Genesys Application Note

Yealink SIP Phones With Genesys SIP Server

Document version 1.6

About Genesys

Genesys is the world's leading provider of customer service and contact center software - with more than 4,000 customers in 80 countries. Drawing on its more than 20 years of customer service innovation and experience, Genesys is uniquely positioned to help companies bring their people, insights and customer channels together to effectively drive today's customer conversation. Genesys software directs more than 100 million interactions every day, maximizing the value of customer engagement and differentiating the experience by driving personalization and multichannel customer service - and extending customer service across the enterprise to optimize processes and the performance of customer-facing employees. Go to www.genesys.com for more information.
Each product has its own documentation for online viewing at the Genesys Documentation website or on the Documentation Library DVD, which is available from Genesys upon request. For more information, contact your sales representative.

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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

3rd 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.

<table>
<tr><th colspan="2">Features Configuration in Genesys Configuration Environment</th></tr>
<tr><th colspan="2">General Features Supported By Phone (1pcc)</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Agent Login from the Phone</td><td>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=true

2. 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.

Notes:
• 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.</td></tr>
<tr><td>Agent State Control from the Phone</td><td>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.

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.</td></tr>
<tr><td>Auto-Answer</td><td>No configuration is required.</td></tr>
</table>

Alternate Ringtones	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** **Notes:** • 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 *Genesys SIP Proxy Deployment Guide* and *Genesys SIP Server High-Availability Deployment Guide*.
DTMF tones generation	No configuration is required.
Multiple calls on one extension	See Call Control using desktop client -> Attended transfer feature.
Message Waiting Indicator	Configure a voice mailbox for an Extension. In the TServer section of the DN object, configure: **gvm_mailbox=<voice mail box number>** For example: gvm_mailbox=12003, where 12003 is a mailbox number.

Shared Call Appearance (SCA)	1. Configure a Primary Shared Line DN: • 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 DNs: • 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	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 alphanumerical 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	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** **Note:** The "talk" value affects the Retrieve feature. See the Hold/Retrieve feature for information about setting the sip-cti-control option.
Conference	Deploy Genesys Media Server with MCU capabilities. See the *SIP Server Deployment Guide* for details.
Hold/Retrieve	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**
Make Outgoing Call	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)	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>** **Note:** It is not recommended to set the timer to "0" (zero).
Unattended transfer (Genesys Single-Step Transfer)	No configuration is required.
Semi-attended transfer (Genesys Blind Transfer)	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** **Note:** This option must be set on the DN object that represents a transfer destination party.

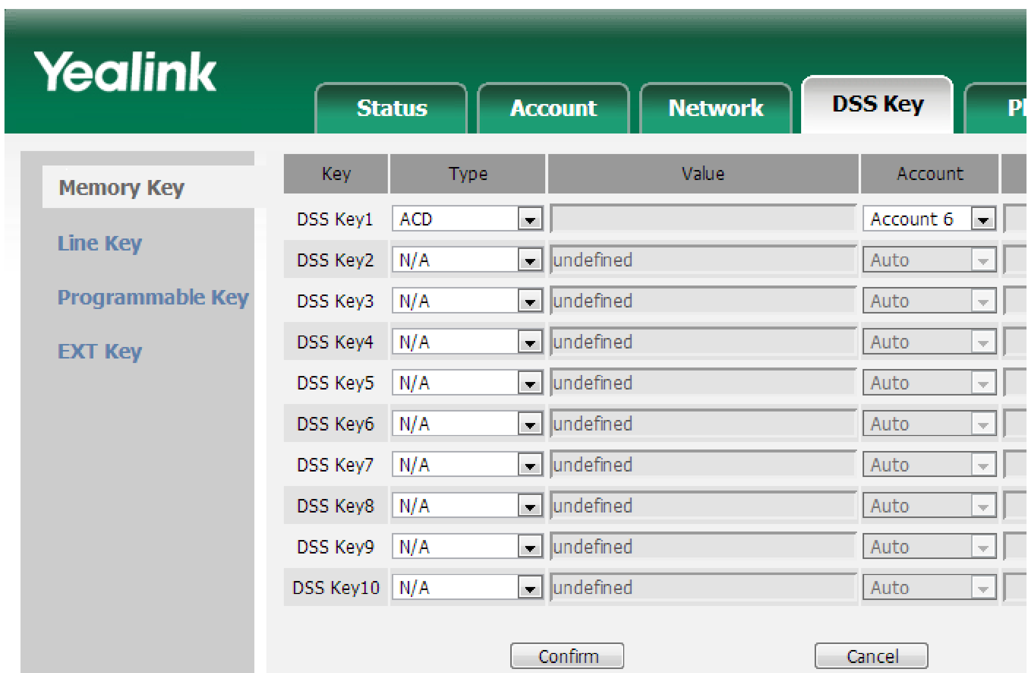
Attended transfer (Genesys Two-Step Transfer)	1. Enable dual-dialog to be supported on a DN for an attended transfer operation requested from a desktop client. In the TServer section of the DN object, configure: **dual-dialog-enabled=true** 2. Specify the call flow to process a make call/initiate consultative 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
Genesys Business Continuity	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**

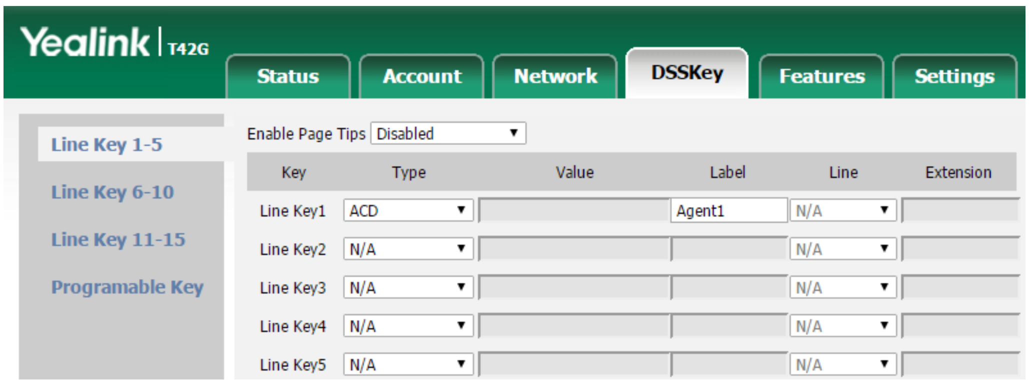
Example of the DN .cfg file:
[TServer]
authenticate-requests=invite,register
blind-transfer-enabled=true
contact=sip:1664@192.168.14.62:5063
dual-dialog-enabled=true
enable-agentlogin-subscribe=true
make-call-rfc3725-flow=1
refer-enabled=false
sip-alert-info=<http://192.168.14.62/Vin/>;info=ring8
sip-cti-control=talk,hold

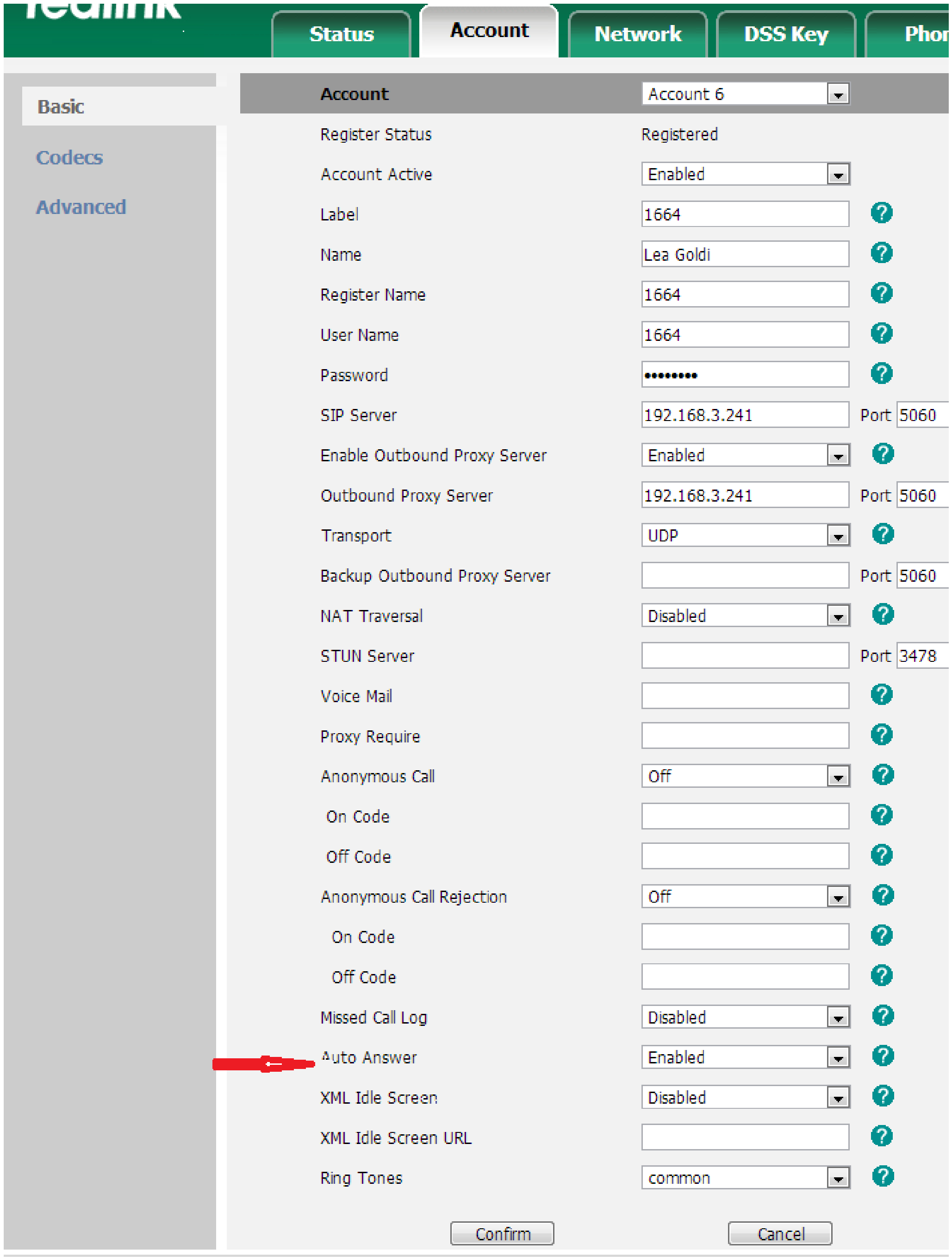
5 Yealink Phone Configuration

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)	1. Using the Web interface, **DSS Key -> Memory Key**, set the ACD feature on a DSS Memory key for an Account in advance: a. Set **DSS Key# -> Type** to **ACD**. b. Set **DSS Key# -> Account** to the Account number.  2. For Login/Logout/Available/Unavailable status, press **Memory Key 1** and enter Login Credentials from the phone touch screen: **User Name** and **Password**.
	1. Using the Web interface, **Settings -> Configuration -> Export CFG Configuration File -> Export**, edit exported the CFG file, and add the following parameters: account.1.acd.enable = 1 account.1.acd.available = 1

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

<table>
<tr>
<td>Auto-Answer</td>
<td>Using the Web interface, Account -> Basic -> Account #, enable Auto Answer for all SIP calls, by setting Auto Answer to Enabled.


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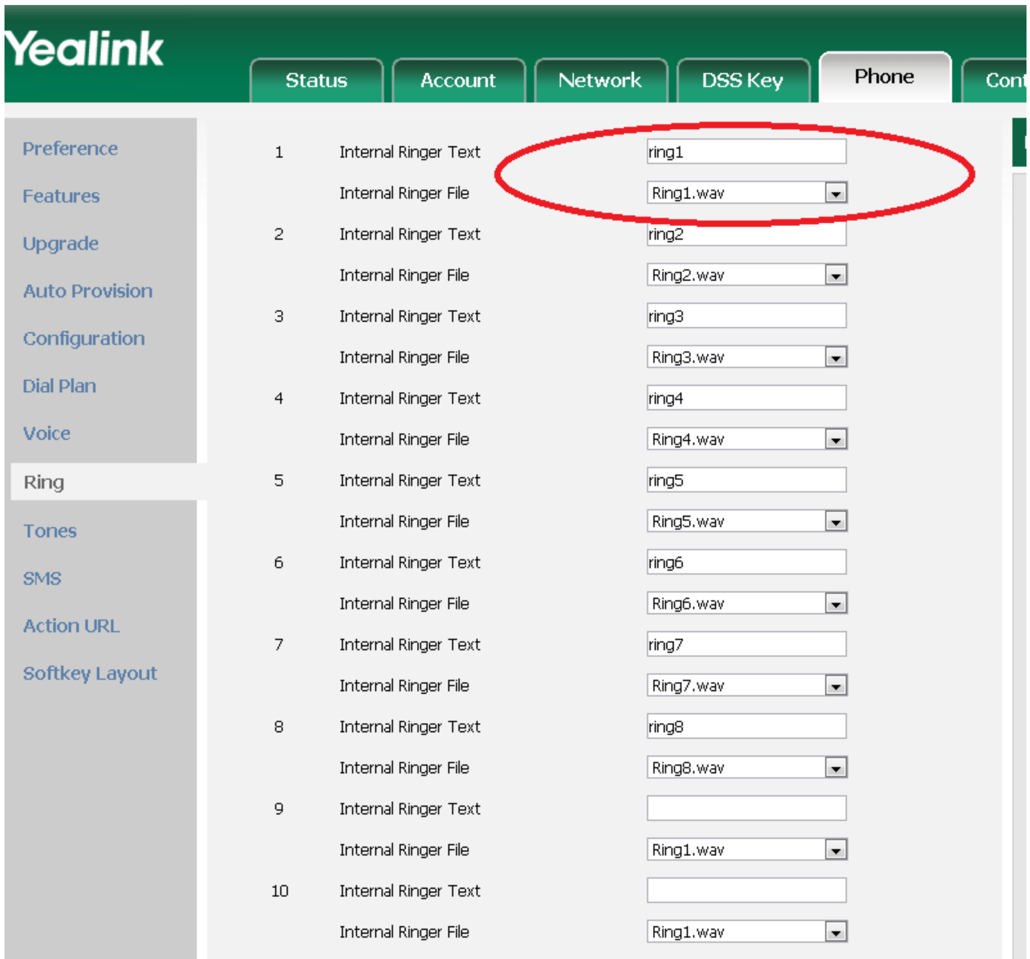
<td>Alternate Ringtones</td>
<td>
Using the Web interface, Phone -> Ring -> 1:

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



Note: A value of the Internal Ringer Text field must be used in the Genesys configuration environment. See Alternate Ringtones.
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<tr>
<td>Caller ID</td>
<td>No configuration is required.</td>
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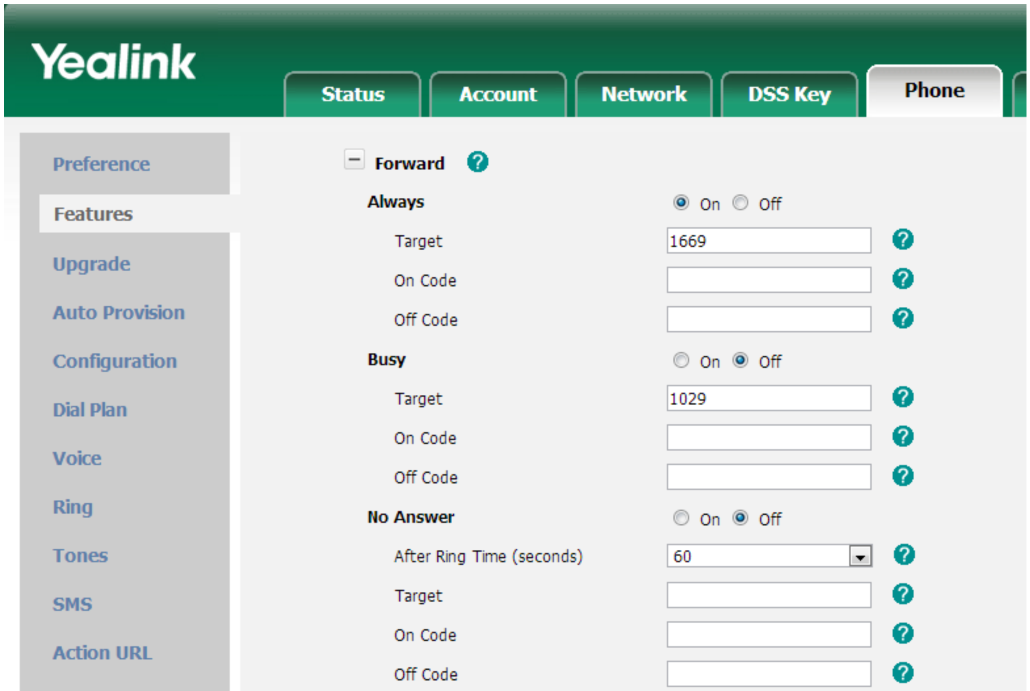
<tr>
<td>Call Forward</td>
<td>Using the Web interface, Phone -> Features -> Forward, configure call forward.



OR:

Using the phone, enable call forward by pressing the Menu button, then selecting Features -> Call Forward.</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>Using the phone, enable DND by pressing the DND button.</td>
</tr>
</table>

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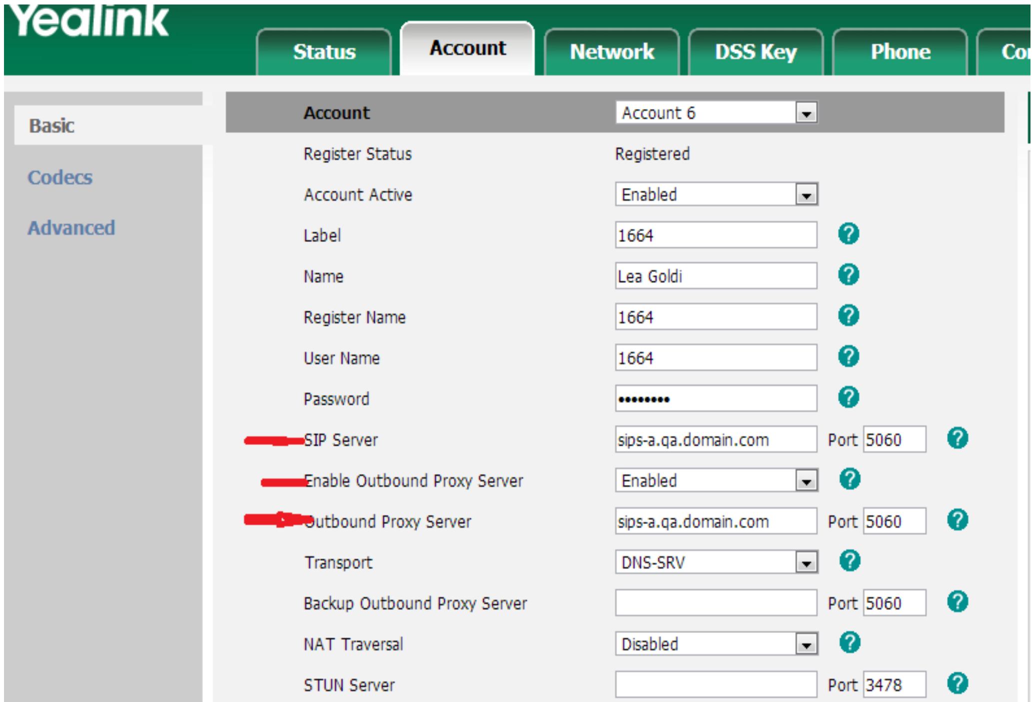
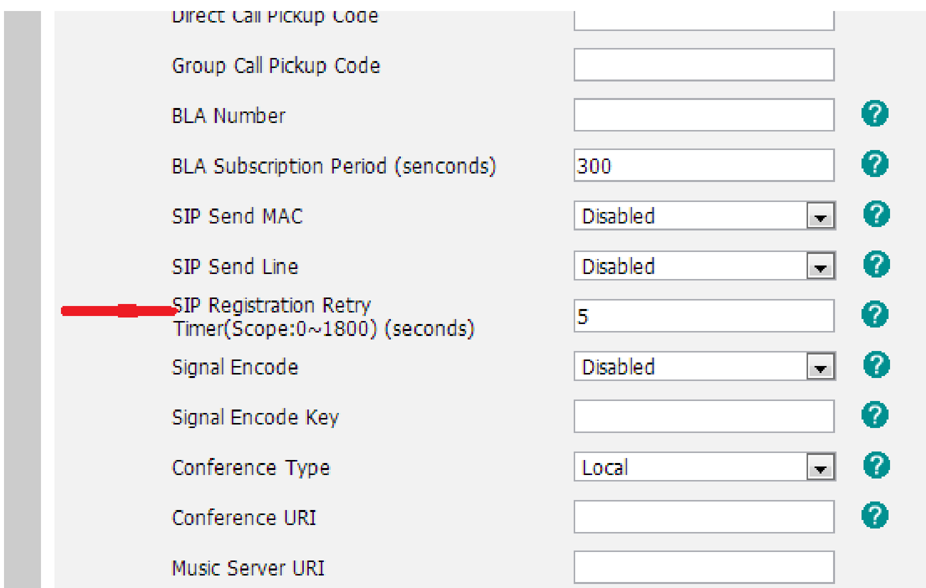
<td>DNS-based redundancy (using SIP Proxy)</td>
<td>

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.



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



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.

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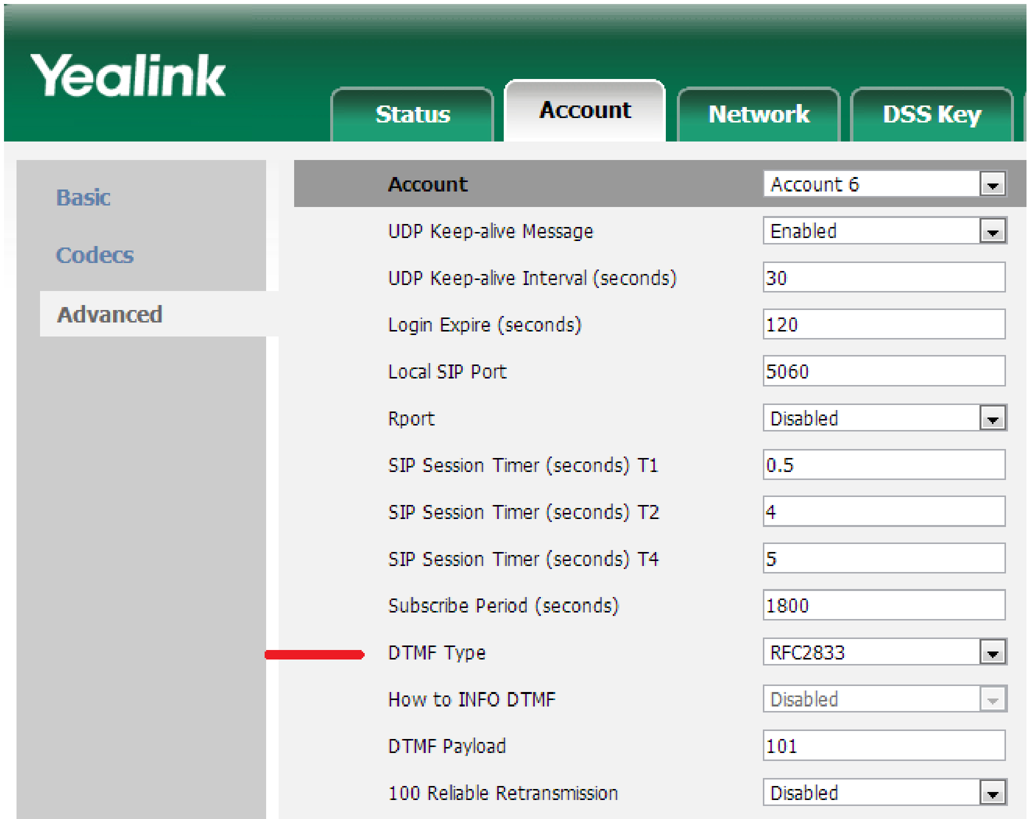
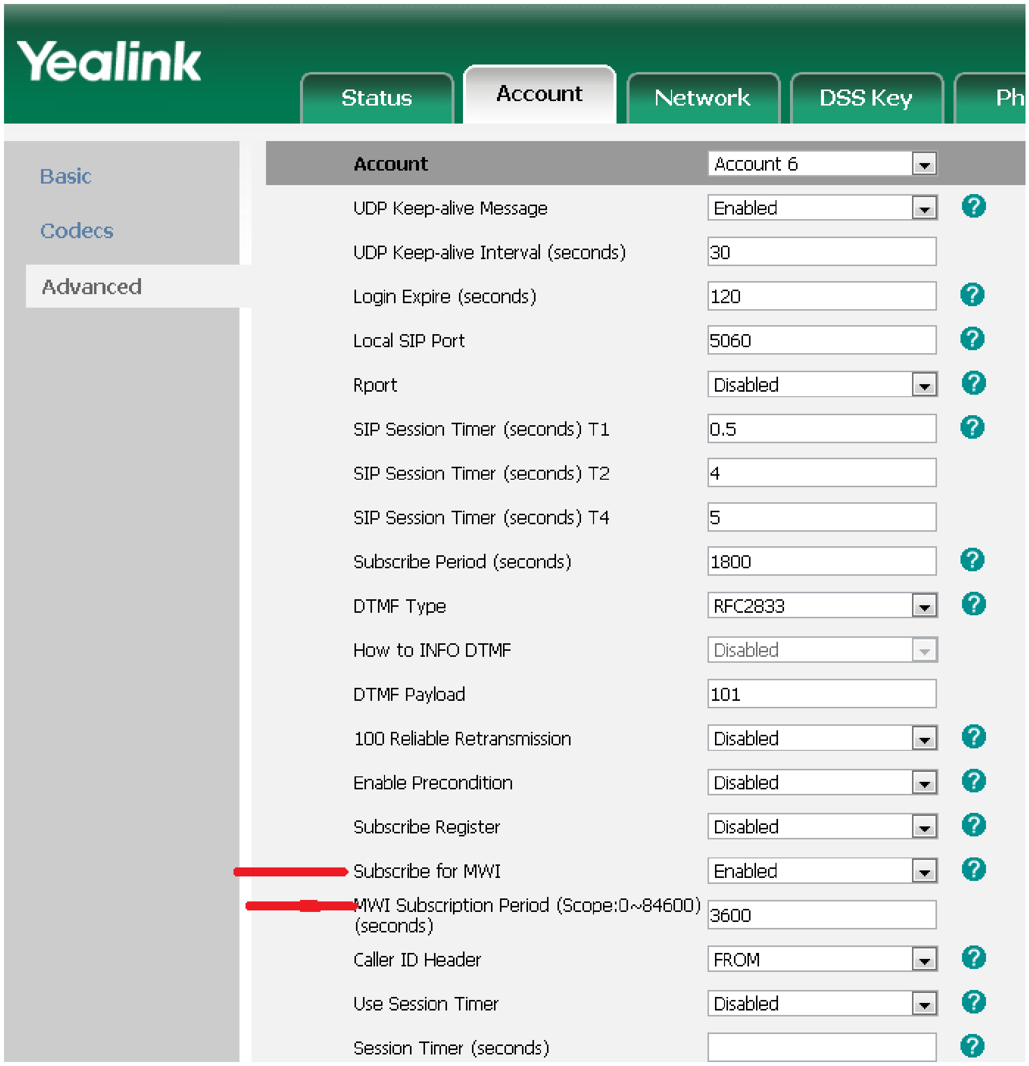
DTMF tones generation	Using the Web Interface, **Account -> Advanced -> Account #,** specify the method for DTMF tone generation in the **DTMF Type** field. 
Multiple calls on one extension	Genesys recommends having no more than two concurrent calls.

<table>
<tr>
<td>Message Waiting Indicator</td>
<td>

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).



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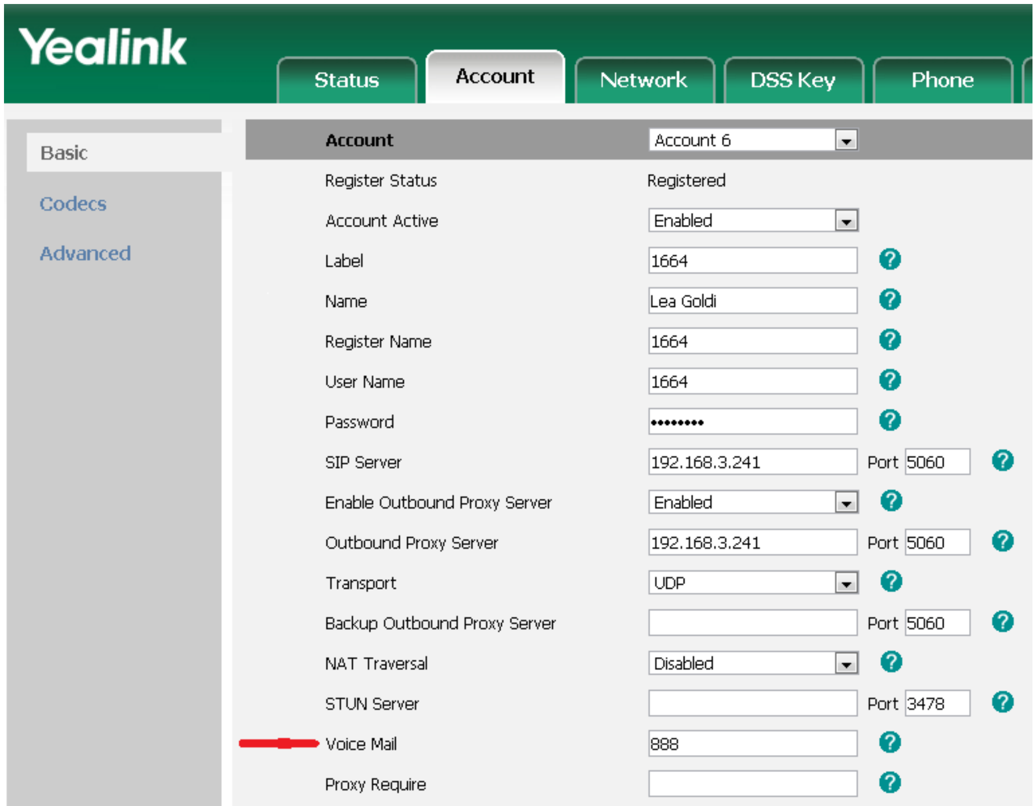
2. Using the Web Interface, **Account -> Basic -> Account #**, specify a number to call a Voice Mail System in the **Voice Mail** field.



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Shared Call Appearance (SCA)

(applies to T38G – 38.70.0.125)

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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).

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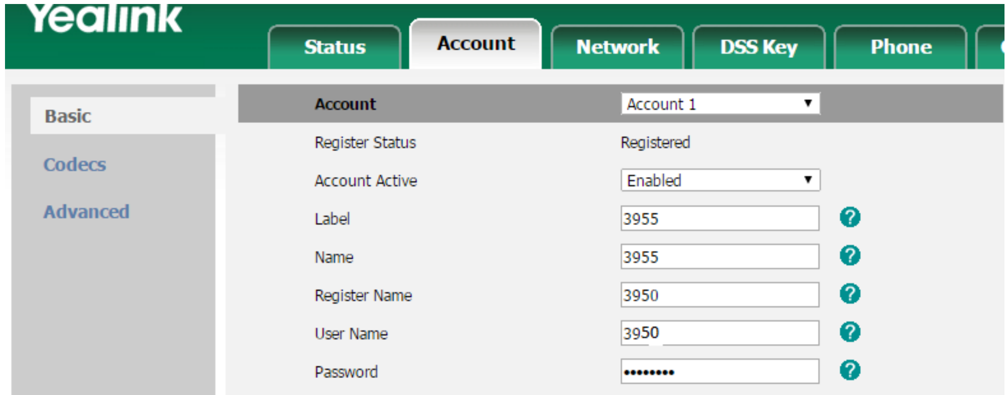


2. Using the Web interface, **Account -> Basic -> Account #**, configure the Secondary Shared Line for Shared Call Appearance:

   a. Specify the following properties:

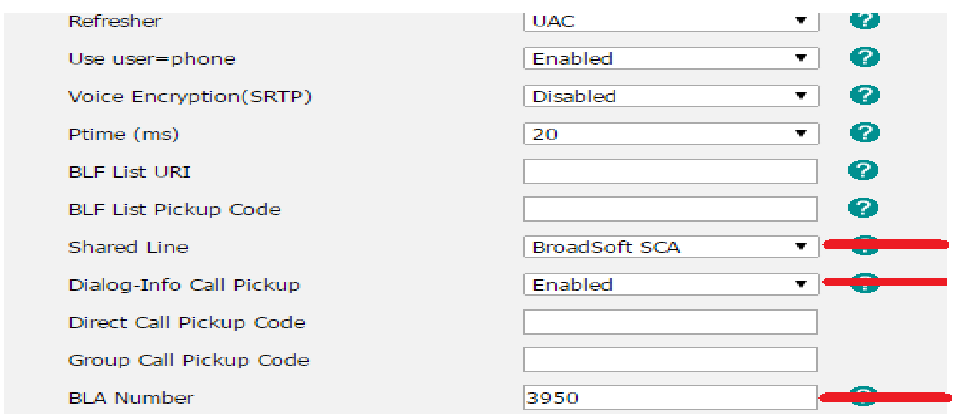
      Label: Specify the number to be shown on the LCD to identify the account.
      Name: Specify the Secondary line number.
      Register Name: Specify the Primary line number.

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

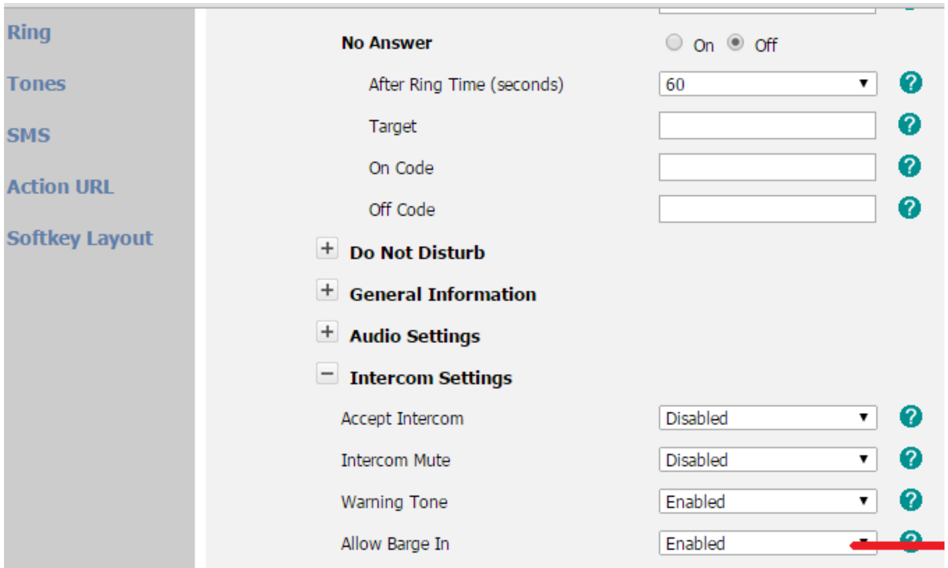
   c. If required, configure SIP authentication for the Shared line. (See SIP authentication in this table.)

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

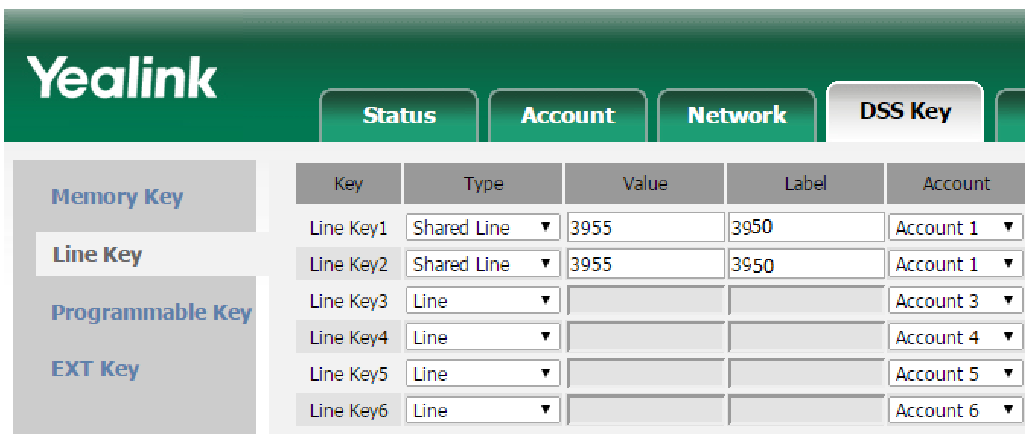
   d. Configure basic calling for the Primary Shared line. See Call Control Using Phone -> Basic calling (Incoming and outgoing calls).



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



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



<table>
<tr>
<td>Shared Call Appearance (SCA)

(applies to T42G-29.80.0.40)</td>
<td>

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

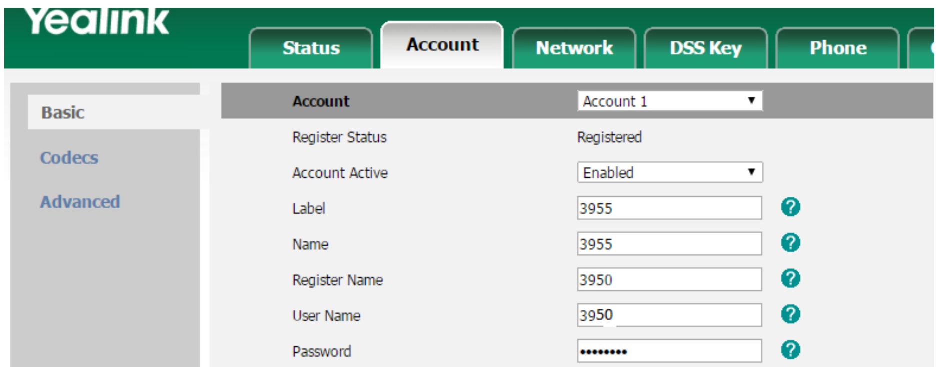
    a. Specify the following properties:

    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:

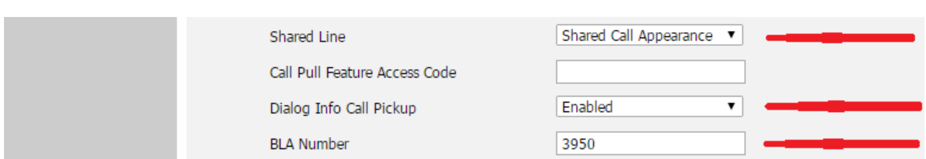
    Label: Specify the number to be shown on the LCD to identify the account.
    Name: Specify the Secondary line number.
    Register Name: Specify the Primary line number.

    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.)

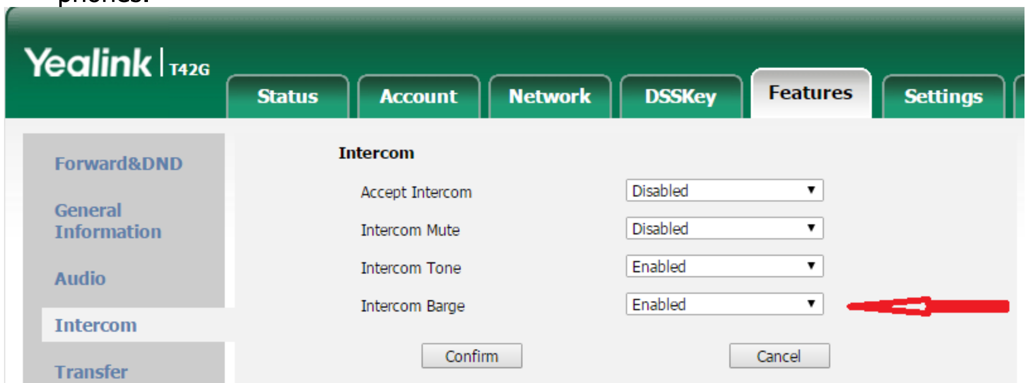
    Note: The Primary User ID must be used in the Authentication when you configure the Secondary Shared Line.

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

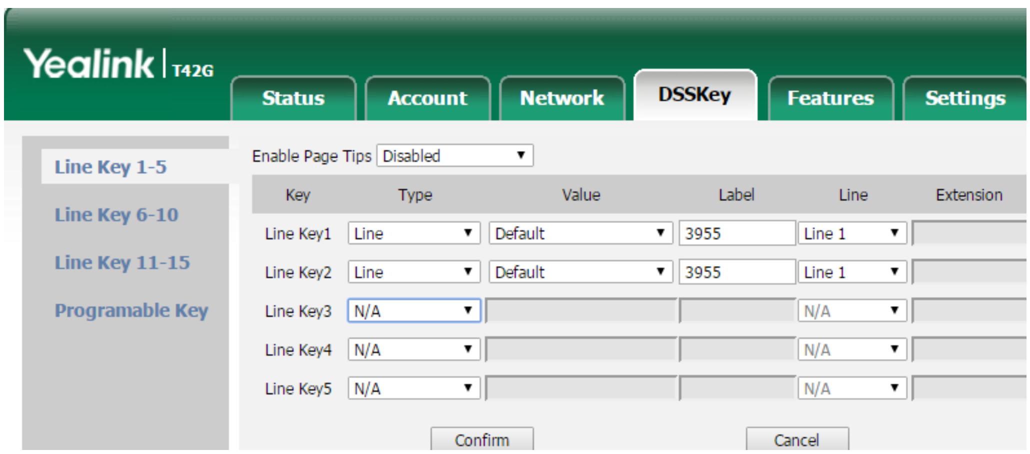


</td>
</tr>
</table>

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



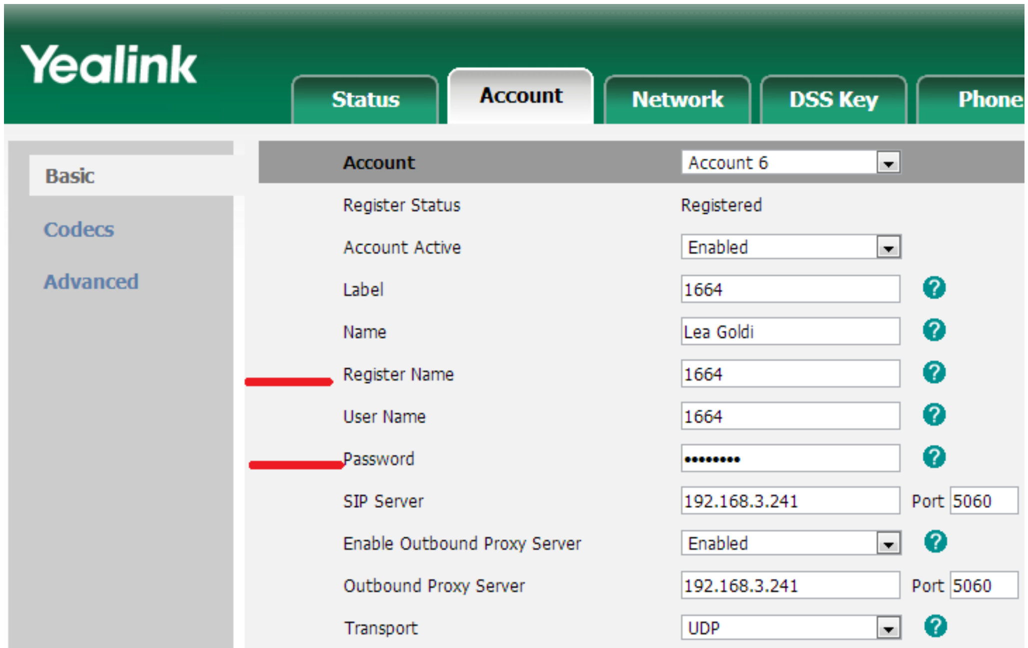
4. Using the Web interface, **DSS Key -> Line Key**, configure Lines (per Account).



5. Barging in an active call:
   When phone A has one active call, do the following:

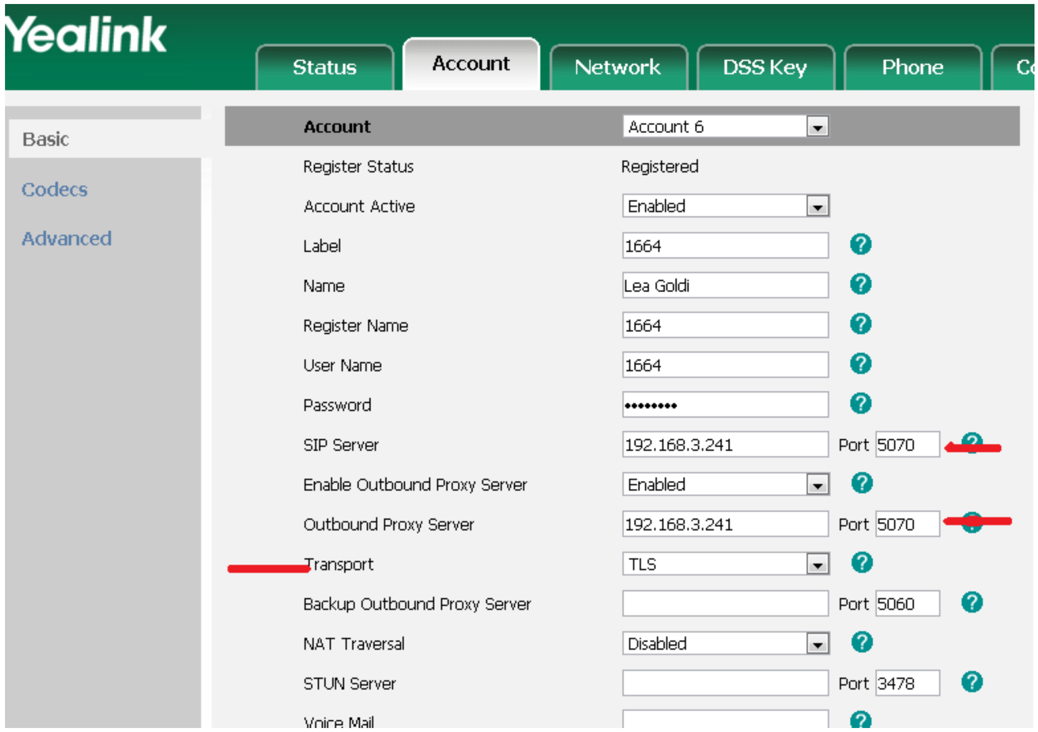
   a. Long press the desired line key on phone B. "Cancel, Call Pull, New Call, and Barge In" soft keys appear on the LCD screen of phone B.

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

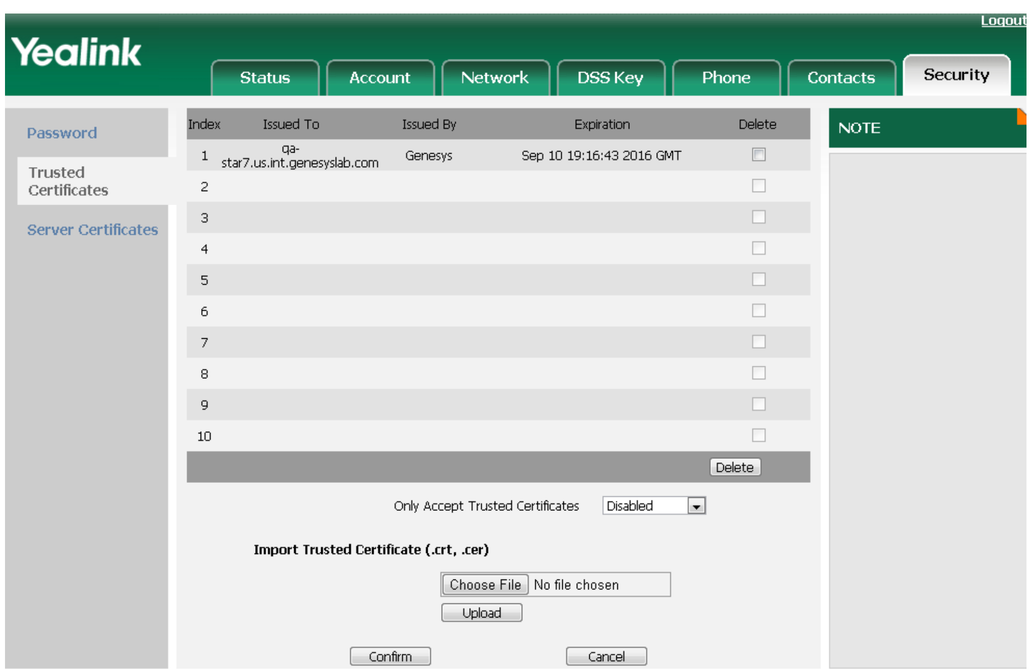
SIP authentication	Using the Web interface, **Account -> Basic -> Account #,** specify login credentials for SIP authentication in the **Register Name** and **Password** fields. **Note:** The **Password** parameter must have the same value as the **password** option configured in the DN object in the Genesys configuration environment. The **Register Name** parameter is used to authenticate line registration or an outgoing INVITE. 

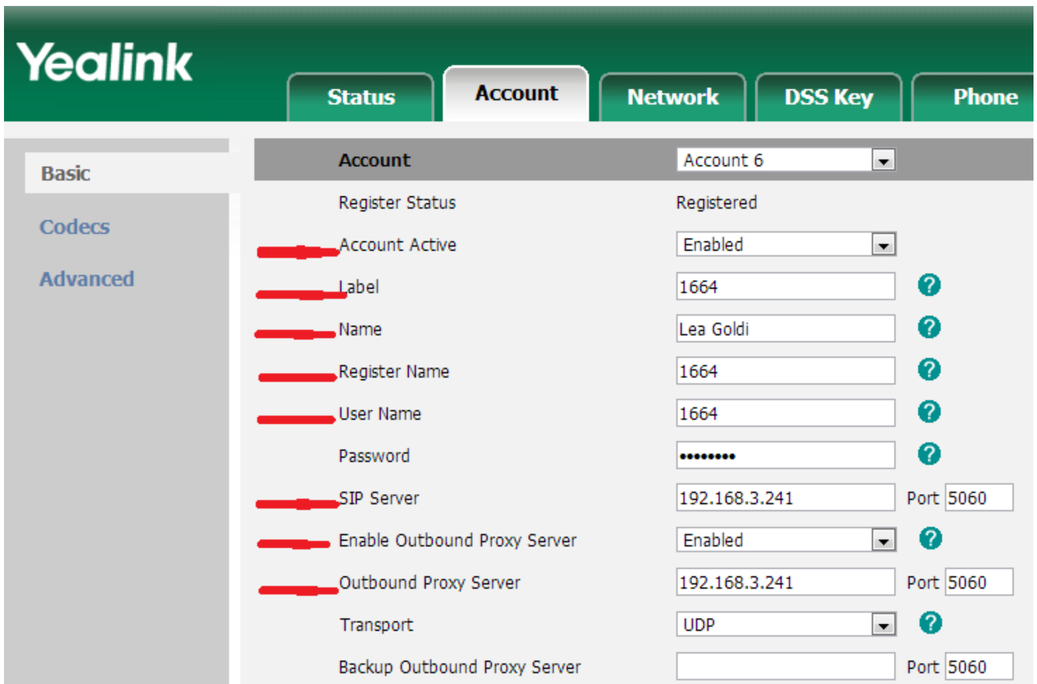
TLS/SRTP

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**.

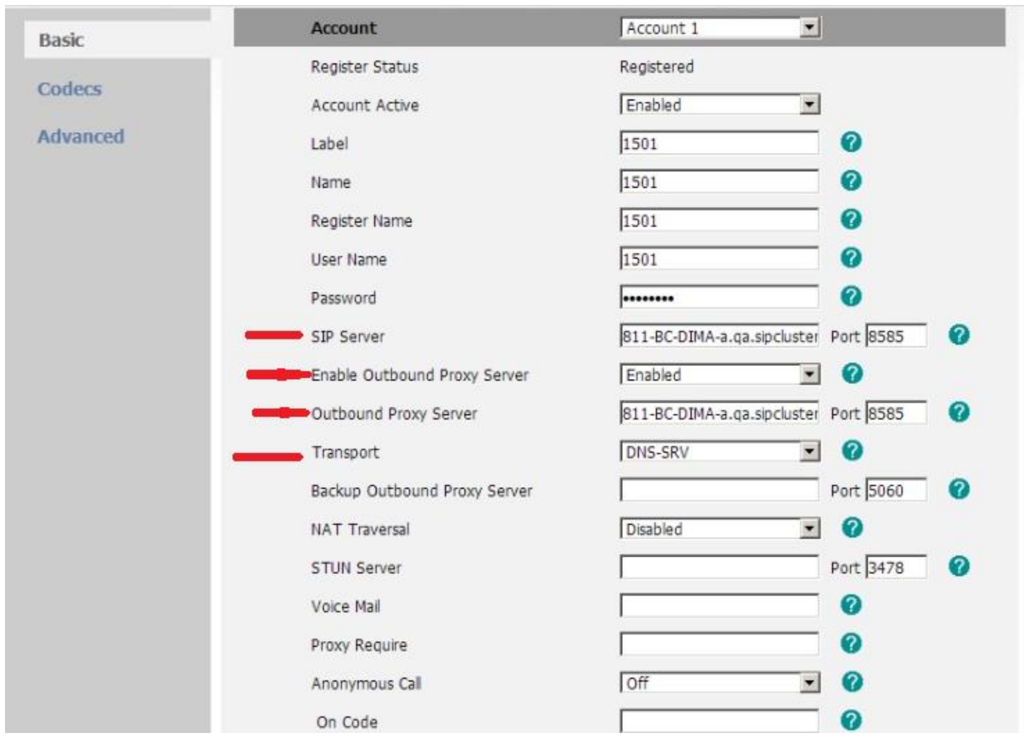


2. Using the Web interface, **Security -> Trusted Certificates:**
   a. Specify the custom CA Certificate (.crt file) by using **Choose File -> Upload**.
   b. Set **Only Accept Trusted Certificates** to **Disabled**.



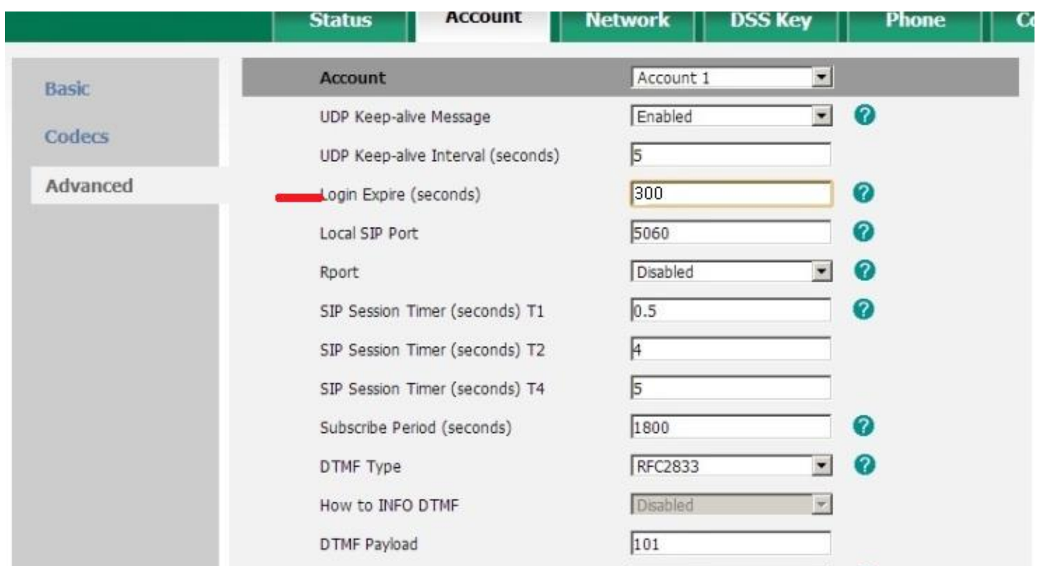
<table>
<tr><th colspan="2">Call Control Using Phone</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Basic calling (incoming and outgoing calls)</td><td>Using the Web interface, Account -> Basic -> Account #:
a. Set Account Active and Enabled Outbound Proxy Server to Enabled.
b. Specify Label, Name, Register Name, and User Name.
c. Specify the IP address (FQDN) and port of SIP Server in the SIP Server and Outbound Proxy Server fields.

</td></tr>
<tr><td>Conference</td><td>No configuration is required.</td></tr>
<tr><td>Hold/Retrieve</td><td>No configuration is required.</td></tr>
<tr><td>Unattended (blind) transfer</td><td>Using the phone, press Transfer, enter the number, press Transfer again.</td></tr>
<tr><td>Semi-attended (two-step) transfer</td><td>Using the phone, press Transfer, enter the number, press OK, and press Transfer while receiving ringback.</td></tr>
<tr><td>Attended (consultative) transfer</td><td>Using the phone, press Transfer, enter the number, press OK, and press Transfer again when the party answers.</td></tr>
</table>

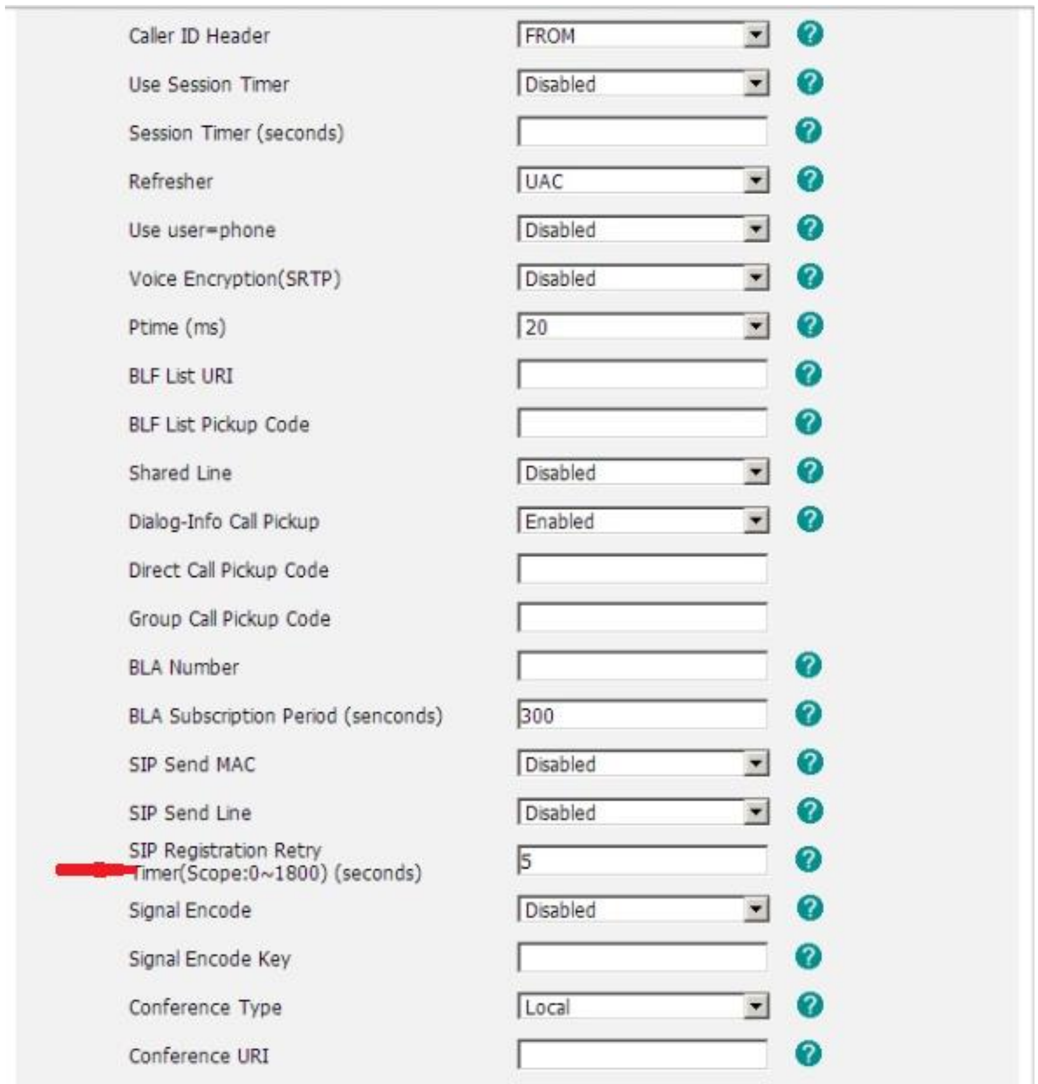
Call Control Using Desktop Client	
Feature	**Key Actions and Procedures**
Answer Incoming Call	No configuration is required.
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Make Outgoing Call	See the Basic calling (incoming and outgoing calls) feature.
Remote Auto-Answer (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	1. Using the Web interface, **Account -> Basic -> Account #:** a. Specify the IP address (FQDN) and port of SIP Server peers in the **SIP Server** and **Outbound Proxy Server** fields. b. Specify **Transport**. **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).



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.




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 (senconds)
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.