</p> <p>features.blf_led_mode</p>
Local	Web User Interface	<p>Assign a BLF key.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p> <p>Specify whether to use visual alert and audio alert for BLF pickup.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-callpickup&q=load</p> <p>Configure BLF LED mode.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>

	Phone User Interface	Assign a BLF key.
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Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.pickup.blf_visual_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to display a visual alert when the monitored user receives an incoming call.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Call Pickup->Visual Alert for BLF Pickup</p> <p>Phone User Interface: None</p>		
features.pickup.blf_audio_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to play an audio alert when the monitored user receives an incoming call.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Features->Call Pickup->Audio Alert for BLF Pickup</p> <p>Phone User Interface: None</p>		
features.blf_led_mode	0, 1, 2 or 3	0
<p>Description: Configures BLF LED mode and provides four kinds of definition for the BLF key LED status.</p> <p>Web User Interface: Features->General Information->BLF LED Mode</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		

BLF Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

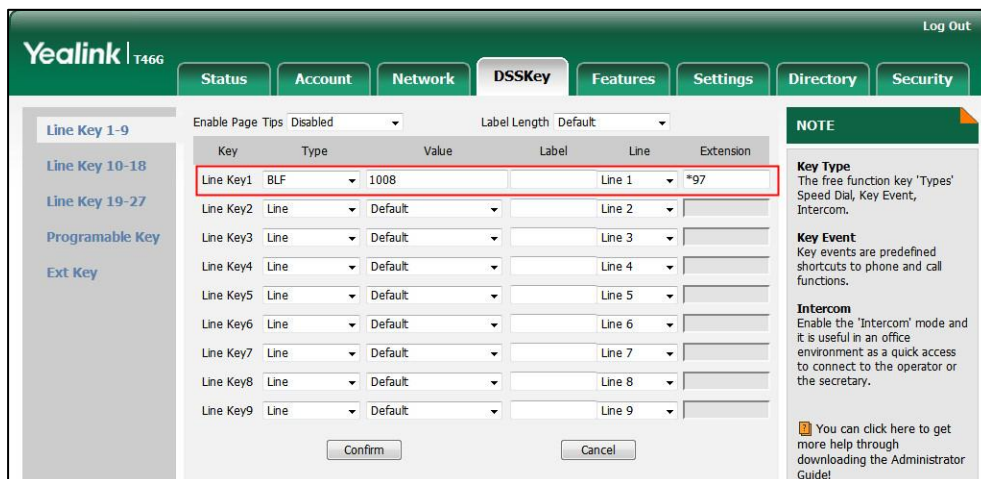
Parameters	Permitted Values	Default
linekey.X.type	16	Refer to the following content
<p>Description: Configures a line key to be a BLF key on the IP phone. The digit 16 stands for the key type BLF. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.type = 16</p> <p>Default:</p> <p>For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Type</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		

Parameters	Permitted Values	Default
linekey.X.line	Integer from 1 to 16	1-6 correspond to the lines 1-6.
<p>Description: Configures the desired line to apply the BLF key. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). The valid values are: 1 to 16 (for SIP-T48G/T46G) 1 to 12 (for SIP-T42G) 1 to 6 (for SIP-T41P) 1-Line 1 2-Line 2 ... 16-Line 16</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Line</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Account ID</p>		
linekey.X.value	String within 99 characters	blank
<p>Description: Configures the number of the monitored user. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.3.value = 1008</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		
linekey.X.pickup_value	String within 256 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the pickup code for BLF feature. This parameter only applies to BLF feature. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.3.pickup_value = *97</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Extension</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Extension</p>		

To configure a BLF key via web user interface:

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

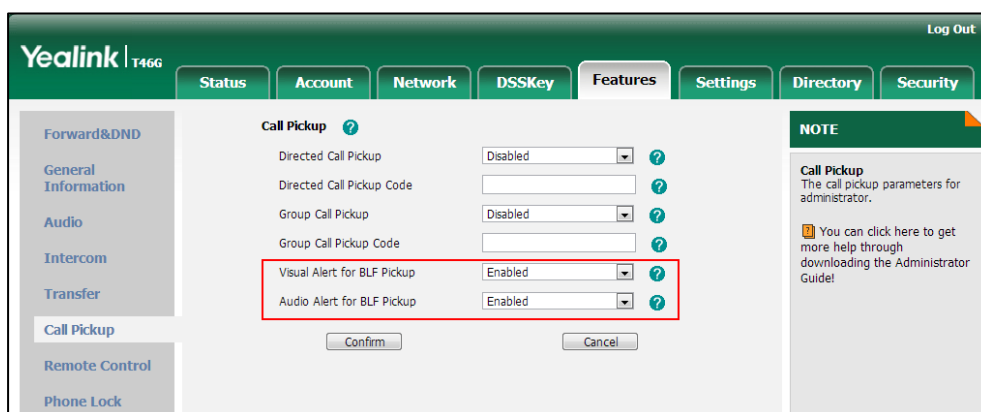


6. Click **Confirm** to accept the change.

To configure visual alert and audio alert features 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**.

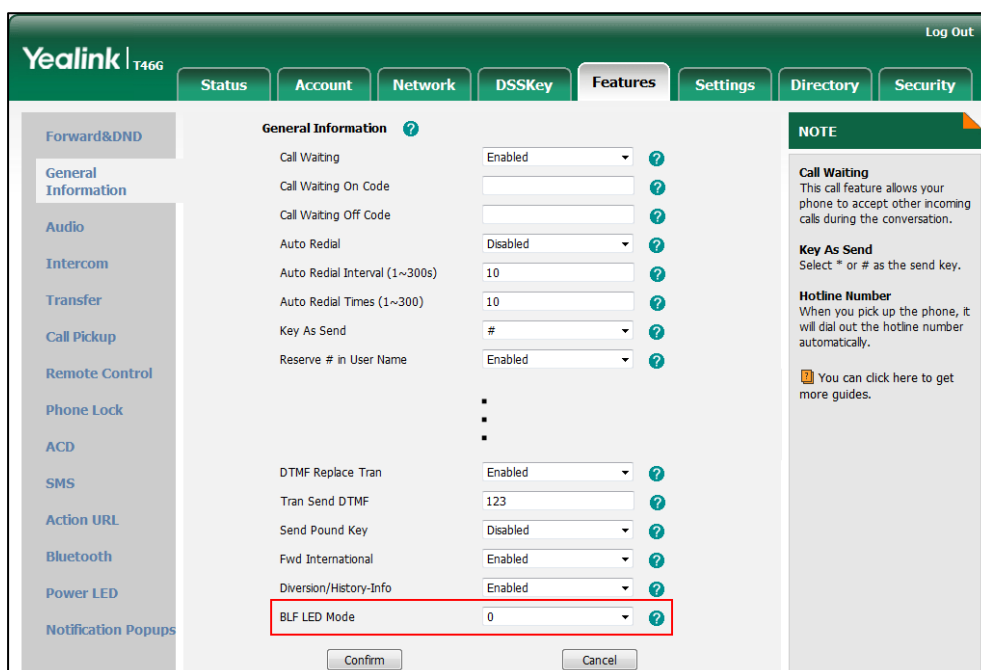
3. Select the desired value from the pull-down list of **Audio Alert for BLF Pickup**.



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**.
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. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the phone number or extension you want to monitor in the **Value** field.
7. (Optional.) Enter the directed call pickup code in the **Extension** field.
8. Press the **Save** soft key to accept the change.

BLF List

Busy Lamp Field (BLF) List allows a list of specific extensions to be monitored for status changes. It enables the monitoring phone to subscribe to a list of users, and receive notifications of the status of monitored users. Different indicators on the monitoring phone show the status of monitored users. The monitoring user can also be notified about calls being parked/no longer parked against any monitored user. IP phones support BLF list using a SUBSCRIBE/NOTIFY mechanism as specified in RFC 3265. This feature depends on support from a SIP server.

Note

BLF list feature is not applicable to SIP-T19P IP phones and applicable to IP phones running firmware version 73 or later in the neutral version.

Procedure

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

Configuration File	y0000000000xx.cfg	<p>Configure BLF List.</p> <p>Parameters:</p> <p>account.X.blf.blf_list_uri</p> <p>account.X.blf_list_code</p> <p>account.X.blf_list_barge_in_code</p> <p>account.X.blf_list_retrieve_call_parked_code</p> <p>Specify whether to automatically configure the BLF list keys.</p> <p>Parameter:</p> <p>phone_setting.auto_blf_list_enable</p> <p>Configure the order of BLF list keys assigned automatically.</p> <p>Parameter:</p> <p>phone_setting.blf_list_sequence_type</p> <p>Assign a BLF List key.</p> <p>Parameters:</p> <p>linekey.X.type</p> <p>linekey.X.line</p>
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Local	Web User Interface	<p>Configure BLF List.</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p> <p>Assign a BLF List key.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=dskey&model=1&q=load&linepage=1</p>
	Phone User Interface	Assign a BLF List key.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.blf.blf_list_uri	String within 256 characters	Blank
<p>Description:</p> <p>Configures the BLF List URI to monitor a list of users for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.blf.blf_list_uri = 4609@pbx.yealink.com</p> <p>Web User Interface:</p> <p>Account->Advanced->BLF List URI</p> <p>Phone User Interface:</p> <p>None</p>		
account.X.blf_list_code	String within 32 characters	Blank
<p>Description:</p> <p>Configures the directed pickup code for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example:</p> <p>account.1.blf_list_code = *97</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: Account->Advanced->BLF List Code</p> <p>Phone User Interface: None</p>		
account.X.blf_list_barge_in_code	String within 32 characters	Blank
<p>Description: Configures the barge-in code for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.blf_list_barge_in_code = *33</p> <p>Note: It is not applicable to SIP-T19P IP phones.</p> <p>Web User Interface: Account->Advanced->BLF List Barge In Code</p> <p>Phone User Interface: None</p>		
account.X.blf_list_retrieve_call_parked_code	String within 32 characters	Blank
<p>Description: Configures the call park retrieve code for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.blf_list_retrieve_call_parked_code = *88</p> <p>Web User Interface: Account->Advanced->BLF List Retrieve call parked Code</p> <p>Phone User Interface: None</p>		
phone_setting.auto_blf_list_enable	0 or 1	1

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to automatically configure the BLF list keys. 0-Disabled 1-Enabled</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
phone_setting.blf_list_sequence_type	0 or 1	0
<p>Description: Configures the order of BLF list keys assigned automatically. 0-Line Key->Ext Key 1-Ext Key->Line Key</p> <p>Note: It is only applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

BLF List Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

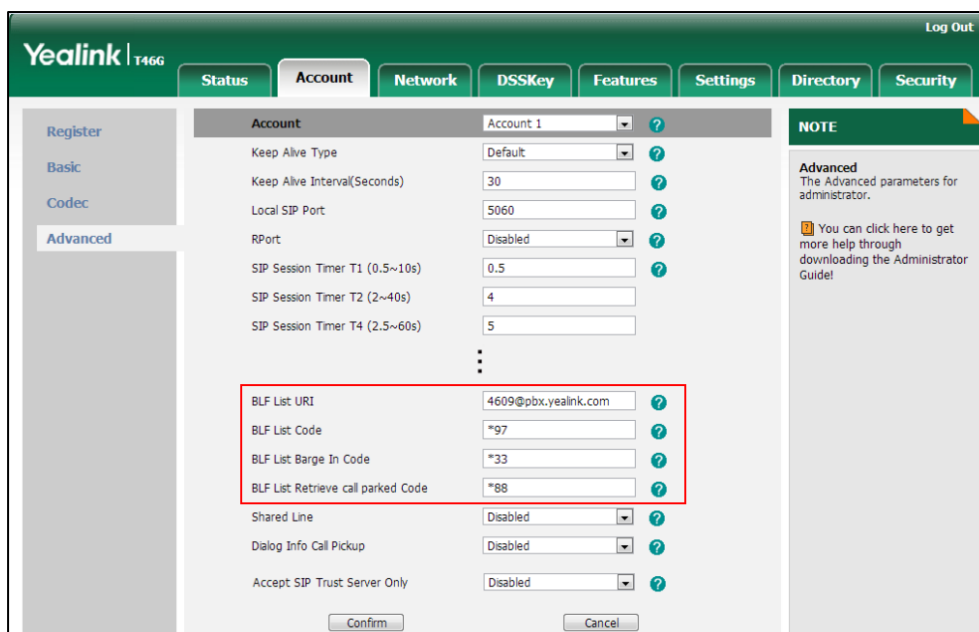
Parameters	Permitted Values	Default
linekey.X.type	39	Refer to the following content
<p>Description: Configures a DSS key as a BLF List key on the IP phone. The digit 39 stands for the key type BLF List. X ranges from 1 to 29 (for SIPT48G). X ranges from 1 to 27 (for SIPT46G). X ranges from 1 to 15 (for SIPT42G/T41P).</p>		

Parameters	Permitted Values	Default
<p>Example: linekey.2.type = 39</p> <p>Default:</p> <p>For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface: DSSKey->Line Key X->Type</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.line	Integer from 1 to 16	1-6 correspond to the lines 1-6
<p>Description: Configures the desired line to apply the BLF List key. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). The valid values are: 1 to 16 (for SIP-T48G/T46G) 1 to 12 (for SIP-T42G) 1 to 6 (for SIP-T41P) 1-Line 1 2-Line 2 ... 16-Line 16</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
DSSKey->Line Key X/Programable Key ->Line		
Phone User Interface:		
Menu->Features->DSS Keys->Line Key X->Account ID		

To configure the BLF List settings 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 BLF List URI in the **BLF List URL** field.
5. (Optional.) Enter the directed pickup code in the **BLF List Code** field.
6. (Optional.) Enter the barge-in code in the **BLF List Barge In Code** field.
7. (Optional.) Enter the retrieve call parked code in the **BLF List Retrieve call parked Code** field.

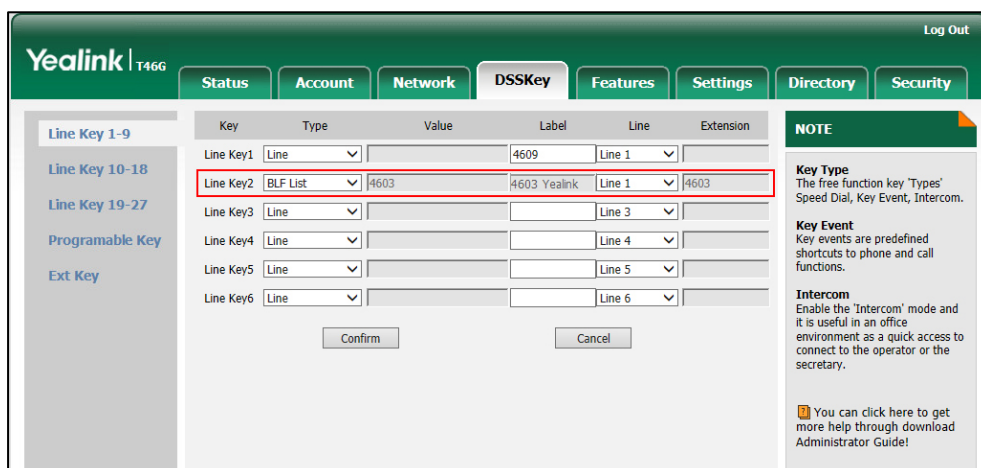


8. Click **Confirm** to accept the change.

To configure BLF List keys manually via web user interface:

1. Click on **DSSKey->Line Key X**.
2. In the desired DSS key field, select **BLF List** from the pull-down list of **Type**.

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



4. Repeat step 2-3, configure more BLF list keys.
5. Click **Confirm** to accept the change.

Hide Features Access Code

Hide Features Access Code feature enables the IP phone to display the feature identifier instead of the dialed feature access code automatically. For example, the dialed call park code will be replaced by the identifier “Call Park” when you park an active call.

The hide feature access codes feature is applicable to the following features:

- Voice Mail
- Pick up
- Group Pick up
- Barge In
- Retrieve
- Call Park
- Group Park

Procedure

The hide feature access codes feature can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure the hide feature access codes feature:</p> <p>Parameters:</p> <p>features.hide_feature_access_codes.enable</p>
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Local	Web User Interface	<p>Configure the hide feature access codes feature.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-general&q=load</p>
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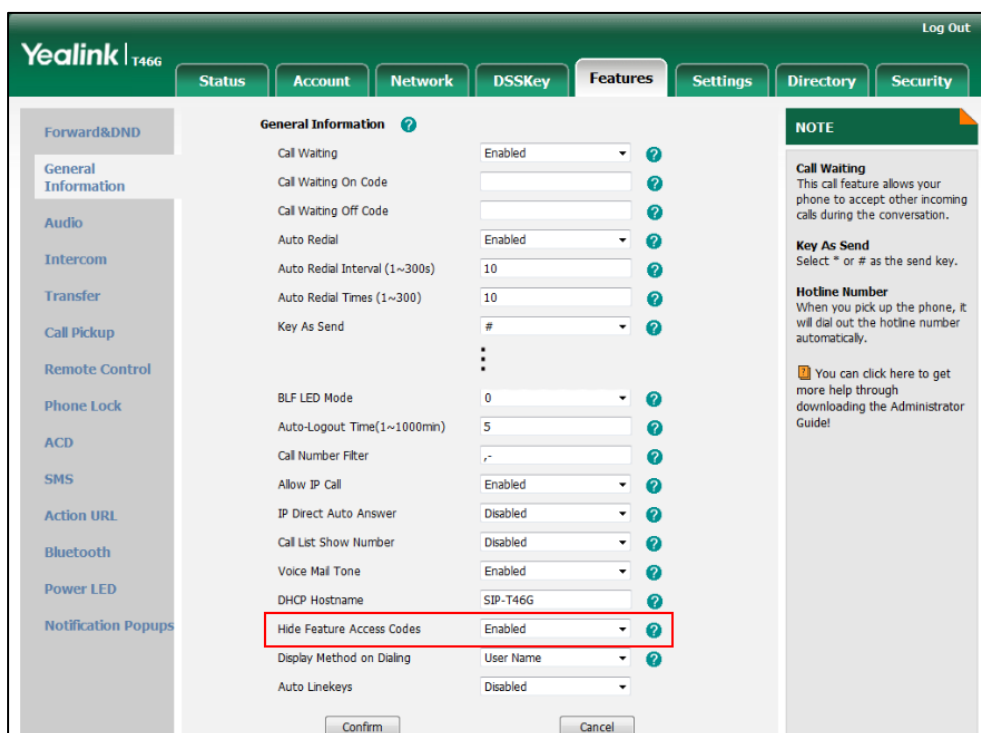
Details of Configuration Parameters:

Parameters	Permitted Values	Default
features.hide_feature_access_codes.enable	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to display feature name instead of the feature access code when dialing and in talk.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface:</p> <p>Features->General Information->Hide Feature Access Codes</p> <p>Phone User Interface:</p> <p>None</p>		

To enable hide feature access codes feature via web user interface:

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

2. Select **Enabled** from the pull-down list of **Hide Feature Access Codes**.



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

Automatic Call Distribution

Automatic Call Distribution (ACD) enables organizations to manage a large number of phone calls on an individual basis. ACD enables 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 IP 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 IP phone status and then decides whether to assign an incoming call to the user’s IP phone. When the IP 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 IP phone status or the ACD auto available timer (if configured) expires. How long the IP phone remains unavailable is configurable by the auto available timer. When the timer expires, the IP phone status is automatically changed to available. ACD auto available timer 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 on the IP phone indicates the ACD status.

Procedure

ACD can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure ACD feature for account: Parameters: account.X.acd.enable account.X.acd.available
	<y0000000000xx>.cfg	Assign an ACD key. Parameters: linekey.X.type Configure ACD auto available. Parameters: acd.auto_available acd.auto_available_timer
Local	Web User Interface	Assign an ACD key. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linenpage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linenpage=1 Configure ACD auto available timer. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-acd&q=load">http://<phoneIPAddress>/servlet?p=features-acd&q=load
	Phone User Interface	Assign an ACD key.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
acd.auto_available	0 or 1	0
<p>Description: Enables or disables the IP phone to automatically change the status of the ACD agent to available after the designated time.</p> <p>0-Disabled 1-Enabled</p>		

Parameters	Permitted Values	Default
<p>Web User Interface:</p> <p>Features->ACD->ACD Auto Available</p> <p>Phone User Interface:</p> <p>None</p>		
acd.auto_available_timer	Integer from 0 to 120	60
<p>Description:</p> <p>Configures the length of time (in seconds) before the status of the ACD agent is automatically changed to available.</p> <p>Web User Interface:</p> <p>Features->ACD->ACD Auto Available Timer (0~120s)</p> <p>Phone User Interface:</p> <p>None</p>		
account.X.acd.enable	0 or 1	0
<p>Description:</p> <p>Enables or disables ACD feature for account X.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p> <p>X ranges from 1 to 12 (for SIP-T42G).</p> <p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
account.X.acd.available	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to display the available and unavailable soft keys for account X after the IP phone logs into the ACD system.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p>		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

ACD Key

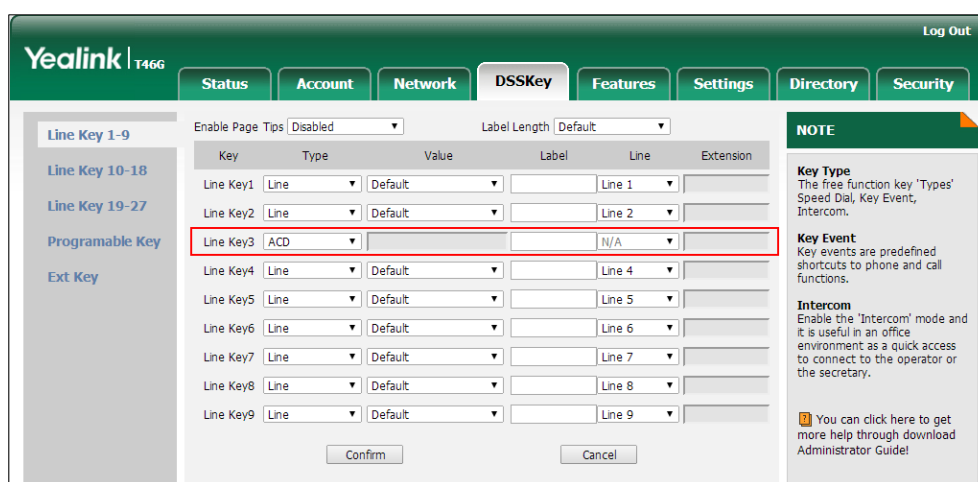
For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameter	Permitted Values	Default
linekey.X.type	42	Refer to the following content
<p>Description: Configures a line key to be an ACD key on the IP phone. The digit 42 stands for the key type ACD. X ranges from 1 to 29 (for SIPT48G). X ranges from 1 to 27 (for SIPT46G). X ranges from 1 to 15 (for SIPT42G/T41P).</p> <p>Example: linekey.2.type = 42</p> <p>Default:</p> <p>For SIPT48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIPT46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIPT42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIPT41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is</p>		

Parameter	Permitted Values	Default
0. Web User Interface: DSSKey->Line Key X->Line KeyX->Type Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type		

To configure an ACD key via web user interface:

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

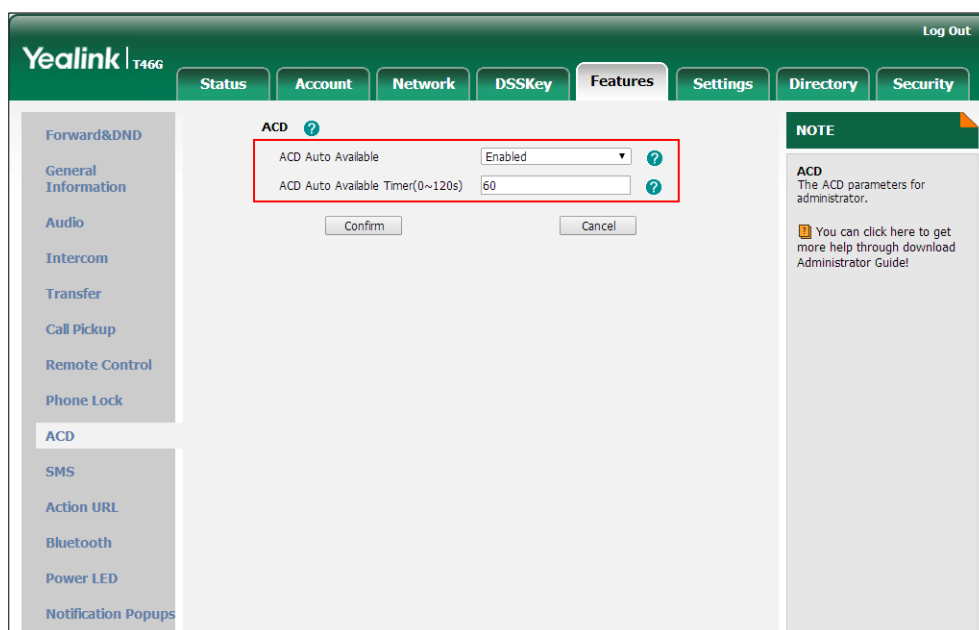


3. Click **Confirm** to accept the change.

To configure the ACD auto available timer feature via web user interface:

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

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



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

To configure an ACD key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **←** or **→** , or the **Switch** soft key to select **ACD** from the **Type** field.
4. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
5. 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.

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 to 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.</p> <p>Parameters:</p> <p>account.X.subscribe_mwi account.X.subscribe_mwi_expires account.X.subscribe_mwi_to_vm</p> <p>Configure subscribe MWI to voice mail.</p> <p>Parameter:</p> <p>voice_mail.number.X</p> <p>Configure the presentation of audio and visual MWI.</p> <p>Parameter:</p> <p>account.X.display_mwi.enable</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure subscribe for MWI.</p> <p>Configure subscribe MWI to voice mail.</p> <p>Configure the presentation of audio and visual MWI.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>

Details of Configuration Parameters:

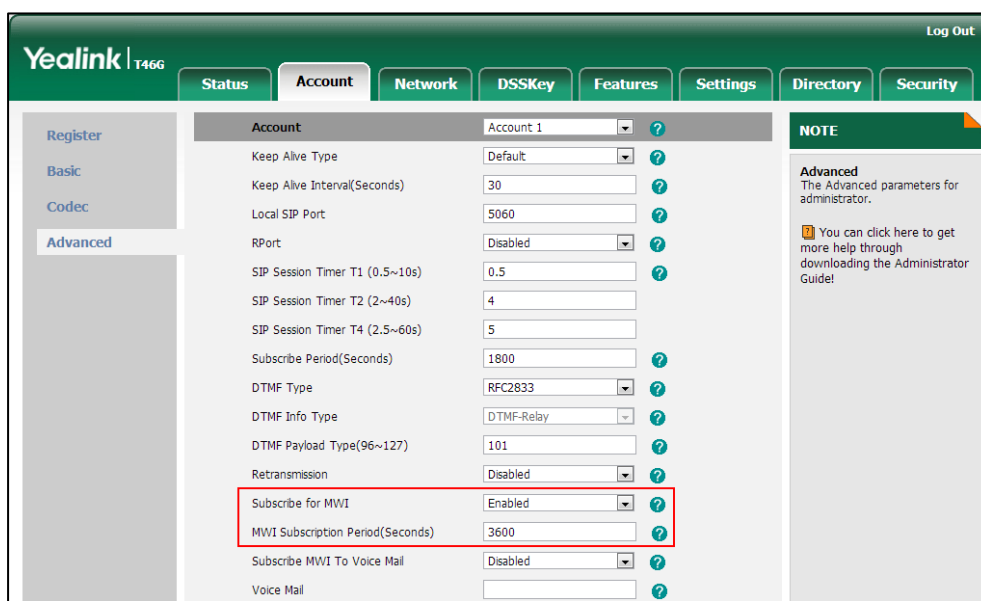
Parameters	Permitted Values	Default
<p>account.X.subscribe_mwi</p>	<p>0 or 1</p>	<p>0</p>
<p>Description:</p> <p>Enables or disables the IP phone to subscribe the message waiting indicator for account X.</p> <p>If it is set to 1 (Enabled), the IP phone will send a SUBSCRIBE message to the server for message-summary updates.</p> <p>0-Disabled 1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p>		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->Subscribe for MWI</p> <p>Phone User Interface: None</p>		
account.X.subscribe_mwi_expires	Integer from 0 to 84600	3600
<p>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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Note: It works only if the value of the parameter "account.X.subscribe_mwi" is set to 1 (Enabled).</p> <p>Web User Interface: Account->Advanced->MWI Subscription Period (Seconds)</p> <p>Phone User Interface: None</p>		
account.X.subscribe_mwi_to_vm	0 or 1	0
<p>Description: Enables or disables the IP phone to subscribe the message waiting indicator to the voice mail number for account X.</p> <p>0-Disabled 1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Note: It works only if the value of the parameters "account.X.subscribe_mwi" is set to 1 (Enabled) and "voice_mail.number.X" is configured.</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
Account->Advanced->Subscribe MWI To Voice Mail Phone User Interface: None		
voice_mail.number.X	String within 99 characters	Blank
Description: Configures the voice mail number for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Example: voice_mail.number.1 = 1234 Note: It works only if the value of the parameter "account.X.subscribe_mwi_to_vm" is set to 1 (Enabled). Web User Interface: Account->Advanced->Voice Mail Phone User Interface: None		
account.X.display_mwi.enable (X ranges from 1 to 6)	0 or 1	1
Description: Enables or disables the IP phone to present audio and visual MWI when it receives new voice mails. 0-Disabled 1-Enabled X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P). Web User Interface: Account->Advanced->Voice Mail Display Phone User Interface: None		

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.



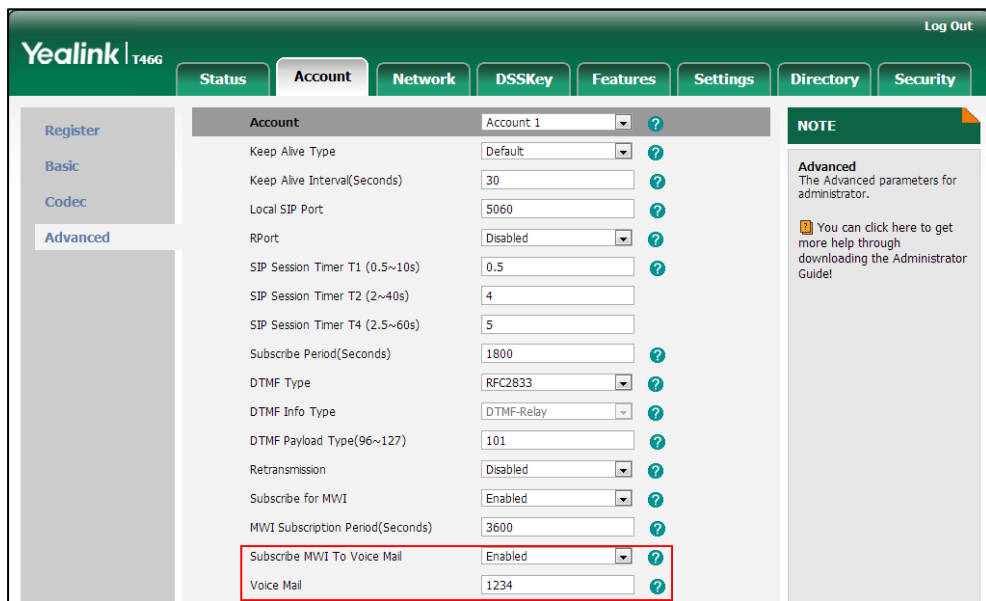
6. Click **Confirm** to accept the change.

The IP phone will subscribe to the account number for MWI service by default.

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

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**.

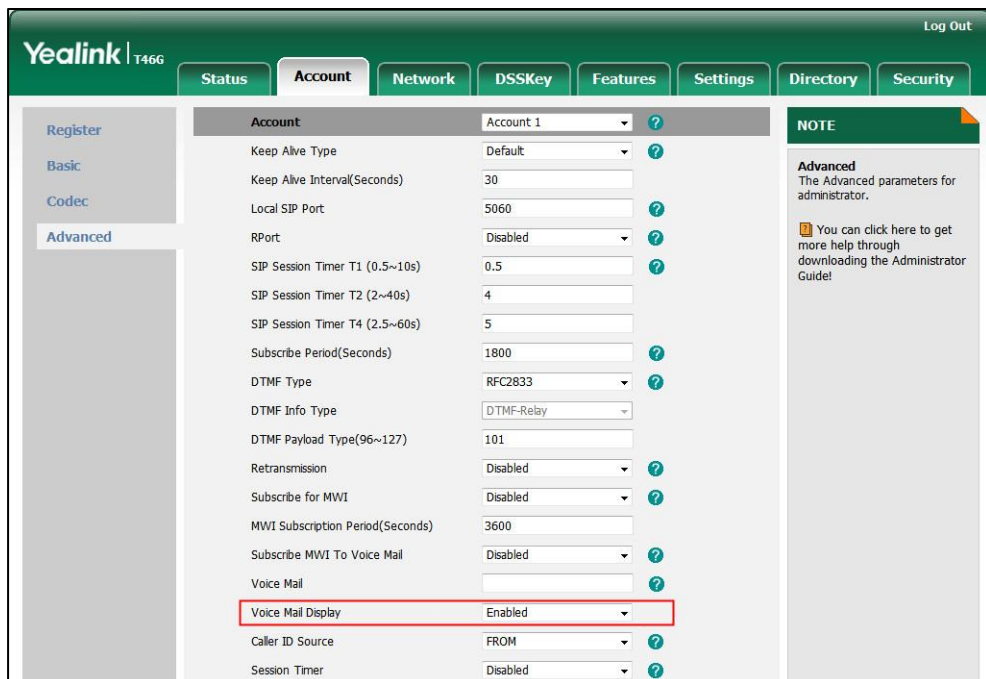
5. Enter the desired voice number in the **Voice Mail** field.



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

To configure the presentation of audio and visual 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 **Voice Mail Display**.



5. 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 or paging list 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 that 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.

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Specify a multicast codec for the IP phone to use for multicast RTP.</p> <p>Parameter:</p> <p>multicast.codec</p> <p>Assign a multicast paging key.</p> <p>Parameters:</p> <p>linekey.X.type</p> <p>linekey.X.value</p> <p>Assign a paging list key.</p> <p>Parameters:</p> <p>linekey.X.type</p> <p>linekey.X.value</p> <p>Configure the multicast IP address and port number for a paging list key.</p> <p>Parameter:</p> <p>multicast.paging_address.X.ip_address</p> <p>Configure the multicast paging group name for a paging list key.</p> <p>Parameter:</p>
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		multicast.paging_address.X.label
Local	Web User Interface	<p>Assign a multicast paging key or a paging list key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p> <p>Specify a multicast codec for the IP phone to send the RTP stream.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=features-general&q=load">http://<phoneIPAddress>/servlet?p=features-general&q=load</p> <p>Configure the multicast IP address and port number for a paging list key.</p> <p>Configure the multicast paging group name for a paging list key.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load">http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load</p>
	Phone User Interface	Assign a multicast paging key or a paging list key.

Details of the Configuration Parameter:

Parameters	Permitted Values	Default
multicast.codec	Refer to the following content	G722
<p>Description: Configures the codec of multicast paging.</p> <p>Example: multicast.codec = G722</p> <p>Permitted Values: PCMU, PCMA, G729, G722</p> <p>Web User Interface: Features->General Information->Multicast Codec</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
multicast.paging_address.X.ip_address	String	Blank
<p>Description: Configures the multicast IP address and port number for a paging list key. X ranges from 1 to 10.</p> <p>Example: multicast.paging_address.1.ip_address = 224.5.6.20:10008</p> <p>Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.</p> <p>Web User Interface: Directory->Multicast IP->Paging List->Paging Address</p> <p>Phone User Interface: Menu->Features->Others->Option->Edit->Address</p>		
multicast.paging_address.X.label	String	Blank
<p>Description: Configures the multicast paging group name for a paging list key. X ranges from 1 to 10.</p> <p>Example: multicast.paging_address.1.label = Product</p> <p>Web User Interface: Directory->Multicast IP->Paging List->Label</p> <p>Phone User Interface: Menu->Features->Others->Option->Edit->Label</p>		

Multicast Paging Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameters	Permitted Values	Default
linekey.X.type	24	Refer to the following content
<p>Description: Configures a line key to be a multicast paging key on the IP phone. The digit 24 stands for the key type Multicast Paging.</p>		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.type = 24</p> <p>Default:</p> <p>For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface: DSSKey->Line Key X->Line Key X->Type</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.value	String within 99 characters	Blank
<p>Description: Configures the multicast IP address and port number. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.</p> <p>Web User Interface: DSSKey->Line Key X->Line Key X->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		

Paging List key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

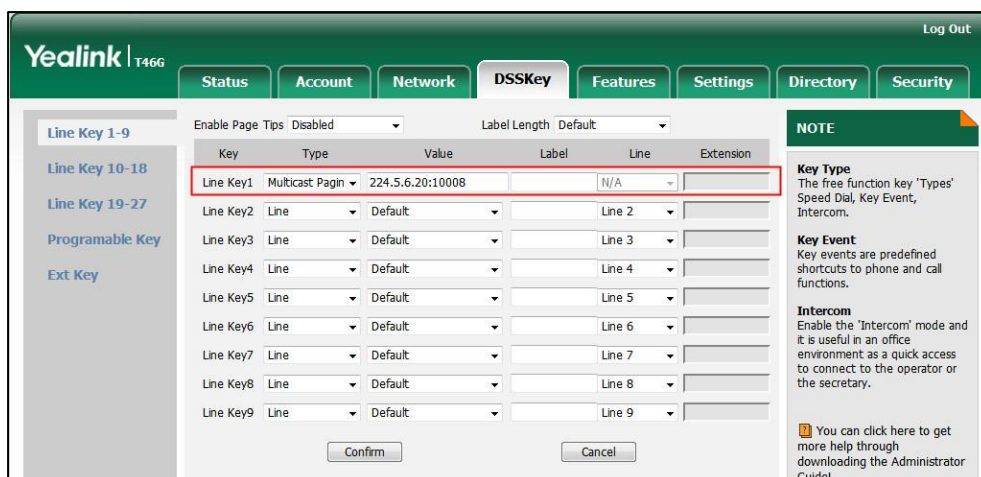
Parameters	Permitted Values	Default
<code>linekey.X.type/programablekey.X.type</code>	66	Refer to the following content
<p>Description: Configures a DSS key as a paging list key on the IP phone. The digit 66 stands for the key type Paging List. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: <code>linekey.1.type = 66</code></p> <p>Default: For line keys: For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0. For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0. For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0. For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0. For programable keys: For SIP-T48G/T46G IP phones: When X=1, the default value is 28 (History). When X=2, the default value is 61 (Directory).</p>		

Parameters	Permitted Values	Default
<p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>Web User Interface:</p> <p>DSSKey->Line Key X/ Programable Key>Type</p> <p>Phone User Interface:</p> <p>Menu->Features->DSS Keys->Line Keys X->Type</p>		

To configure a multicast paging key via web user interface:

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

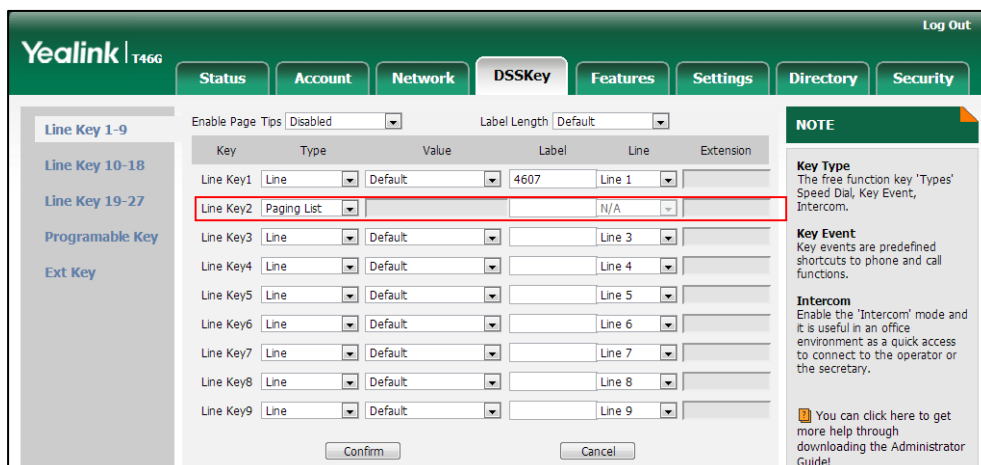
The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.



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

To configure a paging list key via web user interface:

1. Click on **DSSKey->Line Key X**.
2. In the desired DSS key field, select **Paging List** from the pull-down list of **Type**.

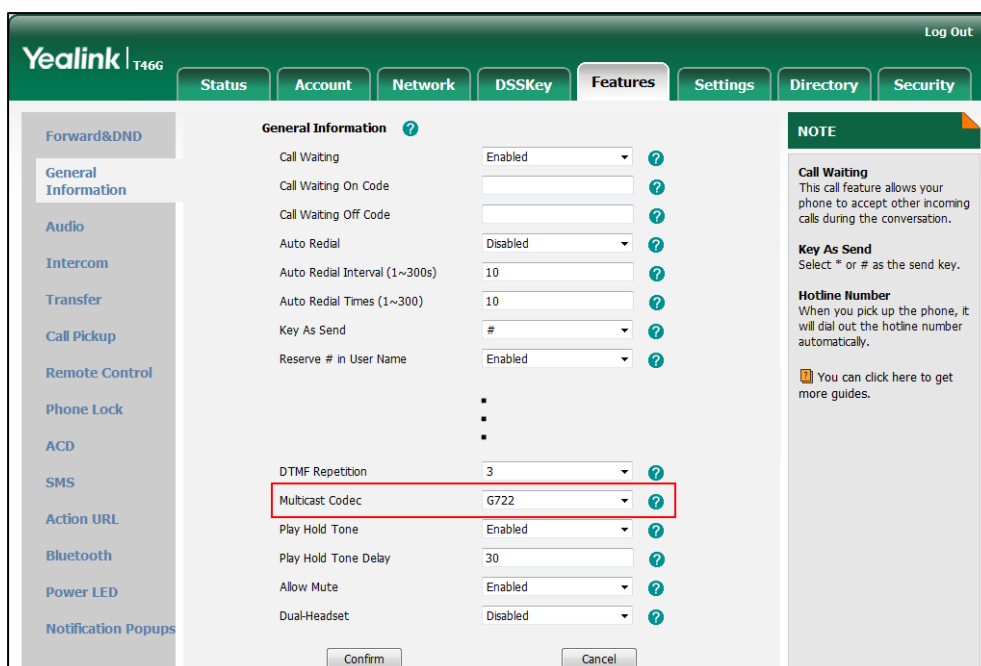


3. Click **Confirm** to accept the change.

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**.

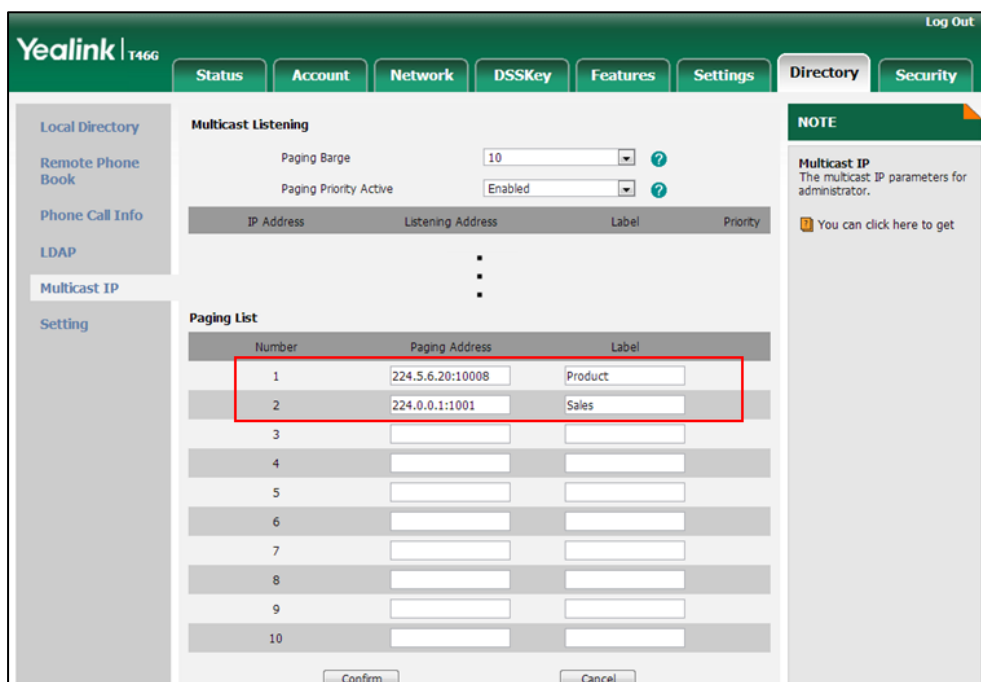


3. Click **Confirm** to accept the change.

To configure multicast addresses for a paging list key via web user interface:





1. Click on **Directory->Multicast IP**.
2. Enter the multicast address and port number in the **Paging Address** field.
3. Enter the label in the **Label** field.

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







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

To configure a multicast paging key via phone user interface:

1. Press **Menu->Features->DSS 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 **Paging** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Enter the multicast IP address and port number in the **Value** field.
7. Press the **Save** soft key to accept the change.

To configure a paging list key via phone user interface:

1. Press **Menu->Features->DSS 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 **Paging List** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

To configure paging list via phone user interface:

1. Press **Menu->Features->Others->Paging List**.
2. Press the **Option** soft key.
3. Press the **Edit** soft key.
4. Enter the multicast IP address and port number (e.g., 224.5.6.20:10008) in the **Address** field.

The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.

5. Enter the group name in the **Label** field.
6. Press the **Save** soft key to accept the change.

Repeat the step 2-6, you can add more paging groups.

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>Parameters: multicast.listen_address.X.label multicast.listen_address.X.ip_address</p> <p>Configure the Paging Barge and Paging Priority Active features.</p> <p>Parameters: multicast.receive_priority.enable multicast.receive_priority.priority</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure the listening multicast address.</p> <p>Configure the Paging Barge and Paging Priority Active features.</p> <p>Navigate to: http://<phoneIPAddress>/servlet?p=contacts-multicastIP&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
<p>multicast.listen_address.X.ip_address (X ranges from 1 to 10)</p>	<p>IP address: port</p>	<p>Blank</p>
<p>Description: Configures the multicast address and port number that the IP phone listens to.</p>		

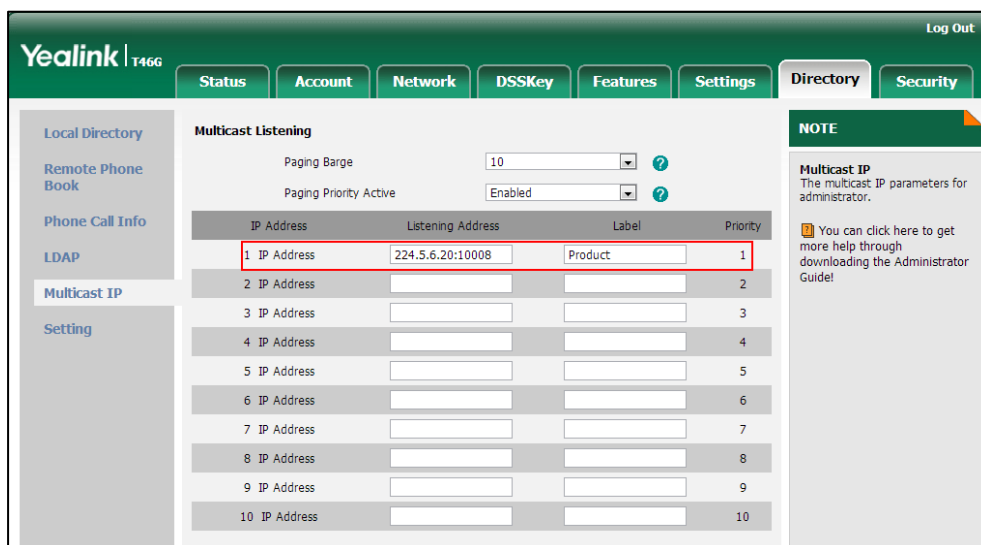
Parameters	Permitted Values	Default
<p>Example: multicast.listen_address.1.ip_address = 224.5.6.20:10008</p> <p>Note: The valid multicast IP addresses range from 224.0.0.0 to 239.255.255.255.</p> <p>Web User Interface: Directory->Multicast IP->Multicast Listening->Listening Address</p> <p>Phone User Interface: None</p>		
<p>multicast.listen_address.X.label (X ranges from 1 to 10)</p>	<p>String within 99 characters</p>	<p>Blank</p>
<p>Description: Configures the label to be displayed on the LCD screen when receiving the RTP multicast.</p> <p>Example: multicast.listen_address.1.label = Product</p> <p>Web User Interface: Directory->Multicast IP->Multicast Listening->Label</p> <p>Phone User Interface: None</p>		
<p>multicast.receive_priority.enable</p>	<p>0 or 1</p>	<p>1</p>
<p>Description: Enables or disables the IP phone to handle the incoming multicast paging calls when there is an active multicast paging call on the IP phone.</p> <p>0-Disabled 1-Enabled</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> <p>Web User Interface: Directory->Multicast IP->Multicast Listening->Paging Priority Active</p> <p>Phone User Interface: None</p>		
<p>multicast.receive_priority.priority</p>	<p>Integer from 0 to 10</p>	<p>10</p>
<p>Description:</p>		

Parameters	Permitted Values	Default
<p>Configures the priority of multicast paging calls.</p> <p>1 is the highest priority, 10 is the lowest priority.</p> <p>If it is set to 0, all incoming multicast paging calls will be automatically ignored.</p> <p>Web User Interface:</p> <p>Directory->Multicast IP->Multicast Listening->Paging Barge</p> <p>Phone User Interface:</p> <p>None</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.

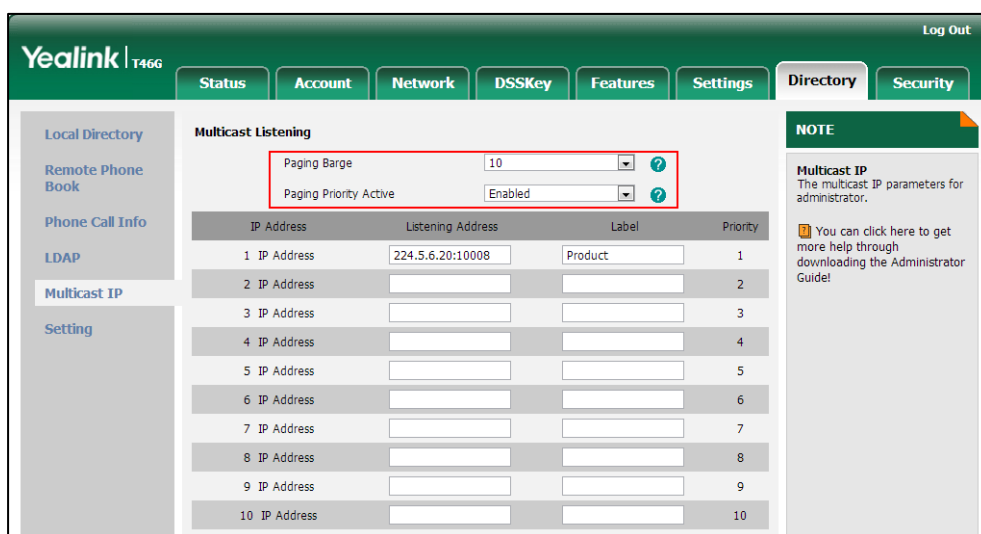


4. Click **Confirm** to accept the change.

To configure the 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.

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-T46G 28.72.0.1
```

```
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-T46G 28.72.0.1
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-T46G 28.72.0.1 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.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Assign a record key. Parameters: linekey.X.type Assign a URL record key. Parameters: linekey.X.type linekey.X.value</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Assign a record key and URL record key. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1">http://<phoneIPAddress>/servlet?p=dsskey&model=1&q=load&linepage=1</p>
	<p>Phone User Interface</p>	<p>Assign a record key and URL record key.</p>

Record Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameters	Permitted Values	Default
linekey.X.type	25	Refer to the following content
<p>Description: Configures a line key to be a record key on the IP phone. The digit 25 stands for the key type Record. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.1.type =25</p> <p>Default:</p> <p>For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface: DSSKey->Line Key X->Line Key X->Type</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		

URL Record Key

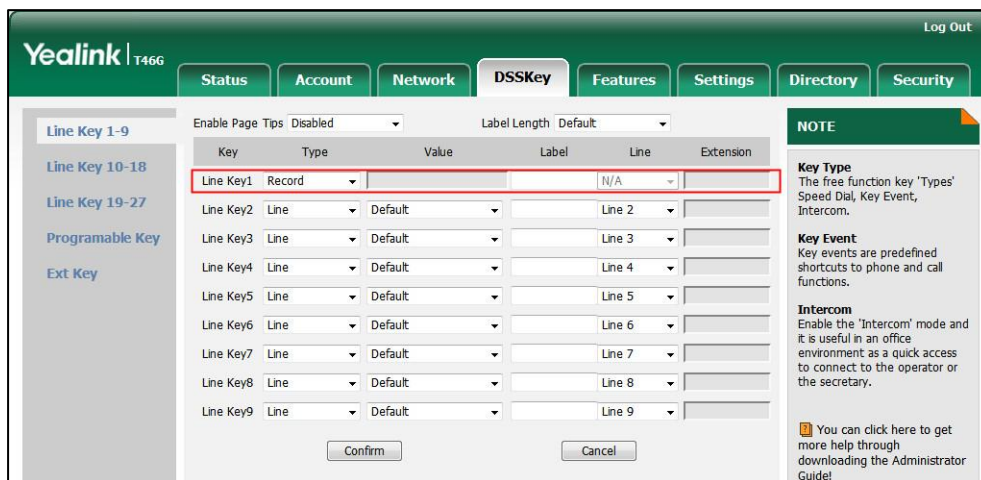
For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

Parameters	Permitted Values	Default
linekey.X.type	35	Refer to the following content
<p>Description: Configures a line key to be a URL record key on the IP phone. The digit 35 stands for the key type URL Record. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.1.type =35</p> <p>Default:</p> <p>For SIP-T48G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Type</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Type</p>		
linekey.X.value	String within 99 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the URL to record a call. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>Example: linekey.2.value = http://10.1.2.224/phonerecording.cgi</p> <p>Web User Interface: DSSKey->Line Key X->Line KeyX->Value</p> <p>Phone User Interface: Menu->Features->DSS Keys->Line Key X->Value</p>		

To configure a record key via web user interface:

1. Click on **DSSKey->Line Key X**.
2. In the desired DSS key field, select **Record** from the pull-down list of **Type**.

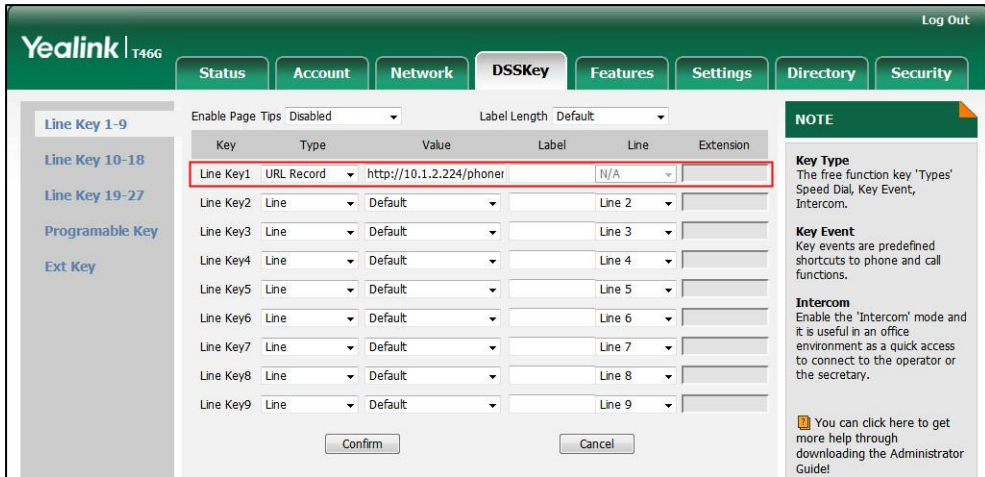


3. Click **Confirm** to accept the change.

To configure a URL record key via web user interface:

1. Click on **DSSKey->Line Key X**.
2. In the desired DSS key field, select **URL Record** from the pull-down list of **Type**.

3. Enter the URL in the **Value** field.



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

To configure a record key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **◀** or **▶**, or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press **◀** or **▶**, or the **Switch** soft key to select **Record** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. 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**.
2. Select the desired DSS key.
3. Press **◀** or **▶**, or the **Switch** soft key to select **URL Record** from the **Type** field.
4. Enter the URL in the **URL Record** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. Press the **Save** soft key to accept the change.

Hot Desking

Hot desking originates from the definition of being the temporary physical occupant of a work station or surface by a particular employee. A primary motivation for hot desking is cost reduction. Hot desking is regularly used in places where not all the employees are in the office at the same time, or not in the office for 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. 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. Parameter: linekey.X.type/ programablekey.X.type
Local	Web User Interface	Assign a hot desking key. Navigate to: http://<phoneIPAddress>/servlet ?p=dsskey&q=load&model=1
	Phone User Interface	Assign a hot desking key.

Hot Desking Key

For more information on how to configure the DSS Key, refer to [Appendix C: Configuring DSS Key](#) on page 514.

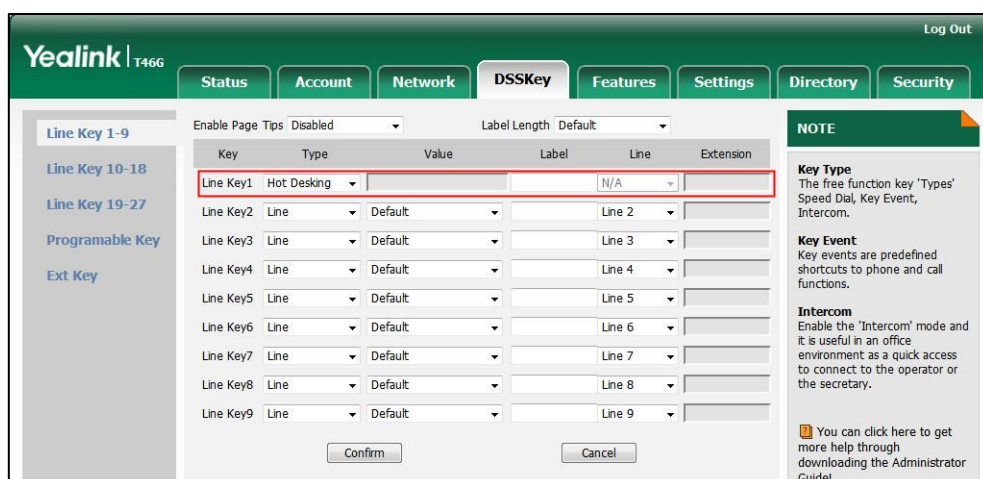
Parameters	Permitted Values	Default
linekey.X.type/programablekey.X.type	34	Refer to the following content
<p>Description: Configures a DSS key as a hot desking key on the IP phone. The digit 34 stands for the key type Hot Desking. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>Example: linekey.2.type = 34</p> <p>Default: For line keys:</p>		

Parameters	Permitted Values	Default
<p>For SIP-T48G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-29 is 0.</p> <p>For SIP-T46G IP phones:</p> <p>The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones:</p> <p>The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones:</p> <p>The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIP-T48G/T46G IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p>		

Parameters	Permitted Values	Default
When X=8, the default value is 52 (Switch Account Down).		
When X=9, the default value is 33 (Status).		
When X=10, the default value is 0 (NA).		
When X=13, the default value is 0 (NA).		
Web User Interface:		
DSSKey->Line Key X/Programable Key->Type		
Phone User Interface:		
None		

To configure a hot desking key via web user interface:

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



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

To configure a hot desking key via phone user interface:

1. Press **Menu->Features->DSS Keys**.
2. Select the desired DSS key.
3. Press **◀** or **▶** , or the **Switch** soft key to select **Key Event** from the **Type** field.
4. Press **◀** or **▶** , or the **Switch** soft key to select **Hot Desking** from the **Key Type** field.
5. (Optional.) Enter the string that will appear on the LCD screen in the **Label** field.
6. 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.
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 phone changes.

Event	Description
Forward Incoming Call	When the IP phone forwards an incoming call.
Reject Incoming Call	When the IP phone rejects an incoming call.
Answer New-In Call	When the IP phone answers a new call.
Transfer Finished	When the IP phone completes to transfer a call.
Transfer Failed	When the IP phone fails to transfer a call.
Idle to Busy	When the state of the IP phone changes from idle to busy.
Busy to Idle	When the state of phone changes from busy to idle.
Autop Finish	When the IP phone completes auto provisioning via power on.

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=$variable value`. Variable name can be customized by users, while the variable value is pre-defined. For example, a URL “`http://192.168.1.10/help.xml?mac=$mac`” is specified for the event Mute, \$mac will be dynamically replaced with the MAC address of the phone when the IP phone mutes a call.

The following table lists the pre-defined variable values.

Variable Value	Description
\$mac	The MAC address of the phone
\$ip	The IP address of the 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

Variable Value	Description
	an incoming call.
\$remote	The SIP URI of the callee when the IP phone places a call. The SIP URI of the caller when the IP phone receives an incoming call.
\$display_local	The display name of the caller when the IP phone places a call. The display name of the callee when the IP phone receives an incoming call.
\$display_remote	The display name of the callee when the IP phone places a call. The display name of the caller when the IP phone receives an incoming call.
\$call_id	The call-id of the active call.

Procedure

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

Configuration File	<y0000000000xx>.cf g	Configure the action URL. Parameters: 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
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		<p>action_url.transfer_call</p> <p>action_url.blind_transfer_call</p> <p>action_url.attended_transfer_call</p> <p>action_url.hold</p> <p>action_url.unhold</p> <p>action_url.mute</p> <p>action_url.unmute</p> <p>action_url.missed_call</p> <p>action_url.call_terminated</p> <p>action_url.busy_to_idle</p> <p>action_url.idle_to_busy</p> <p>action_url.ip_change</p> <p>action_url.forward_incoming_call</p> <p>action_url.reject_incoming_call</p> <p>action_url.answer_new_incoming_call</p> <p>action_url.transfer_finished</p> <p>action_url.transfer_failed</p> <p>action_url.setup_autop_finish</p>
Local	Web User Interface	<p>Configure the action URL.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=features-actionurl&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
action_url.setup_completed	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the action URL the IP phone sends after startup.</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 		

Parameters	Permitted Values	Default
<ul style="list-style-type: none"> • \$firmware • \$active_url • \$active_user • \$active_host • \$local • \$remote • \$display_local • \$display_remote • \$call_id <p>Example: action_url.setup_completed = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Setup Completed</p>		
action_url.registered	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends after an account is registered.</p> <p>Example: action_url.registered = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Note: The old parameter "action_url.log_on" is also applicable to IP phones.</p> <p>Web User Interface: Features->Action URL->Registered</p> <p>Phone User Interface: None</p>		
action_url.unregistered	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends after an account is unregistered.</p> <p>Example: action_url.unregistered = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Note: The old parameter "action_url.log_off" is also applicable to IP phones.</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
<p>Features->Action URL->Unregistered</p> <p>Phone User Interface:</p> <p>None</p>		
action_url.register_failed	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the action URL the IP phone sends after a register failed.</p> <p>Example:</p> <p>action_url.register_failed = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface:</p> <p>Features->Action URL->Register Failed</p> <p>Phone User Interface:</p> <p>None</p>		
action_url.off_hook	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the action URL the IP phone sends when off hook.</p> <p>Example:</p> <p>action_url.off_hook = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface:</p> <p>Features->Action URL->Off Hook</p> <p>Phone User Interface:</p> <p>None</p>		
action_url.on_hook	URL within 511 characters	Blank
<p>Description:</p> <p>Configures the action URL the IP phone sends when on hook.</p> <p>Example:</p> <p>action_url.on_hook = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface:</p> <p>Features->Action URL->On Hook</p>		

Parameters	Permitted Values	Default
<p>Phone User Interface: None</p>		
<p>action_url.incoming_call</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when receiving an incoming call.</p> <p>Example: action_url.incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Incoming Call</p> <p>Phone User Interface: None</p>		
<p>action_url.outgoing_call</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when placing a call.</p> <p>Example: action_url.outgoing_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Outgoing Call</p> <p>Phone User Interface: None</p>		
<p>action_url.call_established</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when establishing a call.</p> <p>Example: action_url.call_established = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Established</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
action_url.dnd_on	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when DND feature is enabled.</p> <p>Example: action_url.dnd_on = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Open DND</p> <p>Phone User Interface: None</p>		
action_url.dnd_off	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when DND feature is disabled.</p> <p>Example: action_url.dnd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Close DND</p> <p>Phone User Interface: None</p>		
action_url.always_fwd_on	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when always forward feature is enabled.</p> <p>Example: action_url.always_fwd_on = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Open Always Forward</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
action_url.always_fwd_off	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when always forward feature is disabled.</p> <p>Example: action_url.always_fwd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Close Always Forward</p> <p>Phone User Interface: None</p>		
action_url.busy_fwd_on	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when busy forward feature is enabled.</p> <p>Example: action_url.busy_fwd_on = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Open Busy Forward</p> <p>Phone User Interface: None</p>		
action_url.busy_fwd_off	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when busy forward feature is disabled.</p> <p>Example: action_url.busy_fwd_off = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Close Busy Forward</p>		

Parameters	Permitted Values	Default
<p>Phone User Interface: None</p>		
action_url.no_answer_fwd_on	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when no answer forward feature is enabled.</p> <p>Example: action_url.no_answer_fwd_on = http://192.168.0.20/help.xml?!IP=\$ip</p> <p>Web User Interface: Features->Action URL->Open No Answer Forward</p> <p>Phone User Interface: None</p>		
action_url.no_answer_fwd_off	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when no answer forward feature is disabled.</p> <p>Example: action_url.no_answer_fwd_off = http://192.168.0.20/help.xml?!IP=\$ip</p> <p>Web User Interface: Features->Action URL->Close No Answer Forward</p> <p>Phone User Interface: None</p>		
action_url.transfer_call	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when performing a transfer.</p> <p>Example: action_url.transfer_call = http://192.168.0.20/help.xml?!IP=\$ip</p> <p>Web User Interface: Features->Action URL->Transfer Call</p>		

Parameters	Permitted Values	Default
<p>Phone User Interface: None</p>		
<p>action_url.blind_transfer_call</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when performing a blind transfer.</p> <p>Example: action_url.blind_transfer_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Blind Transfer</p> <p>Phone User Interface: None</p>		
<p>action_url.attended_transfer_call</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when performing an attended/semi-attended transfer.</p> <p>Example: action_url.attended_transfer_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Attended Transfer</p> <p>Phone User Interface: None</p>		
<p>action_url.hold</p>	<p>URL within 511 characters</p>	<p>Blank</p>
<p>Description: Configures the action URL the IP phone sends when placing a call on hold.</p> <p>Example: action_url.hold = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Hold</p> <p>Phone User Interface:</p>		

Parameters	Permitted Values	Default
None		
action_url.unhold	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when resuming a held call.</p> <p>Example: action_url.unhold = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->UnHold</p> <p>Phone User Interface: None</p>		
action_url.mute	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when muting a call.</p> <p>Example: action_url.mute = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Mute</p> <p>Phone User Interface: None</p>		
action_url.unmute	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when un-muting a call.</p> <p>Example: action_url.unmute = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->UnMute</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
action_url.missed_call	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when missing a call.</p> <p>Example: action_url.missed_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Missed Call</p> <p>Phone User Interface: None</p>		
action_url.call_terminated	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when terminating a call.</p> <p>Example: action_url.call_terminated = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Terminated</p> <p>Phone User Interface: None</p>		
action_url.busy_to_idle	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when changing the state of the IP phone from busy to idle.</p> <p>Example: action_url.busy_to_idle = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Busy To Idle</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
action_url.idle_to_busy	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when changing the state of the IP phone from idle to busy.</p> <p>Example: action_url.idle_to_busy = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Idle To Busy</p> <p>Phone User Interface: None</p>		
action_url.ip_change	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when changing the IP address of the IP phone.</p> <p>Example: action_url.ip_change = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->IP Changed</p> <p>Phone User Interface: None</p>		
action_url.forward_incoming_call	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when forwarding an incoming call.</p> <p>Example: action_url.forward_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Forward Incoming Call</p> <p>Phone User Interface: None</p>		

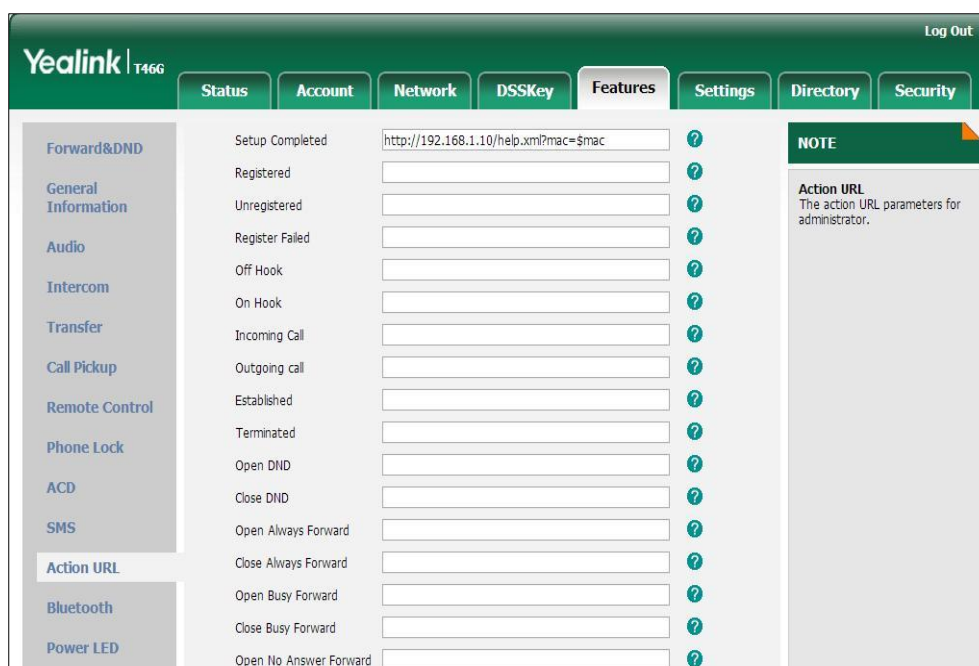
Parameters	Permitted Values	Default
action_url.reject_incoming_call	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when rejecting an incoming call.</p> <p>Example: action_url.reject_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Reject Incoming Call</p> <p>Phone User Interface: None</p>		
action_url.answer_new_incoming_call	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when answering a new incoming call.</p> <p>Example: action_url.answer_new_incoming_call = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Answer New-In Call</p> <p>Phone User Interface: None</p>		
action_url.transfer_finished	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when completing a call transfer.</p> <p>Example: action_url.transfer_finished = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Transfer Finished</p> <p>Phone User Interface: None</p>		
action_url.transfer_failed	URL within 511 characters	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the action URL the IP phone sends when failing to transfer a call.</p> <p>Example: action_url.transfer_failed = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Transfer Failed</p> <p>Phone User Interface: None</p>		
action_url.setup_autop_finish	URL within 511 characters	Blank
<p>Description: Configures the action URL the IP phone sends when completing auto provisioning via power on.</p> <p>Example: action_url.setup_autop_finish = http://192.168.0.20/help.xml?IP=\$ip</p> <p>Web User Interface: Features->Action URL->Autop Finish</p> <p>Phone User Interface: None</p>		

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 the pre-defined variable values.

Variable Value	Phone Action
OK	Press the OK key.
ENTER	Press the Enter soft key
SPEAKER	Press the Speakerphone key.
F_TRANSFER	Transfers a call to another party.
VOLUME_UP	Increase the volume.
VOLUME_DOWN	Decrease the volume.
MUTE	Mute a call.
F_HOLD	Place an active call on hold.

Variable Value	Phone Action
X	Cancel actions or reject incoming calls.
CANCEL	Return to a previous screen or cancel a call.
0-9*/POUND	Press the keypad (0-9, * or #).
L1-LX	Press the line key (for SIP-T48G, X=29, SIP-T46G, X=27, for SIP-T42G/T41P, X=15).
F_CONFERENCE	Press the Conference soft key.
F1-F4	Press the soft keys.
MSG	Press the MESSAGE key.
HEADSET	Press the HEADSET key.
RD	Press the REDIAL key.
UP/DOWN/LEFT/RIGHT	Press the navigation keys.
Reboot	Reboot the phone.
AutoP	Perform auto provisioning.
DNDOn	Activate the DND mode.
DNDOff	Deactivate the DND mode.
number=xxx&outgoing_uri=y	Place a call to xxx from SIP URI y.
OFFHOOK	Pick up the handset.
ONHOOK	Hang up the handset.
ANSWER	Answer a call.
Reset	Reset a phone.
ATrans=xxx	Perform a semi-attended/attended transfer
BTrans=xxx	Perform a blind transfer to xxx.
CALLEND	End a call.

Variable Value	Phone Action
<p>phonecfg=get[&accounts=x][&dnd=x][&fw=x]</p>	<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> <p>Example:</p> <p>http://10.3.20.10/servlet?phonecfg=get[&accounts=1][&dnd=0][&fw=1]</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 specified 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. You can use action URI feature to capture the phone's current screen.

Procedure

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

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Specify the trusted IP address(es) for sending the Action URI to the IP phone.</p> <p>Parameter:</p> <p>features.action_uri_limit_ip</p>
<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>/servl</p>

		et?p=features-remotecontrl&q =load
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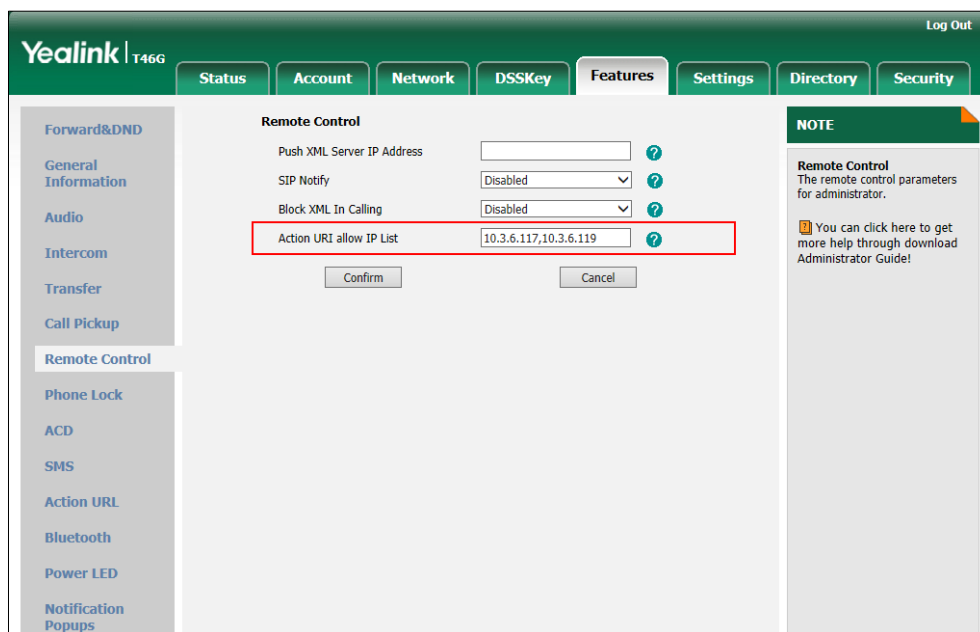
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.action_uri_limit_ip	IP address or any	Blank
<p>Description: Configures the address(es) from which Action URI will be accepted. For discontinuous IP addresses, multiple IP addresses are separated by commas. For continuous IP addresses, the format likes *.*.* and the "*" stands for the values 0~255. For example: 10.10.*.* stands for the IP addresses that range from 10.10.0.0 to 10.10.255.255. If left blank, the IP phone will reject any HTTP GET request. If it is set to "any", the IP phone will accept and handle HTTP GET requests from any IP address.</p> <p>Example: features.action_uri_limit_ip = 10.3.6.117,10.3.6.119</p> <p>Web User Interface: Features->Remote Control->Action URI allow IP List</p> <p>Phone User Interface: None</p>		

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.

Capturing the Current Screen of the Phone

You can capture the screen display of the IP phone using the action URI. IP phones support handling an HTTP or HTTPS GET request. The URI format is `http(s)://<phoneIPAddress>/screencapture`. The captured picture can be saved as a BMP or JPEG file.

You can also use the URI "`http(s)://<phoneIPAddress>/screencapture/download`" to capture the screen display first, and then download the image (which is saved as a JPG file and named with the phone model and the capture time) to the local system. Before capturing the phone's current screen, ensure that the IP address of the computer is included in the trusted IP address for Action URI on the phone.

When you capture the screen display, the IP phone may prompt you to enter the user name and password of the administrator if web browser does not remember the user name and password for web user interface login.

Note

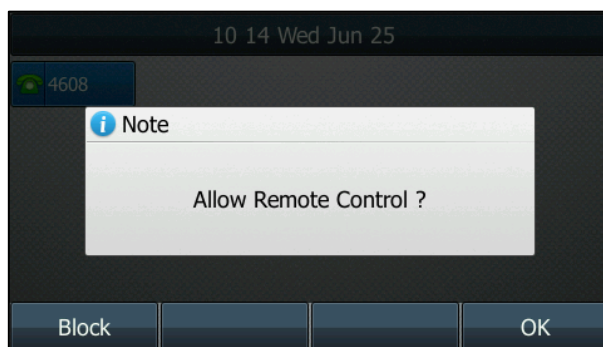
IP phones also support capturing the screen display using the old URI "`http://<phoneIPAddress>/servlet?command=screenshot`".

To capture the current screen of the phone:

1. Enter request URI (e.g., `http://10.3.6.104/screencapture`) in the browser's address

bar and press the Enter key on the keyboard.

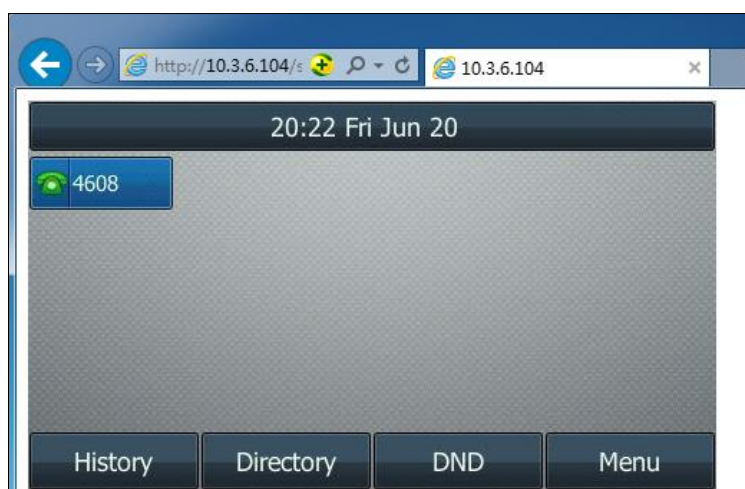
2. Do one of the following:
 - If it is the first time you capture the phone's current screen using the computer, the browser will display "remote control forbidden", and the LCD screen will prompt the message "Allow Remote Control?".



Press the **OK** soft key on the phone to allow remote control. The phone will return to the previous screen.

Refresh the web page.

The browser will display an image showing the phone's current screen. You can then save the image to your local system.



- Else, the browser will display an image showing the phone's current screen display directly. You can save the image to your local system.

Note

Frequent capture may affect the phone performance. Yealink recommend you to capture the phone screen display within a minimum interval of 4 seconds.

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

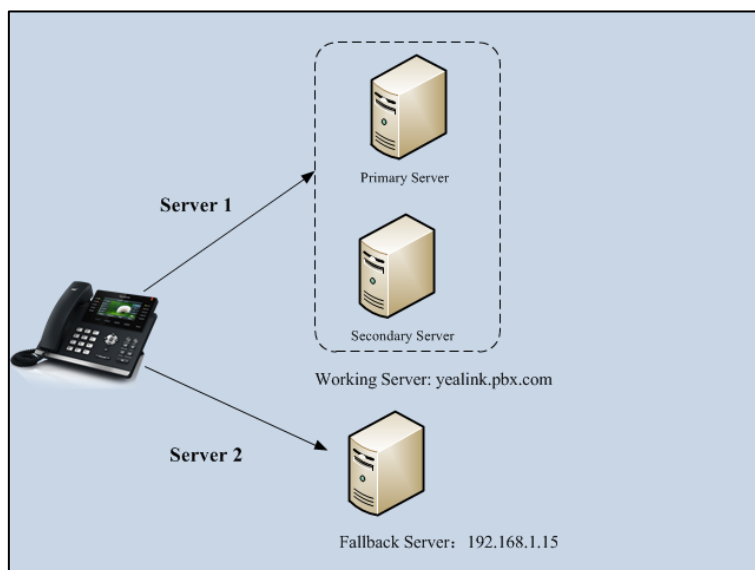
server fails, or the connection between the IP phone and the server fails.

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 mechanisms from the primary to the secondary server.
- **Fallback:** In this mode, a second less featured call server (fallback server) with SIP capability takes over call control to provide basic calling capability, but without some advanced features (for example, shared lines, call recording and MWI) offered by the working server. 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 physical SIP servers for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server is 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 address of the fallback server. For example, 192.168.1.15. A fallback server offers less functionality than the working server.

Phone Registration

Two registration methods for fallback mode:

- **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.

For more information on server redundancy, refer to *Server Redundancy on Yealink IP Phones*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

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

Configuration File	<MAC>.cfg	Configure the server redundancy on the IP phone. Parameters: account.X.sip_server.Y.address account.X.sip_server.Y.port account.X.sip_server.Y.expires account.X.sip_server.Y.retry_counts Fallback Mode: account.X.fallback.redundancy_type account.X.fallback.timeout Failover Mode: account.X.sip_server.Y.failback_mode account.X.sip_server.Y.failback_timeout account.X.sip_server.Y.register_on_enable
Local	Web User Interface	Configure the server redundancy on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.sip_server.Y.address (X ranges from 1 to 16. Y ranges from 1 to 2)	String within 256 characters	Blank
<p>Description: Configures the IP address or domain name of the SIP server Y for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.sip_server.1.address = yealink.pbx.com</p> <p>Web User Interface: Account->Register ->SIP Server Y->Server Host</p> <p>Phone User Interface: None</p>		
account.X.sip_server.Y.port (X ranges from 1 to 16. Y ranges from 1 to 2)	Integer from 0 to 65535	5060
<p>Description: Configures the port of the SIP server Y for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.sip_server.1.port = 5060</p> <p>Web User Interface: Account->Register ->SIP Server Y->Port</p> <p>Phone User Interface: None</p>		
account.X.sip_server.Y.expires (X ranges from 1 to 16. Y ranges from 1 to 2)	Integer from 30 to 2147483647	3600

Parameters	Permitted Values	Default
<p>Description: Configures the registration expiration time (in seconds) of the SIP server Y for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.sip_server.1.expires = 3600</p> <p>Web User Interface: Account->Register ->SIP Server Y->Server Expires</p> <p>Phone User Interface: None</p>		
<p>account.X.sip_server.Y.retry_counts (X ranges from 1 to 16. Y ranges from 1 to 2)</p>	<p>Integer from 0 to 20</p>	<p>3</p>
<p>Description: Configures the retry times for the IP phone to resend requests when the SIP server Y is unavailable or there is no response from the SIP server Y for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Register->SIP Server Y ->Server Retry Counts</p> <p>Phone User Interface: None</p>		
<p>account.X.fallback.redundancy_type (X ranges from 1 to 16)</p>	<p>0 or 1</p>	<p>0</p>
<p>Description: Configures the registration mode for the IP phone in fallback mode. 0-Concurrent Registration 1-Successive Registration X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p>		

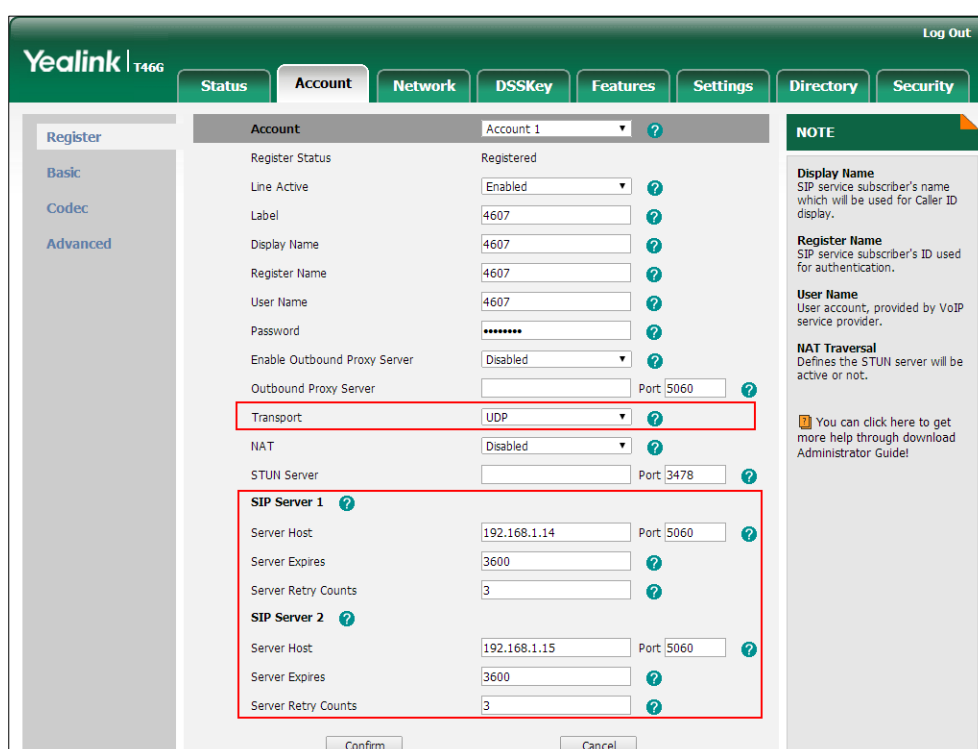
Parameters	Permitted Values	Default
<p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
account.X.fallback.timeout	Integer from 10 to 2147483647	120
<p>Description: Configures the time interval (in seconds) for the IP phone to detect whether the working server is available by sending the registration request after the fallback server takes over call control. It is only applicable to the Successive Registration mode. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
account.X.sip_server.Y.fallback_mode (X ranges from 1 to 16, Y ranges from 1 to 2)	0, 1, 2 or 3	0
<p>Description: Configures the way in which the phone fails back to the primary server for call control in the failover mode.</p> <p>0-newRequests: all requests are sent to the primary server first, regardless of the last server that was used.</p> <p>1-DNSTTL: the IP phone will send requests to the last registered server first. If the time defined by DNSTTL on the registered server expires, the phone will retry to send requests to the primary server.</p> <p>2-registration: the IP phone will send requests to the last registered server first. If the registration expires, the phone will retry to send requests to the primary server.</p> <p>3-duration: the IP phone will send requests to the last registered server first. If the time defined by the account.X.sip_server.Y.fallback_timeout parameter expires, the phone will retry to send requests to the primary server.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G).</p>		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>account.X.sip_server.Y.failback_timeout (X ranges from 1 to 16, Y ranges from 1 to 2)</p>	<p>0, 60 to 65535</p>	<p>3600</p>
<p>Description:</p> <p>Configures the time (in seconds) for the phone to retry to send requests to the primary server after failing over to the current working server when the parameter account.X.sip_server.Y.failback_mode is set to 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 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>account.X.sip_server.Y.register_on_enable (X ranges from 1 to 16, Y ranges from 1 to 2)</p>	<p>0 or 1</p>	<p>0</p>
<p>Description:</p> <p>Enables or disables the IP phone to register to the secondary server when sending requests to the secondary server in the failover mode.</p> <p>0-Disabled 1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: None</p>		

Parameters	Permitted Values	Default
Phone User Interface:		
None		

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. Select the desired value from the pull-down list of **Transport**.
4. Configure parameters of the SIP server 1 in the corresponding fields.
5. Configure parameters of the SIP server 2 in the corresponding fields.

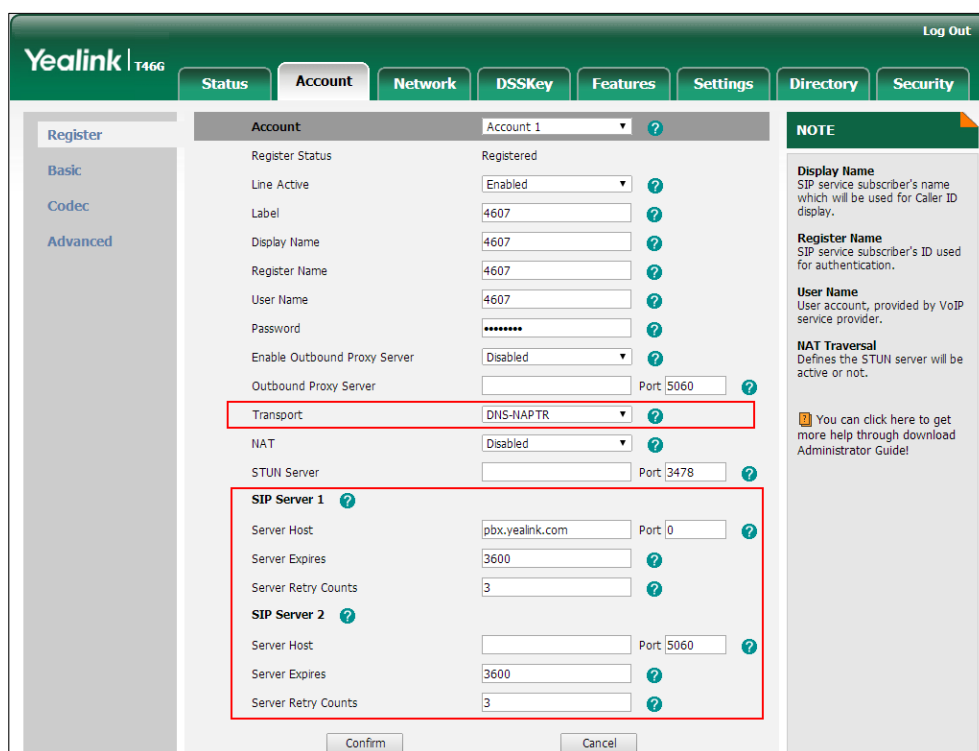


6. Click **Confirm** to accept the change.

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**.
5. 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.



6. 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 discovered 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 the NAPTR query for the SRV 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, 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 the NAPTR query to get the SRV 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 supported by the domain server: 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 the NAPTR query again according to the previous NAPTR query result.

SRV (Service Location Record)

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

SIP Server Domain Name Resolution can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure the transport type on the IP phone. Parameters: account.X.transport account.X.naptr_build
Local	Web User Interface	Configure the transport type on the IP phone. Navigate to: http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.transport	0, 1, 2 or 3	0
<p>Description: Configures the type of transport protocol for account X.</p> <p>0-UDP 1-TCP 2-TLS 3-DNS-NAPTR</p> <p>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.</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface: Account->Register ->Transport</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
account.X.naptr_build	0 or 1	0
<p>Description:</p> <p>Configures the way of SRV query for the IP phone to be performed when no result is returned from NAPTR query for account X.</p> <p>0-SRV query using UDP only</p> <p>1-SRV query using UDP, TCP and TLS</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G).</p> <p>X ranges from 1 to 12 (for SIPT42G).</p> <p>X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		

Static DNS Cache

Failover redundancy can only be utilized when the configured domain name of the SIP server is resolved to multiple IP addresses. If the IP phone is not configured with a DNS server, or the DNS query returns no result from a DNS server, you can configure a set of DNS NAPTR/SRV/A records into the IP phone. The IP phone will attempt to resolve the domain name of the SIP server with static DNS cache.

When the IP phone is configured with a DNS server, the IP phone will behave as follows to resolve domain name of the SIP server:

- The IP phone performs a DNS query to resolve the domain name from the DNS server.
- If the DNS query returns no results for the domain name, or the returned record cannot be contacted, the values in the static DNS cache (if configured) are used when their configured time intervals are not elapsed.
- If the configured time interval is elapsed, the IP phone will attempt to perform a DNS query again.
- If the DNS query returns a result, the IP phone will use the returned record and ignore the statically configured cache values.

When the IP phone is not configured with a DNS server, it will behave as follow:

- The IP phone attempts to resolve the domain name within the static DNS cache.
- The IP phone will always use the results returned from the static DNS cache.

IP phones can be configured to use static DNS cache preferentially. Static DNS cache is configurable on a per-line basis.

Procedure

Static DNS cache can be configured only using the configuration files.

Configuration File	<y0000000000 xx>.cfg	Configure NAPTR/SRV/A records. Parameters: dns_cache_naptr.X.name dns_cache_naptr.X.flags dns_cache_naptr.X.order dns_cache_naptr.X.preference dns_cache_naptr.X.replace dns_cache_naptr.X.service dns_cache_naptr.X.ttl dns_cache_srv.X.name dns_cache_srv.X.port dns_cache_srv.X.priority dns_cache_srv.X.target dns_cache_srv.X.weight dns_cache_srv.X.ttl dns_cache_a.X.name dns_cache_a.X.ip dns_cache_a.X.ttl
	<MAC>.cfg	Configure the IP phone whether to cache the additional DNS records. Parameter: account.X.dns_cache_type Configure the IP phone whether to use static DNS cache preferentially. Parameter: account.X.static_cache_pri

Details of Configuration Parameters:

Parameters	Permitted Values	Default
dns_cache_naptr.X.name (X ranges from 1 to 12)	String within 256 characters	Blank
<p>Description: Configures the domain name to which NAPTR record X refers.</p> <p>Example: dns_cache_naptr.1.name = yealink.pbx.com</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
dns_cache_naptr.X.flags (X ranges from 1 to 12)	S, A, U or P	Blank
<p>Description: Configures the flag of NAPTR record X. (Always "s" for SIP, which means to do an SRV lookup on whatever is in the replacement field).</p> <p>S-Do an SRV lookup next. A-Do an A lookup next. U-No need to do a DNS query next. P-Service customized by the user</p> <p>Example: dns_cache_naptr.1.flags = S</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
dns_cache_naptr.X.order (X ranges from 1 to 12)	Integer from 0 to 65535	0
<p>Description: Configures the order of NAPTR record X. NAPTR record with lower order is more preferred.</p> <p>Example:</p>		

Parameters	Permitted Values	Default
<p>dns_cache_naptr.1.order = 90</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_naptr.X.preference (X ranges from 1 to 12)</p>	<p>Integer from 0 to 65535</p>	<p>0</p>
<p>Description: Configures the preference of NAPTR record X. NAPTR record with lower preference is more preferred.</p> <p>Example: dns_cache_naptr.1.preference = 50</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_naptr.X.replace (X ranges from 1 to 12)</p>	<p>Domain name</p>	<p>Blank</p>
<p>Description: Configures a domain name to be used for the next SRV query in NAPTR record X.</p> <p>Example: dns_cache_naptr.1.replace = _sip_tcp.yealink.pbx.com</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_naptr.X.service (X ranges from 1 to 12)</p>	<p>String within 32 characters</p>	<p>Blank</p>
<p>Description: Configures the transport protocol available for the SIP server in NAPTR record X.</p> <p>SIP+D2U: SIP over UDP</p> <p>SIP+D2T: SIP over TCP</p>		

Parameters	Permitted Values	Default
<p>SIP+D2S: SIP over SCTP</p> <p>SIPS+D2T: SIPS over TCP</p> <p>Example:</p> <p>dns_cache_naptr.1.service = SIP+D2T</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
<p>dns_cache_naptr.X.ttl (X ranges from 1 to 12)</p>	<p>Integer from 30 to 2147483647</p>	<p>300</p>
<p>Description:</p> <p>Configures the time interval (in seconds) that NAPTR record X may be cached before the record should be consulted again.</p> <p>Example:</p> <p>dns_cache_naptr.1.ttl = 3600</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
<p>dns_cache_srv.X.name (X ranges from 1 to 12)</p>	<p>Domain name</p>	<p>Blank</p>
<p>Description:</p> <p>Configures the domain name in SRV record X.</p> <p>Example:</p> <p>dns_cache_srv.1.name = _sip_tcp.yealink.pbx.com</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		
<p>dns_cache_srv.X.port (X ranges from 1 to 12)</p>	<p>Integer from 0 to 65535</p>	<p>0</p>
<p>Description:</p>		

Parameters	Permitted Values	Default
<p>Configures the port to be used in SRV record X.</p> <p>Example; dns_cache_srv.1.port = 5060</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_srv.X.priority (X ranges from 1 to 12)</p>	<p>Integer from 0 to 65535</p>	<p>0</p>
<p>Description: Configures the priority for the target host in SRV record X. Lower priority is more preferred.</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_srv.X.target (X ranges from 1 to 12)</p>	<p>Domain name</p>	<p>Blank</p>
<p>Description: Configures the domain name of the target host for an A query in SRV record X.</p> <p>Example: dns_cache_srv.1.target = server1.yealink.pbx.com</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_srv.X.weight (X ranges from 1 to 12)</p>	<p>Domain name</p>	<p>0</p>
<p>Description: Configures the weight of the target host in SRV record X. When priorities are equal, weight is used to differentiate the preference. Higher weight is more preferred.</p>		

Parameters	Permitted Values	Default
<p>Example: dns_cache_srv.1.weight = 1</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_srv.X.ttl (X ranges from 1 to 12)</p>	<p>Integer from 30 to 2147483647</p>	<p>300</p>
<p>Description: Configures the time interval (in seconds) that SRV record X may be cached before the record should be consulted again.</p> <p>Example: dns_cache_srv.1.ttl = 3600</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_a.X.name (X ranges from 1 to 12)</p>	<p>Domain name</p>	<p>Blank</p>
<p>Description: Configures the domain name in A record X.</p> <p>Example: dns_cache_a.1.name = yealink.pbx.com</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_a.X.ip (X ranges from 1 to 12)</p>	<p>IP address</p>	<p>Blank</p>
<p>Description: Configures the IP address that the domain name in A record X maps to.</p> <p>Example: dns_cache_a.1.ip = 192.168.1.13</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>dns_cache_a.X.ttl (X ranges from 1 to 12)</p>	<p>Integer from 30 to 2147483647</p>	<p>300</p>
<p>Description: Configures the time interval (in seconds) that A record X may be cached before the record should be consulted again.</p> <p>Example: dns_cache_a.1.ttl = 3600</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>account.X.dns_cache_type</p>	<p>0, 1 or 2</p>	<p>1</p>
<p>Description: Configures whether the IP phone uses the DNS cache for domain name resolution of the SIP server and caches the additional DNS records for account X.</p> <p>0-Perform real-time DNS query rather than using DNS cache. 1-Use DNS cache, but do not cache the additional DNS records. 2-Use DNS cache and cache the additional DNS records.</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIPT41P).</p> <p>Example: account.1.dns_cache_type = 1</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		
<p>account.X.static_cache_pri</p>	<p>0 or 1</p>	<p>0</p>

Parameters	Permitted Values	Default
<p>Description: Configures whether preferentially to use the static DNS cache for domain name resolution of the SIP server for account X.</p> <p>0-Use domain name resolution from the DNS server preferentially 1-Use static DNS cache preferentially</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.static_cache_pri = 1</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

LLDP

LLDP (Linker Layer Discovery Protocol) is a vendor-neutral Link Layer protocol, which allows IP phones to receive and/or transmit device-related information from/to directly connected devices on the network that are also using the protocol, and store the information about other devices. LLDP transmits information as packets called LLDP Data Units (LLDPDUs). An LLDPDU consists of a set of Type-Length-Value (TLV) elements, each of which contains a particular type of information about the device or port transmitting it.

LLDP-MED (Media Endpoint Discovery)

LLDP-MED is published by the Telecommunications Industry Association (TIA). It is an extension to LLDP that operates between endpoint devices and network connectivity devices. LLDP-MED provides the following capabilities for IP phones:

- Capabilities Discovery -- allows LLDP-MED 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 phone.
	Port ID	The MAC address of the phone.
	Time To Live	Seconds until data unit expires. The default value is 60s.
	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 phone capabilities. 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) 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,

TLV Type	TLV Name	Description
		Extended Power via MDI-PD, Inventory.
	Network Policy	Port VLAN ID, application type, L2 priority and DSCP value.
	Extended Power-via-MDI	Power type, source, priority and value.
	Inventory – Hardware Revision	Hardware revision of phone.
	Inventory – Firmware Revision	Firmware revision of phone.
	Inventory – Software Revision	Software revision of phone.
	Inventory – Serial Number	Serial number of phone.
	Inventory – Manufacturer Name	Manufacturer name of phone. The default value is “yealink”.
	Inventory – Model Name	Model name of phone.
	Asset ID	Assertion identifier of phone. The default value is “asset”.

Procedure

LLDP can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure LLDP feature. Parameters: network.lldp.enable network.lldp.packet_interval
Local	Web User Interface	Configure LLDP feature. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load
	Phone User Interface	Configure LLDP feature.

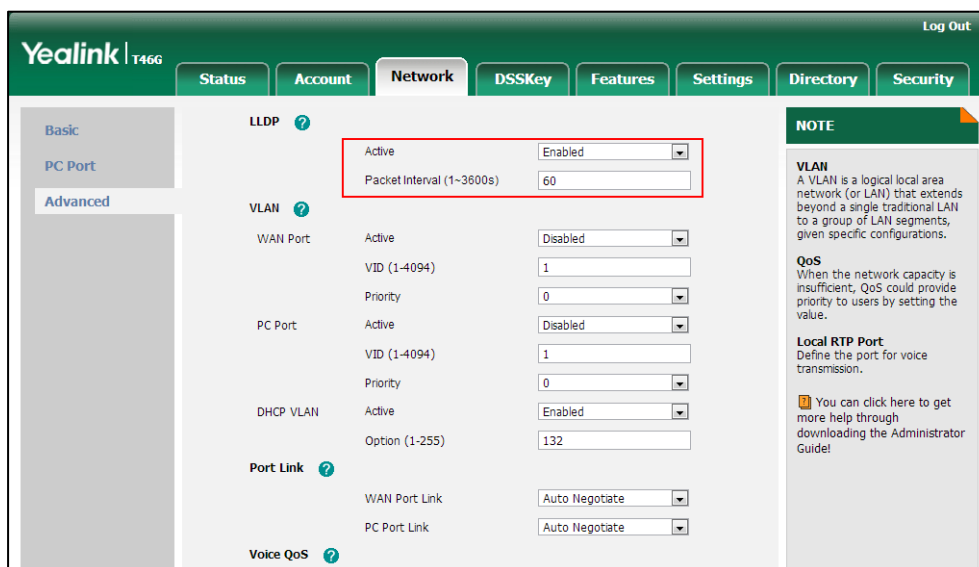
Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.lldp.enable	0 or 1	1
<p>Description: Enables or disables LLDP feature on the IP phone.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->LLDP->Active</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->LLDP->LLDP Status</p>		
network.lldp.packet_interval	Integer from 1 to 3600	60
<p>Description: Configures the interval (in seconds) for the IP phone to send the LLDP request.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect. It works only if the value of the parameter "network.lldp.enable" is set to 1 (Enabled).</p> <p>Web User Interface: Network->Advanced->LLDP->Packet Interval (1~3600s)</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->LLDP->Packet Interval</p>		

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**.

- Enter the desired time interval in the **Packet Interval (1~3600s)** field.



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

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

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

To configure LLDP feature via phone user interface:

- Press **Menu->Advanced** (default password: admin) ->**Network->LLDP->LLDP Status**.
- Press **◀** or **▶** , or the **Switch** soft key to select the desired value from the **LLDP Status** field.
- Enter the desired time interval in the **Packet Interval** field.
- Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

VLAN

VLAN (Virtual Local Area Network) is used to logically divide a physical network into several broadcast domains. VLAN membership can be configured through software instead of physically relocating devices or connections. Grouping devices with a common set of requirements regardless of their physical location can greatly simplify network design. VLANs can address issues such as scalability, security, and network management.

The purpose of VLAN configurations on the IP phone is to insert tag with VLAN information to the packets generated by the IP phone. When VLAN is properly configured for the ports (internet port and PC port) on the IP phone, the IP phone will tag all packets from these ports with the VLAN ID. The switch receives and forwards the

tagged packets to the corresponding VLAN according to the VLAN ID in the tag as described in IEEE Std 802.3.

VLAN on IP phones allows simultaneous access for a regular PC. This feature allows a PC to be daisy chained to an IP phone and the connection for both PC and IP phone to be trunked through the same physical Ethernet cable.

In addition to manual configuration, the IP phone also supports automatic discovery of VLAN via LLDP or DHCP. The assignment takes effect in this order: assignment via LLDP, manual configuration, then assignment via DHCP.

VLAN Discovery via DHCP

IP phones support VLAN discovery via DHCP. When the VLAN Discovery method is set to DHCP, the IP phone will examine DHCP option for a valid VLAN ID. The predefined option 132 is used to supply the VLAN ID by default. You can customize the DHCP option used to request the VLAN ID.

For more information on VLAN, refer to *VLAN Feature on Yealink IP Phones*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

VLAN can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configure VLAN for the Internet port and PC port manually.</p> <p>Parameters:</p> <p>network.vlan.internet_port_enable network.vlan.internet_port_vid network.vlan.internet_port_priority network.vlan.pc_port_enable network.vlan.pc_port_vid network.vlan.pc_port_priority network.vlan.pc_port_mode</p> <p>Configure DHCP VLAN discovery feature.</p> <p>Parameters:</p> <p>network.vlan.dhcp_enable network.vlan.dhcp_option</p> <p>Configure the VLAN assignment method.</p> <p>Parameters:</p> <p>network.vlan.vlan_change.enable</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configure VLAN for the Internet</p>

		<p>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> <p>Configure DHCP VLAN discovery feature.</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.vlan.internet_port_enable	0 or 1	0
<p>Description: Enables or disables VLAN for the Internet (WAN) port.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN ->WAN Port->Active</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->WAN Port->VLAN Status</p>		
network.vlan.internet_port_vid	Integer from 1 to 4094	1
<p>Description: Configures VLAN ID for the Internet (WAN) port.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN ->WAN Port->VID (1-4094)</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->WAN</p>		

Parameters	Permitted Values	Default
Port->VID Number		
network.vlan.internet_port_priority	Integer from 0 to 7	0
<p>Description: Configures VLAN priority for the Internet (WAN) port. 7 is the highest priority, 0 is the lowest priority.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN ->WAN Port->Priority</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->WAN Port->Priority</p>		
network.vlan.pc_port_enable	0 or 1	0
<p>Description: Enables or disables VLAN for the PC (LAN) port.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN >PC Port->Active</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->PC Port->VLAN Status</p>		
network.vlan.pc_port_vid	Integer from 1 to 4094	1
<p>Description: Configures VLAN ID for the PC (LAN) port.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN >PC Port->VID (1-4094)</p>		

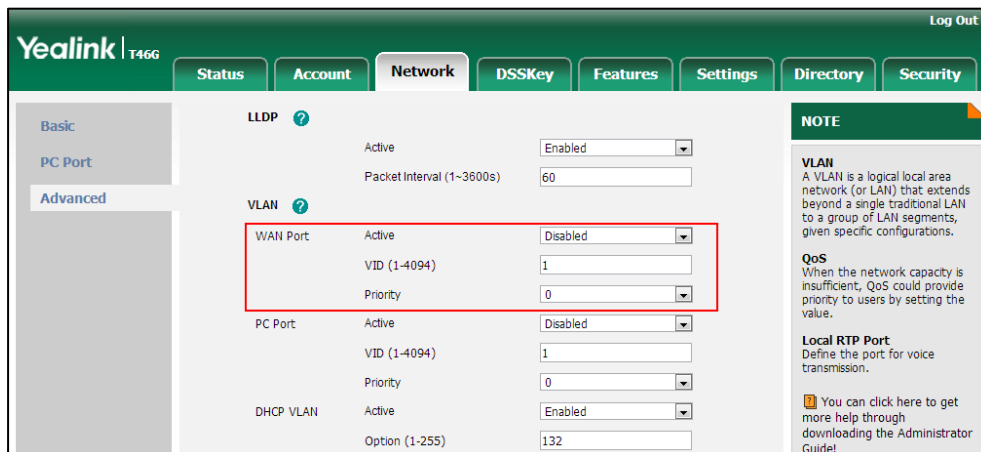
Parameters	Permitted Values	Default
Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->PC Port->VID		
network.vlan.pc_port_priority	Integer from 0 to 7	0
Description: Configures VLAN priority for the PC (LAN) port. Note: If you change this parameter, the IP phone will reboot to make the change take effect. Web User Interface: Network->Advanced->VLAN >PC Port->Priority Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->PC Port->Priority		
network.vlan.pc_port_mode	0 or 1	0
Description: Configures the way the IP phone processes packets for the PC (LAN) port when VLAN is enabled on the PC (LAN) port. 0 -when packets are sent from the PC port to the Internet port, the IP phone will forward the packets directly. 1 -when packets are sent from the PC port to the Internet port, and there is no VLAN tag in the packet, the IP phone will tag the packet with the tag configured for the PC port and then forward it. Note: When packets are sent from the Internet port to the PC port, remove the packet' tag if it is the same as the configured tag for the PC port, else forward the packets directly. If you change this parameter, the IP phone will reboot to make the change take effect. Web User Interface: None Phone User Interface: None		
network.vlan.dhcp_enable	0 or 1	1
Description: Enables or disables DHCP VLAN discovery feature on the IP phone.		

Parameters	Permitted Values	Default
<p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN >DHCP VLAN->Active</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->DHCP VLAN ->DHCP VLAN</p>		
network.vlan.dhcp_option	Integer from 128 to 254	132
<p>Description: Configures the DHCP option from which the IP phone will obtain the VLAN settings. You can configure at most five DHCP options and separate them by commas.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VLAN->DHCP VLAN->Option(1-255)</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VLAN ->DHCP VLAN ->Option</p>		
network.vlan.vlan_change.enable	0 or 1	0
<p>Description: Enables or disables the IP phone to obtain IP address with lower preference of VLAN assignment method or disable VLAN feature when the IP phone cannot obtain IP address with the current VLAN assignment method.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

To configure VLAN for Internet port via web user interface:

1. Click on **Network->Advanced**.

- In the **VLAN** block, select the desired value from the pull-down list of **WAN 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**.



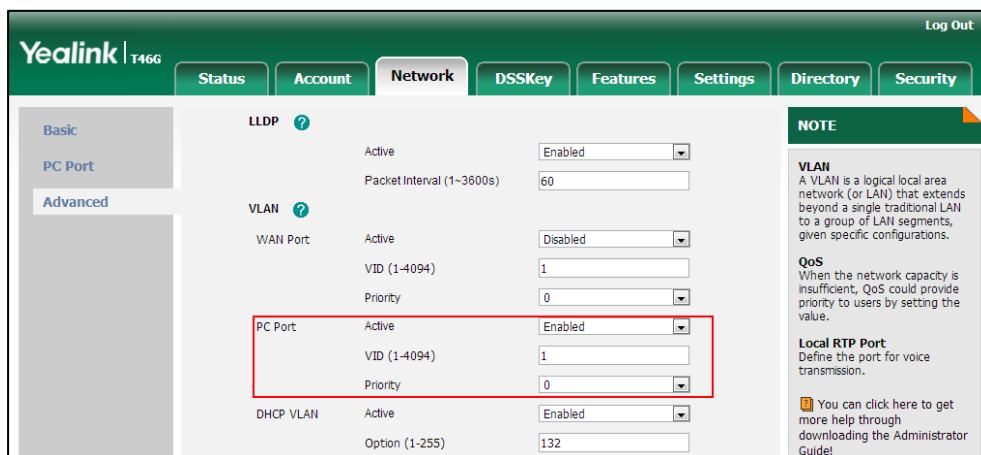
- Click **Confirm** to accept the change.

A dialog box pops up to prompt reboot to make the settings effective.

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



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

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

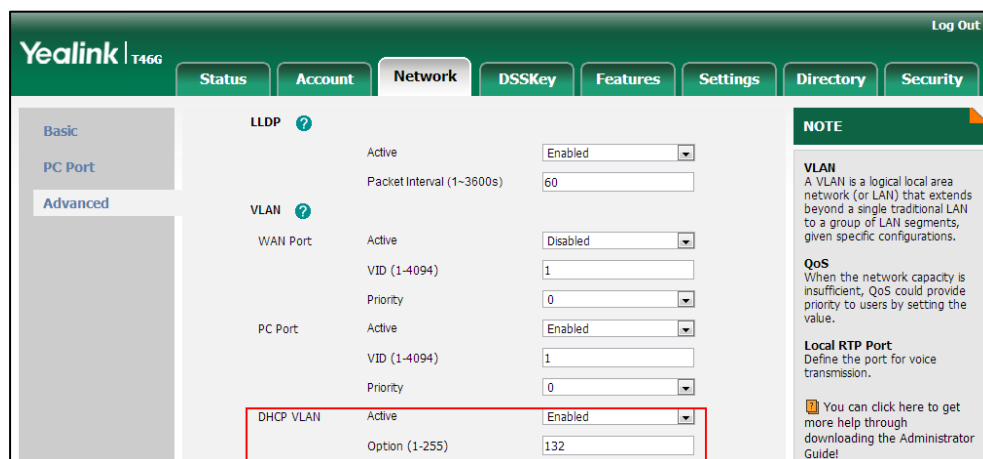
- Click **OK** to reboot the phone.

To configure the DHCP VLAN discovery via web user interface:

- 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 (1-255)** 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 reboot.
5. Click **OK** to reboot the phone.

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

1. Press **Menu->Advanced** (default 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** field.
4. Enter the priority value (0-7) in the **Priority** field.
5. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

To configure DHCP VLAN discovery via phone user interface:

1. Press **Menu->Advanced** (default password: admin) ->**Network->VLAN->DHCP VLAN**.
2. Press **◀** or **▶** , or the **Switch** soft key to select the desired value from the **DHCP VLAN** field.
3. Enter the desired option in the **Option** field.
4. Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

VPN

VPN (Virtual Private Network) is a secured private network connection built on top of public telecommunication infrastructure, such as the Internet. VPN has become more prevalent due to the benefits of scalability, reliability, convenience and security. VPN provides remote offices or individual users with secure access to their organization's network. 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 networking 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 the VPN feature. To prevent disclosure of private information, tunnel endpoints must authenticate each other before secure VPN tunnel is established. After the VPN feature is configured properly on the IP phone, the IP phone acts as a VPN client and uses the certificates to authenticate the VPN server.

To use 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 VPN-related 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*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

VPN can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the OpenVPN feature and upload a TAR file to the IP phone. Parameters: network.vpn_enable openvpn.url
Local	Web User Interface	Configure VPN feature and upload a TAR package to the IP phone.

		<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	Configure VPN feature.

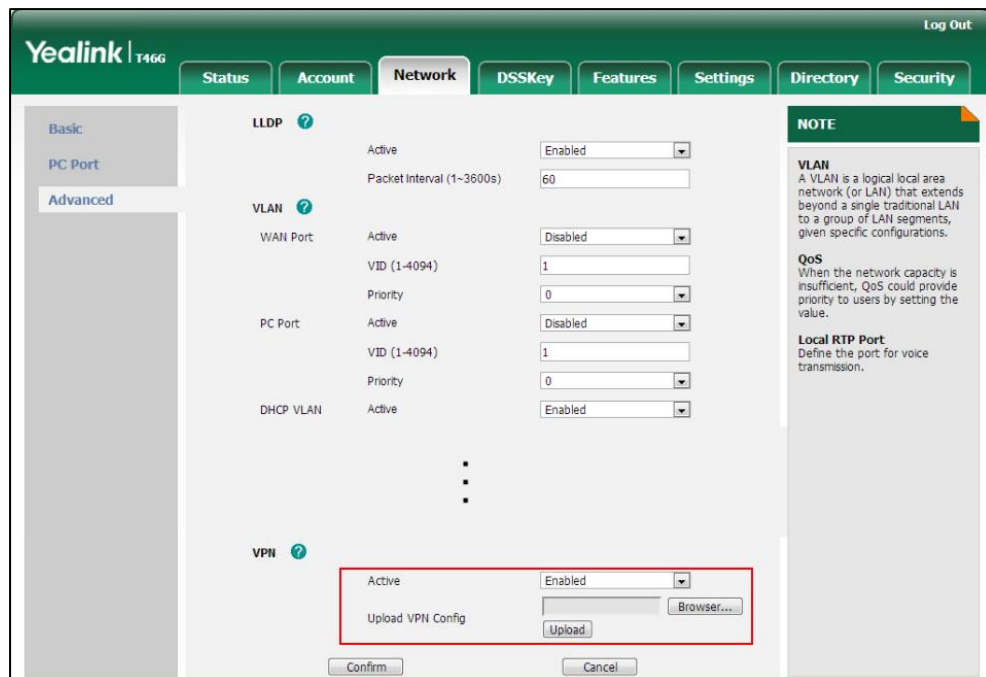
Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.vpn_enable	0 or 1	0
<p>Description: Enables or disables OpenVPN feature on the IP phone. 0-Disabled 1-Enabled Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->VPN ->Active</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->VPN->VPN Active</p>		
openvpn.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the *.tar file for OpenVPN.</p> <p>Example: openvpn.url = http://192.168.10.25/OpenVPN.tar</p> <p>Web User Interface: Network->Advanced->VPN->Upload VPN Config</p> <p>Phone User Interface: None</p>		

To upload the tar file to the phone and configure VPN via web user interface:

1. Click on **Network->Advanced**.
2. Click **Browse** to locate the TAR package from the local system.

3. Click **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 reboot.

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

To configure VPN via phone user interface after uploading the tar file:

1. Press **Menu->Advanced** (default 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 LCD screen will prompt “Reboot now?”. Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

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 the IP networks. It provides no guarantees for data delivering, which means delay, jitter, packet loss and bandwidth allocation are unpredictable. Differentiated Services (DiffServ or DS) is the most widely 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 to 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, 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

DSCPs for voice packets and SIP packets can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the DSCPs for voice packets and SIP packets. Parameters: network.qos.rtplos network.qos.signallos
Local	Web User Interface	Configure the DSCPs for voice packets and SIP packets. Navigate to: http://<phoneIPAddress>/servlet?p=network-adv&q=load

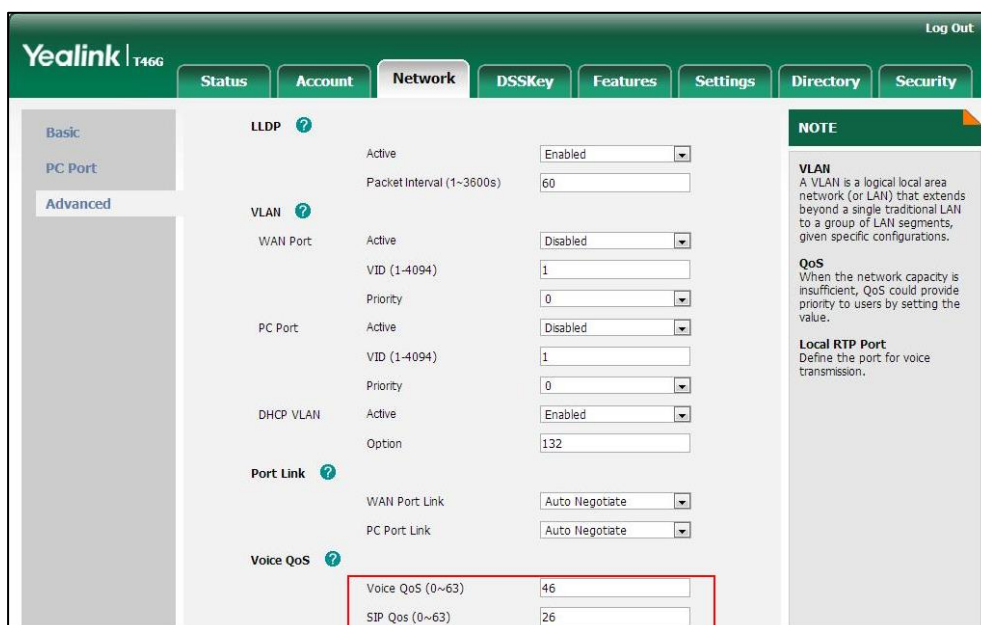
Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.qos.rtplos	Integer from 0 to 63	46
<p>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.</p> <p>Web User Interface: Network->Advanced->Voice QoS (0~63)</p>		

Parameters	Permitted Values	Default
Phone User Interface: None		
network.qos.signaltos	Integer from 0 to 63	26
<p>Description: Configures the DSCP for SIP packets. The default DSCP value for SIP packets is 26 (Assured Forwarding).</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Advanced->SIP QoS (0~63)</p> <p>Phone User Interface: None</p>		

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.
3. Enter the desired value in the **SIP Qos (0~63)** field.



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

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

5. Click **OK** to reboot the 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. The NAT feature 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, sending 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. Parameters: account.X.nat.nat_traversal account.X.nat.stun_server account.X.nat.stun_port
Local	Web User Interface	Configure NAT traversal and STUN server on the IP phone. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0

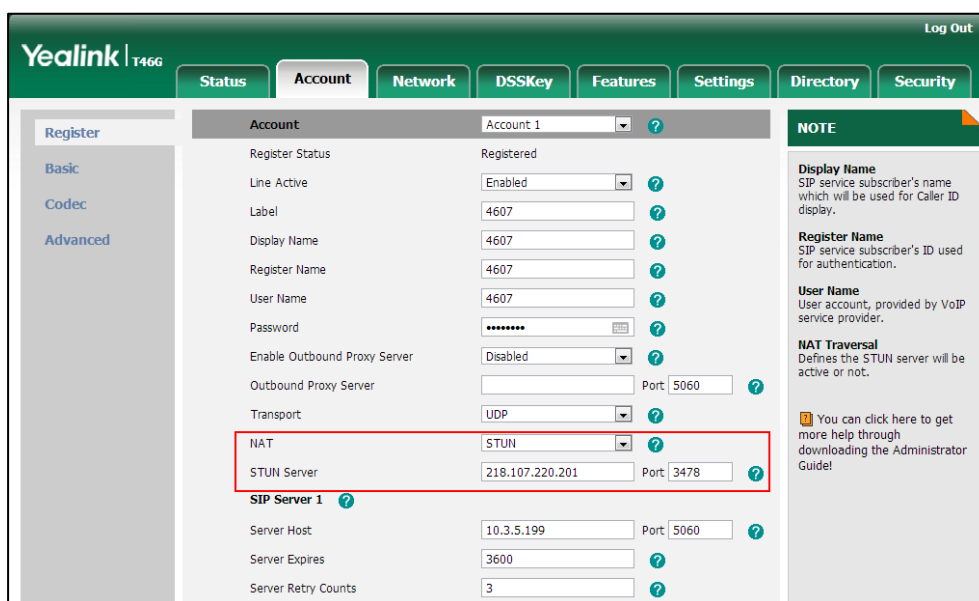
Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.nat.nat_traversal	0 or 1	0
<p>Description: Enables or disables the NAT traversal for account X.</p> <p>0-Disabled 1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Register ->NAT</p> <p>Phone User Interface: None</p>		
account.X.nat.stun_server	IP address or domain name	Blank
<p>Description: Configures the IP address or the domain name of the STUN server for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Example: account.1.nat.stun_server = 218.107.220.201</p> <p>Web User Interface: Account->Register ->STUN Server</p> <p>Phone User Interface: None</p>		
account.X.nat.stun_port	Integer from 1024 to 65000	3478
<p>Description: Configures the port of the STUN server for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p>		

Parameters	Permitted Values	Default
Example:		
account.1.nat.stun_port = 3478		
Web User Interface:		
Account->Register ->STUN Server ->Port		
Phone User Interface:		
None		

To configure the NAT traversal and STUN server via web user interface:

1. Click on **Account**.
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 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 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, EAP-PEAP/MSCHAPv2, EAP-TTLS/EAP-MSCHAPv2, EAP-PEAP/GTC and EAP-TTLS/EAP-GTC for 802.1X authentication.

For more information on 802.1X authentication, refer to *Yealink 802.1X Authentication*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

802.1X authentication can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the 802.1X authentication. Parameters: network.802_1x.mode network.802_1x.identity network.802_1x.md5_password network.802_1x.root_cert_url network.802_1x.client_cert_url.
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.

Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.802_1x.mode	0, 1, 2, 3, 4, 5 or 6	0
Description: Configures the 802.1x authentication method. 0-Disabled 1-EAP-MD5 2-EAP-TLS		

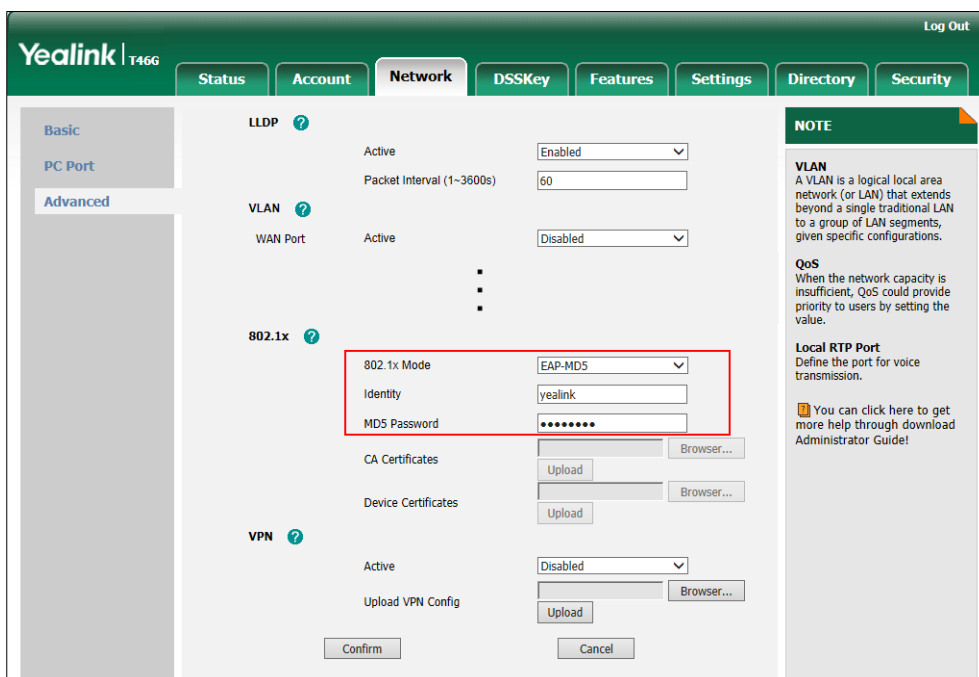
Parameters	Permitted Values	Default
<p>3-EAP-PEAP/MSCHAPv2</p> <p>4-EAP-TTLS/EAP-MSCHAPv2</p> <p>5-EAP-PEAP/GTC</p> <p>6-EAP-TTLS/EAP-GTC</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface:</p> <p>Network->Advanced->802.1x->802.1x Mode</p> <p>Phone User Interface:</p> <p>Menu->Advanced (default password: admin) ->Network->802.1x >802.1x Mode</p>		
network.802_1x.identity	String within 32 characters	Blank
<p>Description:</p> <p>Configures the user name for 802.1x authentication.</p> <p>Example:</p> <p>network.802_1x.identity = yealink</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface:</p> <p>Network->Advanced->802.1x->Identity</p> <p>Phone User Interface:</p> <p>Menu->Advanced (default password: admin) ->Network->802.1x >802.1x Mode ->Identity</p>		
network.802_1x.md5_password	String within 32 characters	Blank
<p>Description:</p> <p>Configures the password for 802.1x authentication.</p> <p>Example:</p> <p>network.802_1x.md5_password = admin123</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is required for all 802.1x authentication methods except EAP-TLS.</p> <p>Web User Interface:</p> <p>Network->Advanced->802.1x->MD5 Password</p> <p>Phone User Interface:</p> <p>Menu->Advanced (default password: admin) ->Network->802.1x ->802.1x Mode</p>		

Parameters	Permitted Values	Default
(EAP-MD5)->MD5 password		
network.802_1x.root_cert_url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the CA certificate when the 802.1x authentication method is configured as EAP-TLS, EAP-PEAP/MSCHAPv2, EAP-TTLS/EAP-MSCHAPv2, EAP-PEAP/GTC or EAP-TTLS/EAP-GTC.</p> <p>Example : network.802_1x.root_cert_url = http://192.168.1.10/ca.pem</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to EAP-TLS, EAP-PEAP/MSCHAPv2, EAP-TTLS/EAP-MSCHAPv2, EAP-PEAP/GTC and EAP-TTLS/EAP-GTC protocols. The format of the certificate must be *.pem, *.crt, *.cer or *.der.</p> <p>Web User Interface: Network->Advanced->802.1x->CA Certificates</p> <p>Phone User Interface: None</p>		
network.802_1x.client_cert_url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the device certificate when the 802.1x authentication method is configured as EAP-TLS.</p> <p>Example: network.802_1x.client_cert_url = http://192.168.1.10/client.pem</p> <p>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.</p> <p>Web User Interface: Network->Advanced->802.1x->Device Certificates</p> <p>Phone User Interface: None</p>		

To configure the 802.1X via web user interface:

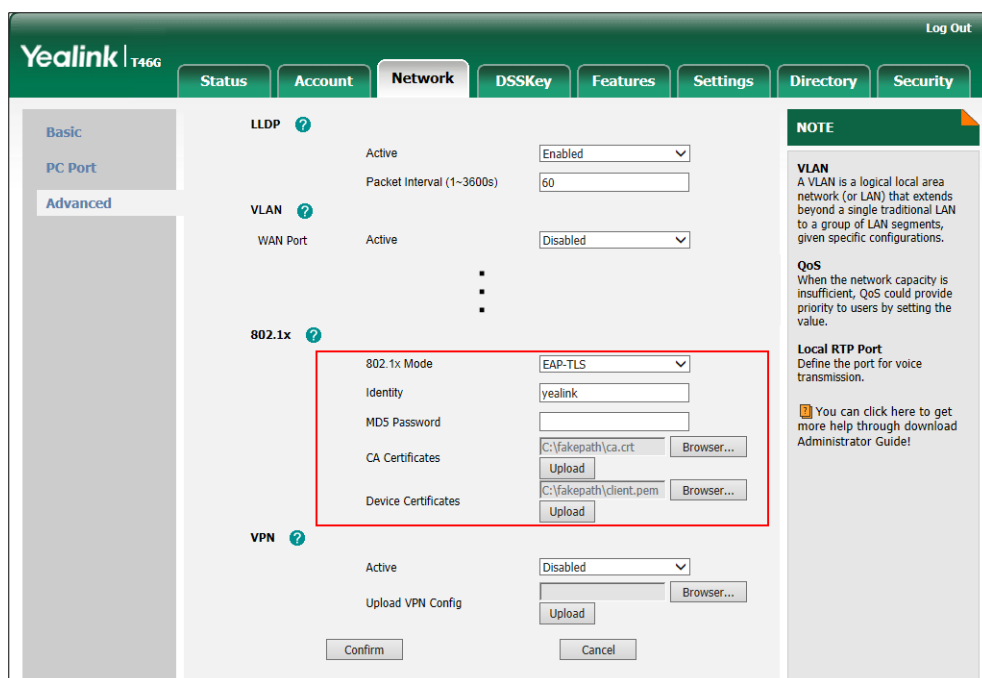
1. Click on **Network->Advanced**.
2. In the **802.1x** block, select the desired protocol from the pull-down list of **Mode 802.1x**.

- 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.



- 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 locate the desired CA certificate (*.pem, *.crt, *.cer or *.der) from your local system.
 - 4) In the **Device Certificates** field, click **Browse** to locate the desired client certificate (*.pem or *.cer) from your local system.

- 5) Click **Upload** to upload the certificates.



- c) If you select **EAP-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 locate the desired certificate (*.pem, *.crt, *.cer or *.der) from your local system.

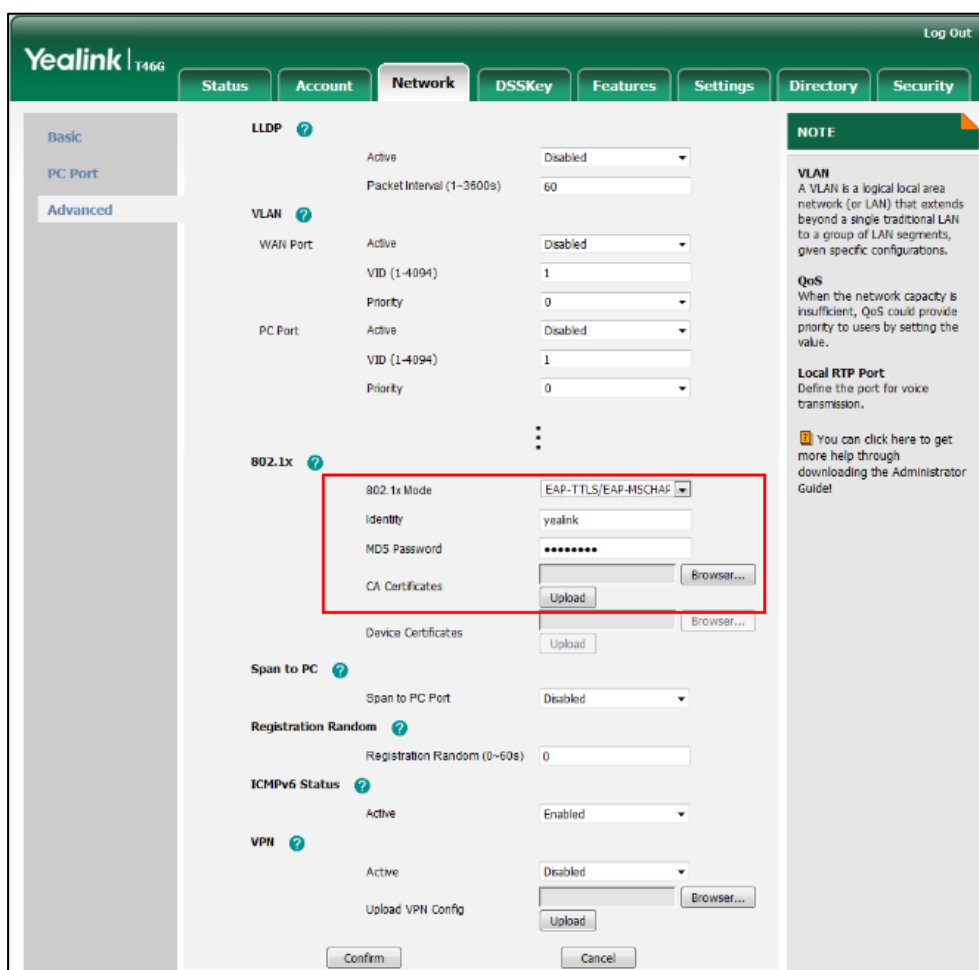
- 4) Click **Upload** to upload the certificate.

The screenshot shows the Yealink T46G configuration interface. The 'Network' tab is selected, and the '802.1x' section is highlighted with a red box. The '802.1x Mode' is set to 'EAP-PEAP/MSCHAPv2'. The 'Identity' field contains 'yealink', and the 'MD5 Password' field is masked with asterisks. The 'CA Certificates' field has an 'Upload' button. Other sections include LLDP, VLAN, Span to PC, Registration Random, ICMPv6 Status, and VPN.

- 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 locate the desired certificate (*.pem, *.crt, *.cer or *.der) from your local system.

- 4) Click **Upload** to upload the certificate.



- e) If you select **EAP-PEAP/GTC**:

- 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.

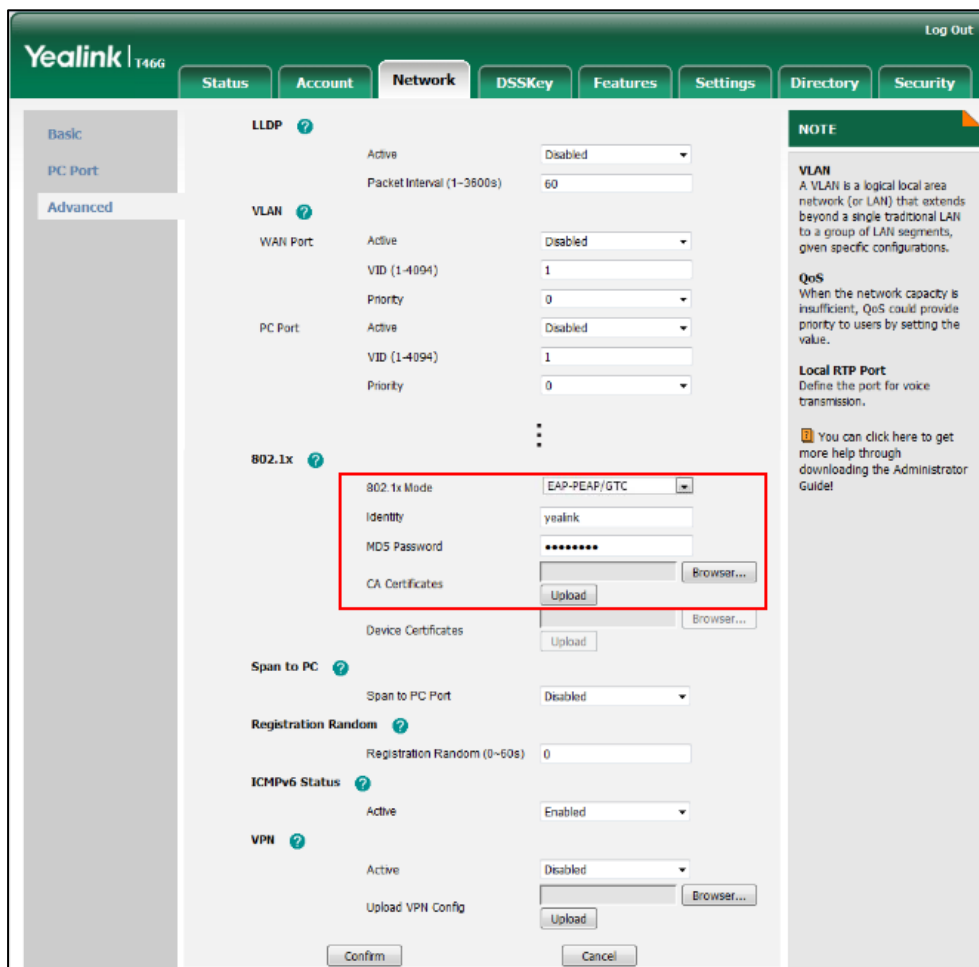
The screenshot displays the Yealink T46G configuration interface. The 'Network' tab is active, and the 'Advanced' section is selected in the sidebar. The '802.1x' configuration section is highlighted with a red box. It includes the following fields and options:

- 802.1x Mode:** A dropdown menu set to 'EAP-PEAP/GTC'.
- Identity:** A text input field containing 'yealink'.
- MD5 Password:** A text input field with masked characters (dots).
- CA Certificates:** A text input field containing 'C:\fakepath\ca.crt' and a 'Browse...' button.
- Device Certificates:** A text input field and an 'Upload' button.

Other visible fields include 'LLDP' (Active, Enabled), 'VLAN' (Active, Packet Interval 60), and 'VPN' (Active, Disabled). The interface also features a 'NOTE' section on the right with information about VLAN, QoS, and Local RTP Port.

- 4) Click **Upload** to upload the certificate.
- f) If you select **EAP-TTLS/EAP-GTC**:
- 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.
3. Click **Confirm** to accept the change.
- A dialog box pops up to prompt that the settings will take effect after reboot.
4. Click **OK** to reboot the phone.

To configure the 802.1X via phone user interface after:

1. Press **Menu->Advanced** (default password: admin) ->**Network->802.1x**.
2. Press **◀** or **▶** , or the **Switch** soft key to select the desired value from the **802.1x Mode** field.
 - 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.
 - b) If you select **EAP-TLS**:
 - 1) Enter the user name for authentication in the **Identity** field.
 - 2) Leave the **MD5 Password** field blank.

- c) If you select **EAP-PEAP/MSCHAPv2**:
 - 1) Enter the user name for authentication in the **Identity** field.
 - 2) Enter the password for authentication in the **MD5 Password** field.
 - 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.
 - e) If you select **EAP-PEAP/GTC**:
 - 1) Enter the user name for authentication in the **Identity** field.
 - 2) Enter the password for authentication in the **MD5 Password** field.
 - f) If you select **EAP-TTLS/EAP-GTC**:
 - 1) 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 LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

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.

RPC Method	Description
GetParameterValues	This method is used to obtain the value of one or more CPE parameters.
GetParameterNames	This method is used to discover the parameters accessible on a particular CPE.
GetParameterAttributes	This method is used to read the attributes associated with one or more CPE parameters.
SetParameterAttributes	This method is used to modify attributes associated with one or more CPE parameters.
Reboot	This method causes the CPE to reboot.
Download	This method is used to cause the CPE to download a specified file from the designated location. File types supported by IP phones are: <ul style="list-style-type: none"> • Firmware Image • Configuration File
Upload	This method is used to cause the CPE to upload a specified file to the designated location. File types supported by IP phones are: <ul style="list-style-type: none"> • Configuration File • Log File
ScheduleInform	This method is used to request the CPE to schedule a one-time Inform method call (separate from its periodic Inform method calls) sometime in the future.
FactoryReset	This method resets the CPE to its factory default state.
TransferComplete	This method informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.
AddObject	This method is used to add a new instance of an object defined on the CPE.
DeleteObject	This method is used to remove a particular instance of an object.

For more information on TR-069, refer to *Yealink TR-069 Technote*, available online: <http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Procedure

TR-069 can be configured using the configuration files or locally.

Configurati on File	<y0000000000xx>.cfg	<p>Configure theTR-069 feature.</p> <p>Parameters:</p> <p>managementserver.enable managementserver.username managementserver.password managementserver.url managementserver.connection_request_use rname managementserver.connection_request_pas sword managementserver.periodic_inform_enable managementserver.periodic_inform_interval</p>
Local	Web User Interface	<p>Configure the TR-069 feature.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=settings -preference&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
managementserver.enable	0 or 1	0
<p>Description: Enables or disables TR-069 feature.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->TR069->Enable TR069</p> <p>Phone User Interface: None</p>		
managementserver.username	String within 128 characters	Blank
<p>Description: Configures the user name for the IP phone to authenticate with the ACS (Auto</p>		

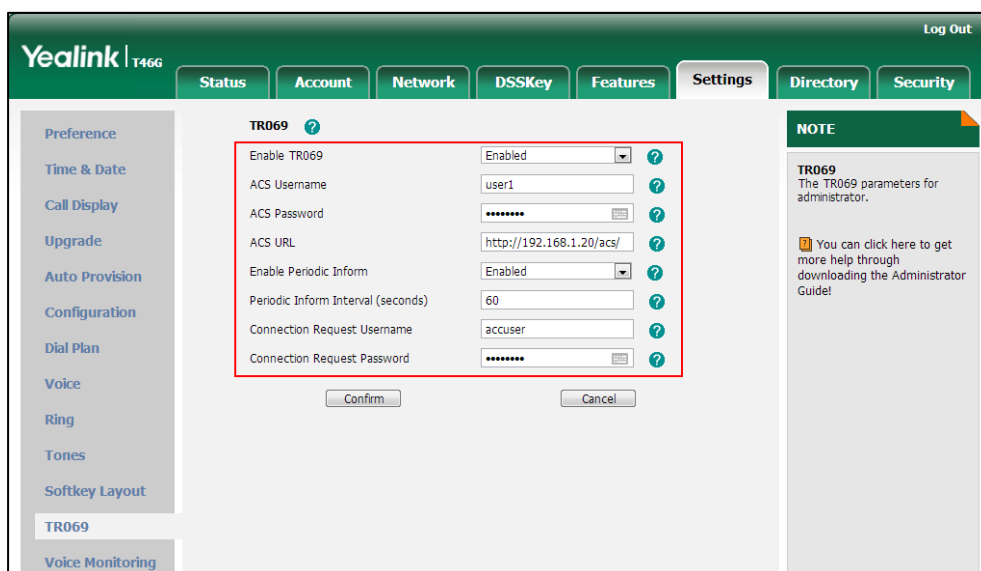
Parameters	Permitted Values	Default
<p>Configuration Servers). This string is set to the empty string if no authentication is required.</p> <p>Example: managementserver.username = user1</p> <p>Web User Interface: Settings->TR069->ACS Username</p> <p>Phone User Interface: None</p>		
managementserver.password	String within 64 characters	Blank
<p>Description: Configures the password for the IP phone to authenticate with the ACS (Auto Configuration Servers). This string is set to the empty string if no authentication is required.</p> <p>Example: managementserver.password = pwd123</p> <p>Web User Interface: Settings->TR069->ACS Password</p> <p>Phone User Interface: None</p>		
managementserver.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the ACS (Auto Configuration Servers).</p> <p>Example: managementserver.url = http://192.168.1.20/acs/</p> <p>Web User Interface: Settings->TR069->ACS URL</p> <p>Phone User Interface: None</p>		
managementserver.connection_request_username	String within 128 characters	Blank
<p>Description:</p>		

Parameters	Permitted Values	Default
<p>Configures the user name for the IP phone to authenticate the incoming connection requests.</p> <p>Example: managementserver.connection_request_username = accuser</p> <p>Web User Interface: Settings->TR069->Connection Request Username</p> <p>Phone User Interface: None</p>		
managementserver.connection_request_password	String within 64 characters	Blank
<p>Description: Configures the password for the IP phone to authenticate the incoming connection requests.</p> <p>Example: managementserver.connection_request_password = acspwd</p> <p>Web User Interface: Settings->TR069->Connection Request Password</p> <p>Phone User Interface: None</p>		
managementserver.periodic_inform_enable	0 or 1	1
<p>Description: Enables or disables the IP phone to periodically report its configuration information to the ACS (Auto Configuration Servers).</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->TR069->Enable Periodic Inform</p> <p>Phone User Interface: None</p>		
managementserver.periodic_inform_interval	Integer from 5 to 4294967295	60
<p>Description:</p>		

Parameters	Permitted Values	Default
Configures the interval (in seconds) for the IP phone to report its configuration to the ACS (Auto Configuration Servers).		
Web User Interface:		
Settings->TR069->Periodic Inform Interval (seconds)		
Phone User Interface:		
None		

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.



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.
- **Stateful DHCPv6:** The Dynamic Host Configuration Protocol for IPv6 (DHCPv6) has been standardized by the IETF through RFC3315. DHCPv6 enables DHCP servers to pass configuration parameters such as IPv6 network addresses to IPv6 nodes. It offers the capability of automatic allocation of reusable network addresses and additional configuration flexibility. This protocol is a stateful counterpart to “IPv6 Stateless Address Autoconfiguration”, and can be used separately or in addition to the stateless autoconfiguration to obtain configuration parameters.

Note

SIPT42G and SIPT41P IP phones do not support the stateful DHCPv6 address assignment method.

If the IP phone enables the SLAAC and DHCPv6 features simultaneously, the IP phone will obtain the IP address via SLAAC and obtain other network parameters via DHCPv6.

Procedure

IPv6 can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><MAC>.cfg</p>	<p>Configure the IPv6 address assignment method.</p> <p>Parameters:</p> <p>network.ip_address_mode</p> <p>network.ipv6_internet_port.type</p> <p>network.ipv6_internet_port.ip</p> <p>network.ipv6_prefix</p> <p>network.ipv6_internet_port.gateway</p>
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		<p>network.ipv6_icmp_v6.enable</p> <p>Configure the IPv6 static DNS address.</p> <p>network.ipv6_primary_dns</p> <p>network.ipv6_secondary_dns</p>
	<p><y0000000000xx>.cf</p> <p>g</p>	<p>Configure the IPv6 static DNS.</p> <p>Parameter:</p> <p>network.ipv6_static_dns_enable</p>
Local	Web User Interface	<p>Configure the IPv6 address assignment method.</p> <p>Configure the IPv6 static DNS.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?parameter=network&q=load</p>
	Phone User Interface	<p>Configure the IPv6 address assignment method.</p> <p>Configure the IPv6 static DNS.</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
network.ip_address_mode	0, 1 or 2	0
<p>Description:</p> <p>Configures the IP address mode.</p> <p>0-IPv4</p> <p>1-IPv6</p> <p>2-IPv4&IPv6</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface:</p> <p>Network->Basic->Internet Port->Mode (IPv4/IPv6)</p> <p>Phone User Interface:</p> <p>Menu->Advanced (default password: admin) ->Network->WAN Port->IP Address Mode</p>		
network.ipv6_internet_port.type	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Configures the Internet (WAN) port type for IPv6 when the IP address mode is configured as IPv6 or IPv4&IPv6.</p> <p>0-DHCP 1-Static IP Address</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type</p>		
network.ipv6_static_dns_enable	0 or 1	0
<p>Description: Enables or disables the IP phone to use manually configured static IPv6 DNS when Internet (WAN) port type for IPv6 is configured as DHCP.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config->IPv6 Static DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Static DNS</p>		
network.ipv6_internet_port.ip	IPv6 address	Blank
<p>Description: Configures the IPv6 address when the IP address mode is configured as IPv6 or IPv4&IPv6, and the Internet (WAN) port type for IPv6 is configured as Static IP Address.</p> <p>Example: network.ipv6_internet_port.ip = 2005:1:1:1::12</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p>		

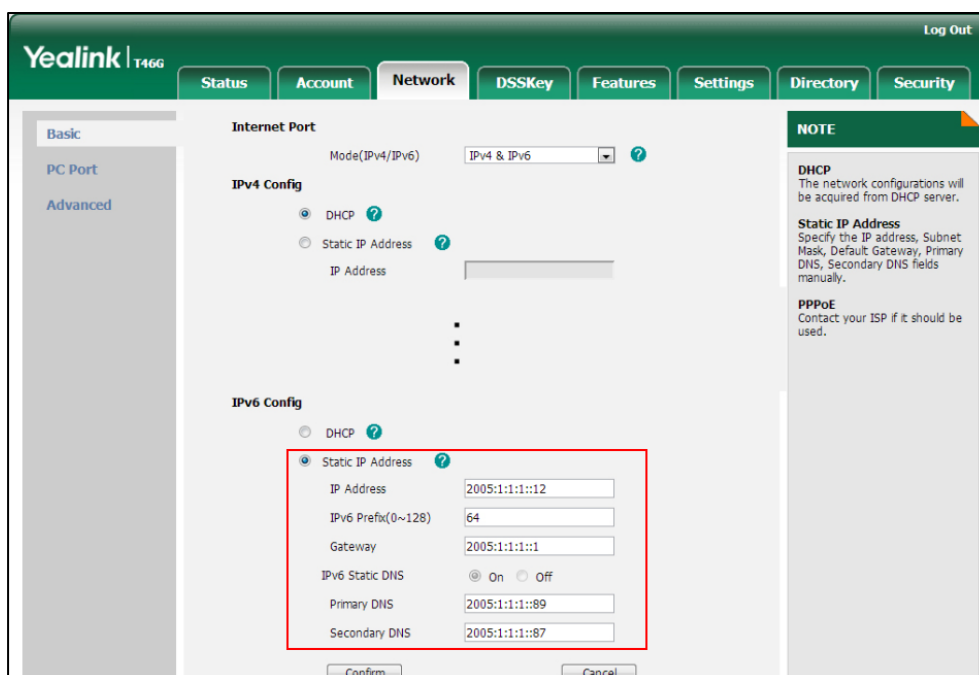
Parameters	Permitted Values	Default
<p>Web User Interface: Network->Basic->IPv6 Config->Static IP Address->IP Address</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(Static IP) ->IP Address</p>		
network.ipv6_prefix	Integer from 0 to 128	64
<p>Description: Configures the IPv6 prefix when the IP address mode is configured as IPv6 or IPv4&IPv6, and the Internet (WAN) port type for IPv6 is configured as Static IP Address.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config->Static IP Address->IPv6 Prefix (0~128)</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(Static IP) ->IPv6 IP Prefix</p>		
network.ipv6_internet_port.gateway	IPv6 address	Blank
<p>Description: Configures the IPv6 default gateway when the IP address mode is configured as IPv6 or IPv4&IPv6, and the Internet (WAN) port type for IPv6 is configured as Static IP Address.</p> <p>Example: network.ipv6_internet_port.gateway = 2005:1:1:1::1</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config->Static IP Address->Gateway</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(Static IP) ->Gateway</p>		
network.ipv6_primary_dns	IPv6 address	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the primary IPv6 DNS server when the IP address mode is configured as IPv6 or IPv4&IPv6, and the Internet (WAN) port type for IPv6 is configured as Static IP Address, or and the Internet (WAN) port type for IPv6 is configured as DHCP and Staic DNS is configured as Enabled.</p> <p>Example: network.ipv6_primary_dns = 2005:1:1:1::89</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config->IPv6 Static DNS->Primary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(Static IP) ->Primary DNS</p> <p>Or Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(DHCP) ->Staic DNS (Enabled) ->Primary DNS</p>		
network.ipv6_secondary_dns	IPv6 address	Blank
<p>Description: Configures the secondary IPv6 DNS server when the IP address mode is configured as IPv6 or IPv4&IPv6, and the Internet (WAN) port type for IPv6 is configured as Static IP Address, or and the Internet (WAN) port type for IPv6 is configured as DHCP and Staic DNS is configured as Enabled.</p> <p>Example: network.ipv6_secondary_dns = 2005:1:1:1::87</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Network->Basic->IPv6 Config->IPv6 Static DNS->Secondary DNS</p> <p>Phone User Interface: Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(Static IP) ->Secondary DNS</p> <p>Or Menu->Advanced (default password: admin) ->Network->WAN Port->IPv6->Type(DHCP) ->Staic DNS (Enabled) ->Secondary DNS</p>		
network.ipv6_icmp_v6.enable	0 or 1	1

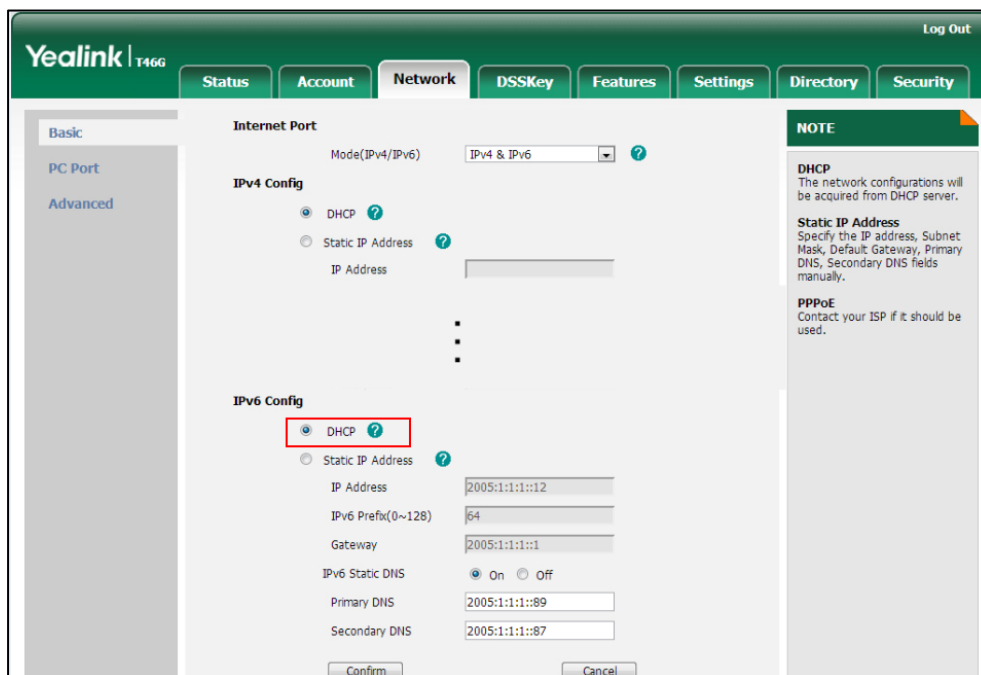
Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to obtain IPv6 network settings via SLAAC (Stateless Address Autoconfiguration) method.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect. It is only applicable to SIP-T48G/T46G IP phones.</p> <p>Web User Interface: Network->Advanced->ICMPv6 Status->Active</p> <p>Phone User Interface: None</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, mark the **DHCP** or the **Static IP Address** radio box.
 - 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.

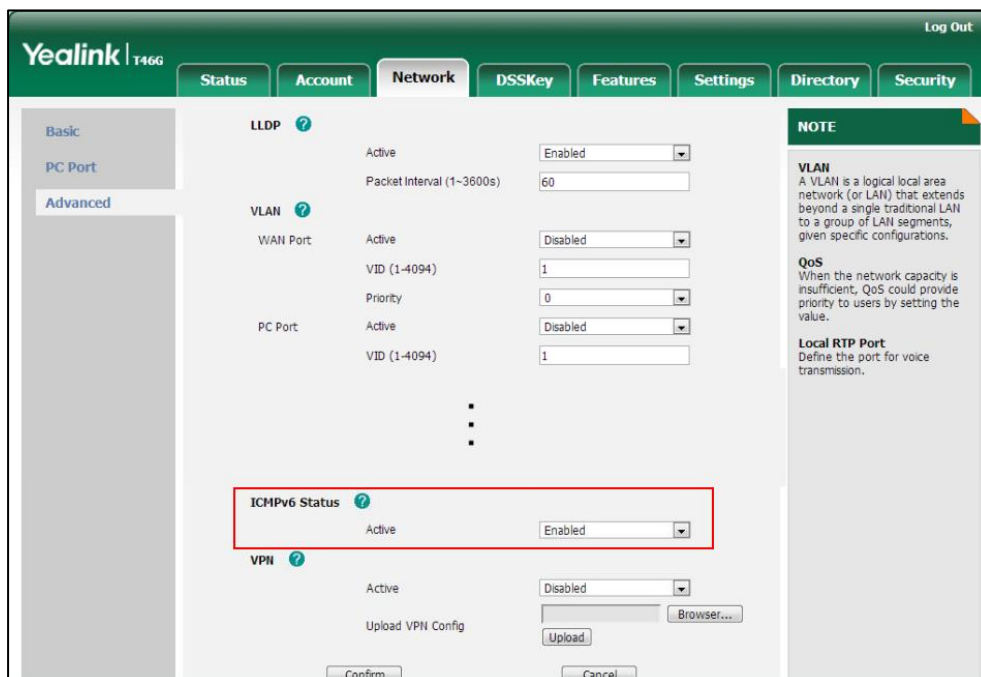
A dialog box pops up to prompt that the settings will take effect after reboot.

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

To configure SLAAC feature via web user interface (not applicable to SIP-T42G/T41P):

1. Click on **Network->Advanced**.

- In the **ICMPv6 Status** block, select the desired value from the pull-down list of **Active**.



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

To configure IPv6 address via phone user interface:

- Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port**.
- Press ◀ or ▶ to select the desired address mode from the **IP Address Mode** field.
- Press ▲ or ▼ to highlight **IPv6** and press the **Enter** soft key.
- Press ▲ or ▼ to select the desired IPv6 address assignment method.

If you select the **Static IP**, configure the IPv6 address and other configuration parameters in the corresponding fields.

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

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

To configure static IPv6 DNS when DHCPv6 is used via phone user interface:

- Press **Menu->Advanced** (default password: admin) ->**Network->WAN Port->IPv6->Static DNS**.
- Press ◀ or ▶, or the **Switch** soft key to select **Enabled** from the **Static DNS** field.
- Enter the desired values in the **Primary DNS** and **Secondary DNS** fields respectively.
- Press the **Save** soft key to accept the change.

The LCD screen will prompt "Reboot now?". Press **OK** soft key to reboot the phone and make settings effective or the **Cancel** soft key to cancel.

Configuring Audio Features

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

- [Headset Prior](#)
- [Dual Headset](#)
- [Voice Quality Monitoring](#)
- [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. Parameter: features.headset_prior
Local	Web User Interface	Configure headset prior. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

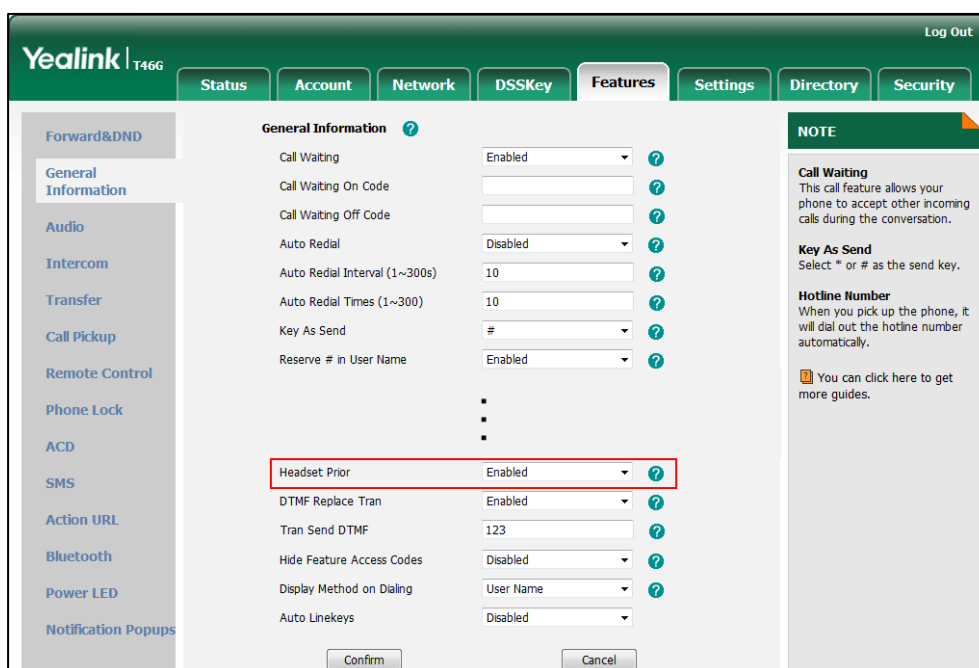
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.headset_prior	0 or 1	0
Description: Enables or disables headset prior feature. 0-Disabled 1-Enabled		

Parameter	Permitted Values	Default
<p>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.</p> <p>Web User Interface: Features->General Information->Headset Prior</p> <p>Phone User Interface: None</p>		

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 IP 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.

Procedure

Dual headset can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure dual headset. Parameter: features.headset_training
Local	Web User Interface	Configure dual headset. Navigate to: http://<phoneIPAddress>/servlet ?p=features-general&q=load

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
features.headset_training	0 or 1	0
<p>Description: Enables or disables dual headset feature.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), users can use two headsets on one phone. When the IP phone joins in a call, the users with the headset connected to the headset jack have a full-duplex conversation, while the users with the headset connected to the handset jack are only allowed to listen to.</p> <p>Web User Interface: Features->General Information->Dual-Headset</p> <p>Phone User Interface: None</p>		

To configure dual headset via web user interface:

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

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

The screenshot shows the Yealink T466 web interface. The 'Features' tab is selected, and the 'General Information' section is expanded. The 'Dual-Headset' option is highlighted with a red box, and its value is set to 'Enabled'. Other features visible include Call Waiting (Enabled), Call Waiting On Code, Call Waiting Off Code, Auto Redial (Disabled), Auto Redial Interval (1~300s) (10), Auto Redial Times (1~300) (10), Key As Send (#), Reserve # in User Name (Enabled), Auto-Answer Delay (1~4s) (1), Headset Prior (Enabled), Hide Feature Access Codes (Disabled), Display Method on Dialing (User Name), and Auto Linekeys (Disabled). A 'NOTE' section on the right provides information about Call Waiting, Key As Send, and Hotline Number.

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

Voice Quality Monitoring

Voice quality monitoring feature allows the IP phones to generate various quality metrics for listening quality and conversational quality. These metrics can be sent between the phones in RTCP-XR packets. These metrics can also be sent in SIP PUBLISH messages to a central voice quality report collector.

Two methods for voice quality monitoring are supported by Yealink IP phones:

- RTCP-XR
- VQ-RTCPXR

RTCP-XR

The RTCP-XR method, compliant with [RFC 3611-RTP Control Extended Reports \(RTCP-XR\)](#), provides the metrics contained in RTCP-XR packets for monitoring and diagnosing the call quality and performance. These metrics include network packet loss, delay metrics, analog metrics and voice quality metrics.

Procedure

RTCP-XR can be configured using the configuration files.

Configuration File	<y0000000000xx>.cfg	Configure the generation of RTCP-XR packets. Parameter: phone_setting.rtcp_xr_report.enable
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Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.rtcp_xr_report.enable	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to periodically (every 5 seconds) send RTCP-XR packets to another participating party during a call for call quality monitoring and diagnosing.</p> <p>Web User Interface:</p> <p>None</p> <p>Phone User Interface:</p> <p>None</p>		

VQ-RTCPXR

The VQ-RTCPXR method, compliant with [RFC 6035](#), sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector. Three types of quality reports can be enabled:

- **Session:** Generated at the end of a call.
- **Interval:** Generated during a call at a configurable period.
- **Alert:** Generated when the call quality degrades below a configurable threshold.

A wide range of performance metrics are generated in the following two ways:

- Based on current values, such as jitter, jitter buffer max and round trip delay.
- Computed using other metrics as input, such as listening Mean Opinion Score (MOS-LQ) and conversational Mean Opinion Score (MOS-CQ).

To operate with central report collector, IP phones must be configured to forward their voice quality reports to the specified report collector. You can specify the report collector on a per-line basis.

Users can check the voice quality data of the last call via web user interface or phone user interface. Users can also specify the options of the RTP status to be displayed on the phone user interface. Options of the RTP status to be displayed on the web user interface cannot be specified.

Note

When using voice quality monitoring feature, some problems will occur:

1. GapDuration always equals to 0 while no burst duration.
2. JitterBufferAdaptive always equals to 2 (non-adaptive/fixed), even if it's configured adaptive.
3. MOSLQ/MOSCCQ may be lower or higher than what VQMon calculates sometimes (error of [1, +0.5]).

The problems will be fixed in firmware version 80.

Procedure

RTCP-XR can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	<p>Configure the generation of session packets.</p> <p>Parameter:</p> <p>phone_setting.vq_rtcpxr.session_report.enable</p> <p>Configure the generation of interval packets.</p> <p>Parameters:</p> <p>phone_setting.vq_rtcpxr.interval_report.enable</p> <p>phone_setting.vq_rtcpxr_interval_period</p> <p>Configure the generation of alert packets.</p> <p>Parameters:</p> <p>phone_setting.vq_rtcpxr_moslq_threshold_warning</p> <p>phone_setting.vq_rtcpxr_moslq_threshold_critical</p> <p>phone_setting.vq_rtcpxr_delay_threshold_warning</p> <p>phone_setting.vq_rtcpxr_delay_threshold_critical</p> <p>Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface.</p> <p>Parameter:</p> <p>phone_setting.vq_rtcpxr.states_show_on_web.enable</p> <p>Configure the phone to display RTP status</p>
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		<p>showing the voice quality report of the last call or the current call on the phone user interface.</p> <p>Parameter:</p> <p>phone_setting.vq_rtcpxr.states_show_on_gui.enable</p> <p>Configure the options of the RTP status displayed on the phone user interface.</p> <p>Parameters:</p> <p>phone_setting.vq_rtcpxr_display_start_time.enable</p> <p>phone_setting.vq_rtcpxr_display_stop_time.enable</p> <p>phone_setting.vq_rtcpxr_display_local_call_id.enable</p> <p>phone_setting.vq_rtcpxr_display_remote_call_id.enable</p> <p>phone_setting.vq_rtcpxr_display_local_codec.enable</p> <p>phone_setting.vq_rtcpxr_display_remote_codec.enable</p> <p>phone_setting.vq_rtcpxr_display_jitter.enable</p> <p>phone_setting.vq_rtcpxr_display_jitter_buffer_max.enable</p> <p>phone_setting.vq_rtcpxr_display_packets_lost.enable</p> <p>phone_setting.vq_rtcpxr_display_symm_one_way_delay.enable</p> <p>phone_setting.vq_rtcpxr_display_round_trip_delay.enable</p> <p>phone_setting.vq_rtcpxr_display_moslq.enable</p> <p>phone_setting.vq_rtcpxr_display_mosqcq.enable</p>
	<MAC>.cfg	<p>Configure the central report collector.</p> <p>Parameters:</p> <p>account.X.vq_rtcpxr.collector_name</p> <p>account.X.vq_rtcpxr.collector_server_host</p> <p>account.X.vq_rtcpxr.collector_server_port</p>
Local	Web User	Configure VQ-RTCPXR.

	Interface	<p>Configure the phone to display RTP status showing the voice quality report of the last call on the web user interface.</p> <p>Configure the phone to display RTP status showing the voice quality report of the last call or the current call on the phone user interface.</p> <p>Configure the options of the RTP status displayed on the phone user interface.</p> <p>Navigate to:</p> <p><a href="http://<phoneIPAddress>/servlet?p=settings-voicemonitoring&q=load">http://<phoneIPAddress>/servlet?p=settings-voicemonitoring&q=load</p> <p>Configure the central report collector.</p> <p>Navigate to:</p> <p><a href="http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0">http://<phoneIPAddress>/servlet?p=account-adv&q=load&acc=0</p>
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Details of Configuration Parameters:

Parameters	Permitted Values	Default
phone_setting.vq_rtcpxr.session_report.enable	0 or 1	0
<p>Description:</p> <p>Enables or disables the IP phone to send a session quality report to the central report collector at the end of each call.</p> <p>0-Disabled</p> <p>1-Enabled</p> <p>Web User Interface:</p> <p>Settings->Voice Monitoring->VQ RTCP-XR Session Report</p> <p>Phone User Interface:</p> <p>None</p>		
phone_setting.vq_rtcpxr.interval_report.enable	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to send an interval quality report to the central report collector periodically throughout a call.</p> <p>0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice Monitoring->VQ RTCP-XR Interval Report</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_interval_period	Integer from 5 to 20	20
<p>Description: Configures the interval (in seconds) for the IP phone to send an interval quality report to the central report collector periodically throughout a call.</p> <p>Web User Interface: Settings->Voice Monitoring->Period for Interval Report</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_moslq_threshold_warning	15 to 40	Blank
<p>Description: Configures the threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a warning alert quality report to the central report collector.</p> <p>For example, a configured value of 35 corresponds to the MOS score 3.5. When the MOS-LQ value computed by the phone is less than or equal to 3.5, the phone will send a warning alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 3.5, the phone will not send a warning alert quality report to the central report collector.</p> <p>If it is set to blank, warning alerts are not generated due to MOS-LQ.</p> <p>Web User Interface: Settings->Voice Monitoring->Warning threshold for Moslq</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
phone_setting.vq_rtcp_r_moslq_threshold_critical	15 to 40	Blank
<p>Description:</p> <p>Configures the desired threshold value of listening MOS score (MOS-LQ) multiplied by 10. The threshold value of MOS-LQ causes the phone to send a critical alert quality report to the central report collector.</p> <p>For example, a configured value of 28 corresponds to the MOS score 2.8. When the MOS-LQ value computed by the phone is less than or equal to 2.8, the phone will send a critical alert quality report to the central report collector. When the MOS-LQ value computed by the phone is greater than 2.8, the phone will not send a critical alert quality report to the central report collector.</p> <p>If it is set to blank, critical alerts are not generated due to MOS-LQ.</p> <p>Web User Interface:</p> <p>Settings->Voice Monitoring->Critical threshold for Moslq</p> <p>Phone User Interface:</p> <p>None</p>		
phone_setting.vq_rtcp_r_delay_threshold_warning	10 to 2000	Blank
<p>Description:</p> <p>Configures the threshold value of one way delay (in ms) that causes the phone to send a warning alert quality report to the central report collector.</p> <p>For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a warning alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a warning alert quality report to the central report collector.</p> <p>If it is set to blank, warning alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.</p> <p>Web User Interface:</p> <p>Settings->Voice Monitoring->Warning threshold for Delay</p> <p>Phone User Interface:</p> <p>None</p>		
phone_setting.vq_rtcp_r_delay_threshold_critical	10 to 2000	Blank

Parameters	Permitted Values	Default
<p>Description: Configures the threshold value of one way delay (in ms) that causes phone to send a critical alert quality report to the central report collector. For example, If it is set to 500, when the value of one way delay computed by the phone is greater than or equal to 500, the phone will send a critical alert quality report to the central report collector; when the value of one way delay computed by the phone is less than 500, the phone will not send a critical alert quality report to the central report collector. If it is set to blank, critical alerts are not generated due to one way delay. One-way delay includes both network delay and end system delay.</p> <p>Web User Interface: Settings->Voice Monitoring->Critical threshold for Delay</p> <p>Phone User Interface: None</p>		
<p>phone_setting.vq_rtcp.rstates_show_on_web.enable</p>	<p>0 or 1</p>	<p>0</p>
<p>Description: Enables or disables the voice quality data of the last call to be displayed on web interface at path Status->RTP Status. 0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice Monitoring->Display Report options on Web</p> <p>Phone User Interface: None</p>		
<p>phone_setting.vq_rtcp.rstates_show_on_gui.enable</p>	<p>0 or 1</p>	<p>0</p>
<p>Description: Enables or disables the voice quality data of the last call or current call to be displayed on the LCD screen. You can view the voice quality data of the last call by pressing Menu->Status->RTP Status. You can view the voice quality data of the current call by pressing RTP Status soft key during a call. 0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice Monitoring->Display Report options on phone</p>		

Parameters	Permitted Values	Default
Phone User Interface: None		
phone_setting.vq_rtcp_xr_display_start_time.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Start Time on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Start Time</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_display_stop_time.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Current Time or Stop Time on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Current Time</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_display_local_call_id.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Local User on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1".</p>		

Parameters	Permitted Values	Default
<p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Local User</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_display_remote_call_id.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Remote User on the LCD screen.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Remote User</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_display_local_codec.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Local Codec on the LCD screen.</p> <p>0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Local Codec</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcp_xr_display_remote_codec.enable	0 or 1	1

Parameters	Permitted Values	Default
<p>Description: Enables or disables the phone to display Remote Codec on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1". Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Remote Codec Phone User Interface: None</p>		
phone_setting.vq_rtcpxr_display_jitter.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Jitter on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1". Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Jitter Phone User Interface: None</p>		
phone_setting.vq_rtcpxr_display_jitter_buffer_max.enable	0 or 1	1
<p>Description: Enables or disables the phone to display JitteBufferMax on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1". Web User Interface: Settings->Voice Monitoring->Report options on phone UI->JitteBufferMax Phone User Interface: None</p>		

Parameters	Permitted Values	Default
phone_setting.vq_rtcpxr_display_packets_lost.enable	0 or 1	1
<p>Description: Enables or disables the phone to display Packet lost on the LCD screen. 0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->Packet lost</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcpxr_display_symm_oneway_delay.enable	0 or 1	0
<p>Description: Enables or disables the phone to display SymmOneWayDelay on the LCD screen. 0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->SymmOneWayDelay</p> <p>Phone User Interface: None</p>		
phone_setting.vq_rtcpxr_display_round_trip_delay.enable	0 or 1	0
<p>Description: Enables or disables the phone to display RoundTripDelay on the LCD screen. 0-Disabled 1-Enabled</p> <p>Note: It works only if the value of the parameter "phone_setting.vq_rtcpxr.states_show_on_gui.enable" is set to "1".</p> <p>Web User Interface: Settings->Voice Monitoring->Report options on phone UI->RoundTripDelay</p>		

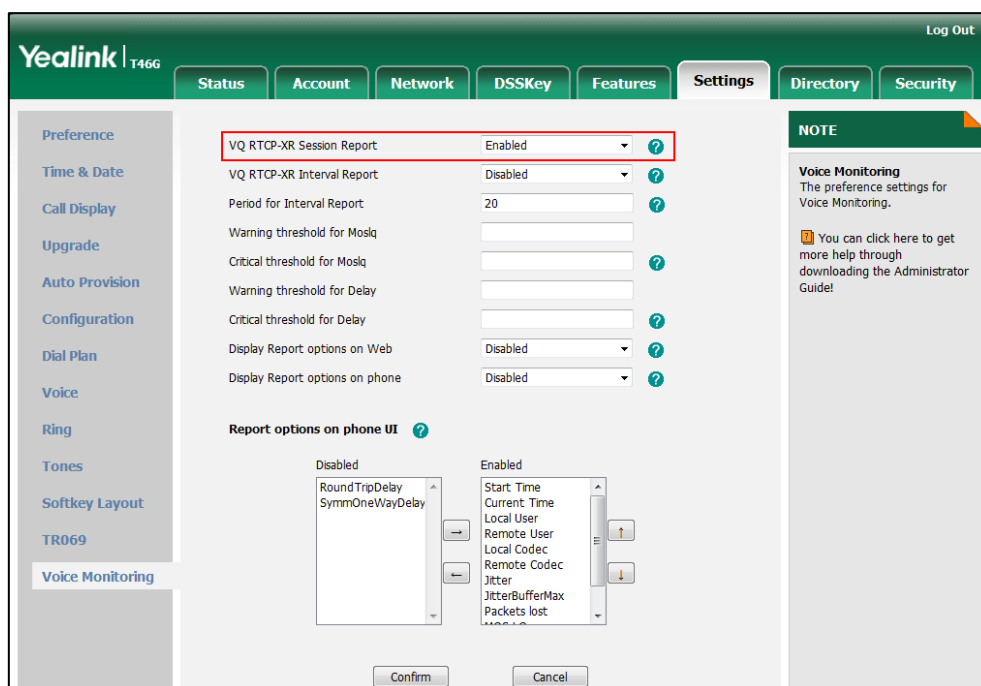
Parameters	Permitted Values	Default
Phone User Interface: None		
phone_setting.vq_rtcp_xr_display_moslq.enable	0 or 1	1
Description: Enables or disables the phone to display MOS-LQ on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1". Web User Interface: Settings->Voice Monitoring->Report options on phone UI->MOS-LQ Phone User Interface: None		
phone_setting.vq_rtcp_xr_display_moscq.enable	0 or 1	1
Description: Enables or disables the phone to display MOS-CQ on the LCD screen. 0-Disabled 1-Enabled Note: It works only if the value of the parameter "phone_setting.vq_rtcp_xr_states_show_on_gui.enable" is set to "1". Web User Interface: Settings->Voice Monitoring->Report options on phone UI->MOS-CQ Phone User Interface: None		
account.X.vq_rtcp_xr_collector_name	String within 32 characters	Blank
Description: Configures the host name of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G).		

Parameters	Permitted Values	Default
<p>X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->VQ RTCP-XR Collector name</p> <p>Phone User Interface: None</p>		
account.X.vq_rtcpxr.collector_server_host	IPv4 Address	Blank
<p>Description: Configures the IP address of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->VQ RTCP-XR Collector address</p> <p>Phone User Interface: None</p>		
account.X.vq_rtcpxr.collector_server_port	Integer from 1 to 65535	5060
<p>Description: Configures the port of the central report collector that accepts voice quality reports contained in SIP PUBLISH messages for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->VQ RTCP-XR Collector port</p> <p>Phone User Interface: None</p>		

To configure session report for VQ-RTCPXR via web user interface:

1. Click on **Settings->Voice Monitoring**.

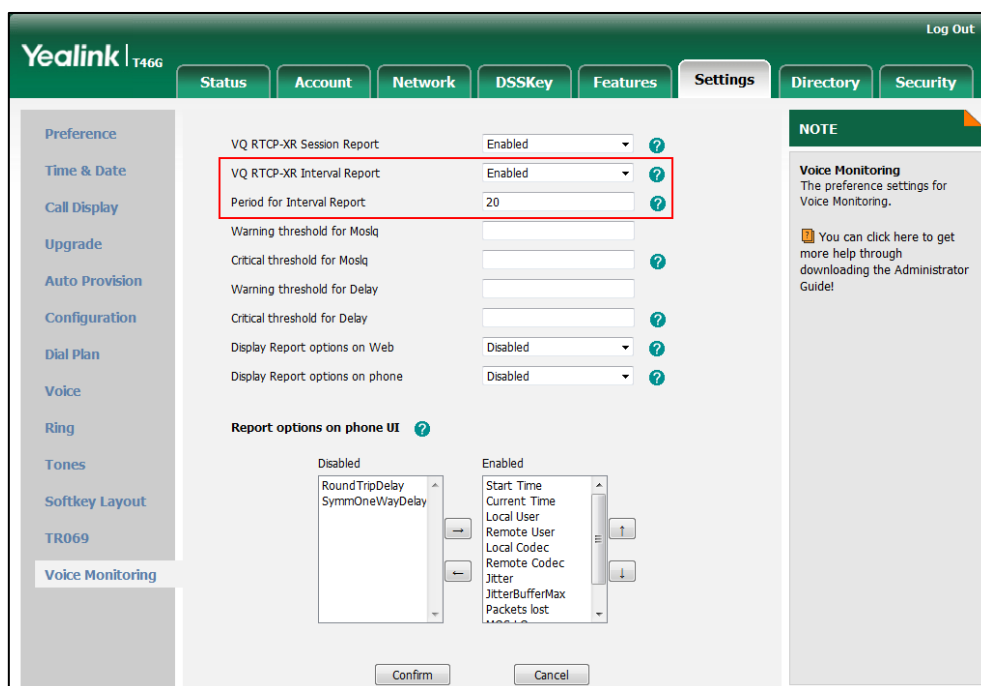
2. Select the desired value from the pull-down list of **VQ RTCP-XR Session Report**.



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

To configure interval report for VQ RTCP-XR via web user interface:

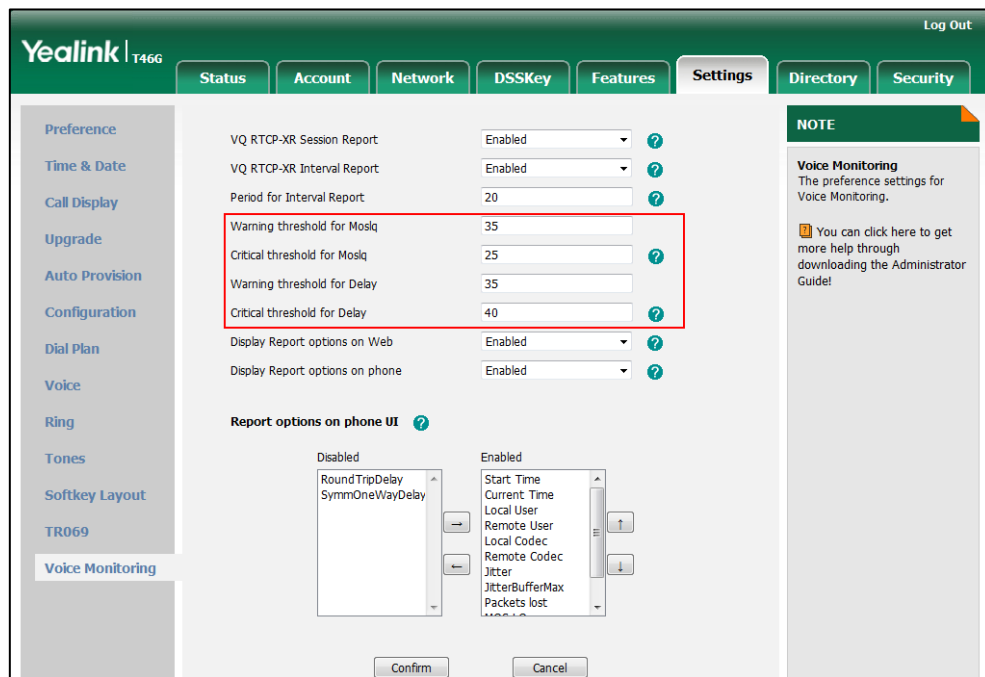
1. Click on **Settings->Voice Monitoring**.
2. Select the desired value from the pull-down list of **VQ RTCP-XR Interval Report**.
3. Enter the desired value in the **Period for Interval Report** field.



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

To configure alert report for VQ RTPC-XR via web user interface:

1. Click on **Settings->Voice Monitoring**.
2. Enter the desired value in the **Warning threshold for Moslq** field.
3. Enter the desired value in the **Critical threshold for Moslq** field.
4. Enter the desired value in the **Warning threshold for Delay** field.
5. Enter the desired value in the **Critical threshold for Delay** field.

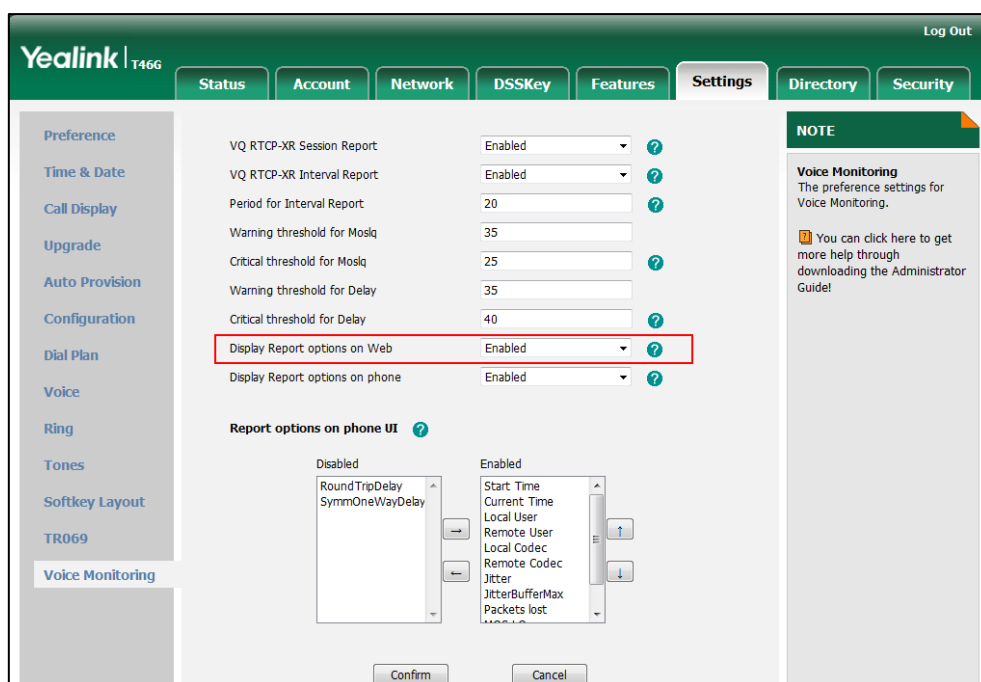


6. Click **Confirm** to accept the change.

To configure RTP status displayed on the web page via web user interface:

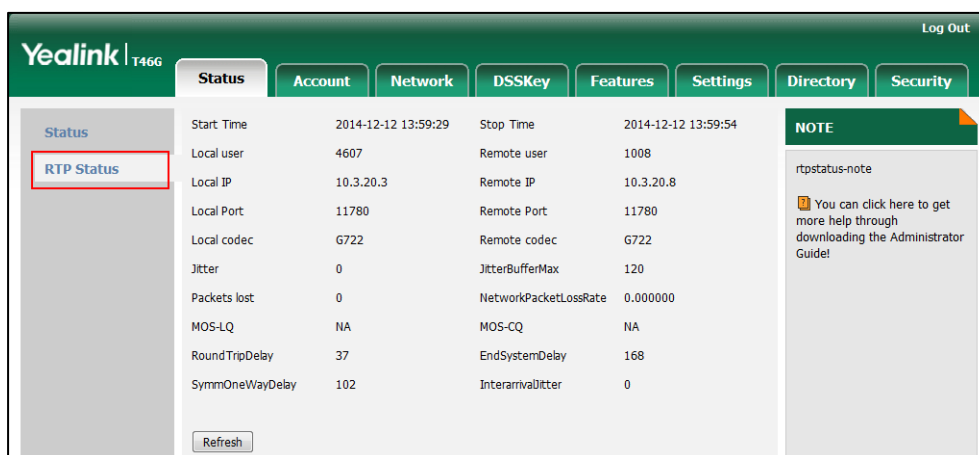
1. Click on **Settings->Voice Monitoring**.

2. Select the desired value from the pull-down list of **Display Report options on Web**.



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

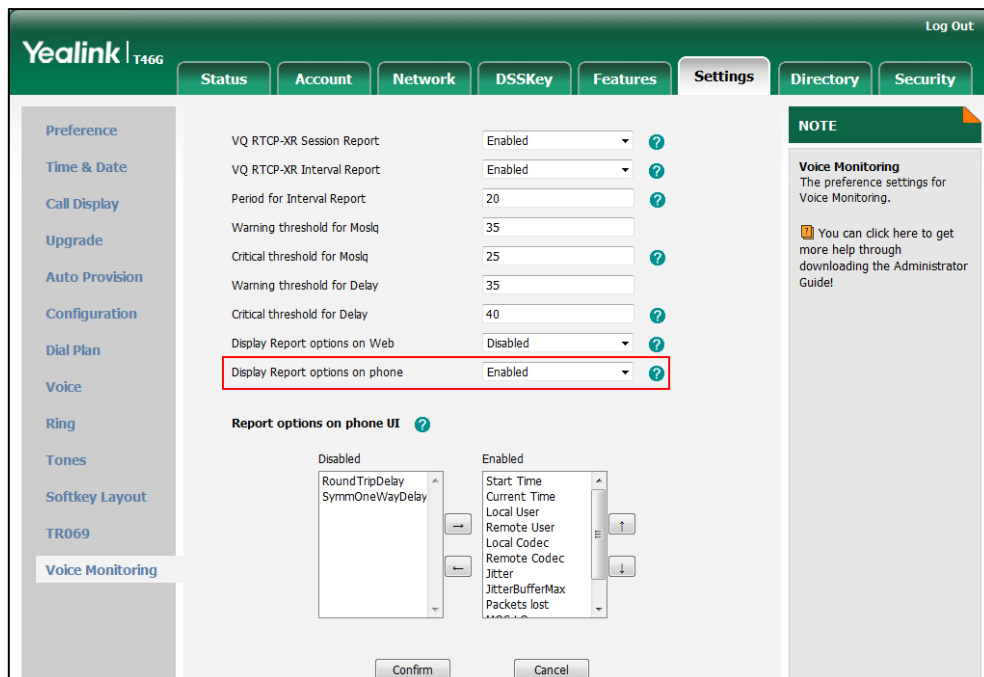
The RTP status will appear on the web user interface at the path: **Status**.



To configure RTP status displayed on the LCD screen via web user interface:

1. Click on **Settings->Voice Monitoring**.

2. Select the desired value from the pull-down list of **Display Report** options on phone.




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

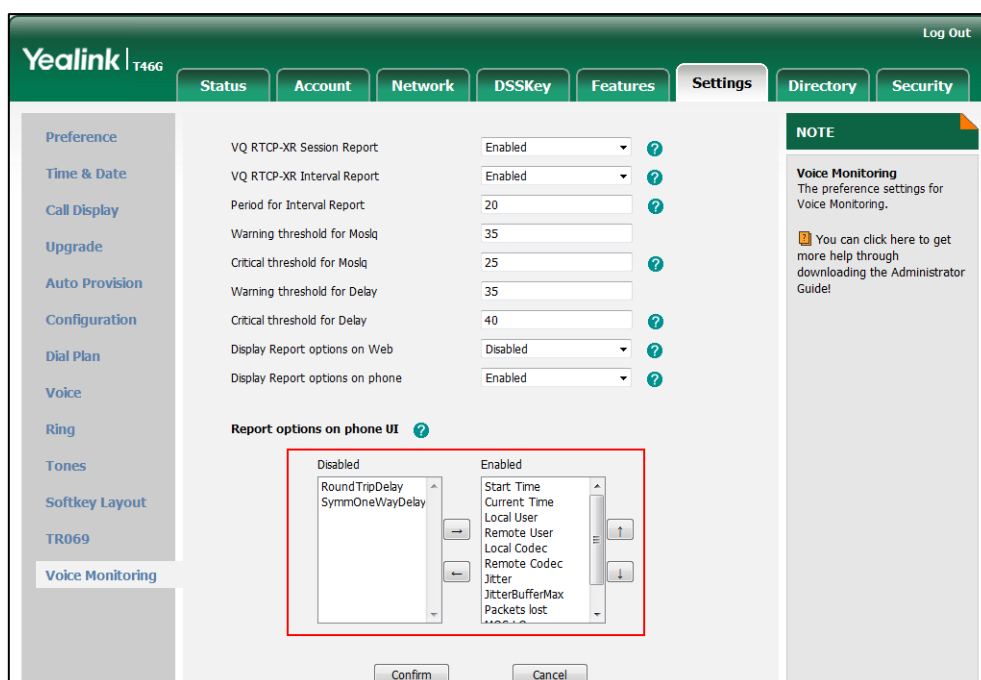
The RTP status will appear on the phone user interface at the path:




Menu->Status->More....

To configure the options of the RTP status displayed on the LCD screen via web user interface:

1. Click on **Settings->Voice Monitoring**.
2. In the **Report options on phone UI** block, select the desired list from the **Disabled** column and then click .

The selected list appears in the **Enabled** column.



3. Repeat step 2 to add more items to the **Enabled** column.
4. To remove an item from the **Enabled** column, select the desired item and then click .
5. To adjust the display order of enabled items, select the desired item and then click  or .

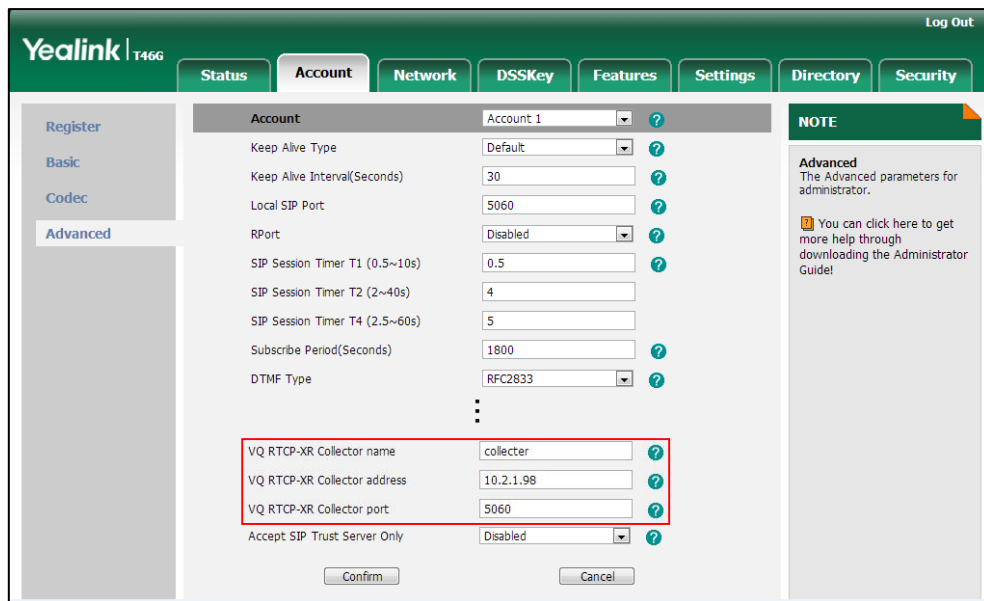
The LCD screen will display the item(s) in the adjusted order.

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

To configure the central report collector via web user interface:

1. Click on **Account->Advanced**.
2. Enter the host name of the central report collector in the **VQ RTCP-XR Collector name** field.
3. Enter the IP address of the central report collector in the **VQ RTCP-XR Collector address** field.

- Enter the port of the central report collector in the **VQ RTCP-XR Collector port** field.



- Click **Confirm** to 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 following table lists the audio codecs supported by SIP-T4X IP phones:

Supported Audio Codecs	Default Audio Codecs
G722, PCMA, PCMU, G729, G723_53, G723_63, G726-16, G726-24, G726-32, G726-40, iLBC	G722, PCMA, PCMU, G729

The following table summarizes the supported audio codecs on IP phones:

Codec	Algorithm	Reference	Bit Rate	Sample	Packetization
G722	G.722	RFC 3551	64 Kbps	16 Ksps	20ms
PCMA	G.711	RFC 3551	64 Kbps	8 Ksps	20ms
PCMU	G.711	RFC 3551	64 Kbps	8 Ksps	20ms
G729	G.729	RFC 3551	8 Kbps	8 Ksps	20ms
G726-16	G.726	RFC 3551	16 Kbps	8 Ksps	20ms

Codec	Algorithm	Reference	Bit Rate	Sample	Packetization
G726-24	G.726	RFC 3551	24 Kbps	8 Ksps	20ms
G726-32	G.726	RFC 3551	32 Kbps	8 Ksps	20ms
G726-40	G.726	RFC 3551	40 Kbps	8 Ksps	20ms
G723_53/ G723_63	G.723.1	RFC 3951	5.3kbps 6.3kbps	8 Ksps	30ms
iLBC	iLBC	RFC 3952	13.33 Kbps 15.2 Kbps	8 Ksps	20ms 30ms

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.

Codecs and priorities of these codecs are configurable on a per-line basis. The attribute "rtpmap" is used to define a mapping from RTP payload codes to a codec, clock rate and other encoding parameters.

The corresponding attributes of the codec are listed as follows:

Codec	Configuration Methods	Priority	RTPmap
G722	Configuration Files Web User Interface	1	9
PCMU	Configuration Files Web User Interface	2	0
PCMA	Configuration Files Web User Interface	3	8
G729	Configuration Files Web User Interface	4	18
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

Codec	Configuration Methods	Priority	RTPmap
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

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>Parameters:</p> <p>account.X.codec.Y.enable account.X.codec.Y.payload_type</p> <p>Configure the priority and rtpmap for the enabled codec.</p> <p>Parameters:</p> <p>account.X.codec.Y.priority account.X.codec.Y.rtpmap</p> <p>Configure the ptime.</p> <p>Parameter:</p> <p>account.X.ptime</p>
Local	Web User Interface	<p>Configure the codecs 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>

Details of the Configuration Parameter:


Parameters	Permitted Values	Default
<p>account.X.codec.Y.enable (X ranges from 1 to 16. Y ranges from 1 to 11)</p>	<p>0 or 1</p>	<p>Refer to the following content</p>
<p>Description: Enables or disables the specified codec for account X.</p> <p>0-Disabled 1-Enabled</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Default: When Y=1, the default value is 1; When Y=2, the default value is 1; When Y=3, the default value is 0; When Y=4, the default value is 0; When Y=5, the default value is 1; When Y=6, the default value is 1; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0; When Y=10, the default value is 0; When Y=11, the default value is 0;</p> <p>Web User Interface: Account->Codec</p> <p>Phone User Interface: None</p>		
<p>account.X.codec.Y.payload_type (X ranges from 1 to 16. Y ranges from 1 to 11)</p>	<p>Refer to the following content</p>	<p>Refer to the following content</p>
<p>Description: Configures the codec for account X.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p>		

Parameters	Permitted Values	Default
<p>Permitted Values: PCMU, PCMA, G729, G722, G723_53, G723_63, G726-16, G726-24, G726-32, G726-40, iLBC</p> <p>Default: 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>Example: account.1.codec.1.payload_type = PCMU</p> <p>Web User Interface: Account->Codec</p> <p>Phone User Interface: None</p>		
<p>account.X.codec.Y.priority (X ranges from 1 to 16. Y ranges from 1 to 11)</p>	<p>Integer from 0 to 11</p>	<p>Refer to the following content</p>
<p>Description: Configures the priority of the enabled codec for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Default: When Y=1, the default value is 2; When Y=2, the default value is 3; When Y=3, the default value is 0; When Y=4, the default value is 0;</p>		




Parameters	Permitted Values	Default
<p>When Y=5, the default value is 4; When Y=6, the default value is 1; When Y=7, the default value is 0; When Y=8, the default value is 0; When Y=9, the default value is 0; When Y=10, the default value is 0; When Y=11, the default value is 0;</p> <p>Web User Interface: Account->Codec</p> <p>Phone User Interface: None</p>		
<p>account.X.codec.Y.rtpmap (X ranges from 1 to 16. Y ranges from 1 to 11)</p>	<p>Integer from 0 to 127</p>	<p>Refer to the following content</p>
<p>Description: Configures the rtpmap of the audio codec for account X. X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Default: 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>Web User Interface: None</p> <p>Phone User Interface: None</p>		

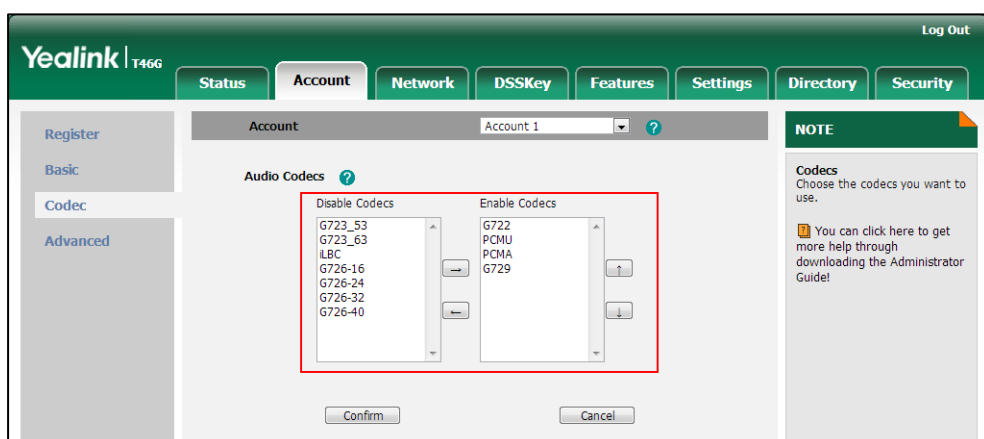
Parameters	Permitted Values	Default
account.X.ptime	0 (Disabled), 10, 20, 30, 40, 50 or 60	20
<p>Description: Configures the ptime (in milliseconds) for the codec for account X. X ranges from 1 to 16 (for SIPT48G/T46G). X ranges from 1 to 12 (for SIPT42G). X ranges from 1 to 6 (for SIPT41P).</p> <p>Example: account.1.ptime = 20</p> <p>Web User Interface: Account->Advanced->PTime(ms)</p> <p>Phone User Interface: None</p>		

To configure the codecs 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 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. Click  to remove the codec from the **Enable Codecs** column.
7. Click  or  to adjust the priority of the enabled codecs.



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

To configure the PTime 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. Select the desired value from the pull-down list of **PTime (ms)**.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The 'PTime(ms)' field is highlighted with a red box, showing a value of 20. Other fields include 'Keep Alive Type' (Default), 'Keep Alive Interval (Seconds)' (30), 'Local SIP Port' (5062), 'RPort' (Disabled), 'SIP Session Timer T1 (0.5~10s)' (0.5), 'RTP Encryption(SRTP)' (Forbin SRTP), 'Shared Line' (Disabled), 'Dialog Info Call Pickup' (Disabled), 'Directed Call Pickup Code', 'Group Call Pickup Code', 'Distinctive Ring Tones' (Disabled), 'Unregister When Reboot' (Disabled), and 'Out Dialog BLF' (Disabled). A 'NOTE' section on the right states: 'Advanced The Advanced parameters for administrator.' Buttons for 'Confirm' and 'Cancel' are at the bottom.

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

Acoustic Clarity Technology

Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) is used to reduce acoustic echo from a voice call to provide natural full-duplex communication patterns. 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. AEC is not normally required for calls via the handset. In certain situation, where echo is experienced by the remote party, AEC may be used to reduce/avoid echo when the user uses the handset.

Procedure

AEC can be configured using the configuration files or locally.

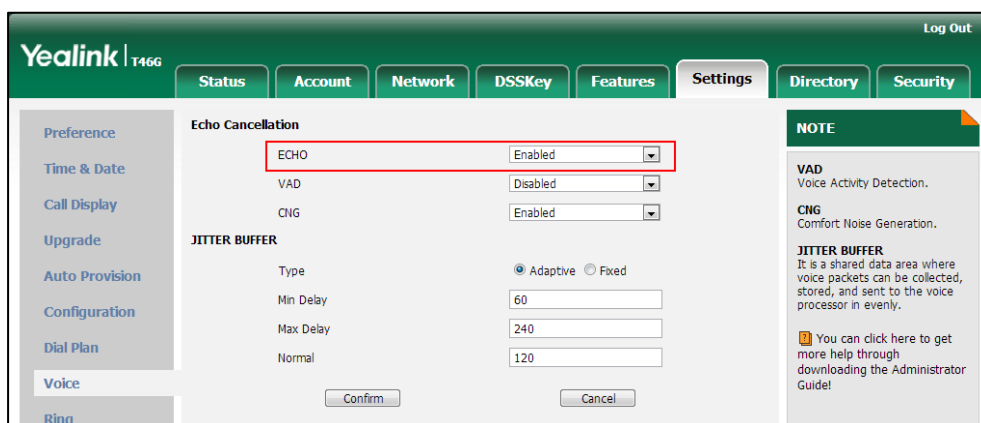
Configuration File	<y0000000000xx>.cfg	Configure AEC. Parameter: voice.echo_cancellation
Local	Web User Interface	Configure AEC. Navigate to: http://<phoneIPAddress>/servlet?p=settings-voice&q=load

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
voice.echo_cancellation	0 or 1	1
<p>Description: Enables or disables AEC (Acoustic Echo Canceller) feature on the IP phone. 0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice->Echo Cancellation->ECHO</p> <p>Phone User Interface: None</p>		

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.

Background Noise Suppression

Background noise suppression (BNS) is designed primarily for hands-free operation and reduces background noise to enhance communication in noisy environments.

Automatic Gain Control

Automatic Gain Control (AGC) is applicable to hands-free operation and is used to keep audio output at nearly a constant level by adjusting the gain of signals in certain circumstances. This increases the effective user-phone radius and helps with the intelligibility of talkers.

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.

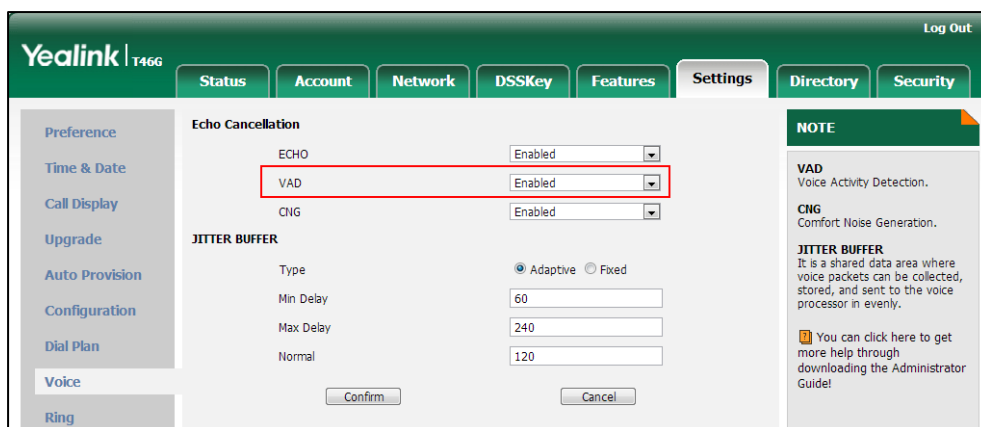
Configuration File	<y0000000000xx>.cfg	Configure VAD. Parameter: voice.vad
Local	Web User Interface	Configure VAD. Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?p=settings-voice&q=load

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
voice.vad	0 or 1	0
<p>Description: Enables or disables VAD (Voice Activity Detection) feature on the IP phone. 0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice->Echo Cancellation ->VAD</p> <p>Phone User Interface: None</p>		

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. Parameter: voice.cng
Local	Web User Interface	Configure CNG. Navigate to: http://<phoneIPAddress>/servl et?p=settings-voice&q=load

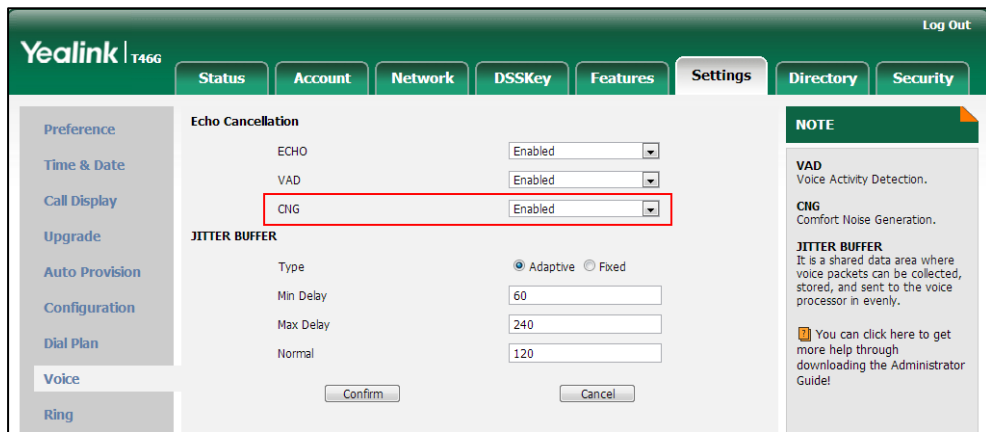
Details of the Configuration Parameter:

Parameter	Permitted Values	Default
voice.cng	0 or 1	1
<p>Description: Enables or disables CNG (Comfortable Noise Generator) feature on the IP phone. 0-Disabled 1-Enabled</p> <p>Web User Interface: Settings->Voice->Echo Cancellation ->CNG</p> <p>Phone User Interface: None</p>		

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: fixed and adaptive. A fixed jitter buffer adds the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. An adaptive jitter buffer is capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can 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. Parameters: voice.jib.adaptive voice.jib.min voice.jib.max voice.jib.normal
Local	Web User Interface	Configure the mode of jitter buffer and the delay time for jitter buffer.

		<p>Navigate to: <a href="http://<phoneIPAddress>/servlet?<phoneIPAddress>/servlet?p=settings-voice&q=load">http://<phoneIPAddress>/servlet?<phoneIPAddress>/servlet?p=settings-voice&q=load</p>
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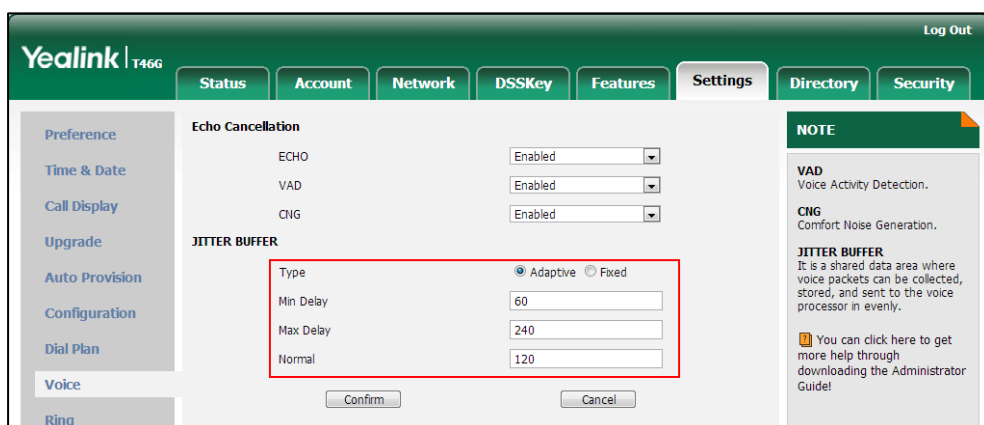
Details of Configuration Parameters:

Parameters	Permitted Values	Default
voice.jib.adaptive	0 or 1	1
<p>Description: Configures the type of jitter buffer. 0-Fixed 1-Adaptive</p> <p>Web User Interface: Settings->Voice->JITTER BUFFER->Type</p> <p>Phone User Interface: None</p>		
voice.jib.min	Integer from 0 to 400	60
<p>Description: Configures the minimum delay time (in milliseconds) of jitter buffer.</p> <p>Note: It works only if the value of the parameter "voice.jib.adaptive" is set to 1 (Adaptive).</p> <p>Web User Interface: Settings->Voice->JITTER BUFFER->Min Delay</p> <p>Phone User Interface: None</p>		
voice.jib.max	Integer from 0 to 400	240
<p>Description: Configures the maximum delay time (in milliseconds) of jitter buffer.</p> <p>Note: It works only if the value of the parameter "voice.jib.adaptive" is set to 1 (Adaptive).</p> <p>Web User Interface: Settings->Voice->JITTER BUFFER->Max Delay</p> <p>Phone User Interface: None</p>		

Parameters	Permitted Values	Default
voice.jib.normal	Integer from 0 to 400	120
<p>Description: Configures the normal delay time (in milliseconds) of jitter buffer.</p> <p>Note: It works only if the value of the parameter “voice.jib.adaptive” is set to 0 (Fixed).</p> <p>Web User Interface: Settings->Voice->JITTER BUFFER->Normal</p> <p>Phone User Interface: None</p>		

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.
5. Enter the fixed delay time for fixed jitter buffer in the **Normal** field.
Valid values range from 0 to 300.



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

Configuring Security Features

This chapter provides information for making configuration changes for the following security-related features:

- [Transport Layer Security](#)
- [Secure Real-Time Transport Protocol](#)
- [Encrypting Configuration Files](#)

Transport Layer Security

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.

SIP-T4X IP phones support TLS 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. SIP-T4X 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:

The image shows a Wireshark network capture of a TLS handshake. The main pane displays a list of packets with the following data:

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

The packet details pane for the selected packet (No. 13) shows:

- 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: The 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 will appear 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 up to 10 custom certificates to the IP phone. The format of the certificates 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 installed by default and is unique to an IP phone (based on the MAC address) and issued by the Yealink Certificate Authority (CA).
 - A generic server certificate:** It is installed by default and is 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. And Security verification rules are compliant with RFC 2818.

Note

For TLS feature, we use the terms trusted and server certificates. 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.

Procedure

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

Configuration File	<MAC>.cfg	<p>Configure TLS on a per-line basis.</p> <p>Parameter: account.X.transport</p>
	<y000000000xx>.cfg	<p>Configure the trusted certificates feature.</p> <p>Parameters: security.trust_certificates security.ca_cert security.cn_validation</p> <p>Configure the server certificates feature.</p> <p>Parameters: security.dev_cert</p> <p>Upload the trusted certificates.</p> <p>Parameter: trusted_certificates.url</p> <p>Upload the server certificates.</p> <p>Parameter: server_certificates.url</p> <p>Configure the custom</p>

		<p>certificates.</p> <p>Parameter:</p> <p>phone_setting.reserve_certs_enable</p>
Local	Web User Interface	<p>Configure TLS on a per-line basis.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=account-register&q=load&acc=0</p> <p>Configure the trusted certificates feature.</p> <p>Upload the trusted certificates.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=trusted-cert&q=load</p> <p>Configure the server certificates feature.</p> <p>Upload the server certificates.</p> <p>Navigate to:</p> <p>http://<phoneIPAddress>/servlet?p=server-cert&q=load</p>

Details of Configuration Parameters:

Parameters	Permitted Values	Default
account.X.transport	0, 1, 2 or 3	0
<p>Description:</p> <p>Configures the type of transport protocol for account X.</p> <p>0-UDP</p> <p>1-TCP</p> <p>2-TLS</p> <p>3-DNS-NAPTR</p> <p>X ranges from 1 to 16 (for SIPT48G/T46G).</p> <p>X ranges from 1 to 12 (for SIPT42G).</p> <p>X ranges from 1 to 6 (for SIPT41P).</p> <p>Web User Interface:</p>		

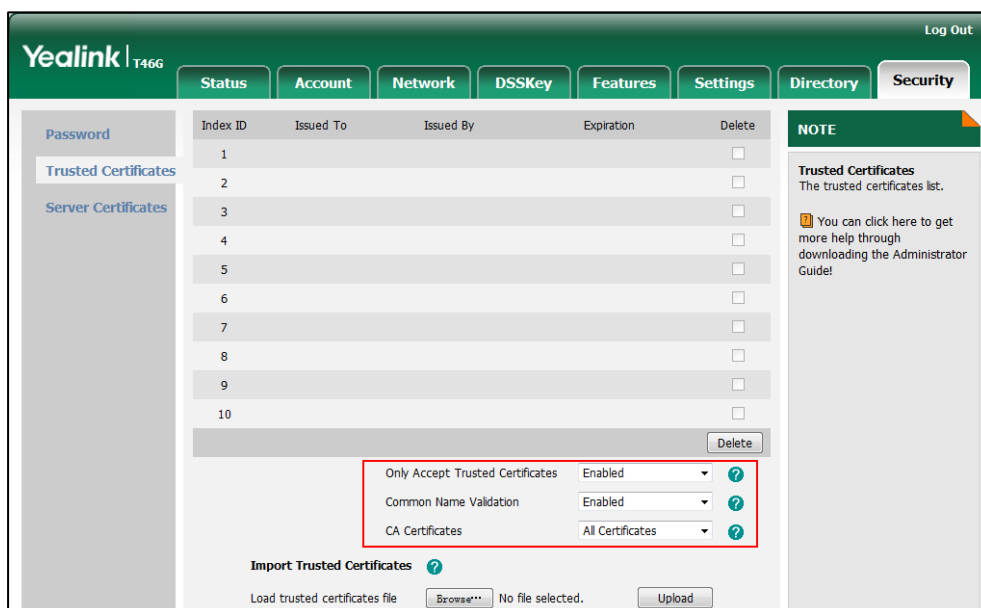
Parameters	Permitted Values	Default
Account->Register ->Transport Phone User Interface: None		
security.trust_certificates	0 or 1	1
<p>Description: Enables or disables the IP phone to only trust the server certificates in the Trusted Certificates list.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone will authenticate the server certificate based on the trusted certificates list. Only when the authentication succeeds, the IP phone will trust the server.</p> <p>If it is set to 0 (Disabled), the IP phone will trust the server no matter whether the certificate sent by the server is valid or not.</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Security->Trusted Certificates->Only Accept Trusted Certificates</p>		
security.ca_cert	0, 1 or 2	2
<p>Description: Configures the type of certificates in the Trusted Certificates list for the IP phone to authenticate for TLS connection.</p> <p>0-Default certificates 1-Custom certificates 2-All certificates</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Security->Trusted Certificates->CA Certificates</p> <p>Phone User Interface: None</p>		
security.cn_validation	0 or 1	0

Parameters	Permitted Values	Default
<p>Description: Enables or disables the IP phone to mandatorily validate the CommonName or SubjectAltName of the certificate sent by the server.</p> <p>0-Disabled 1-Enabled</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Security->Trusted Certificates->Common Name Validation</p> <p>Phone User Interface: None</p>		
security.dev_cert	0 or 1	0
<p>Description: Configures the type of the device certificates for the IP phone to send for TLS authentication.</p> <p>0-Default certificates 1-Custom certificates</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Security->Server Certificates->Device Certificates</p> <p>Phone User Interface: None</p>		
trusted_certificates.url	URL within 511 characters	Blank
<p>Description: Configures the access URL of the custom trusted certificate used to authenticate the connecting server.</p> <p>Example: trusted_certificates.url = http://192.168.1.20/tc.crt</p> <p>Note: The certificate you want to upload must be in *.pem, *.crt, *.cer or *.der format.</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
Security->Trusted Certificates->Load trusted certificates file Phone User Interface: None		
server_certificates.url	URL within 511 characters	Blank
Description: Configures the access URL of the certificate the IP phone sends for authentication. Example: server_certificates.url = http://192.168.1.20/ca.pem Note: The certificate you want to upload must be in *.pem or *.cer format. Web User Interface: Security->Server Certificates->Load server cer file Phone User Interface: None		

To configure the trusted certificates feature via web user interface:

1. Click on **Security->Trusted Certificates**.
2. Select the desired value from the pull-down list of **Only Accept Trusted Certificates**.
3. Select the desired value from the pull-down list of **Common Name Validation**.
4. Select the desired value from the pull-down list of **CA Certificates**.



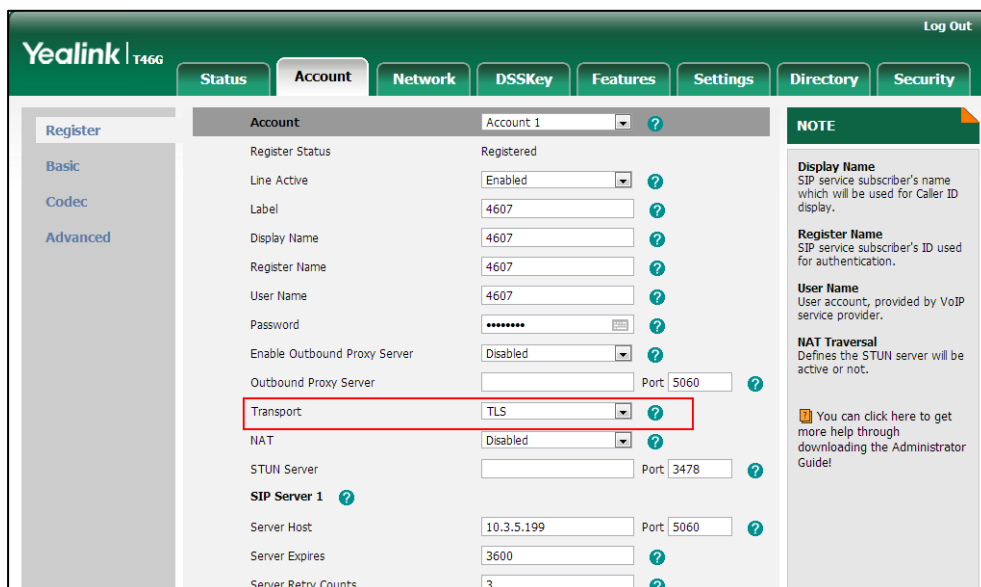
5. Click **Confirm** to accept the change.

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

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

To configure TLS 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. Select **TLS** from the pull-down list of the **Transport**.

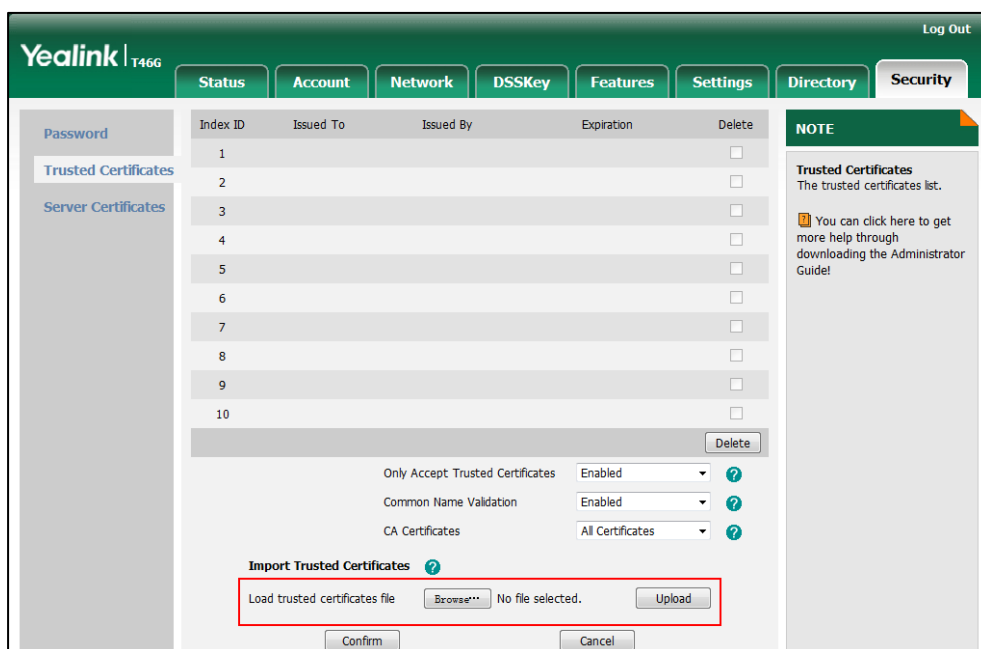


4. Click **Confirm** to accept the change.

To upload a trusted certificate via web user interface:

1. Click on **Security->Trusted Certificates**.

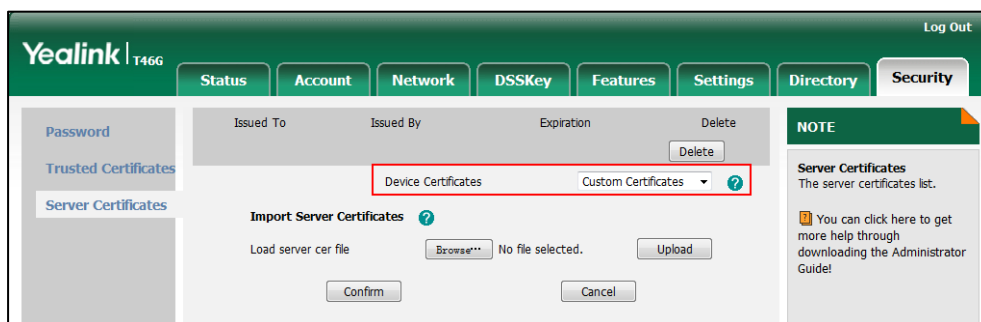
- Click **Browse** to locate the certificate (*.pem,*.crt, *.cer or *.der) from your local system.



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

To configure the server certificates feature via web user interface:

- Click on **Security->Server Certificates**.
- Select the desired value from the pull-down list of **Device Certificates**.



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

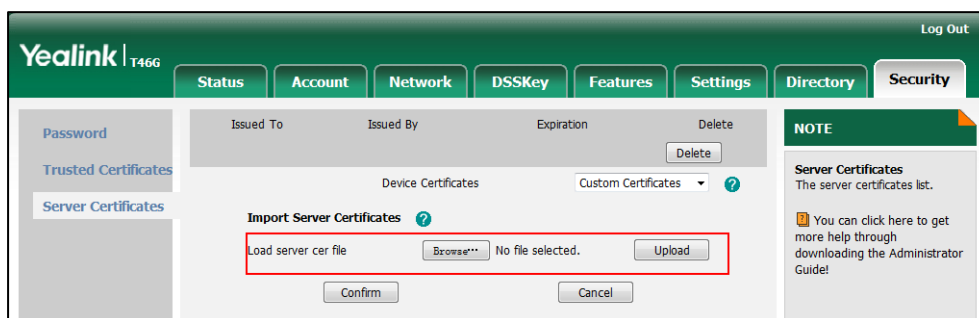
A dialog box pops up to prompt that the settings will take effect after reboot.

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

To upload a server certificate via web user interface:

- Click on **Security->Server Certificates**.

- Click **Browse** to locate the certificate (*.pem or *.cer) from your local system.



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

The 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 RTP streams during VoIP phone calls to avoid interception and eavesdropping. The parties participating in the call must enable SRTP simultaneously. When this feature is enabled on both phones, the encryption algorithm utilized for the session is negotiated between 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:NzFINTUwZDk2OGVIOTc3YzNkYTkWZWkMTM1YWFj
a=crypto:2 AES_CM_128_HMAC_SHA1_32
inline:NzkyM2FjNzQ2ZDgxYjg0MzQwMGVmMGUxMzdmNWFm
a=crypto:3 F8_128_HMAC_SHA1_80 inline:NDIiMWIwZGE1ZTAwZjA5ZGFhNjQ5YmEANTMzYzA0
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 feature 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 461.

Procedure

SRTP can be configured using the configuration files or locally.

Configuration File	<MAC>.cfg	Configure SRTP feature on a per-line basis. Parameter: account.X.srtp_encryption account.X.srtp_auth_tag_mode
Local	Web User Interface	Configure SRTP feature on a per-line basis. Navigate to: http://<phoneIPAddress>/servlet ?p=account-adv&q=load&acc=0

Details of the Configuration Parameter:

Parameters	Permitted Values	Default
account.X.srtp_encryption	0, 1 or 2	0
<p>Description: Configures whether to use voice encryption service for account X.</p> <p>0-Disabled 1-Optional 2-Compulsory</p> <p>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.</p> <p>If it is set to 2 (Compulsory), the IP phone is forced to use SRTP during a call.</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->RTP Encryption (SRTP)</p> <p>Phone User Interface: None</p>		
account.X.srtp_auth_tag_mode	0, 1 or 2	0
<p>Description: Configures the encryption algorithm carried in the SIP message when using voice encryption service for account X.</p> <p>0-AES-80&&AES-32 1-AES-80 2-AES-32</p> <p>X ranges from 1 to 16 (for SIP-T48G/T46G). X ranges from 1 to 12 (for SIP-T42G). X ranges from 1 to 6 (for SIP-T41P).</p> <p>Web User Interface: Account->Advanced->SRTP Auth-tag</p> <p>Phone User Interface: None</p>		

To configure SRTP 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. Select the desired key type from the pull-down list of **SRTP Auth-tag**.

The screenshot shows the Yealink T466 web interface. The 'Account' tab is active, and the 'Advanced' sub-tab is selected. The configuration table is as follows:

Parameter	Value
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(96~127)	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
RTP Encryption(SRTP)	Optional
SRTP Auth-tag	AES-80&&AES-32

A red box highlights the 'RTP Encryption(SRTP)' and 'SRTP Auth-tag' rows. A 'NOTE' box on the right states: 'Advanced: The Advanced parameters for administrator. You can click here to get more help through download Administrator Guide!'.

6. 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 supplies 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, y000000000028_Security.enc for y000000000028.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 supplies a configuration encryption tool (yealinkencrypt) for Linux platform if required. For more information, refer to *Yealink Configuration Encryption Tool User Guide*.

For the 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 IP 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.

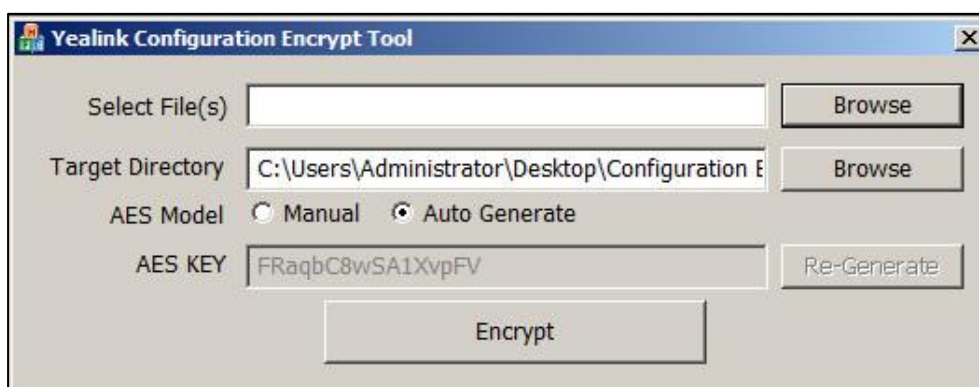
The way the IP phone processes the <MAC>.cfg file is the same to that of the <y0000000000xx>.cfg file.

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., y000000000028.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.

- (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.

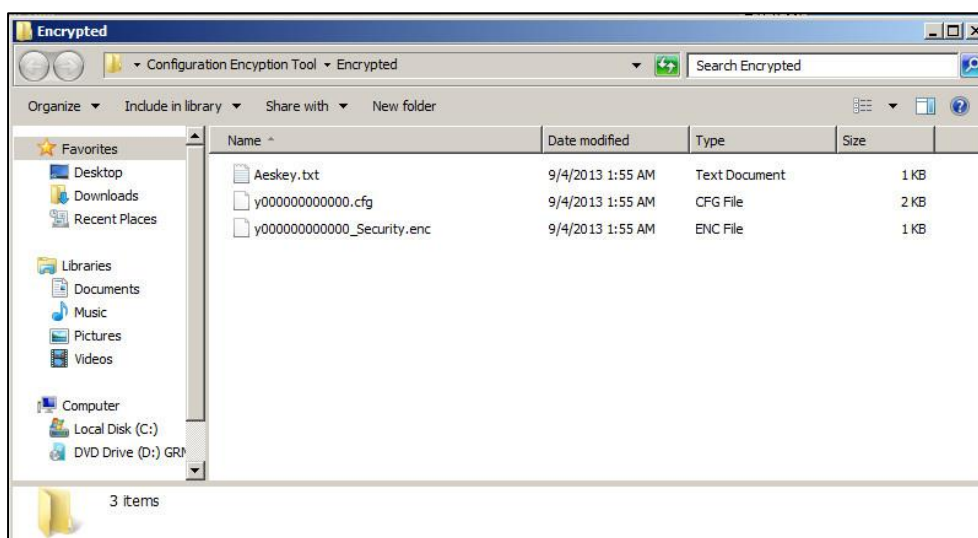
Note AES keys must be 16 characters and the supported characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % * + , - . : = ? @ [] ^ _ { } ~.

- Click **Encrypt** to encrypt the configuration file(s).



- Click **OK**.

The target directory will be automatically opened. You can find the encrypted configuration file(s), encrypted key file(s) and an Aeskey.txt file storing plaintext AES key(s).



Procedure

Encryption method and AES keys can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure the decryption method and AES keys. Parameters: auto_provision.aes_key_in_file auto_provision.aes_key_16.com auto_provision.aes_key_16.mac auto_provision.update_file_mode
Local	Web User Interface	Configure the AES keys. Navigate to: http://<phoneIPAddress>/servlet?p=settings-autop&q=load

Details of Configuration Parameters:

Parameters	Permitted Values	Default
auto_provision.aes_key_in_file	0 or 1	0
<p>Description: Enables or disables the IP phone to decrypt configuration files using the encrypted AES keys.</p> <p>0-Disabled 1-Enabled</p> <p>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).</p> <p>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.</p> <p>Web User Interface: None</p> <p>Phone User Interface: None</p>		

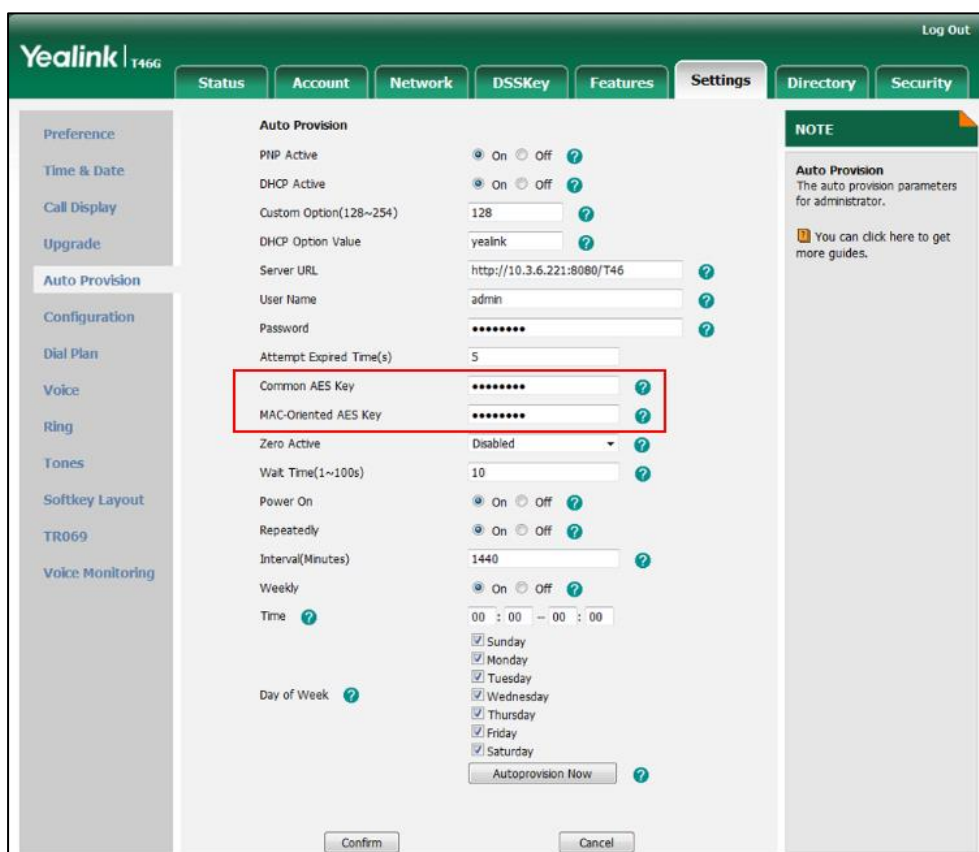
Parameters	Permitted Values	Default
auto_provision.aes_key_16.com	16 characters	Blank
<p>Description: Configures the plaintext AES key for decrypting the Common CFG file. The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % * + , - . : = ? @ [] ^ _ { } ~.</p> <p>Example: auto_provision.aes_key_16.com = 0123456789abcdef</p> <p>Note: It works only if the value of the parameter "auto_provision.aes_key_in_file" is set to 0 (Disabled).</p> <p>Web User Interface: Settings->Auto Provision->Common AES Key</p> <p>Phone User Interface: None</p>		
auto_provision.aes_key_16.mac	16 characters	Blank
<p>Description: Configures the plaintext AES key for decrypting the MAC-Oriented CFG file. The valid characters contain: 0 ~ 9, A ~ Z, a ~ z and the following special characters are also supported: # \$ % * + , - . : = ? @ [] ^ _ { } ~.</p> <p>Example: auto_provision.aes_key_16.mac = 0123456789abmins</p> <p>Note: It works only if the value of the parameter "auto_provision.aes_key_in_file" is set to 0 (Disabled).</p> <p>Web User Interface: Settings->Auto Provision->MAC-Oriented AES Key</p> <p>Phone User Interface: None</p>		
auto_provision.update_file_mode	0 or 1	0
<p>Description: Enables or disables the IP phone to update encrypted configuration settings only during auto provisioning. 0-Disabled</p>		

Parameters	Permitted Values	Default
1-Enabled		
Web User Interface:		
None		
Phone User Interface:		
None		

To configure the AES keys via web user interface:

1. Click on **Settings->Auto Provision**.
2. Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.

AES keys must be 16 characters and the supported characters contain: 0-9, A-Z, a-z and the following special characters are also supported: # \$ % * +, - . : = ? @ [] ^ _ { } ~.



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

Resource Files

When configuring particular features, you may need to upload resource files (e.g., local contact directory, remote phone book) to the IP phone. 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.

The names of the Yealink-supplied template files are (You can rename the filename as required):

Template File	File Name
Replace Rule Template	dialplan.xml
Dial-now Template	dialnow.xml
Softkey Layout Template	CallFailed.xml CallIn.xml Connecting.xml Dialing.xml (not applicable to SIPT48G) RingBack.xml Talking.xml
Directory Template	favorite_setting.xml
Super Search Template	super_search.xml
Local Contact File	contact.xml
Remote XML Phone Book	Department.xml Menu.xml

This chapter provides the detailed information on how to customize the following resource files:

- [Replace Rule Template](#)
- [Dial-now Template](#)

- [Softkey Layout Template](#)
- [Directory Template](#)
- [Super Search Template](#)
- [Local Contact File](#)
- [Remote XML Phone Book](#)

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 file, learn the following:

- `<DialRule>` indicates the start of the template file and `</DialRule>` indicates the end of the template file.
- 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.
- At most 100 replace rules can be added to the IP phone.
- The expression syntax in the replace rule template is the same as that introduced in the section [Dial Plan](#) on page 115.

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 leaving it blank, 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 shows an example of a replace rule file:

```
<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.
- At most 100 dial-now rules can be added to the IP phone.
- The expression syntax in the dial-now rule template is the same as that introduced in the section [Dial Plan](#) on page 115.

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 leaving it blank, this rule will apply to all lines.

3. Specify the values within double quotes.
4. Save the change and place this file to the provisioning server.

The following shows 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 assigning different soft key layouts to different call states. The call states include CallFailed, CallIn, Connecting, Dialing (not applicable to SIP-T48G), 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. Save the change and place this file to the provisioning server.

The following shows an example of the CallFailed template file:

```
<CallFailed>
  <Disable>
    <Key Type="Empty"/>
    <Key Type="Switch"/>
    <Key Type="Cancel"/>
  </Disable>
  <Enable>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Enable>
  <Default>
    <Key Type="NewCall"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
    <Key Type="Empty"/>
  </Default>
</CallFailed>
```

Directory Template

Directory provides easy access to frequently used lists. Users can access lists by pressing the Directory soft key when the IP phone is idle. The lists may contain Local Directory, History, Remote Phone Book and LDAP. You can add the desired list(s) to Directory using the supplied directory template (favorite_setting.xml). After setup, place the directory template to the provisioning server and specify the access URL in the configuration files.

When editing a directory template, learn the following:

- `<root_favorite_set>` indicates the start of a template and `</root_favorite_set>` indicates the end of a template.
- The default display names of directory lists are Local Directory, History, Remote Phone Book and LDAP.
- 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 for Directory, 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, edit the corresponding string in the file. For example, you want to configure the local directory list, edit the following strings:

```
<item id_name="localdirectory" display_name="Local Directory" priority="1"
enable="1" />
```

Where:

`id_name=""` specifies the existing directory list ("localdirectory" for the local directory list). Do not edit this field.

`display_name=""` specifies the display name of the directory list. We recommend you do not edit this field.

`priority=""` specifies the display priority of the directory list.

`enable=""` enables or disables the directory list for Directory.

3. Edit the values within double quotes.
4. Place this file to the provisioning server.

The following shows an example of a directory template:

```
<root_favorite_set>
  <item id_name="localdirectory" display_name="Local Directory"
priority="1" enable="1" />
  <item id_name="history" display_name="History" priority="2"
enable="0" />
  <item id_name="remotedirectory" display_name="Remote Phone Book"
priority="3" enable="0" />
  <item id_name="ldap" display_name="LDAP" priority="4" enable="0" />
</root_favorite_set>
```

Super Search Template

Search source list in dialing allows the IP phone to search for entries from the desired lists based on the entered string when in the pre-dialing screen, and then the user can select the desired entry to dial out quickly. The lists may contain Local Directory, History,

Remote Phone Book and LDAP. You can configure the search source list in dialing using the supplied super search template (super_search.xml). After setup, place the super search template to the provisioning server and specify the access URL in the configuration files.

When editing a super search template, learn the following:

- `<root_super_search>` indicates the start of a template and `</root_super_search>` indicates the end of a template.
- The default display names of directory lists are Local Directory, History, Remote Phone Book and LDAP.
- 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, edit the corresponding string in the file. For example, you want to configure the local directory list, edit the following strings:

```
<item id_name="local_directory_search" display_name="Local Directory" priority="1"
enable="1" />
```

Where:

`id_name=""` specifies the directory list ("local_directory_search" for the local directory list). Do not edit this field.

`display_name=""` specifies the display name of the directory list. We recommend you do not edit this field.

`priority=""` specifies the priority of search results.

`enable=""` enables or disables the IP phone to search the directory list.

3. Edit the values within double quotes.
4. Place this file to the provisioning server.

The following shows an example of a super search template:

```
<root_super_search>
  <item id_name="local_directory_search" display_name="Local
Directory" priority="1" enable="1" />
  <item id_name="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>

```

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 file, 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: Silent.wav, Resource: Splash.wav or Resource: RingN.wav (the default system ring tone, integer N ranges from 1 to 8) or Custom: Name.wav (the custom ring tone).
- When specifying a desired line for a contact, valid values are -1~15. Multiple line IDs are separated by commas.

The following table lists valid values for each phone model.

Phone Model	Values	Description
SIP-T41P	-1~5	-1 stands for Auto (the first registered line) 0~5 stand for line1~line6
SIP-T42G	-1~11	-1 stands for Auto (the first registered line) 0~11 stand for line1~line12
SIP-T46G/T48G	-1~15	-1 stands for Auto (the first registered line) 0~15 stand for line1~line16

- When specifying an avatar for a contact, valid values are Default: avatar name (the built-in avatar) and Config: avatar name (the custom avatar). It is not applicable to SIP-T42G and SIP-T41P IP phones.

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="" default_photo="" />
```

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.

default_photo="" specifies the avatar for this contact.

4. Specify the values within double quotes.
5. Save the change and place this file to the provisioning server.

The following shows an example of a local contact file:

```
<root_group>
  <group display_name="Friend" ring="" />
  <group display_name="Family" ring="Resource:Ring1.wav" />
</root_group>
<root_contact>
  <contact display_name="John" office_number="1001"
mobile_number="12345678910" other_number="" line="0" ring="Auto"
group_id_name="All Contacts"
default_photo="Default:default_contact_image.png"/>
  <contact display_name="Alice" office_number="1008" mobile_number=""
other_number="" line="2" ring="Resource:Silent.wav"
group_id_name="Friend" default_photo="Config:custom.png"/>
```

```
</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. You can also add multiple remote contacts at a time and/or share remote contacts between IP phones using the supplied template files (Menu.xml and Department.xml). The Menu.xml file defines departments of a remote phone book. The Department.xml file defines contact lists for a department, which is nested in Menu.xml file. After setup, place the files (Menu.xml and Department.xml) to the provisioning server, and specify the access URL of the file (Menu.xml) in the configuration files.

When creating a Menu.xml file, learn the following:

- `<YealinkIPPhoneMenu>` indicates the start of a remote phone book file and `</YealinkIPPhoneMenu>` indicates the end of a remote phone book file.
- Create the title of a remote phone book between `<Title>` and `</Title>`.
- `<MenuItem>` indicates the start of specifying a department file and `</MenuItem>` indicates the end of specifying a department file.
- `<SoftKeyItem>` indicates the start of specifying a XML file and `</SoftKeyItem>` indicates the end of specifying a XML file.

Procedure

Use the following procedures to customize an XML phone book.

To customize a Menu.xml file:

1. Open the template file using an ASCII editor.
2. For each department that you want to add, add the following strings to the file. Each starts on a separate line:

```
<MenuItem>  
    <Name> Department1 </Name>  
    <URL> http://10.3.6.117:8080/Department1.xml </URL>  
</MenuItem>
```

Where:

Specify the name of a department between `<Name>` and `</Name>`.

Specify the access URL of a department file between `</URL>` and `</URL>`.

3. For each XML file that you want to add, add the following strings to the file. Each starts on a separate line:

```
<SoftKeyItem>  
<Name> # </Name>
```

```
<URL> http://10.3.6.128:8080/TextMenu.xml</URL>
</SoftKeyItem>
```

Where:

Specify the key between <Name> and </Name>.

Specify the access URL of a XML file between </URL> and </URL>.

4. Save the file and place this file to the provisioning server.

The following shows an example of a Menu.xml file:

```
<YealinkIPPhoneMenu>
  <Title>XiaMen Yealink</Title>
  <MenuItem>
    <Name>Department1</Name>
    <URL>http://10.2.9.1:99/Department.xml</URL>
  </MenuItem>
  <MenuItem>
    <Name>Department2</Name>
    <URL>http://10.2.9.1:99/Department.xml</URL>
  </MenuItem>
  <SoftKeyItem>
    <Name>#</Name>
    <URL>http://10.2.9.1:99/Department.xml</URL>
  </SoftKeyItem>
  <SoftKeyItem>
    <Name>*</Name>
    <URL>http://10.2.9.1:99/Department.xml</URL>
  </SoftKeyItem>
  <SoftKeyItem>
    <Name>1</Name>
    <URL>http://10.2.9.1:99/Department.xml</URL>
  </SoftKeyItem>
</YealinkIPPhoneMenu>
```

When creating a Department.xml file, learn the following:

- <YealinkIPPhoneDirectory> indicates the start of a department file and </YealinkIPPhoneDirectory> indicates the end of a department file.
- Create contact lists for a department between <DirectoryEntry> and </DirectoryEntry>.

To customize a Department.xml file:

1. Open the template file using an ASCII editor.
2. For each contact that you want to add, add the following strings to the file. 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>.

3. Save the file and place this file to the provisioning server.

The following shows an example of a Department.xml file:

```
<YealinkIPPhoneDirectory>  
  <DirectoryEntry>  
    <Name>Test1</Name>  
    <Telephone>23000</Telephone>  
  </DirectoryEntry>  
  <DirectoryEntry>  
    <Name>Test2</Name>  
    <Telephone>303</Telephone>  
    <Telephone>915980830849</Telephone>  
  </DirectoryEntry>  
  <DirectoryEntry>  
    <Name>Test3</Name>  
    <Telephone>6650</Telephone>  
    <Telephone>915980830849</Telephone>  
  </DirectoryEntry>  
</YealinkIPPhoneDirectory>
```

Note

Yealink supplies a phone book generation tool to quickly generate a remote XML phone book. For more information, refer to *Yealink Phonebook Generation Tool User Guide*, available online:

<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

Troubleshooting

This chapter provides an administrator with general information for troubleshooting some common problems that he (or she) may encounter while using SIP-T4X 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 the Watch Dog Feature](#)
- [Getting Information from Status Indicators](#)
- [Analyzing Configuration Files](#)

Viewing Log Files

If your IP phone encounters some problems, commonly the log files are needed. You can export the log files to a syslog server or the local system. You can also specify the severity level of the log to be reported to a log file. The default system log level is 3.

In the configuration files, you can use the following parameters to configure system log settings:

- **syslog.mode** – Specify the system log to be exported to a server or local system.
- **syslog.server** -- Specify the IP address or domain name of the syslog server to which the log will be exported.
- **syslog.log_level** -- Specify the system log level. The following lists the log level of events you can log:

0: system is unusable

1: action must be taken immediately

2: critical condition

3: error conditions

4: warning conditions

5: normal but significant condition

6: informational

Procedure

Log setting can be configured using the configuration files or locally.

<p>Configuration File</p>	<p><y0000000000xx>.cfg</p>	<p>Configures the syslog mode.</p> <p>Parameters: syslog.mode</p> <p>Configures the IP address or domain name of the syslog server where to export the log files.</p> <p>Parameters: syslog.server</p> <p>Configures the severity level of the logs to be reported to a log file.</p> <p>Parameters: syslog.log_level</p>
<p>Local</p>	<p>Web User Interface</p>	<p>Configures the syslog mode.</p> <p>Configures the IP address or domain name of the syslog server where to export the log files.</p> <p>Configures the severity level of the logs to be reported to a log file.</p> <p>Navigate to: <a href="http://<phoneIPAddress>/servlet?p=settings-config&q=load">http://<phoneIPAddress>/servlet?p=settings-config&q=load</p>

Details of Configuration Parameters:

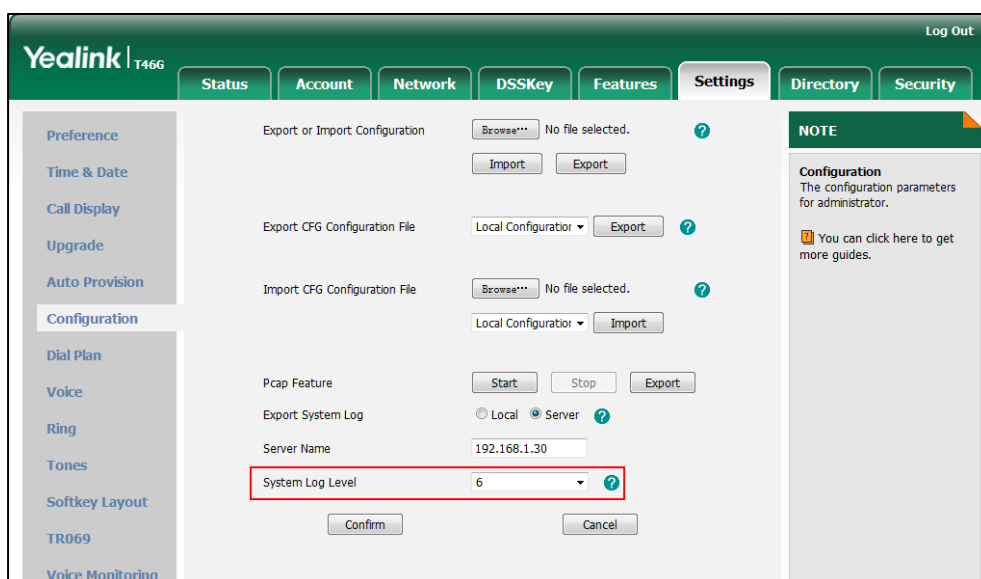
Parameters	Permitted Values	Default
syslog.mode	0 or 1	0
<p>Description: Configures the IP phone to export log files to a syslog server or the local system.</p> <p>0-Local</p>		

Parameters	Permitted Values	Default
<p>1-Server</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface: Settings->Configuration->Export System Log</p> <p>Phone User Interface: None</p>		
syslog.server	IP address or domain name	Blank
<p>Description: Configures the IP address or domain name of the syslog server when exporting log to the syslog server.</p> <p>Example: syslog.server = 192.168.1.30</p> <p>Note: It works only if the value of 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.</p> <p>Web User Interface: Settings->Configuration->Server Name</p> <p>Phone User Interface: None</p>		
syslog.log_level	Integer from 0 to 6	3
<p>Description: Configures the detail level of syslog information to be exported.</p> <p>0: system is unusable</p> <p>1: action must be taken immediately</p> <p>2: critical condition</p> <p>3: error conditions</p> <p>4: warning conditions</p> <p>5: normal but significant condition</p> <p>6: informational</p> <p>Note: If you change this parameter, the IP phone will reboot to make the change take effect.</p> <p>Web User Interface:</p>		

Parameters	Permitted Values	Default
Settings->Configuration->System Log Level		
Phone User Interface:		
None		

To configure the system log level 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.

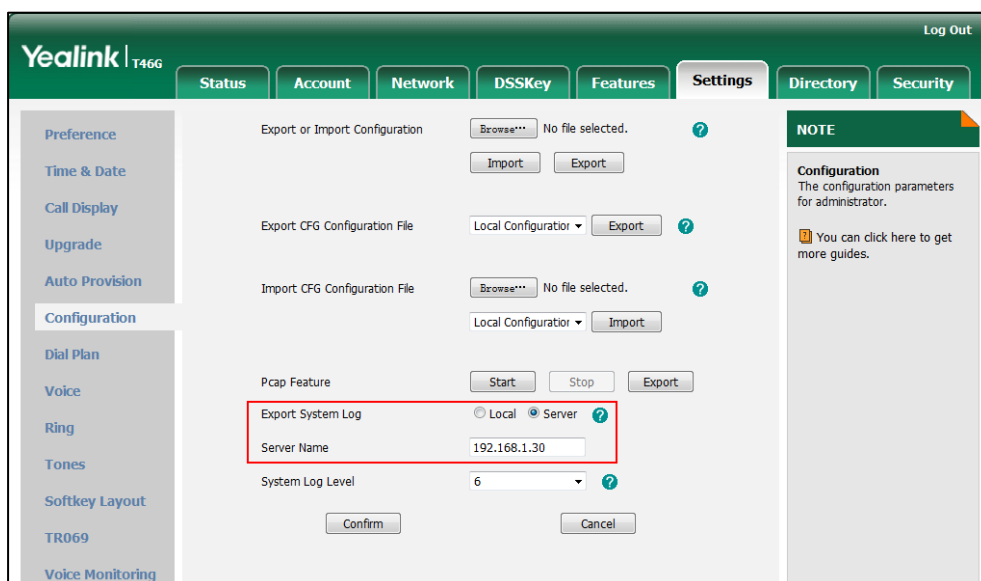
The system log level is set as 6, the informational level.

Note Informational level may make some sensitive information accessible (e.g., password-dial number), we recommend that you reset the system log level to 3 after providing the syslog file.

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.

- Enter the IP address or domain name of the syslog server in the **Server Name** field.



- 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.

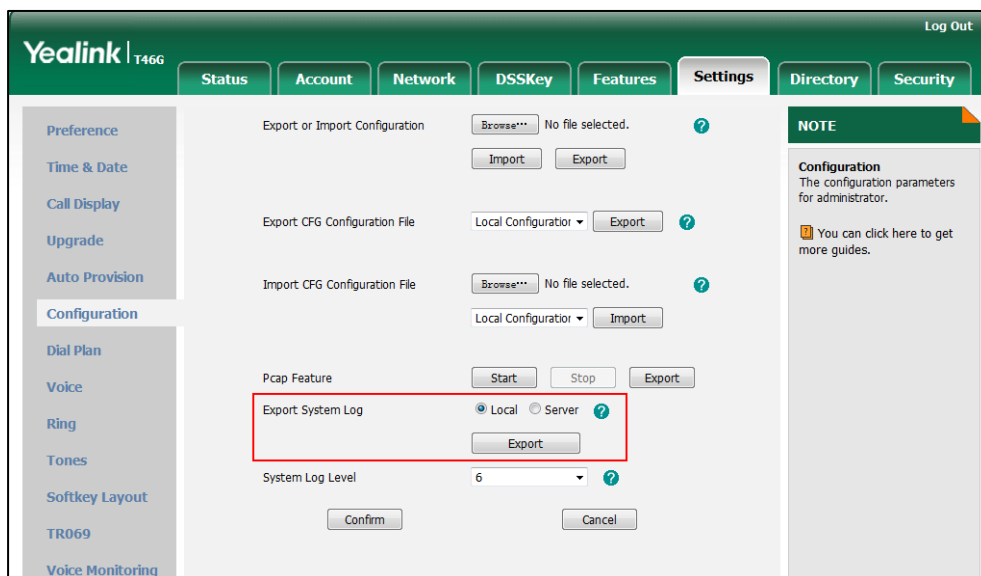
- Click **OK** to reboot the phone.

The system log will be exported successfully to the desired syslog server after a reboot.

- Reproduce the issue.

To export a log file to the local system via web user interface:

- Click on **Settings->Configuration**.
- Mark the **Local** radio box in the **Export System Log** field.
- 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:

```

190 root      2856 S      /bin/sh /boot/script/netapp.sh
197 root      22484 S     /boot/bin/rtServer.exx
210 root      2856 S      /usr/sbin/telnetd
211 root      17448 S     /boot/bin/autoServer.exx
249 root      3440 S      ./sbin/lighttpd -f /phone/bin/lighttpd/config/lighttpd
252 root      18924 S     /phone/www/WEB-INFO/bin/cgiServer.exx
263 root      2856 S      /sbin/syslogd -S -O /tmp/log/000000000000.log -s 200
275 root      2856 S      /bin/sh /phone/scripts/phoneapp.sh
276 root      6092 S      ./pcap.exx
291 root      140m S     /phone/bin/dskPhone.exx -qws
300 root          0 SW<     [ethTx/0]
301 root          0 SW<     [ethStatus/0]
309 root      5060 S      /boot/bin/lldpd
310 root      5572 S      /boot/bin/lldpd
357 root          0 DW      [hwthread]
358 root          0 DW      [hausioctl]
359 root          0 SW<     [frameProfiler]
360 root          0 DW<     [Cadence]
369 root      14016 S     /phone/bin/vaServer -q -w -m ANY=5
388 root      2920 S      /phone/bin/snmpd -c /etc/snmpd.conf
389 root      2856 S      /bin/sh /phone/scripts/sipapp.sh
396 root      41396 S N    /phone/bin/sipServer.exx
415 root      1628 S     /phone/bin/busybox udhcpc -b -i eth0 -a -s /boot/bin/
487 root      2856 S      sh -c cd /tmp;ifconfig >> log/000000000000.log;ps >>
489 root      3180 R      ps
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::write() Begin. size= 1;count=1024
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::commitHeader() Begin
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::commitHeader() End2
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::write() End.write 1024 bytes
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::write() Begin. size= 1;count=1024
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::commitHeader() Begin
Mar 12 03:32:58 fcgiServer.exx: HttpResponseImpl::commitHeader() End

```

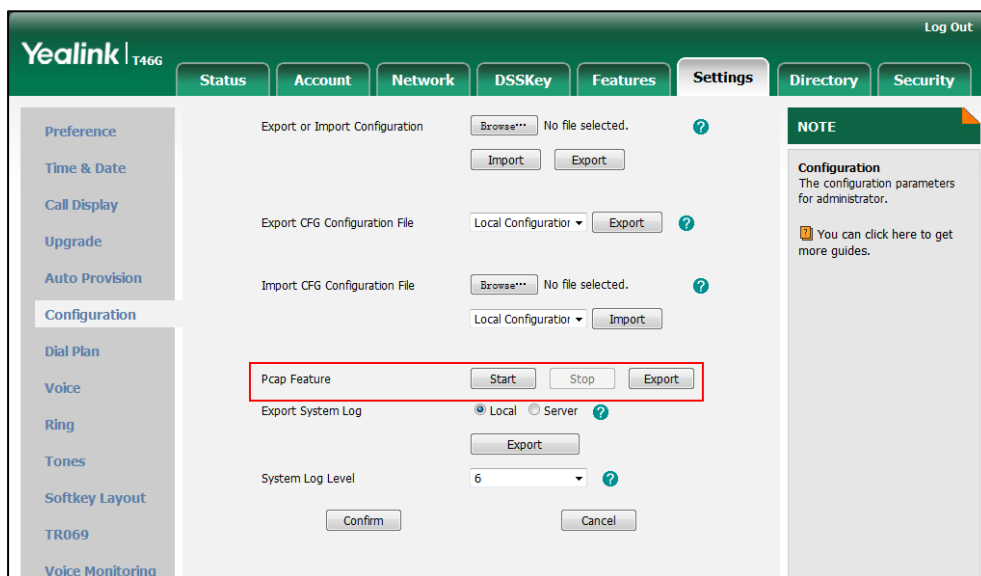
Capturing Packets

You can capture packets in two ways: capturing the packets via web user interface or using the Ethernet software. You can analyze the packets captured for troubleshooting 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.

- 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 the Watch Dog Feature

The IP phone provides a troubleshooting feature called “Watch Dog”, which helps you monitor the IP phone status and provides the ability to get stack traces from the last time the IP phone failed. 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 in the configuration files.

Procedure

Watch Dog can be configured using the configuration files or locally.

Configuration File	<y0000000000xx>.cfg	Configure Watch Dog feature. Parameter: watch_dog.enable
Local	Web User Interface	Configure Watch Dog feature. Navigate to: http://<phoneIPAddress>

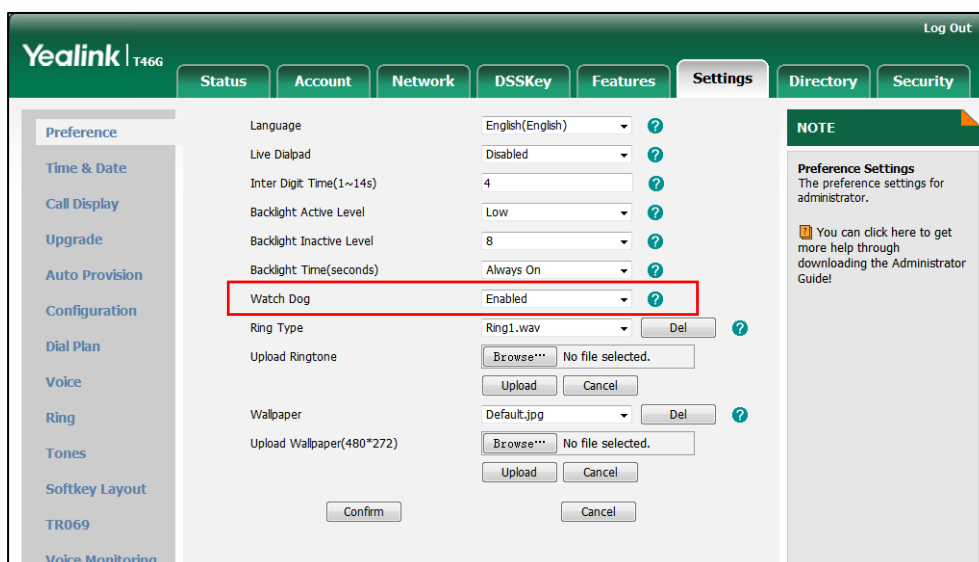
		/servlet?p=settings-preference&q=load
--	--	---------------------------------------

Details of the Configuration Parameter:

Parameter	Permitted Values	Default
watch_dog.enable	0 or 1	1
<p>Description : Enables or disables Watch Dog feature.</p> <p>0-Disabled 1-Enabled</p> <p>If it is set to 1 (Enabled), the IP phone will reboot automatically when the system is broken down.</p> <p>Web User Interface: Settings->Preference->Watch Dog</p> <p>Phone User Interface: None</p>		

To configure watch dog 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, line key indicator, headset key indicator, mute key indicator and the on-screen icon.

The following shows two examples of obtaining the IP phone information from status indicators on SIP-T46G IP phones:

- 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 an active call on the IP phone is on mute, the Mute key LED illuminates.

For more information on the icons, refer to [Reading Icons](#) on page 14.

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.

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 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 and 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 noise, 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 or 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 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 between 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 message is formed as below:

```
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 T4 Series IP Phones Auto Provisioning Guide, available online:
<http://www.yealink.com/DocumentDownload.aspx?CatId=142&flag=142>.

How to increase or decrease the volume?

You can press the volume key to increase or decrease the ringer volume and receiver volume. Press the volume key to adjust the ringer volume when the IP phone is idle, or to adjust the volume of the 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 use the PoE preferentially.

What is auto provisioning?

Auto provisioning refers to the update of IP phones, including update on configuration parameters, local phonebook, 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 configuration 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 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.

For anonymous call/anonymous call rejection feature, the phone will send either the on code or off code to the server according to the value of Send Anonymous Code/Send Rejection Code. For more information, refer to [Anonymous Call](#) on page 154 and [Anonymous Call Rejection](#) on page 158.

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->Advanced** (default password: admin) ->**Network->WAN Port->IPv4 (or IPv6)**. If the Static IP is selected, select DHCP instead.

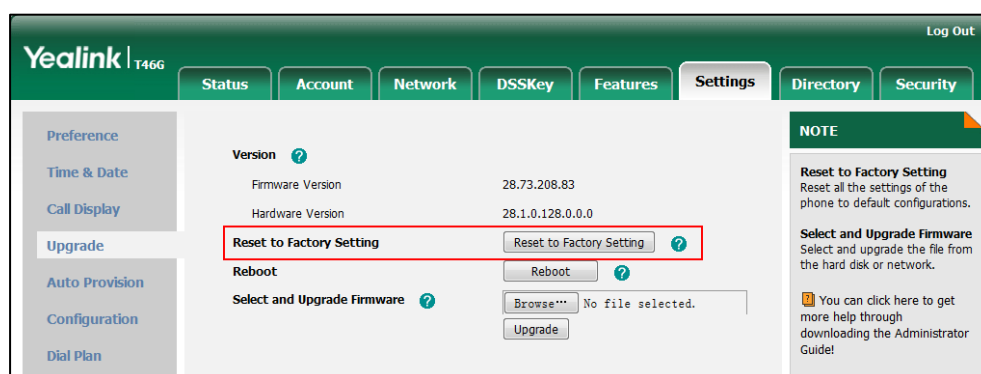
How to reset your phone to factory configurations?

Reset your phone to factory configurations after you have tried all troubleshooting suggestions but do not resolve the problem. Note that all custom settings will be overwritten after resetting.

To reset your phone via web user interface:

1. Click on **Settings->Upgrade**.
2. Click **Reset to Factory** 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 IP phone will be reset to factory successfully after startup.

Note Reset of the phone may take a few minutes. Do not power off until the IP 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.

Analyzing Configuration Files

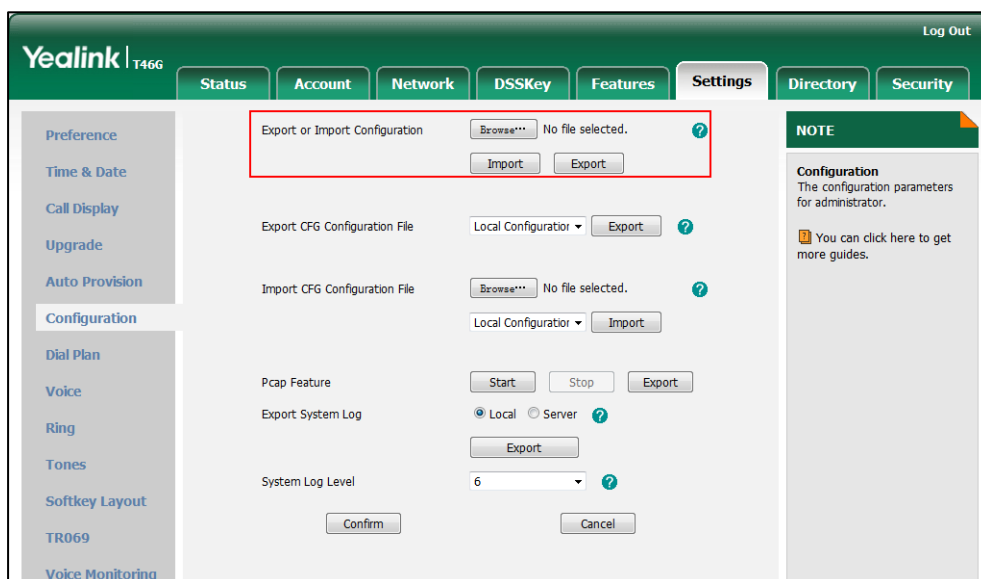
Wrong configurations may have an impact on your phone use. You can export configuration files to check the current configuration of the IP phone and troubleshoot if necessary. You can also import configuration files for a quick and easy configuration.

Three types of configuration files can be exported to your local system: config.bin, <mac>-all.cfg and <mac>-local.cfg. The <mac>-all.cfg configuration file contains all changes made via phone user interface, web user interface and using configuration files. The <mac>-local.cfg configuration file contains changes made via phone user interface and web user interface. The config.bin file is an encrypted file. For more information on config.bin file, contact your Yealink reseller.

To export a BIN configuration file via web user interface:

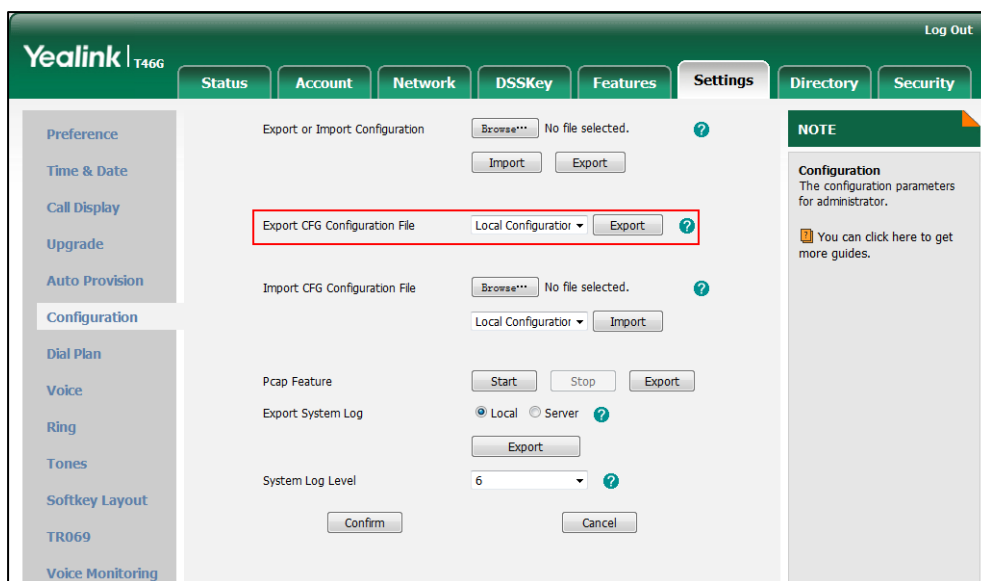
1. Click on **Settings->Configuration**.

- In the **Export or Import Configuration** block, click **Export** to open the file download window, and then save the file to your local system.



To export CFG configuration files via web user interface:

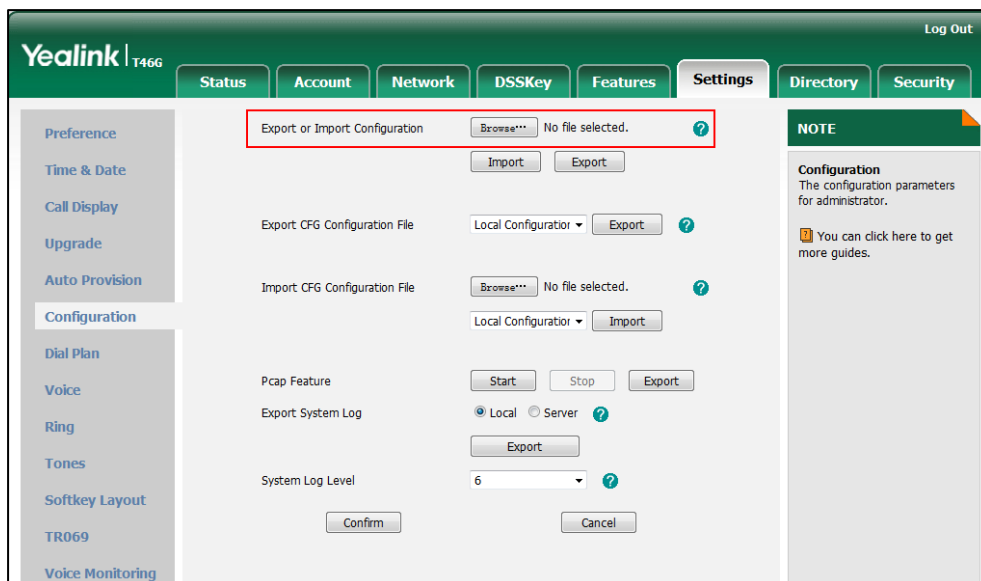
- Click on **Settings->Configuration**.
- Select **Local Configuration** or **All Configuration** from the pull-down list of **Export CFG Configuration File** and then click **Export** to save the file to your local system.



To import a BIN configuration file via web user interface:

- Click on **Settings->Configuration**.

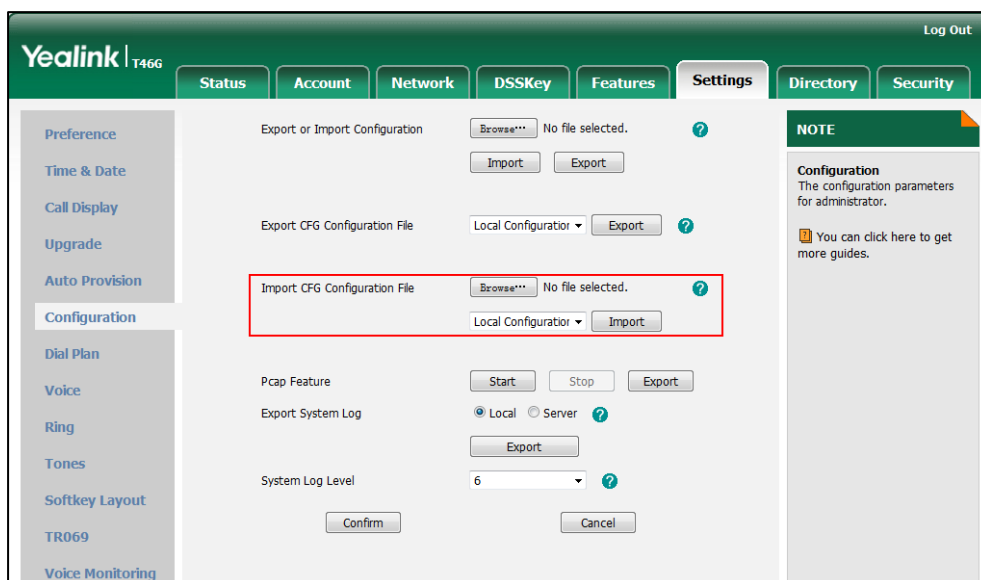
- In the **Export or Import Configuration** block, click **Browse** to locate a BIN configuration file from your local system.



- Click **Import** to import the configuration file.

To import CFG configuration files via web user interface:

- Click on **Settings->Configuration**.
- In the **Import CFG Configuration File** block, click **Browse** to locate a CFG configuration file from your local system.



- Click **Import** to import the configuration file.

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 offers an authentication mechanism for devices to connect 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
-09:30	French Polynesia
-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)
-02:30	Newfoundland and Labrador
-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)

Time Zone	Time Zone Name
0	Spain-Canary Islands(Las Palmas)
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)

Time Zone	Time Zone Name
+04:00	Russia(Samara)
+04:30	Afghanistan(Kabul)
+05:00	Kazakhstan(Aqtobe)
+05:00	Kyrgyzstan(Bishkek)
+05:00	Pakistan(Islamabad)
+05:00	Russia(Chelyabinsk)
+05:30	India(Calcutta)
+05:45	Nepal(Katmandu)
+06:00	Kazakhstan(Astana, Almaty)
+06:00	Russia(Novosibirsk, Omsk)
+06:30	Myanmar(Naypyitaw)
+07:00	Russia(Krasnoyarsk)
+07:00	Thailand(Bangkok)
+08:00	China(Beijing)
+08:00	Singapore(Singapore)
+08:00	Australia(Perth)
+08:00	Russian(Irkutsk, Ulan-Ude)
+08:45	Eucla
+09:00	Korea(Seoul)
+09:00	Japan(Tokyo)
+09:00	Russian(Yakutsk, Chita)
+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)
+11:00	Russia(Srednekolymsk Time)
+11:30	Norfolk Island
+12:00	New Zealand(Wellington, Auckland)
+12:00	Russian(Kamchatka Time)
+12:45	New Zealand(Chatham Islands)
+13:00	Tonga(Nukualofa)
+13:30	Chatham Islands
+14:00	Kiribati

Appendix C: Configuring DSS Key

This appendix describes the DSS key parameters you can configure on IP phones. DSS keys can be assigned with various key features. The DSS key consists of line key and programable key. The following table lists the number of DSS keys you can configure for each phone model:

Parameter- linekey.X.type	Configuration File <y0000000000xx>.cfg
Parameter- programablekey.X.type	
Description	<p>Configures the key feature for the DSS key.</p> <p>For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>For line keys: Valid types are:</p> <ul style="list-style-type: none"> • N/A • Conference • Forward • Transfer • Hold • DND • ReCall • SMS (not applicable to SIP-T48G/T42G/T41P) • Directed Pickup • Call Park • DTMF • Voice Mail • Speed Dial • Intercom

	<ul style="list-style-type: none">• Line• BLF• URL• Group Listening• Hot Desking• XML Group• Group Pickup• Multicast Paging• Record• XML Browser• URL Record• LDAP• Prefix• Zero Touch• ACD• Local Group• Phone Lock(not applicable to SIP-T48G)• Directory <p>For programable keys:</p> <p>Valid types are:</p> <ul style="list-style-type: none">• N/A• Forward• DND• ReCall• SMS (not applicable to SIP-T48G/T42G/T41P)• Directed Pickup• Speed Dial• XML Group• Group Pickup• XML Browser• History• Menu• New SMS (not applicable to
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	<p>SIP-T48G/T42G/T41P)</p> <ul style="list-style-type: none"> • Status • Hot Desking (only applicable to SIP-T48G/T46G) • LDAP • Prefix • Local Directory • Local Group • XML Directory • Phone Lock (not applicable to SIP-T48G) • Switch Account Up • Switch Account Down • Directory
Format	Integer
Default Value	<p>For line keys:</p> <p>For SIP-T46G IP phones: The default value of the line key 1-16 is 15, and the default value of the line key 17-27 is 0.</p> <p>For SIP-T42G IP phones: The default value of the line key 1-12 is 15, and the default value of the line key 13-15 is 0.</p> <p>For SIP-T41P IP phones: The default value of the line key 1-6 is 15, and the default value of the line key 7-15 is 0.</p> <p>For programable keys:</p> <p>For SIP-T46G IP phones: When X=1, the default value is 28 (History). When X=2, the default value is 61 (Directory). When X=3, the default value is 5 (DND). When X=4, the default value is 30 (Menu). When X=5, the default value is 28 (History). When X=6, the default value is 61</p>

	<p>(Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=12, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p> <p>When X=14, the default value is 2 (Forward).</p> <p>For SIP-T42G/T41P IP phones:</p> <p>When X=1, the default value is 28 (History).</p> <p>When X=2, the default value is 61 (Directory).</p> <p>When X=3, the default value is 5 (DND).</p> <p>When X=4, the default value is 30 (Menu).</p> <p>When X=5, the default value is 28 (History).</p> <p>When X=6, the default value is 61 (Directory).</p> <p>When X=7, the default value is 51 (Switch Account Up).</p> <p>When X=8, the default value is 52 (Switch Account Down).</p> <p>When X=9, the default value is 33 (Status).</p> <p>When X=10, the default value is 0 (NA).</p> <p>When X=13, the default value is 0 (NA).</p>
<p>Range</p>	<p>Valid values are:</p> <p>0-N/A</p> <p>1-Conference</p> <p>2-Forward</p> <p>3-Transfer</p> <p>4-Hold</p> <p>5-DND</p> <p>7-ReCall</p> <p>8-SMS (not applicable to SIP-T48G/T42G/T41P)</p> <p>9-Directed Pickup</p>

	<p>10-Call Park</p> <p>11-DTMF</p> <p>12-Voice Mail</p> <p>13-SpeedDial</p> <p>14-Intercom</p> <p>15-Line</p> <p>16-BLF</p> <p>17-URL</p> <p>18-Group Listening</p> <p>22-XML Group</p> <p>23-Group Pickup</p> <p>24-Multicast Paging</p> <p>25-Record</p> <p>27-XML browser</p> <p>28-History</p> <p>30-Menu</p> <p>32-New SMS (not applicable to SIP-T48G/T42G/T41P)</p> <p>33-Status</p> <p>34-Hot Desking</p> <p>35-URL Record</p> <p>38-LDAP</p> <p>40-Prefix</p> <p>41-Zero Touch</p> <p>42-ACD</p> <p>43-Local Directory</p> <p>45-Local Group</p> <p>47-XML Directory</p> <p>50-Phone Lock (not applicable to SIP-T48G)</p> <p>51-Switch Account Up</p> <p>52-Switch Account Down</p> <p>61-Directory</p>
Example	linekey.1.type = 15

Parameter- linekey.X.line	Configuration File <y0000000000xx>.cfg
Parameter- programablekey.X.line	
Description	<p>Configures the desired line to apply the key feature.</p> <p>For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>When assigning the following features, you don't need to configure this parameter:</p> <ul style="list-style-type: none"> • DTMF • Prefix • XML Browser • LDAP • Conference • Forward • Hold • DND • ReCall • SMS (not applicable to SIP-T48G/T42G/T41P) • Record • URL Record • Multicast Paging • Group Listening • Local Group • XML Group • Zero Touch • URL

	<ul style="list-style-type: none"> • ACD • Hot Desking • Phone Lock • Directory
Format	Integer
Default Value	<p>For programable keys, the default value is not applicable.</p> <p>For line keys, when x=1, the default value is 1.</p> <p>When x=2, the default value is 2.</p> <p>...</p> <p>When x=6, the default value is 6.</p>
Range	<p>Valid values are:</p> <p>1 to 16 (for SIP-T48G/T46G)</p> <p>1 to 12 (for SIP-T42G)</p> <p>1 to 6 (for SIP-T41P)</p> <p>1-Line 1</p> <p>2-Line 2</p> <p>...</p> <p>16-Line 16</p>
Example	linekey.1.line = 2

Parameter- linekey.X.value	Configuration File <y0000000000xx>.cfg
Parameter- programablekey.X.value	
Description	<p>Configures the value for some key features.</p> <p>For line keys:</p> <p>X ranges from 1 to 29 (for SIP-T48G).</p> <p>X ranges from 1 to 27 (for SIP-T46G).</p> <p>X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys:</p> <p>X=1-10, 12-14 (for SIP-T48G/T46G)</p> <p>X=1-10, 13 (for SIP-T42G/T41P)</p>
Format	String
Default Value	Blank

Range	String within 99 characters
Example	When you assign the Speed Dial to the line key, this parameter is used to specify the number you want to dial out. linekey.1.value = 1001

Parameter- linekey.X.label	Configuration File <y0000000000xx>.cfg
Parameter- programablekey.X.label	
Description	Configures the label displaying on the LCD screen for each line key and each soft key. This is an optional configuration. For line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P). For programable keys, x ranges from 1 to 4.
Format	String
Default Value	Blank
Range	String within 99 characters
Example	linekey.1.label = Dir

Parameter- linekey.X.pickup_value	Configuration File <y0000000000xx>.cfg
Description	Configures the pickup code for BLF feature. This parameter is only applicable to BLF feature. X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).
Format	String
Default Value	Blank
Range	String within 256 characters

Example	linekey.1.pickup_value = *88
----------------	------------------------------

Parameter- linekey.X.xml_phonebook	Configuration File <y0000000000xx>.cfg
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 line keys: X ranges from 1 to 29 (for SIP-T48G). X ranges from 1 to 27 (for SIP-T46G). X ranges from 1 to 15 (for SIP-T42G/T41P).</p> <p>For programable keys: X=1-10, 12-14 (for SIP-T48G/T46G) X=1-10, 13 (for SIP-T42G/T41P)</p> <p>When the key feature is configured as Local Group, valid values are:</p> <p>0-All contacts 1-First local group ... 48-Forty-eighth 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 1-Second XML group ... 4-Fifth XML group</p>
Format	Integer
Default Value	0
Range	0 to 48
Example	Configures the second remote phone

	book. linekey.1.xml_phonebook = 1
--	--------------------------------------

Appendix D: SIP (Session Initiation Protocol)

This section describes how Yealink SIP-T4X 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 2976—The SIP INFO Method
- 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 3863—Presence Information Data Format
- 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 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
- RFC 5763—Framework for Establishing a Secure Real-time Transport Protocol (SRTP)
- RFC 5806—Diversion Indication in SIP
- 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-00.txt—SIP Extensions for Caller Identity and Privacy, November
- 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
- draft-ietf-sip-connect-reuse-06.txt—Connection Reuse in the Session Initiation Protocol (SIP)
- draft-ietf-bliss-shared-appearances-15.txt—Shared Appearances of a Session Initiation Protocol (SIP) Address of Record (AOR)
- draft-anil-sipping-bla-04.txt—Implementing Multiple Line Appearances using the Session Initiation Protocol (SIP)

To find the applicable Request for Comments (RFC) document, go to <http://www.ietf.org/rfc.html> and enter the RFC number.

SIP Request

The following SIP request messages are supported:

Method	Supported	Notes
REGISTER	Yes	
INVITE	Yes	Yealink SIP-T4X IP phones support mid-call changes such as putting a call on hold as signaled by a new INVITE that contains an existing Call-ID.
ACK	Yes	
CANCEL	Yes	
BYE	Yes	
OPTIONS	Yes	
SUBSCRIBE	Yes	
NOTIFY	Yes	
REFER	Yes	
PRACK	Yes	
INFO	Yes	
MESSAGE	Yes	
UPDATE	Yes	
PUBLISH	Yes	

SIP Header

The following SIP request headers are supported:

Method	Supported	Notes
Accept	Yes	
Alert-Info	Yes	
Allow	Yes	
Allow-Events	Yes	
Authorization	Yes	
Call-ID	Yes	
Call-Info	Yes	
Contact	Yes	
Content-Length	Yes	
Content-Type	Yes	
CSeq	Yes	
Diversion	Yes	
Event	Yes	
Expires	Yes	
From	Yes	
Max-Forwards	Yes	
Min-SE	Yes	
P-Asserted-Identity	Yes	
P-Preferred-Identity	Yes	
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	
RAck	Yes	
Record-Route	Yes	
Refer-To	Yes	
Referred-By	Yes	
Remote-Party-ID	Yes	
Replaces	Yes	

Method	Supported	Notes
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	

3xx Response	Supported	Notes
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	
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	

4xx Response	Supported	Notes
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

SDP Headers	Supported
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

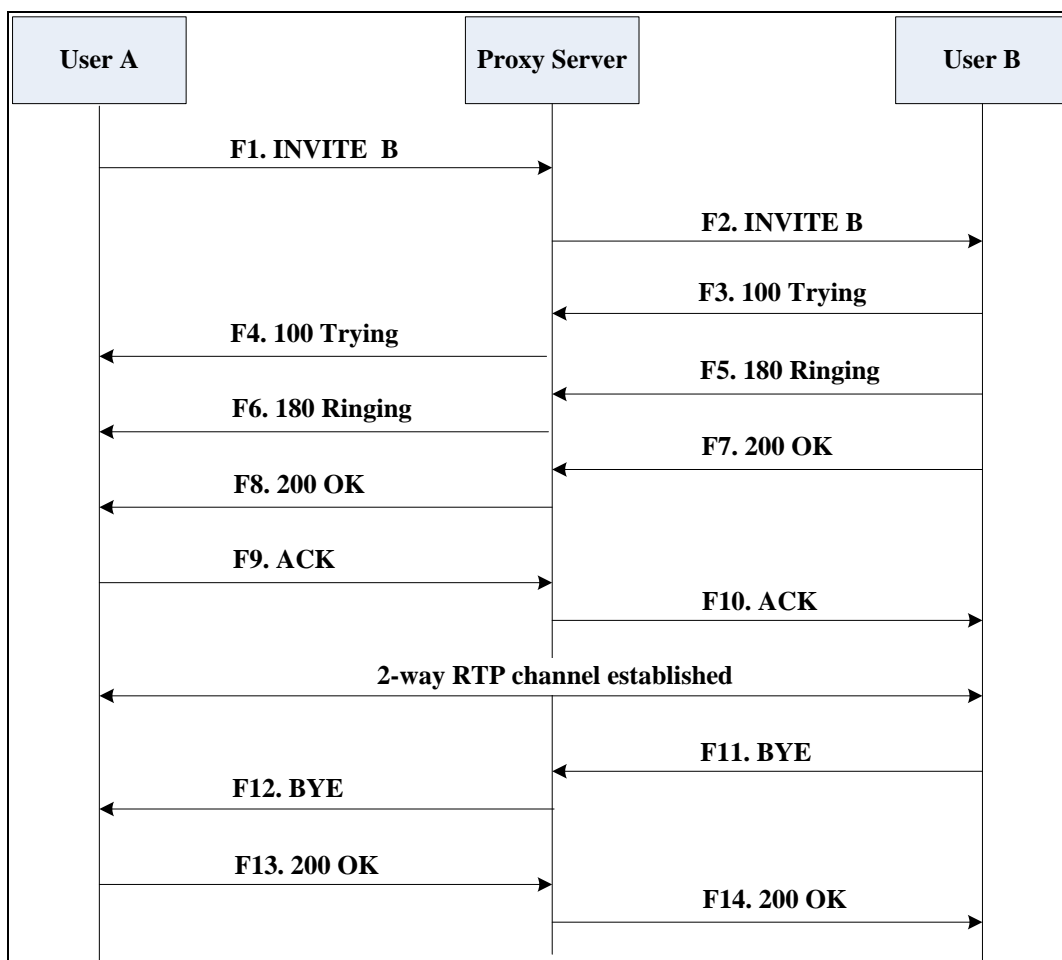
Successful Call Setup and Disconnect

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 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.

Step	Action	Description
		<ul style="list-style-type: none"> 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 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

Step	Action	Description
		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.

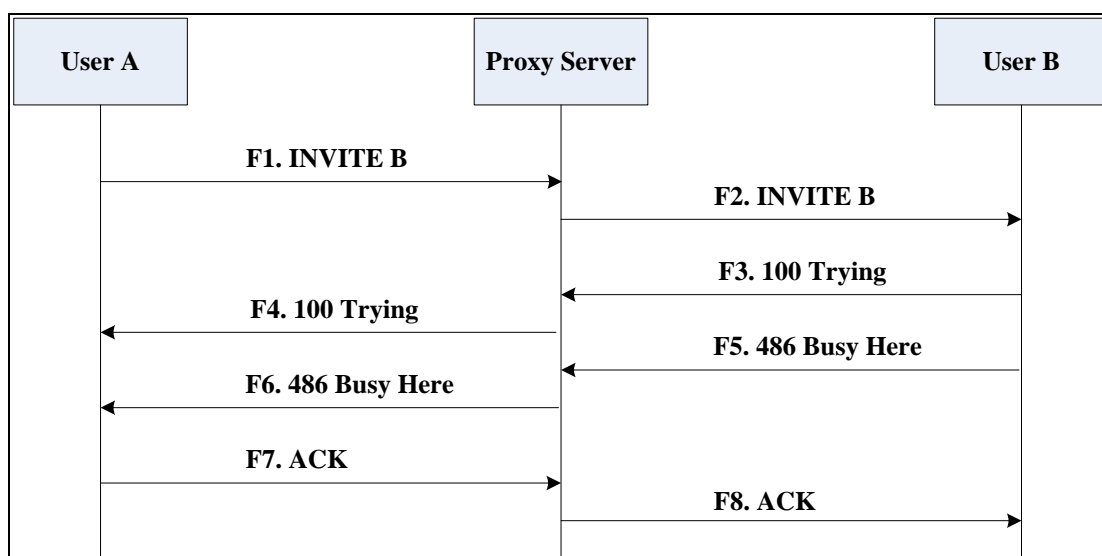
Unsuccessful Call Setup—Called User is Busy

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 specified.
F2	INVITE—Proxy Server to User B	<p>The proxy server maps the SIP URI in the To field to User B. Proxy server forwards the INVITE message to User B.</p>

Step	Action	Description
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.

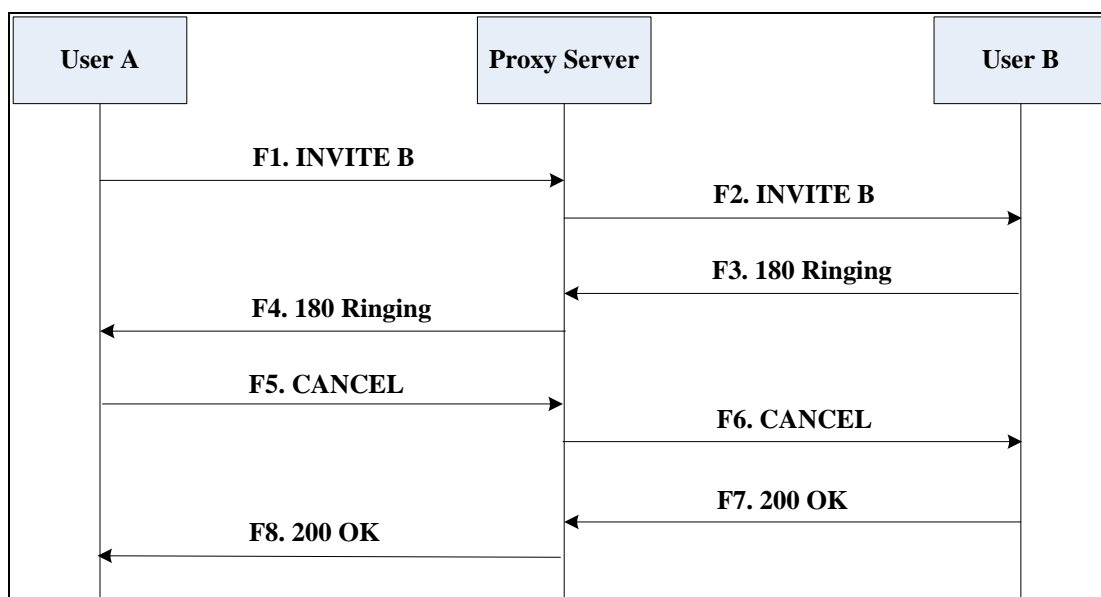
Unsuccessful Call Setup—Called User Does Not Answer

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

Step	Action	Description
		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 User B	The proxy server forwards the SIP CANCEL request to notify User B that 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.

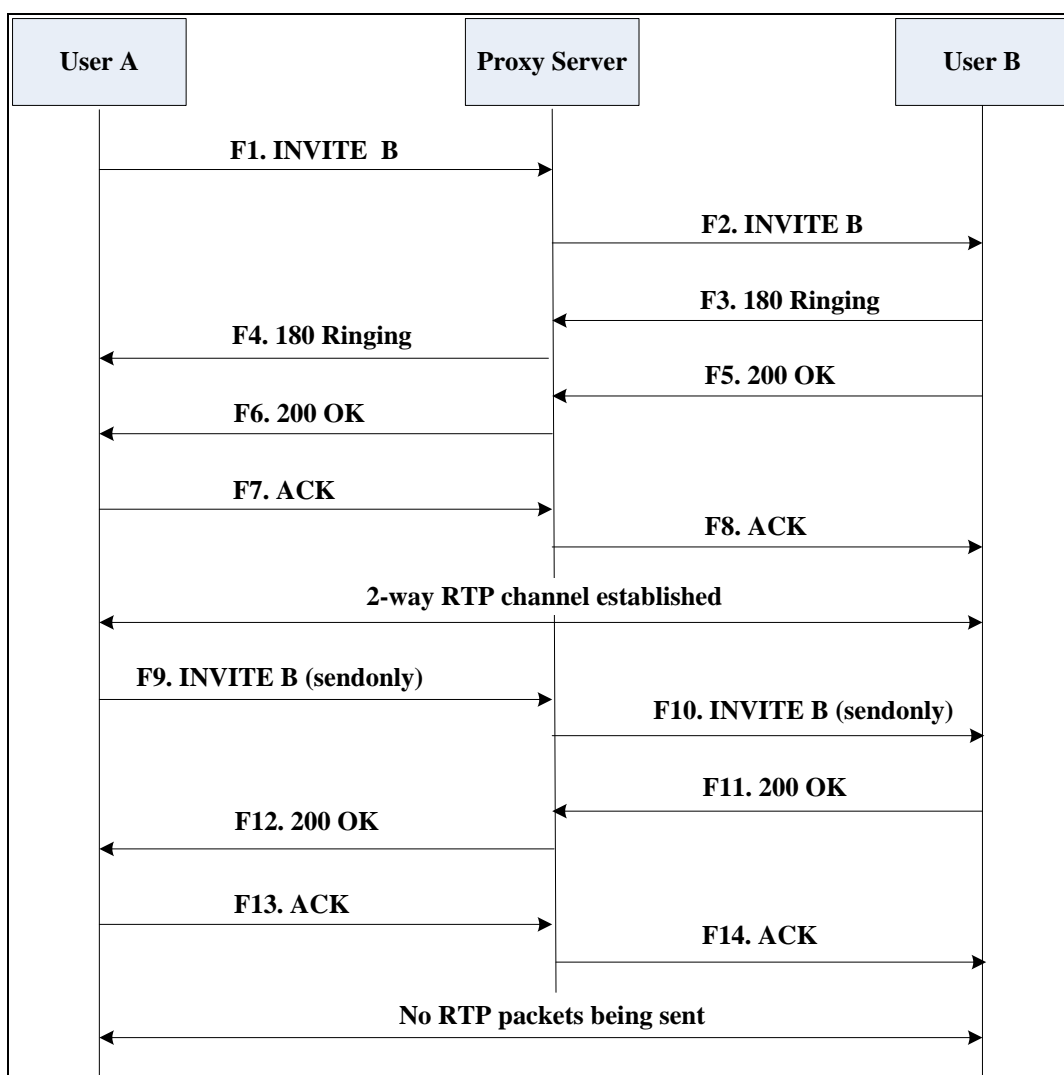
Successful Call Setup and Call Hold

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 puts 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.

Step	Action	Description
		<ul style="list-style-type: none"> 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.
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

Step	Action	Description
		place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE is successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.

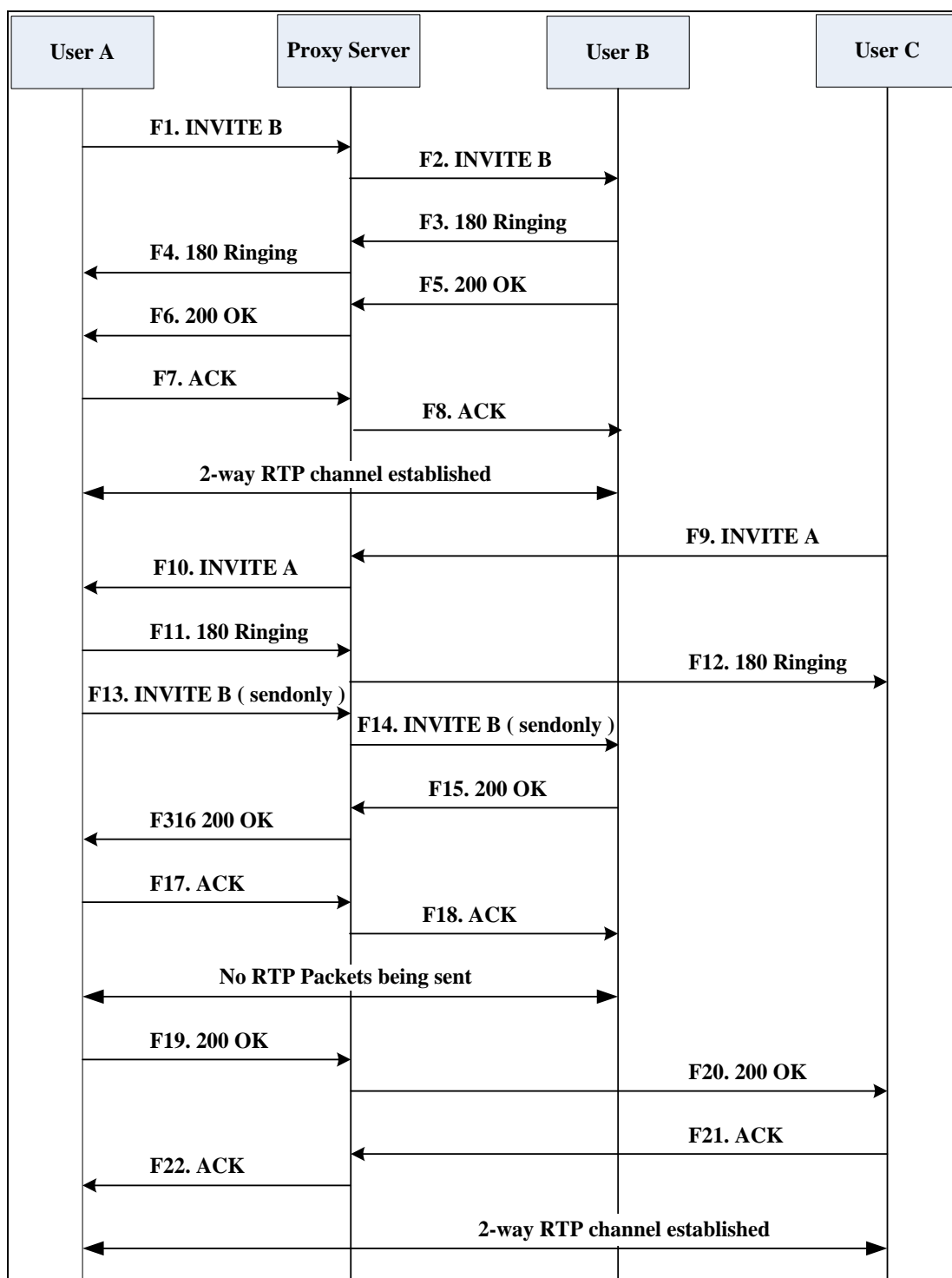
Successful Call Setup and Call Waiting

The following figure illustrates a successful call between Yealink SIP IP phones in which 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	User A sends an INVITE message to a proxy server. The INVITE request is an invitation to User B to participate in a

Step	Action	Description
		<p>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.
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server, The ACK confirms that User A

Step	Action	Description
		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.
F13	INVITE—User A to Proxy	User A sends a mid-call INVITE request

Step	Action	Description
	Server	to the proxy server with new SDP session parameters, which are used to place the call on hold.
F14	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F15	200 OK—User B to Proxy Server	User B sends a 200 OK to the proxy server. The 200 OK response indicates that the INVITE was successfully processed.
F16	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F17	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F18	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F19	200 OK—User A to Proxy Server	User A sends a 200 OK response to the proxy server. The 200 OK response notifies that the connection has been made.
F20	200 OK—Proxy Server User C	The proxy server forwards the 200 OK message to User C.
F21	ACK—User C to Proxy Server	User C sends a SIP ACK to the proxy server. The ACK confirms that User C has received the 200 OK response. The call session is now active.
F22	ACK—Proxy Server to User A	The proxy server forwards the SIP ACK to User A to confirm that User C has received the 200 OK response.

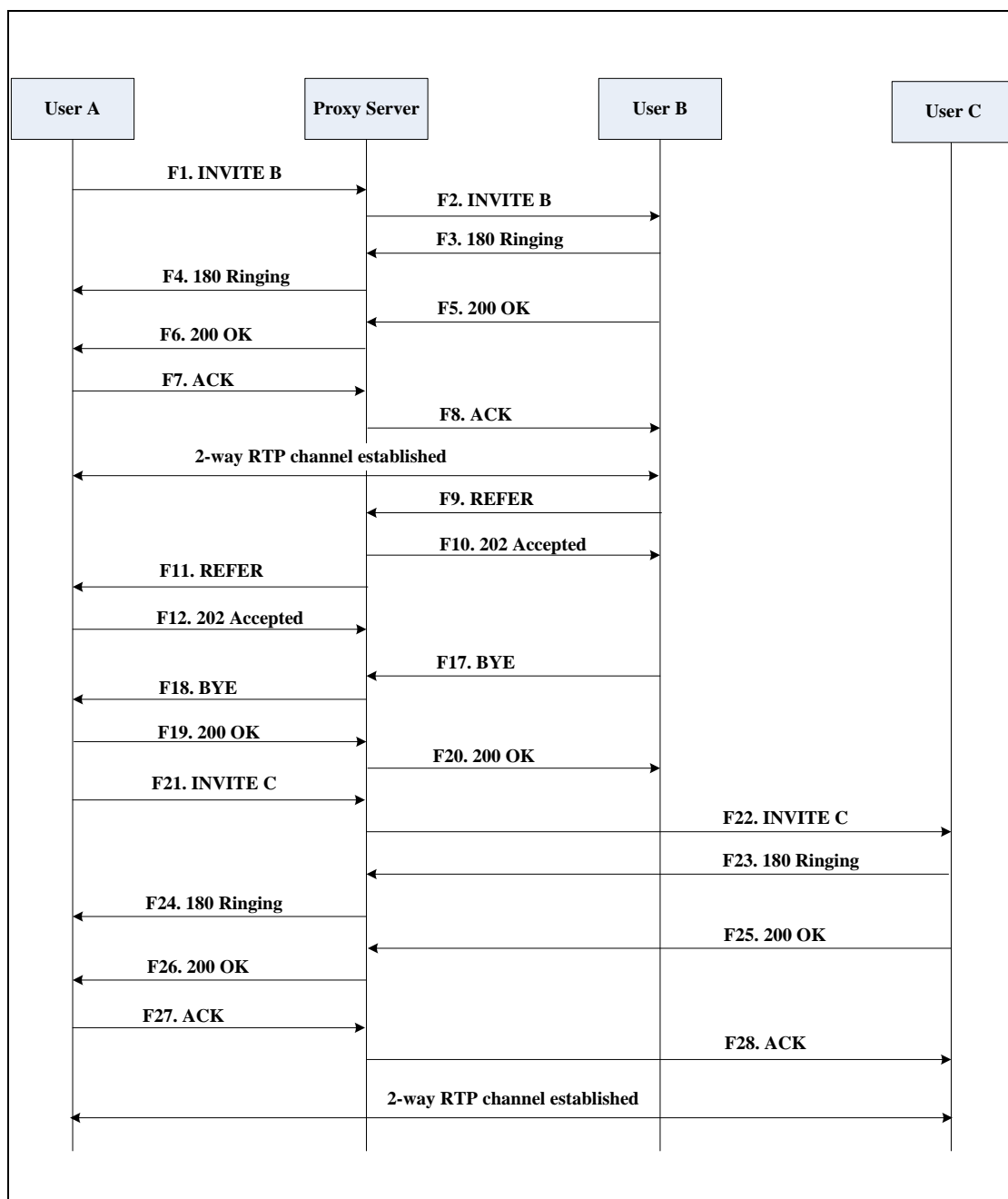
Call Transfer without Consultation

The following figure illustrates a successful call between Yealink SIP IP phones in which two parties are in a call and then one of the parties transfers the call to a third party without 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

Step	Action	Description
		<p>in the Request-URI field.</p> <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	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.
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

Step	Action	Description
		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 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

Step	Action	Description
		Ringling response indicates that the user is being alerted.
F20	180 Ringling—Proxy Server to User A	The proxy server forwards the 180 Ringling 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.

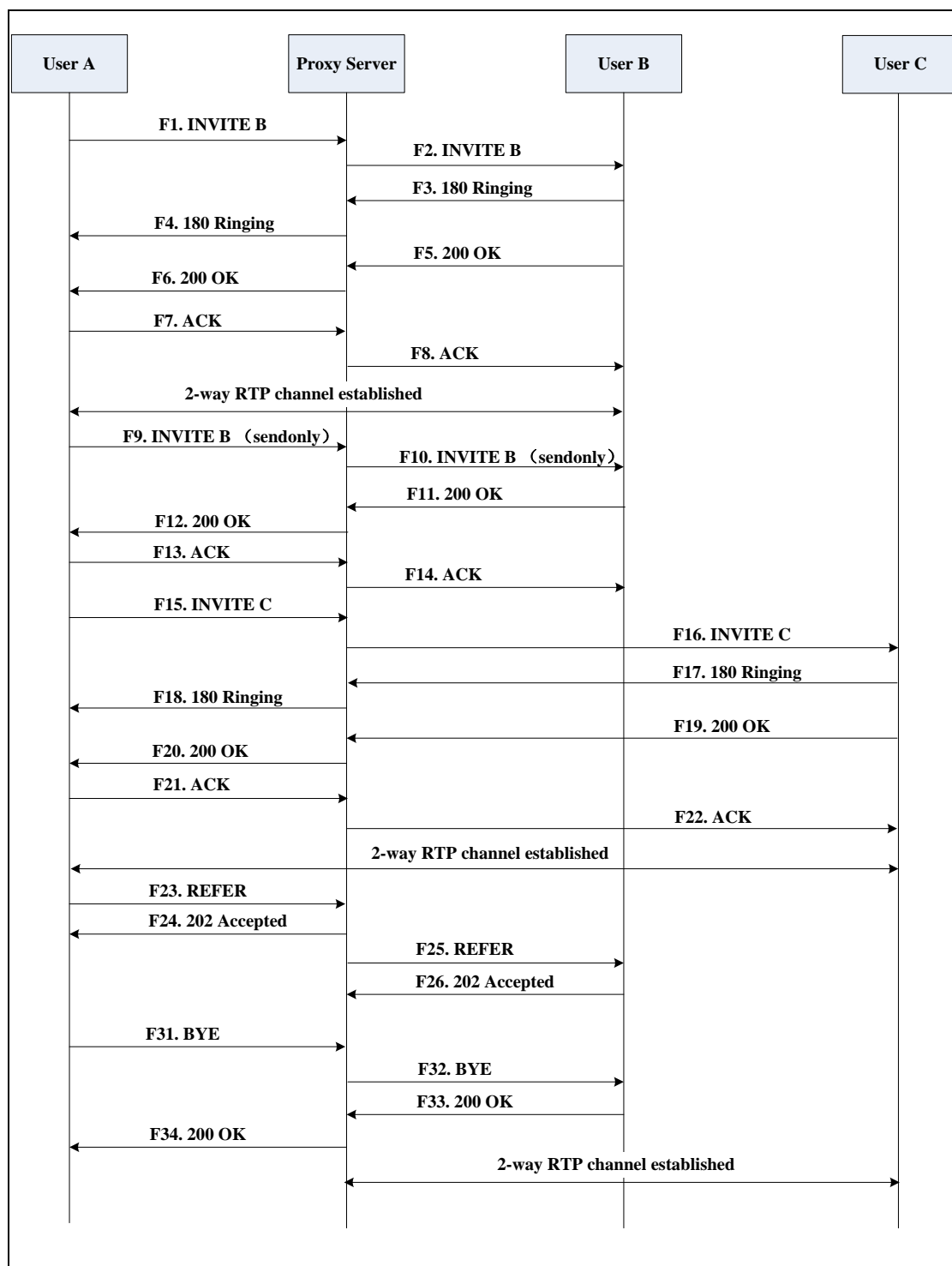
Call Transfer with Consultation

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

Step	Action	Description
		connection has been made.
F7	ACK—User A to Proxy Server	User A sends a SIP ACK to the proxy server, The ACK confirms that User A has received the 200 OK response. The call session is now active.
F8	ACK—Proxy Server to User B	The proxy server sends the SIP ACK to User B. The ACK confirms that the proxy server has received the 200 OK response. The call session is now active.
F9	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE was successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends an ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.

Step	Action	Description
F16	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 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.

Step	Action	Description
F26	202 Accepted—User B to Proxy Server	User B sends a SIP 202 Accept response to the proxy server. The 202 Accepted 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

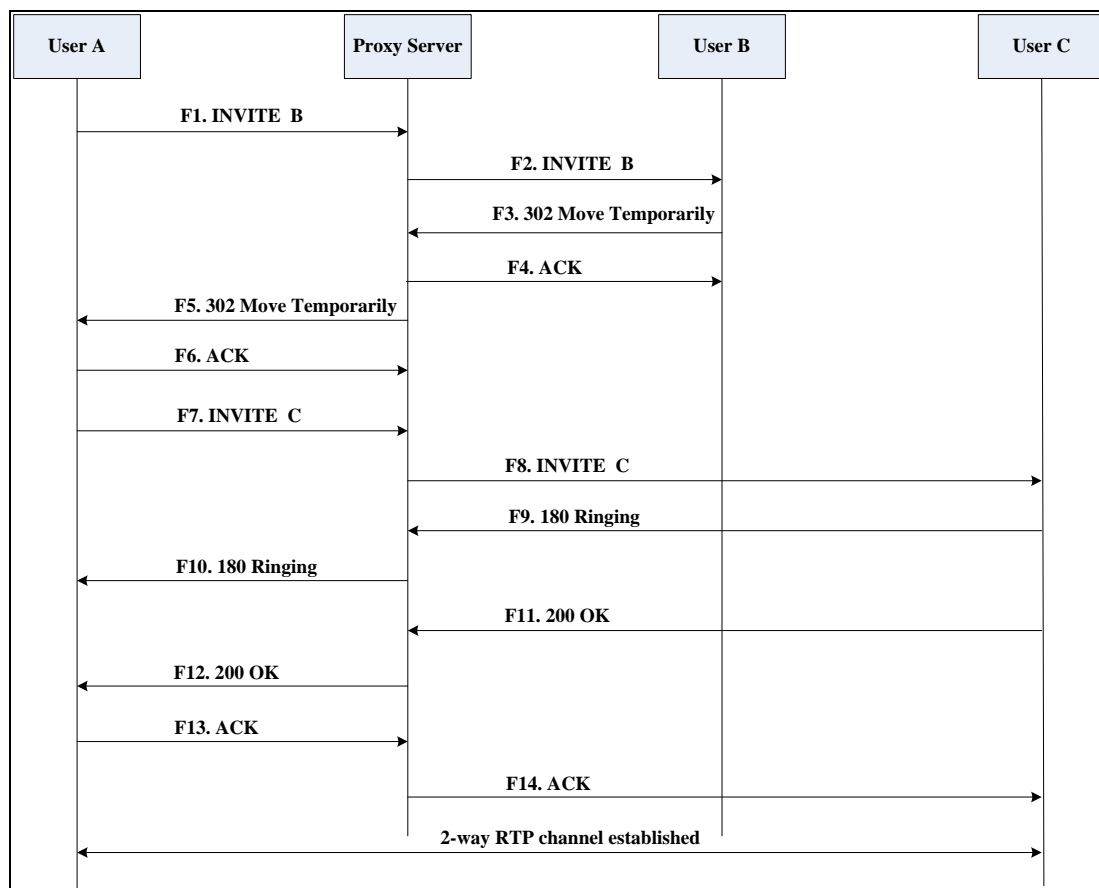
The following figure illustrates successful call forwarding between Yealink SIP IP phones in which User B has enabled always call forward. The incoming call is immediately forwarded to User C when User A calls User B. In this call flow scenario, the end users are User A, User B, and User C. They are 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

Step	Action	Description
		<p>CSeq field.</p> <ul style="list-style-type: none"> 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.
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

Step	Action	Description
		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.

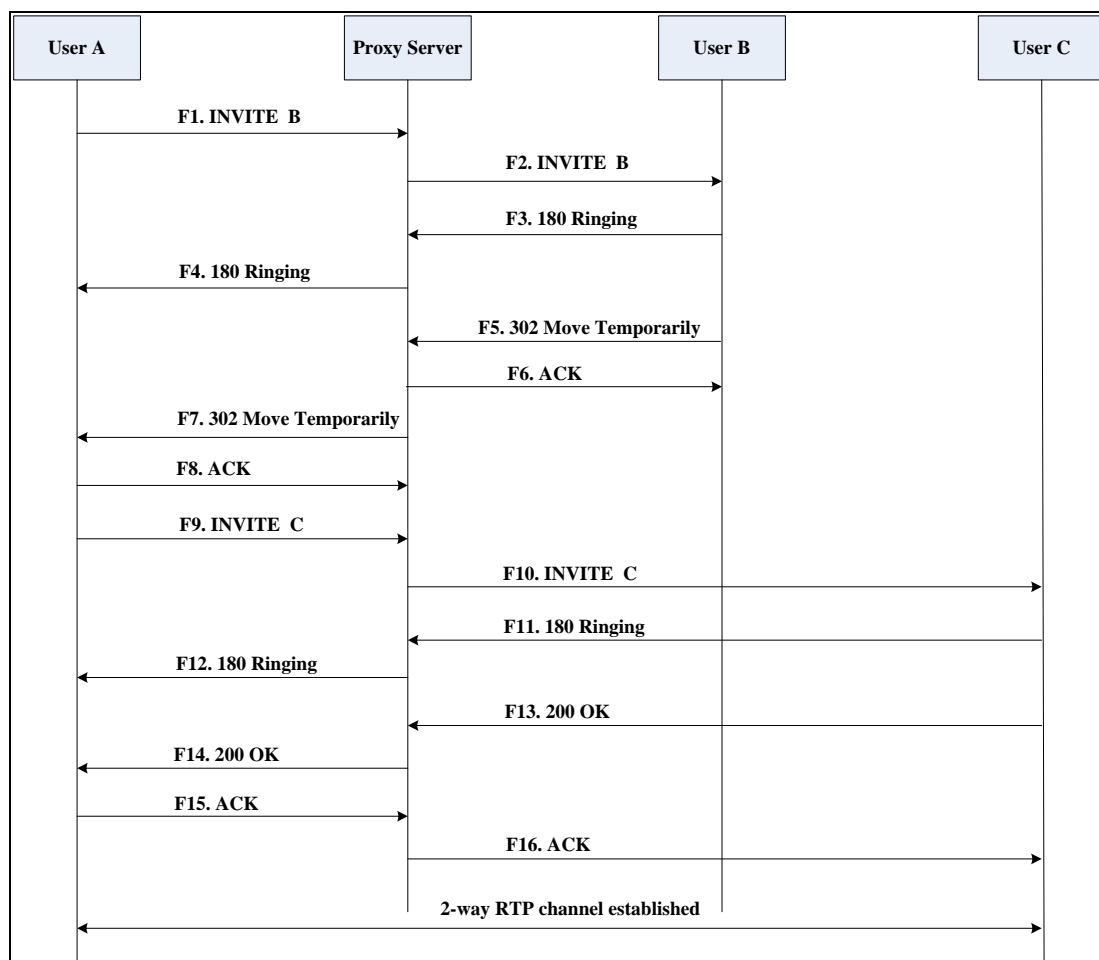
Busy Call Forward

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.

Step	Action	Description
		<ul style="list-style-type: none"> 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 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.

Step	Action	Description
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.

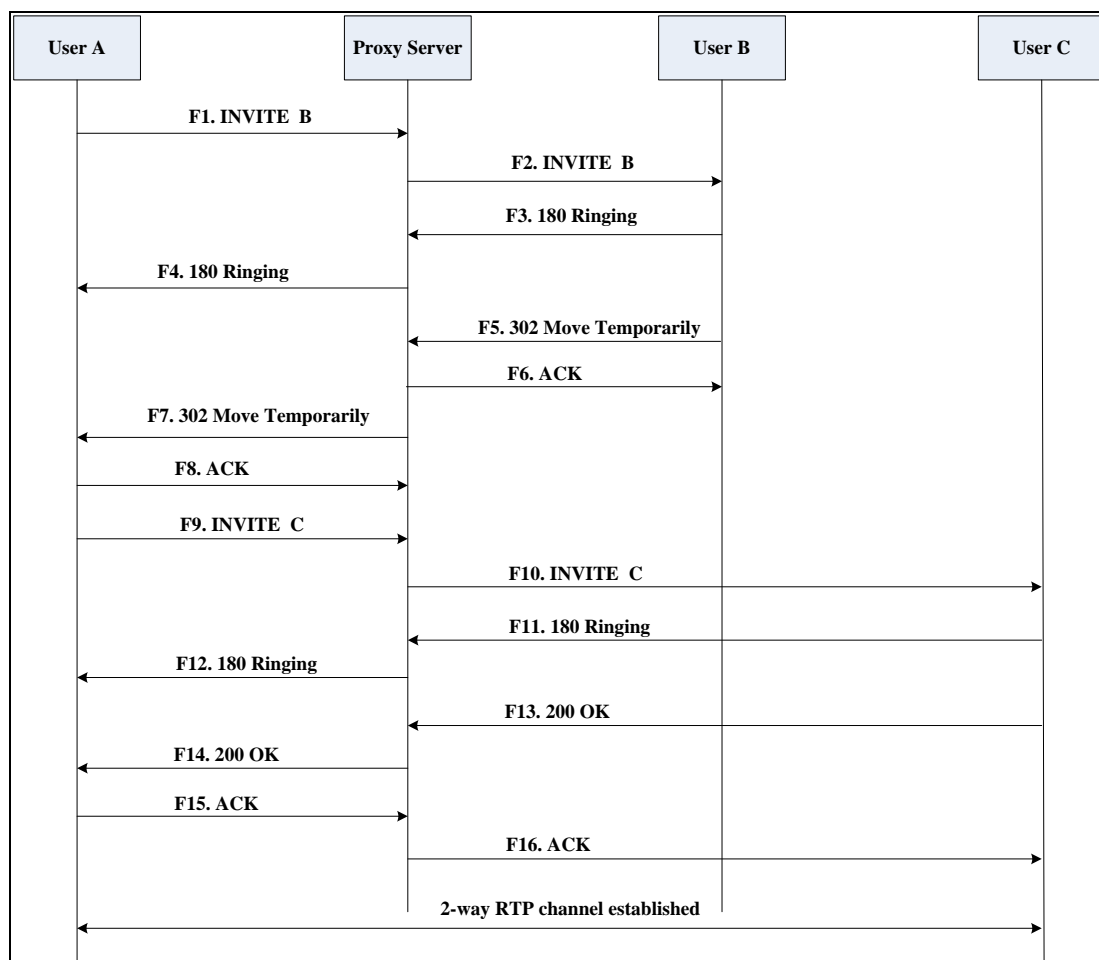
No Answer Call Forward

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.

Step	Action	Description
		<ul style="list-style-type: none"> 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 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.

Step	Action	Description
F11	180 Ringing—User C to Proxy Server	User C sends a SIP 180 Ringing response to the proxy server. The 180 Ringing response indicates that the user is being alerted.
F12	180 Ringing—Proxy Server to User A	The proxy server forwards the 180 Ringing response to User A. User A hears the ring-back tone indicating that User C is being alerted.
F13	200OK—User C to Proxy Server	User C sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the connection has been made.
F14	200OK—Proxy Server to User A	The proxy server forwards the SIP 200 OK response to User A. The 200 OK response notifies User A that the connection has been made.
F15	ACK— User A to Proxy Server	User A sends a SIP ACK to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now active.
F16	ACK—Proxy Server to User C	The proxy server sends the ACK message to User C. The ACK confirms that the proxy server has received the 200 OK response.

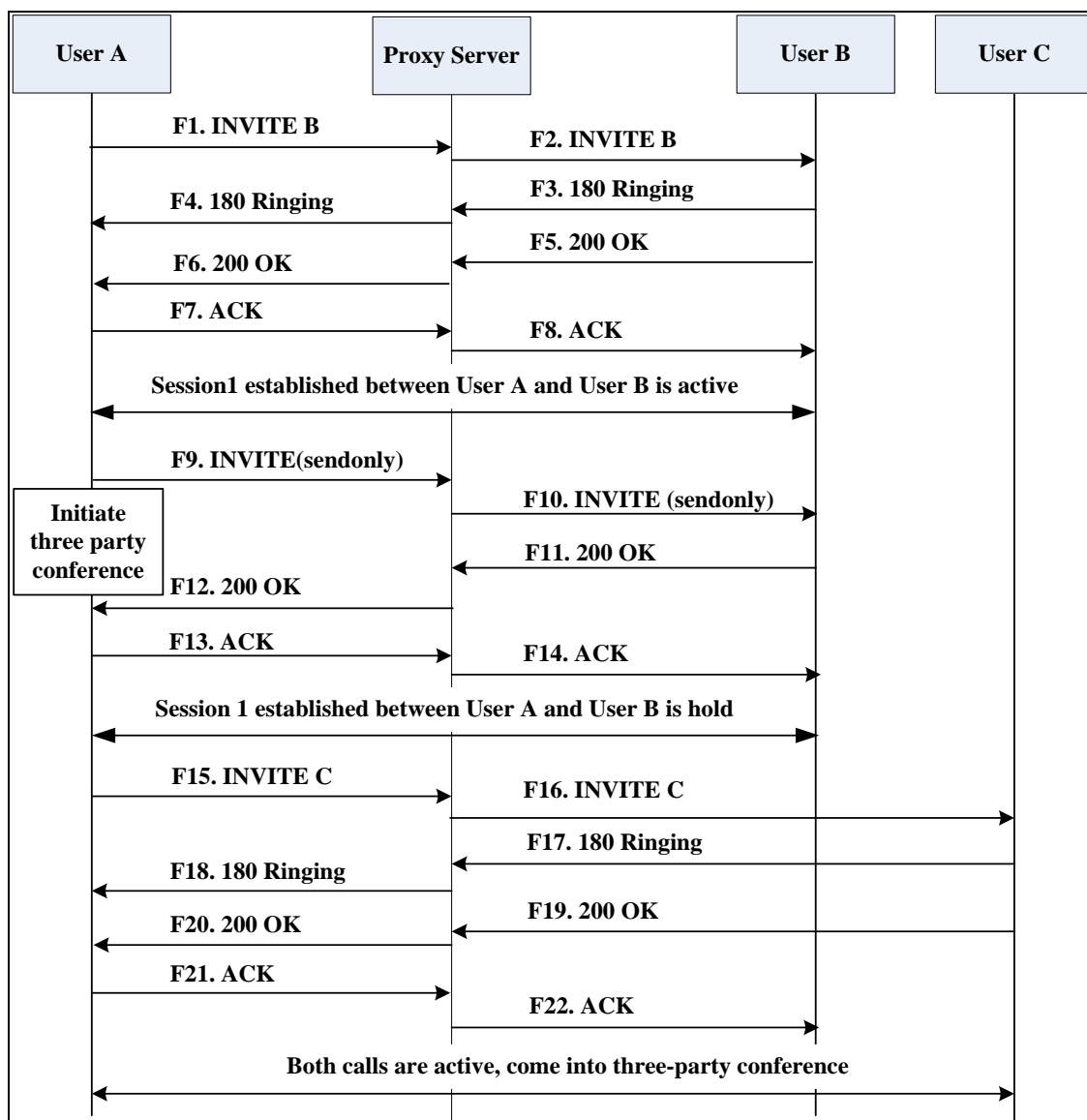
Call Conference

The following figure illustrates successful 3-way calling between Yealink SIP-T4X 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 puts 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.

Step	Action	Description
		<ul style="list-style-type: none"> • 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.
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.

Step	Action	Description
F9	INVITE—User A to Proxy Server	User A sends a mid-call INVITE request to the proxy server with new SDP session parameters, which are used to place the call on hold.
F10	INVITE—Proxy Server to User B	The proxy server forwards the mid-call INVITE message to User B.
F11	200 OK—User B to Proxy Server	User B sends a SIP 200 OK response to the proxy server. The 200 OK response notifies User A that the INVITE is successfully processed.
F12	200 OK—Proxy Server to User A	The proxy server forwards the 200 OK response to User A. The 200 OK response notifies User A that User B is successfully put on hold.
F13	ACK—User A to Proxy Server	User A sends the ACK message to the proxy server. The ACK confirms that User A has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
F14	ACK—Proxy Server to User B	The proxy server sends the ACK message to User B. The ACK confirms that the proxy server has received the 200 OK response.
F15	INVITE—User A to Proxy Server	User A sends a SIP INVITE request to the proxy server. In the INVITE request, a unique Call-ID is generated and the Contact-URI field indicates that User A requests the call.
F16	INVITE—Proxy Server to User 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.
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

Step	Action	Description
		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.

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