



Genesys Application Note

Yealink SIP Phones With Genesys SIP Server

Document version 1.6

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1 Summary

Yealink phones are recommended as SIP “hard phones” to be integrated and used with the Genesys SIP solution. All voice features, from simple calls to voicemail integration to agent-login, have been successfully validated during extensive testing.

Yealink phones run common firmware across the models. The following Yealink models are supported:

Yealink IP Phone Model	Firmware Version
SIP-T38G SIP-T32G	V70 (x.70.0.125)
SIP-T28P SIP-T26P SIP-T22P SIP-T20P	V70, V71 (x.71.169.x)
SIP-T42G	V80 (x.80.0.40) and later
SIP-T23G SIP-T21P E2	V80 (x.80.0.33) and later
SIP-T19P E2 SIP-T23P SIP-T27P SIP-T29G SIP-T41P SIP-T46G SIP-T48G	V80 and later

The supporting versions of Genesys components include SIP Server v8.1.x (8.1.1 recommended), SIP Feature Server v8.1.x (8.1.2 recommended), Media Server (v8.1.x and v8.5.x), and SIP Proxy (v8.1.x).

2 SIP Endpoint Features

2.1 Feature Chart

Feature Name	
General Features Supported By Phone (1pcc)	Supported
Agent Login from the Phone	Yes
Agent State Control from the Phone	Yes*
Auto-Answer	Yes
Alternate Ringtones	Yes
Caller ID	Yes
Call Forward	Yes
Do Not Disturb	Yes
DNS-based redundancy (using SIP Proxy)	Yes
DTMF tones generation	Yes
IPv6 support	Yes
Multiple calls on one extension	Yes
Message Waiting Indicator	Yes
Shared Call Appearance	Yes
SIP authentication	Yes
TLS/SRTP	Yes
Call Control Using Phone (1pcc)	Supported
Basic calling (incoming and outgoing calls)	Yes
Conference	Yes*
Hold/Retrieve	Yes
Unattended transfer	Yes
Semi-attended transfer	Yes
Attended transfer	Yes
Call Control Using Desktop Client (3pcc)	Supported
Answer Incoming Call	Yes
Make Outgoing Call	Yes
Hold/Retrieve	Yes
Conference	Yes
Remote Auto-Answer (based on SIP header)	Yes
Unattended transfer (Genesys Single-Step Transfer)	Yes
Semi-attended transfer (Genesys Blind Transfer)	Yes
Attended transfer (Genesys Two-Step Transfer)	Yes*
DTMF tone generation	No
Video Support	Supported
Basic Video Calls	No
Push Video	No
Video Call on Hold/Retrieve	No
Video Call Transfer	No
Video Conference	No
Support of Genesys Solutions	Supported
Genesys Business Continuity	Yes
Genesys Voice Mail Solution	Yes

* See [section 6](#) for known limitations.

2.2 Feature Chart Glossary

2.2.1 General Features Supported by Phone

1pcc: First-Party Call Control is a method to handle calls using the phone keypad.

3pcc: Third-Party Call Control is a method to handle calls using T-Library desktop connected to SIP Server.

Agent Login from the Phone: Agent sets login/logout from the phone. Agent state ready/not ready can be set from the phone or it can be pushed from the server to the phone after agent logs in from the phone. Functionality is supported based on RFC3863 using presence states open/closed.

Agent State Control from the Phone: This feature enables an agent to perform agent-related operations from the phone: login/logout, change of the state to ready/not ready/ACW, reason code for not ready state. Available for phones that support BroadSoft's Application Server Feature Event Package and Hoteling Event Package.

Alternate Ringtones: Phone provides distinctive ringtones requested by SIP Server. Functionality is supported based on RFC3261 using the Alert-Info header.

Auto-Answer: Phone can be configured to answer all calls automatically.

Caller ID: Phone is able to display the number and name of the calling party.

Call Forward: Phone can forward calls unconditionally or based on internal state (e.g. 'busy').

Do Not Disturb: Phone can reject all incoming calls.

DNS-based redundancy: Phone can toggle between SIP Servers provisioned by single FQDN if current SIP Server becomes unavailable. This functionality is required to deploy a phone with Genesys SIP Proxy. It also may be used for Genesys Business Continuity.

DTMF tones generation: Phone can pass DTMF tones in-band (RFC2833, RFC4733) or using SIP INFO messages.

IPv6 support: Phone can support the IPv6 protocol.

Message Waiting Indicator: SIP MWI support (RFC3842).

Multiple calls on one line: Phone can process multiple incoming/outgoing calls simultaneously on the same line.

Shared Call Appearance (SCA): This feature enables a group of SIP phones to receive inbound calls directed to a single destination (shared line); that way, any phone from this group can answer the call, barge-in to the active call, or retrieve the call placed on hold. The shared line has sub-lines called appearances.

SIP authentication: Phone can authenticate with SIP Server using the HTTP Digest algorithm (RFC3261 and RFC2617).

TLS/SRTP: Phone supports secure SIP environment that uses TLS and SRTP.

2.2.2 Call Control Using Phone (1pcc)

Basic calling: Incoming and outgoing calls.

Conference: Phone can bridge two or more calls without using MCU.

Hold/Retrieve: Phone can put a call on hold and then retrieve it.

Transfer:

- **Unattended transfer:** Call transfer using REFER.
- **Semi-attended transfer:** Completing the transfer when one party is on hold and the other party is ringing, using REFER with Replaces.
- **Attended transfer:** Completing the transfer using REFER with Replaces when one party is on hold and the other party has answered the call.

2.2.3 Call Control Using Desktop Client (3pcc)

Answer Incoming Call: Phone can answer the call using the BroadSoft extension 'talk' passed in SIP NOTIFY.

Make Outgoing Call: Phone can make an outgoing call initiated by SIP Server through the Genesys T-Library interface.

Conference: Phone supports server side single-step or two-step conference.

Hold/Retrieve: Phone can put a call on hold and retrieve it using the BroadSoft extensions 'hold' and 'talk' passed in SIP NOTIFY.

Remote Auto-Answer: Phone can answer a call automatically based on Auto-Answer (RFC5373) or Alert-Info headers.

Transfer:

- **Unattended transfer (Genesys Single-Step Transfer):** Phone supports unattended transfer initiated by SIP Server using REFER or re-INVITE.
- **Semi-attended transfer (Genesys Blind Transfer):** Phone supports completion of two-step transfer initiated by SIP Server when one party is on hold and the other party is ringing.
- **Attended transfer (Genesys Two-Step Transfer):** Phone supports completion of two-step transfer initiated by SIP Server when one party is on hold and the other party has answered the call.

DTMF tone generation: Phone can generate DTMF tones through RTP when the tone generation is requested by SIP Server through the Genesys T-Library interface.

2.2.4 Video Support

Basic Video Calls: Incoming and outgoing video calls.

Push Video: Agent can show a video clip to the customer.

Video Hold/Treatment: Playing video file when call is put on hold or treatment is applied from routing strategy.

Video Call Transfer: Transferring video calls.

Video Conference: Video Conference with active speaker detection using Genesys Media Server.

2.2.5 Support of Genesys Solutions

Genesys Business Continuity: Phone is certified to be used in the Genesys Business Continuity environment in one of two modes. It can switch between the two geo-redundant sites, or it can stay connected to both of them at the same time.

Genesys Voice Mail Solution: Phone is certified to be used with the Genesys Voice Mail solution. Optional advanced features support group Voice Mail Boxes, enable multiple Voice Mail Boxes to be configured for one line, and provide easy access to all configured Voice Mail Boxes.

3 Software and Hardware Versions Validated

The following Genesys components and Yealink phones were validated for reference configuration examples.

3.1 Genesys Components

Genesys Components		
Component	Version	Notes
SIP Server	8.1.1	Genesys SIP Server performs call switching and control. SIP Server communicates via SIP with SIP Endpoints.
Genesys Media Server	8.5.1	Used to handle media interactions such as call treatments (ring back, busy tones and music on hold); also used as MCU.
Genesys SIP Feature Server	8.1.2	Used as a SIP Voicemail Server.
SIP Proxy	8.1.1	Used for HA deployment.

3.2 Yealink SIP phones

3 rd Party Hardware Components		
Model	Version	Notes
SIP-T38G	38.70.0.125	v38.70.0.125 or later supported
SIP-T42G	29.80.0.40	
SIP-T23G SIP-T21P E2	52.80.0.33	

For a full listing of 3rd party hardware/software supported by Genesys, see the [Genesys Supported Media Interface Guide \(SMI\)](#).

4 Features Configuration in Genesys Configuration Environment

This section describes how to configure features represented in the [Feature Chart](#) (Section 2.1, above) within a Genesys configuration environment.

Features can be configured in the SIP Server Switch on a DN object with type Extension (or ACD Position) representing SIP Endpoint devices and/or on an Agent Login object.

Note: It is assumed the reader has Genesys knowledge and is familiar with deploying a basic Genesys environment.

Features Configuration in Genesys Configuration Environment	
General Features Supported By Phone (1pcc)	
Feature	Key Actions and Procedures
Agent Login from the Phone	<ol style="list-style-type: none">1. Enable SIP Server mapping of agent-status from SUBSCRIBE or NOTIFY messages from a SIP Endpoint into T-LIB Events. In the TServer section of the DN object, configure: enable-agentlogin-subscribe=true2. If required, configure the password used for User authorization during the ACD login operation on the phone. Enter the password in the "Enter password" field on the Advanced tab of the Agent Login object. <p>Notes:</p> <ul style="list-style-type: none">• The name of the Agent Login object must match the User ID value entered from the phone when you enter Login credentials.• The value of the password field on the Advanced tab must match the password value entered on the phone when you enter Login credentials.
Agent State Control from the Phone	<p>If required, configure the password used for User authorization during the ACD login operation on the phone. Enter the password in the "Enter password" field on the Advanced tab of the Agent Login object.</p> <p>The name of the Agent Login object must match the User ID value entered from the phone when you enter Login credentials. The value of the password field on the Advanced tab must match the password value entered on the phone when you enter Login credentials.</p>
Auto-Answer	No configuration is required.

Alternate Ringtones	<ol style="list-style-type: none"> 1. If required, specify the ring type for an incoming external call. For an external call, SIP Server will add the specified value to the Alert-Info header of the INVITE message that it sends to the SIP Endpoint. In the TServer section of the DN object, configure: sip-alert-info-external= <http://192.168.14.62/Vin/>;info=ring1 2. If required, specify the ring type for a consultation call. For a consultation call, SIP Server will add the specified value to the Alert-Info header of the INVITE message that it sends to the SIP Endpoint. In the TServer section of the DN object, configure: sip-alert-info-consult= <http://192.168.14.62/Vin/>;info=ring5 3. If required, specify the ring type for any type of call. SIP Server will add the specified value to the Alert-Info header of the INVITE message that it sends to the SIP Endpoint. In the TServer section of the DN object, configure: sip-alert-info=<http://192.168.14.62/Vin/>;info=ring8 <p>Notes:</p> <ul style="list-style-type: none"> • Settings in the sip-alert-info-external or sip-alert-info-consult options take precedence over sip-alert-info settings. • The values of these options have URL-access to a customized ringtone .wav file (<http://192.168.14.62/Vin/>) concatenated with the value set in the Internal Ringer Text field. (Using the phone Web interface, see Phone -> Ring.)
Caller ID	No configuration is required.
Call Forward	No configuration is required.
Do Not Disturb	No configuration is required.
DNS-based redundancy (using SIP Proxy)	Requires HA deployment using SIP Proxy deployment. SIP Proxy can be used in SIP Server standalone deployment or Genesys Business Continuity with SIP Proxy deployment. Refer to the <i>Genesys SIP Proxy Deployment Guide</i> and <i>Genesys SIP Server High-Availability Deployment Guide</i> .
DTMF tones generation	No configuration is required.
Multiple calls on one extension	See Call Control using desktop client -> Attended transfer feature.
Message Waiting Indicator	<p>Configure a voice mailbox for an Extension. In the TServer section of the DN object, configure: gvm_mailbox=<voice mail box number></p> <p>For example: gvm_mailbox=12003, where 12003 is a mailbox number.</p>

Shared Call Appearance (SCA)	<ol style="list-style-type: none"> 1. Configure a Primary Shared Line DN: <ul style="list-style-type: none"> • Create a DN of type Extension with the number where all incoming calls will be delivered. • Specify that this DN is used as a Primary shared line number. In the TServer section of the DN object, configure: shared-line=true • Specify a number of shared line appearances. In the TServer section of the DN object, configure: shared-line-capacity=<maximum number of simultaneous calls per shared line> • If required, configure SIP authentication. (See SIP authentication in this table.) 2. Configure Secondary Shared Line DN: <ul style="list-style-type: none"> • Create a DN of type Extension with the number to be used as a Secondary DN. • Specify a number of the Primary DN. In the TServer section of the DN object, configure: shared-line-number=<number of Primary shared line DN>
SIP authentication	<ol style="list-style-type: none"> 1. Specify SIP requests (REGISTER, INVITE), which are sent by the phone to be authenticated by SIP Server. In the TServer section of the DN object, configure: authenticate-requests=register,invite 2. If required, configure the password used for authentication of incoming REGISTER or INVITE messages to SIP Server. In the TServer section of the DN object, configure: password=<Any alphanumeric string> Note: String must match the phone setting in Account -> Basic -> Account # -> Password.
TLS/SRTP	See the Transport Layer Security for SIP Traffic chapter in the Genesys 8.1 SIP Server Deployment Guide for details.
Call Control Using Phone (1pcc)	
Feature	Key Actions and Procedures
Basic calling (incoming and outgoing calls)	See the Make Outgoing Call feature.
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Unattended transfer	No configuration is required.
Semi-attended transfer	No configuration is required.
Attended transfer	No configuration is required.

Call Control Using Desktop Client (3pcc)	
Feature	Key Actions and Procedures
Answer Incoming Call	<p>Enable SIP Server to send the SIP NOTIFY (event talk) message when desktop client requests to answer the incoming call. In the TServer section of the DN object, configure: sip-cti-control=talk</p> <p>Note: The "talk" value affects the Retrieve feature. See the Hold/Retrieve feature for information about setting the sip-cti-control option.</p>
Conference	<p>Deploy Genesys Media Server with MCU capabilities. See the <i>SIP Server Deployment Guide</i> for details.</p>
Hold/Retrieve	<p>Enable SIP Server to send the SIP NOTIFY (event hold) message when desktop client requests to hold the call, and the SIP NOTIFY (event talk) message when desktop client requests to retrieve the call. In the TServer section of the DN object, configure: sip-cti-control=talk,hold</p>
Make Outgoing Call	<ol style="list-style-type: none"> 1. Create a DN object with type Extension or ACD Position in the Genesys configuration environment under Switch object and DNs folders. This object represents the SIP phone. 2. To activate required features described in this Table, configure options in the DN object > TServer section. 3. Configure a phone to make basic calls (incoming, outgoing) with SIP Server. 4. Restart the phone. 5. After successful SIP registration the phone is ready for making outgoing calls and receiving incoming calls. 6. Run your desktop client to make a test call.
Remote Auto-Answer (based on SIP header)	<p>If required, specify the timer that SIP Server will add to the answer-after parameter in the Call-Info header of the INVITE message, which it sends to the SIP Endpoint. The phone will wait this time interval before automatically answering a call. In the TServer section of the DN object, configure: auto-answer-after=<time in sec></p> <p>Note: It is not recommended to set the timer to "0" (zero).</p>
Unattended transfer (Genesys Single-Step Transfer)	<p>No configuration is required.</p>
Semi-attended transfer (Genesys Blind Transfer)	<p>Enable completion of transfer when the destination is in alerting state. In the TServer section of the DN object (transfer target DN), configure: blind-transfer-enabled=true</p> <p>Note: This option must be set on the DN object that represents a transfer destination party.</p>

<p>Attended transfer (Genesys Two-Step Transfer)</p>	<ol style="list-style-type: none"> 1. Enable dual-dialog to be supported on a DN for an attended transfer operation requested from a desktop client. In the TServer section of the DN object, configure: dual-dialog-enabled=true 2. Specify the call flow to process a make call/initiate consultative call operation initiated from a desktop client. In the TServer section of the DN object, configure: make-call-rfc3725-flow=2 Note: A value of 1 or 2 is sufficient for the phone. 3. Specify the INVITE method to be used to create a simple call or a consultation call when operation is requested from a desktop client. In the TServer section of the DN object, configure: refer-enabled=false Note: Only INVITE method can be used to create a simple call or a consultation call when operation is requested from a desktop client
<p>Genesys Business Continuity</p>	<p>Configure SIP Server to forward an incoming call to the second SIP Server peer if an Endpoint is in an Out-Of-Service (OOS) state. In the TServer section of the DN object, configure: dr-forward=oos</p>

Example of the DN .cfg file:

[TServer]

authenticate-requests=invite,register

blind-transfer-enabled=true

contact=sip:1664@192.168.14.62:5063

dual-dialog-enabled=true

enable-agentlogin-subscribe=true

make-call-rfc3725-flow=1

refer-enabled=false

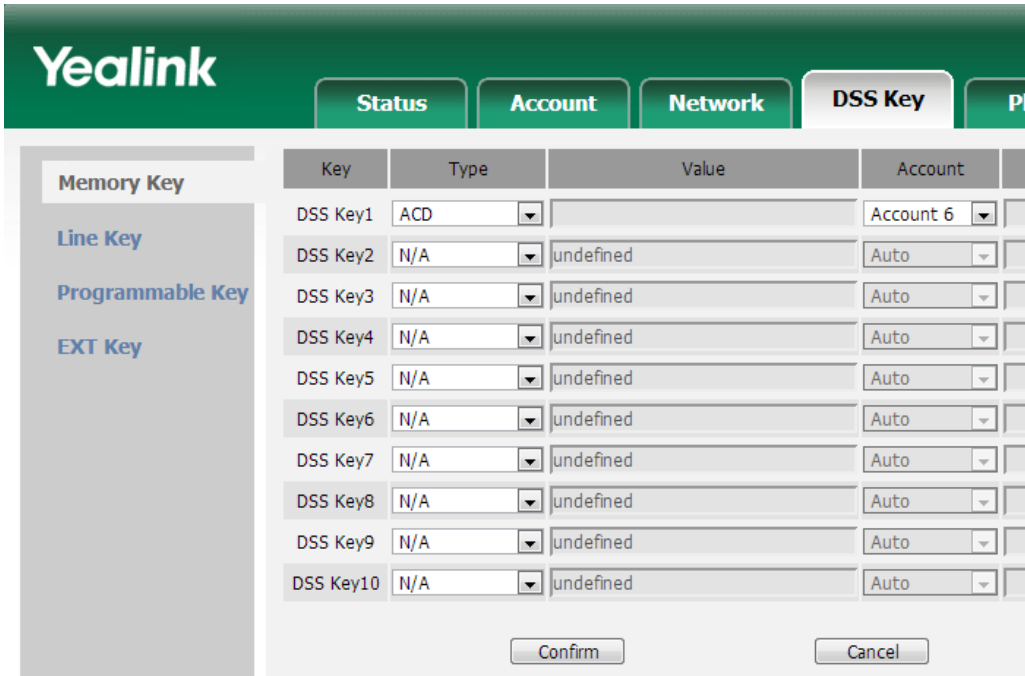
sip-alert-info=<http://192.168.14.62/Vin/>;info=ring8

sip-cti-control=talk,hold

5 Yealink Phone Configuration

This section describes how to configure features represented in the [Feature Chart](#) (Section 2.1, above) using the phone Web interface.

The following table displays screenshots of the Web interface of the Yealink SIP-T38G except where T42G configuration is described.

Yealink Phone Configuration	
General Features Supported By Phone	
Feature	Key Actions and Procedures
Agent Login from the Phone (applies to T38G-38.70.0.125)	<ol style="list-style-type: none"> Using the Web interface, DSS Key -> Memory Key, set the ACD feature on a DSS Memory key for an Account in advance: <ol style="list-style-type: none"> Set DSS Key# -> Type to ACD. Set DSS Key# -> Account to the Account number.  For Login/Logout/Available/Unavailable status, press Memory Key 1 and enter Login Credentials from the phone touch screen: User Name and Password.
	<ol style="list-style-type: none"> Using the Web interface, Settings -> Configuration -> Export CFG Configuration File -> Export, edit exported the CFG file, and add the following parameters: <pre>account.1.acd.enable = 1 account.1.acd.available = 1</pre>

Agent State
Control from
the Phone

(applies to
T42G-
29.80.0.40)

```
bw.enable = 1
account.1.hoteling.enable = 0
bw.feature_key_sync = 1
account.1.sip_server_type = 10
account.1.acd.initial_state = 1
account.1.acd.unavailable_reason_enable = 1
acd.auto_available = 0
```

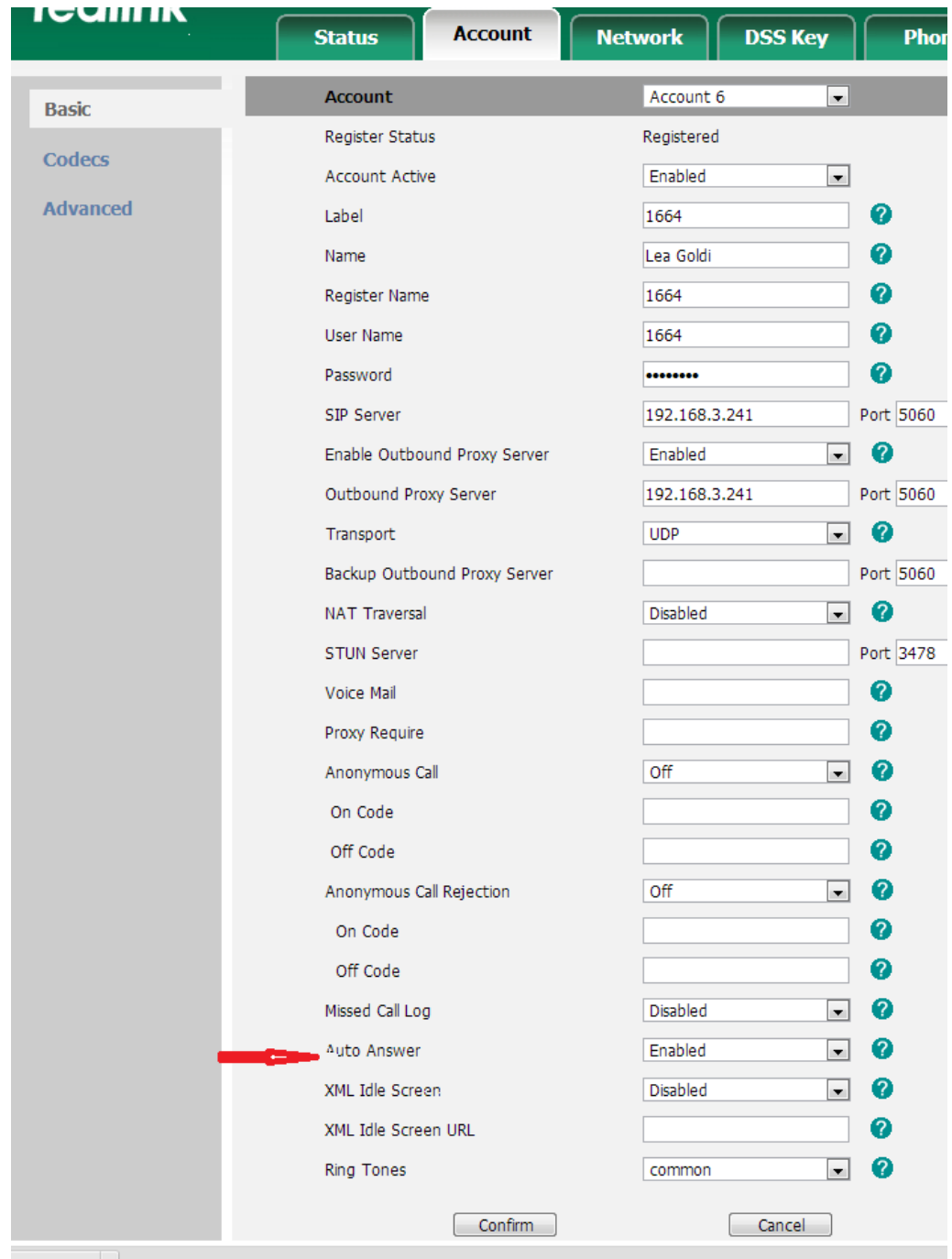
2. Using the Web interface, **DSS Key -> Line Key 1-5**, set the ACD feature on DSS Line Key 1:
 - Set **DSS Key# -> Type** to **ACD**.

Key	Type	Value	Label	Line	Extension
Line Key1	ACD		Agent1	N/A	
Line Key2	N/A			N/A	
Line Key3	N/A			N/A	
Line Key4	N/A			N/A	
Line Key5	N/A			N/A	

3. For Login/Logout/Available/Unavailable status, press **Line Key 1** and enter Login Credentials from the phone touch screen: **User Name** and **Password**.

Using the Web interface, **Account -> Basic -> Account #**, enable Auto Answer for all SIP calls, by setting **Auto Answer** to **Enabled**.

Auto-Answer



The screenshot shows the Yealink SIP Phone web interface. At the top, there are tabs for Status, Account, Network, DSS Key, and Phone. The 'Account' tab is selected, and the 'Basic' sub-tab is active. The left sidebar contains links for Basic, Codecs, and Advanced. The main content area displays configuration settings for 'Account 6'. A red arrow points to the 'Auto Answer' setting, which is currently set to 'Enabled'. Other settings include Register Status (Registered), Account Active (Enabled), Label (1664), Name (Lea Goldi), Register Name (1664), User Name (1664), Password (masked), SIP Server (192.168.3.241), Port (5060), Enable Outbound Proxy Server (Enabled), Outbound Proxy Server (192.168.3.241), Port (5060), Transport (UDP), Backup Outbound Proxy Server, NAT Traversal (Disabled), STUN Server, Voice Mail, Proxy Require, Anonymous Call (Off), On Code, Off Code, Anonymous Call Rejection (Off), Missed Call Log (Disabled), XML Idle Screen (Disabled), XML Idle Screen URL, and Ring Tones (common). Each setting has a corresponding help icon (?) on the right. At the bottom, there are 'Confirm' and 'Cancel' buttons.

Account		Account 6
Register Status	Registered	
Account Active	Enabled	
Label	1664	?
Name	Lea Goldi	?
Register Name	1664	?
User Name	1664	?
Password	?
SIP Server	192.168.3.241	Port 5060
Enable Outbound Proxy Server	Enabled	?
Outbound Proxy Server	192.168.3.241	Port 5060
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060
NAT Traversal	Disabled	?
STUN Server		Port 3478
Voice Mail		?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?
On Code		?
Off Code		?
Missed Call Log	Disabled	?
Auto Answer	Enabled	?
XML Idle Screen	Disabled	?
XML Idle Screen URL		?
Ring Tones	common	?

Confirm Cancel

Alternate Ringtones

Using the Web interface, **Phone -> Ring -> 1:**

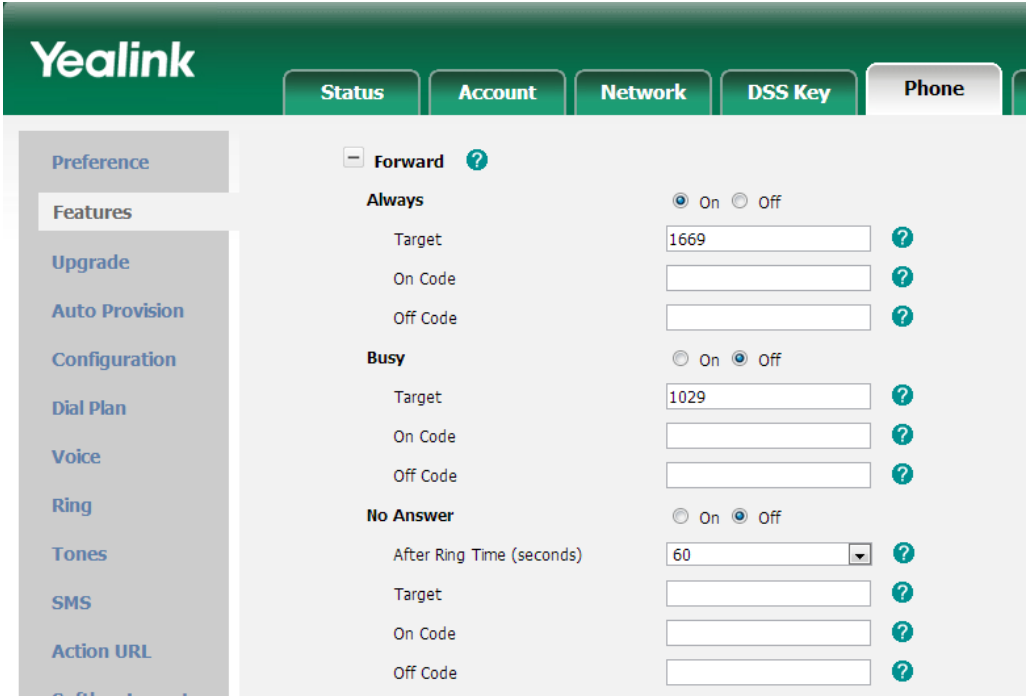
1. Specify a name for a .wav ringer file in the **Internal Ringer Text** field. Use Yealink IP Phone built-in system ringtones.
2. Specify the **Internal Ringer File**. For example, Ring1.wav.
3. In the same way, specify the internal ringer text for other .wav files.

Index	Internal Ringer Text	Internal Ringer File
1	ring1	Ring1.wav
2	ring2	Ring2.wav
3	ring3	Ring3.wav
4	ring4	Ring4.wav
5	ring5	Ring5.wav
6	ring6	Ring6.wav
7	ring7	Ring7.wav
8	ring8	Ring8.wav
9		Ring1.wav
10		Ring1.wav

Note: A value of the **Internal Ringer Text** field must be used in the Genesys configuration environment. See [Alternate Ringtones](#).

Caller ID

No configuration is required.

Call Forward	<p>Using the Web interface, Phone -> Features -> Forward, configure call forward.</p>  <p>OR:</p> <p>Using the phone, enable call forward by pressing the Menu button, then selecting Features -> Call Forward.</p>
Do Not Disturb	Using the phone, enable DND by pressing the DND button.

DNS-based
redundancy
(using SIP
Proxy)

1. Using the Web interface, **Account -> Basic -> Account #**, configure the Address (FQDN of the SIP Proxy pool) and Port in the **SIP Server** and **Outbound Proxy Server** fields.

The screenshot shows the Yealink web interface with the 'Account' tab selected. The 'Basic' sub-tab is active, and 'Account 6' is selected from the dropdown. The configuration table is as follows:

Field	Value	Port
Register Status	Registered	
Account Active	Enabled	
Label	1664	
Name	Lea Goldi	
Register Name	1664	
User Name	1664	
Password	*****	
SIP Server	sips-a.qa.domain.com	5060
Enable Outbound Proxy Server	Enabled	
Outbound Proxy Server	sips-a.qa.domain.com	5060
Transport	DNS-SRV	
Backup Outbound Proxy Server		5060
NAT Traversal	Disabled	
STUN Server		3478

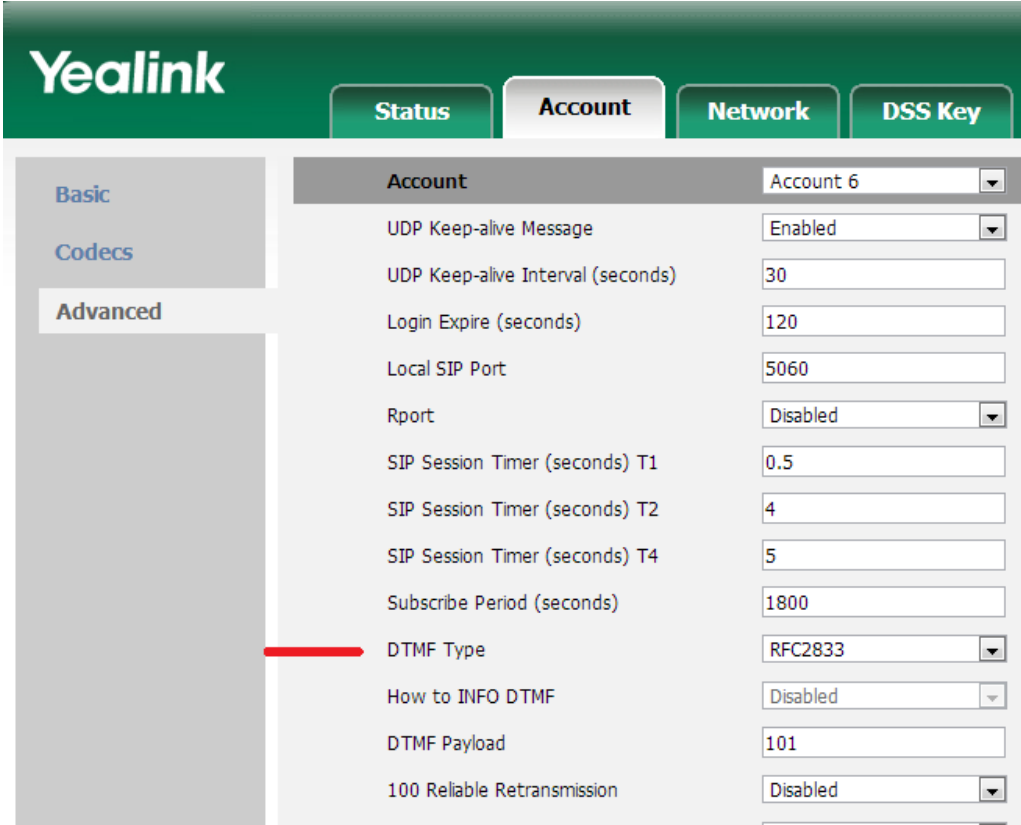
2. Using the Web interface, **Account -> Basic -> Account #**, configure **SIP Registration Retry Timer**.

The screenshot shows the 'SIP Registration Retry Timer' field highlighted with a red arrow. The value is set to '5'.

Field	Value
Direct Call Pickup Code	
Group Call Pickup Code	
BLA Number	
BLA Subscription Period (seconds)	300
SIP Send MAC	Disabled
SIP Send Line	Disabled
SIP Registration Retry Timer(Scope:0~1800) (seconds)	5
Signal Encode	Disabled
Signal Encode Key	
Conference Type	Local
Conference URI	
Music Server URI	

Note: The Address field has the FQDN (sips-a.qa.domain.com) of the SIP Proxy pool that must be resolved in multiple a-records.

Each SIP Proxy in the pool has the same SIP port configured in the Genesys configuration environment.

DTMF tones generation	<p>Using the Web Interface, Account -> Advanced -> Account #, specify the method for DTMF tone generation in the DTMF Type field.</p>  <p>The screenshot shows the Yealink web interface with the 'Account' tab selected. The 'Advanced' sub-tab is active, and the 'DTMF Type' field is highlighted with a red line. The 'DTMF Type' is set to 'RFC2833'.</p>
Multiple calls on one extension	Genesys recommends having no more than two concurrent calls.

Message
Waiting
Indicator

1. Using the Web interface, **Account -> Advanced -> Account #:**
 - a. Set **Subscribe for MWI** to **Enabled**.
 - b. If required, set **MWI Subscription Period** (default 3600 seconds).

The screenshot shows the Yealink web interface with the 'Account' tab selected. The 'Advanced' sub-tab is active, displaying configuration options for 'Account 6'. The 'Subscribe for MWI' option is highlighted with a red line and is set to 'Enabled'. The 'MWI Subscription Period (Scope:0~84600) (seconds)' is set to 3600. Other visible settings include 'UDP Keep-alive Message' (Enabled), 'UDP Keep-alive Interval (seconds)' (30), 'Login Expire (seconds)' (120), 'Local SIP Port' (5060), 'Rport' (Disabled), 'SIP Session Timer (seconds) T1' (0.5), 'SIP Session Timer (seconds) T2' (4), 'SIP Session Timer (seconds) T4' (5), 'Subscribe Period (seconds)' (1800), 'DTMF Type' (RFC2833), 'How to INFO DTMF' (Disabled), 'DTMF Payload' (101), '100 Reliable Retransmission' (Disabled), 'Enable Precondition' (Disabled), 'Subscribe Register' (Disabled), 'Caller ID Header' (FROM), 'Use Session Timer' (Disabled), and 'Session Timer (seconds)'.

Account		Account 6
UDP Keep-alive Message	Enabled	?
UDP Keep-alive Interval (seconds)	30	
Login Expire (seconds)	120	?
Local SIP Port	5060	?
Rport	Disabled	?
SIP Session Timer (seconds) T1	0.5	?
SIP Session Timer (seconds) T2	4	
SIP Session Timer (seconds) T4	5	
Subscribe Period (seconds)	1800	?
DTMF Type	RFC2833	?
How to INFO DTMF	Disabled	
DTMF Payload	101	
100 Reliable Retransmission	Disabled	?
Enable Precondition	Disabled	?
Subscribe Register	Disabled	?
Subscribe for MWI	Enabled	?
MWI Subscription Period (Scope:0~84600) (seconds)	3600	
Caller ID Header	FROM	?
Use Session Timer	Disabled	?
Session Timer (seconds)		?

2. Using the Web Interface, **Account -> Basic -> Account #**, specify a number to call a Voice Mail System in the **Voice Mail** field.

The screenshot shows the Yealink web interface with the 'Account' tab selected. Under the 'Basic' sub-tab, the 'Account' configuration page for 'Account 6' is displayed. The 'Voice Mail' field is highlighted with a red bar and is set to '888'. Other fields include 'Register Status' (Registered), 'Account Active' (Enabled), 'Label' (1664), 'Name' (Lea Goldi), 'Register Name' (1664), 'User Name' (1664), 'Password' (masked), 'SIP Server' (192.168.3.241), 'Port' (5060), 'Enable Outbound Proxy Server' (Enabled), 'Outbound Proxy Server' (192.168.3.241), 'Transport' (UDP), 'Backup Outbound Proxy Server' (empty), 'NAT Traversal' (Disabled), 'STUN Server' (empty), and 'Proxy Require' (empty). Each field has a help icon (?) next to it.

Shared Call Appearance (SCA)

(applies to T38G – 38.70.0.125)

1. Using the Web interface, **Account -> Basic -> Account #**, configure the Primary Shared Line for Shared Call Appearance:
 - a. Specify the following properties:
 - Label:** Specify the number to be shown on the LCD to identify the account.
 - Name:** Specify the Primary line number.
 - Register Name:** Specify the Primary line number.
 - b. Set **Shared Line** to **BroadSoft SCA** from the drop-down menu.
 - c. If required, configure SIP authentication for the Primary Shared line. (See [SIP authentication](#) in this table.)
 - d. Configure basic calling for the Primary Shared line. See Call Control Using Phone -> [Basic calling](#) (Incoming and outgoing calls).

Yealink

Status Account Network DSS Key Phone

Basic Account 1

Register Status Registered

Account Active Enabled

Label 3955

Name 3955

Register Name 3950

User Name 3950

Password

2. Using the Web interface, **Account -> Basic -> Account #**, configure the Secondary Shared Line for Shared Call Appearance:

a. Specify the following properties:

Label: Specify the number to be shown on the LCD to identify the account.

Name: Specify the Secondary line number.

Register Name: Specify the Primary line number.

b. Set **Shared Line** to **BroadSoft SCA** from the drop-down menu.

c. If required, configure SIP authentication for the Shared line. (See [SIP authentication](#) in this table.)

Note: The Primary User ID must be used in the Authentication when you configure the Secondary Shared Line.

d. Configure basic calling for the Primary Shared line. See Call Control Using Phone -> [Basic calling](#) (Incoming and outgoing calls).

Refresher UAC

Use user=phone Enabled

Voice Encryption(SRTP) Disabled

Ptime (ms) 20

BLF List URI

BLF List Pickup Code

Shared Line BroadSoft SCA

Dialog-Info Call Pickup Enabled

Direct Call Pickup Code

Group Call Pickup Code

BLA Number 3950

3. Using the Web interface, **Phone -> Features**, enable **Allow Barge In** under the Intercom Settings of the Primary and Secondary Shared Line phones.

The screenshot shows the 'Features' configuration page for a Yealink phone. On the left is a sidebar with menu items: Ring, Tones, SMS, Action URL, and Softkey Layout. The main content area is titled 'No Answer' and includes a toggle for 'On' (radio button) and 'Off' (radio button). Below this are fields for 'After Ring Time (seconds)' (set to 60), 'Target', 'On Code', and 'Off Code', each with a help icon. Further down are expandable sections: '+ Do Not Disturb', '+ General Information', '+ Audio Settings', and '- Intercom Settings'. The 'Intercom Settings' section contains four items: 'Accept Intercom' (Disabled), 'Intercom Mute' (Disabled), 'Warning Tone' (Enabled), and 'Allow Barge In' (Enabled). The 'Allow Barge In' dropdown is highlighted with a red box and a red arrow.

4. Using the Web interface, **DSS Key -> Line Key**, configure Lines (per Account).

The screenshot shows the 'DSS Key' configuration page in the Yealink web interface. The top navigation bar includes 'Status', 'Account', 'Network', and 'DSS Key'. On the left is a sidebar with menu items: Memory Key, Line Key (selected), Programmable Key, and EXT Key. The main content area is a table with the following columns: Key, Type, Value, Label, and Account.

Key	Type	Value	Label	Account
Line Key1	Shared Line	3955	3950	Account 1
Line Key2	Shared Line	3955	3950	Account 1
Line Key3	Line			Account 3
Line Key4	Line			Account 4
Line Key5	Line			Account 5
Line Key6	Line			Account 6

Shared Call Appearance (SCA)
(applies to T42G-29.80.0.40)

1. Using the Web interface, **Account -> Account #**, configure the Primary Shared Line for Shared Call Appearance:
 - a. Specify the following properties:

Label: Specify the number to be shown on the LCD to identify the account.
Name: Specify the Primary line number.
Register Name: Specify the Primary line number.
 - b. Set **Shared Line** to **Share Call Appearance** from the drop-down menu.
 - c. If required, configure SIP authentication for the Primary Shared line. (See [SIP authentication](#) in this table.)
 - d. Configure basic calling for the Primary Shared line. See Call Control Using Phone -> [Basic calling](#) (Incoming and outgoing calls).

2. Using the Web interface, **Account -> Basic -> Account #**, configure the Secondary Shared Line for Shared Call Appearance:
 - a. Specify the following properties:

Label: Specify the number to be shown on the LCD to identify the account.
Name: Specify the Secondary line number.
Register Name: Specify the Primary line number.
 - b. Set **Shared Line** to **Share Call Appearance** from the drop-down menu.
 - c. If required, configure SIP authentication for the Shared line. (See [SIP authentication](#) in this table.)

Note: The Primary User ID must be used in the Authentication when you configure the Secondary Shared Line.

 - d. Configure basic calling for the Primary Shared line. See Call Control Using Phone -> [Basic calling](#) (Incoming and outgoing calls).

3. Using the Web interface, **Phone -> Features -> Intercom**, enable **Intercom Barge** under the Intercom Settings of the Primary and Secondary Shared Line phones.

The screenshot shows the Yealink T42G web interface. The 'Features' tab is selected, and the 'Intercom' sub-tab is active. On the left sidebar, 'Intercom' is highlighted. The main content area shows the following settings:

- Accept Intercom: Disabled
- Intercom Mute: Disabled
- Intercom Tone: Enabled
- Intercom Barge: Enabled (highlighted with a red arrow)

At the bottom, there are 'Confirm' and 'Cancel' buttons.

4. Using the Web interface, **DSS Key -> Line Key**, configure Lines (per Account).

The screenshot shows the Yealink T42G web interface. The 'DSSKey' tab is selected, and the 'Line Key' sub-tab is active. On the left sidebar, 'Line Key 1-5' is highlighted. The main content area shows the following configuration:

Enable Page Tips: Disabled

Key	Type	Value	Label	Line	Extension
Line Key1	Line	Default	3955	Line 1	
Line Key2	Line	Default	3955	Line 1	
Line Key3	N/A			N/A	
Line Key4	N/A			N/A	
Line Key5	N/A			N/A	

At the bottom, there are 'Confirm' and 'Cancel' buttons.

5. Barging in an active call:
When phone A has one active call, do the following:
 - a. Long press the desired line key on phone B. "Cancel, Call Pull, New Call, and Barge In" soft keys appear on the LCD screen of phone B.



- b. Press the Barge In soft key to join the active call of phone A.

SIP
authentication

Using the Web interface, **Account -> Basic -> Account #**, specify login credentials for SIP authentication in the **Register Name** and **Password** fields.

Note: The **Password** parameter must have the same value as the **password** option configured in the DN object in the Genesys configuration environment.
The **Register Name** parameter is used to authenticate line registration or an outgoing INVITE.

Yealink

Status Account Network DSS Key Phone

Basic Codecs Advanced

Account Account 6

Register Status	Registered
Account Active	Enabled
Label	1664
Name	Lea Goldi
Register Name	1664
User Name	1664
Password
SIP Server	192.168.3.241 Port 5060
Enable Outbound Proxy Server	Enabled
Outbound Proxy Server	192.168.3.241 Port 5060
Transport	UDP

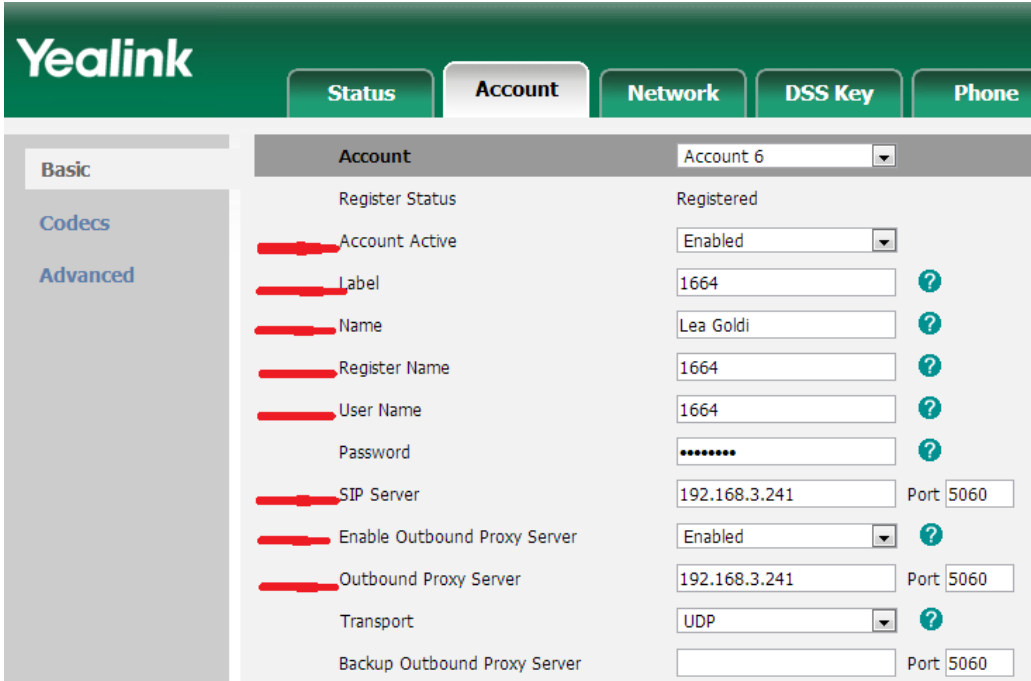
TLS/SRTP

1. Using the Web interface, **Account -> Basic -> Account #:**
 - a. Specify the SIP Server IP address and port in the **SIP Server** and **Port** fields.
 - b. Set **Transport** to **TLS**.

The screenshot shows the Yealink web interface with the 'Account' tab selected. Under the 'Basic' sub-tab, the 'Account' dropdown is set to 'Account 6'. The 'Transport' field is highlighted in red and set to 'TLS'. The 'SIP Server' field is set to '192.168.3.241' and the 'Port' field is set to '5070'. Other fields include 'Register Status' (Registered), 'Account Active' (Enabled), 'Label' (1664), 'Name' (Lea Goldi), 'Register Name' (1664), 'User Name' (1664), 'Password' (masked), 'Enable Outbound Proxy Server' (Enabled), 'Outbound Proxy Server' (192.168.3.241), 'Port' (5070), 'Backup Outbound Proxy Server' (empty), 'Port' (5060), 'NAT Traversal' (Disabled), 'STUN Server' (empty), 'Port' (3478), and 'Voicemail Mail' (empty).

2. Using the Web interface, **Security -> Trusted Certificates:**
 - a. Specify the custom CA Certificate (.crt file) by using **Choose File -> Upload**.
 - b. Set **Only Accept Trusted Certificates** to **Disabled**.

The screenshot shows the Yealink web interface with the 'Security' tab selected. Under the 'Trusted Certificates' sub-tab, the 'Only Accept Trusted Certificates' dropdown is set to 'Disabled'. The 'Import Trusted Certificate (.crt, .cer)' section shows a 'Choose File' button and an 'Upload' button. The 'Delete' button is also visible. The 'NOTE' section is empty.

Call Control Using Phone	
Feature	Key Actions and Procedures
Basic calling (incoming and outgoing calls)	<p>Using the Web interface, Account -> Basic -> Account #:</p> <ol style="list-style-type: none"> Set Account Active and Enabled Outbound Proxy Server to Enabled. Specify Label, Name, Register Name, and User Name. Specify the IP address (FQDN) and port of SIP Server in the SIP Server and Outbound Proxy Server fields. 
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Unattended (blind) transfer	Using the phone, press Transfer , enter the number, press Transfer again.
Semi-attended (two-step) transfer	Using the phone, press Transfer, enter the number, press OK , and press Transfer while receiving ringback.
Attended (consultative) transfer	Using the phone, press Transfer , enter the number, press OK , and press Transfer again when the party answers.

Call Control Using Desktop Client	
Feature	Key Actions and Procedures
Answer Incoming Call	No configuration is required.
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Make Outgoing Call	See the Basic calling (incoming and outgoing calls) feature.
Remote Auto-Answer (based on SIP header)	No configuration is required.
Unattended transfer	No configuration is required.
Semi-attended transfer	No configuration is required.
Attended (consultative) transfer	No configuration is required.
Genesys Business Continuity	<ol style="list-style-type: none"> Using the Web interface, Account -> Basic -> Account #: <ol style="list-style-type: none"> Specify the IP address (FQDN) and port of SIP Server peers in the SIP Server and Outbound Proxy Server fields. Specify Transport. <p>Note: The Address field has the FQDN (811-BC-DIMA-a.qa.sipcluster.genesyslab.com) of SIP Server peers that must be resolved in multiple a-records (each record has an address of the SIP Server peer).</p>

Basic		Account		Account 1	
Register Status	Registered				
Account Active	Enabled				
Label	1501				?
Name	1501				?
Register Name	1501				?
User Name	1501				?
Password	*****				?
SIP Server	811-BC-DIMA-a.qa.sipcluster	Port	8585		?
Enable Outbound Proxy Server	Enabled				?
Outbound Proxy Server	811-BC-DIMA-a.qa.sipcluster	Port	8585		?
Transport	DNS-SRV				?
Backup Outbound Proxy Server		Port	5060		?
NAT Traversal	Disabled				?
STUN Server		Port	3478		?
Voice Mail					?
Proxy Require					?
Anonymous Call	Off				?
On Code					?

2. Using the Web interface, **Account -> Advanced -> Account #:**
 - a. Set **Login Expire (seconds)** to **300**.
 - b. Set **SIP Registration Retry Timer (Scope:0~1800) (seconds)** to **5**.

For Genesys Business Continuity deployment, the Yealink phone registers (SIP REGISTER) with one SIP Server peer and registers on another SIP Server peer when the first one becomes unavailable.

Status	Account	Network	DSS Key	Phone	Co																																																																																				
<table border="1"> <thead> <tr> <th colspan="2">Basic</th> <th colspan="2">Account</th> <th colspan="2">Account 1</th> </tr> </thead> <tbody> <tr> <td>UDP Keep-alive Message</td> <td>Enabled</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>UDP Keep-alive Interval (seconds)</td> <td>5</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Login Expire (seconds)</td> <td>300</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>Local SIP Port</td> <td>5060</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>Rport</td> <td>Disabled</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>SIP Session Timer (seconds) T1</td> <td>0.5</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>SIP Session Timer (seconds) T2</td> <td>4</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>SIP Session Timer (seconds) T4</td> <td>5</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Subscribe Period (seconds)</td> <td>1800</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>DTMF Type</td> <td>RFC2833</td> <td></td> <td></td> <td></td> <td>?</td> </tr> <tr> <td>How to INFO DTMF</td> <td>Disabled</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>DTMF Payload</td> <td>101</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>DTMF Subcode</td> <td>Disabled</td> <td></td> <td></td> <td></td> <td>?</td> </tr> </tbody> </table>						Basic		Account		Account 1		UDP Keep-alive Message	Enabled				?	UDP Keep-alive Interval (seconds)	5					Login Expire (seconds)	300				?	Local SIP Port	5060				?	Rport	Disabled				?	SIP Session Timer (seconds) T1	0.5				?	SIP Session Timer (seconds) T2	4					SIP Session Timer (seconds) T4	5					Subscribe Period (seconds)	1800				?	DTMF Type	RFC2833				?	How to INFO DTMF	Disabled					DTMF Payload	101					DTMF Subcode	Disabled				?
Basic		Account		Account 1																																																																																					
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Caller ID Header	FROM	?
Use Session Timer	Disabled	?
Session Timer (seconds)		?
Refresher	UAC	?
Use user=phone	Disabled	?
Voice Encryption(SRTP)	Disabled	?
Ptime (ms)	20	?
BLF List URI		?
BLF List Pickup Code		?
Shared Line	Disabled	?
Dialog-Info Call Pickup	Enabled	?
Direct Call Pickup Code		
Group Call Pickup Code		
BLA Number		?
BLA Subscription Period (seconds)	300	?
SIP Send MAC	Disabled	?
SIP Send Line	Disabled	?
SIP Registration Retry Timer(Scope:0~1800) (seconds)	5	?
Signal Encode	Disabled	?
Signal Encode Key		?
Conference Type	Local	?
Conference URI		?

6 Known Issues and Limitations

6.1 Issues and Limitations Identified with Genesys Products

When SIP Server is operating with Yealink phones:

- Three-way conferences initiated on any SIP Phone will not be reported as a conference.
- The Agent State Control from the Phone feature is not supported in Business Continuity deployments.
- When Call Forwarding is set on the phone, that phone might send the SUBSCRIBE request to SIP Server containing the tag "SetForwarding" in the XML body. SIP Server is not able to process this subscription request and will reject it. However, it will not affect further processing of Call Forwarding or any other functionality of the phone or SIP Server.

6.2 Issues and Limitations Identified with Third-Party Products

When Yealink phones are operating with SIP Server:

- Only the INVITE method can be used to create a simple call or a consultation call when the operation is requested from a desktop client.
- The Agent State Control from the Phone feature is supported on T42G, T21P E2, and T23G models with firmware V80.