Genesys Application Note

AudioCodes SIP Phones
With Genesys SIP Server

Document version 2.1

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Table of Contents

1 Summary

AudioCodes 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.

This application note details the supported features of AudioCodes 440HD with 2.2.16 version of firmware, and includes reference configuration examples.

AudioCodes 405, 405HD, 420HD, and 430HD with 2.2.16 version of firmware are also supported as the phone runs the same version of firmware.

The supporting versions of Genesys components include SIP Server 8.1.x (8.1.1 recommended), SIP Feature Server 8.1.x (8.1.2 recommended), Media Server (8.1.x and 8.5.x), and SIP Proxy 8.1.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	No
Alternate Ringtones	No
Caller ID	Yes
Call Forward	Yes
Do Not Disturb	Yes
DNS-based redundancy (using SIP Proxy)	Yes
DTMF tones generation	Yes
IPv6 support	No
Multiple calls on one extension	Yes
Message Waiting Indicator	Yes
Shared Call Appearance (SCA)	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	Yes
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 which 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: A phone can generate DTMF tone through RTP when tone generation was 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 AudioCodes 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 AudioCodes SIP Phones

3rd Party Hardware Components		
Model	**Version**	**Notes**
AudioCodes 440HD, 430HD, 420HD, 405	2.2.16.428	2.2.16 or later supported
AudioCodes 440HD, 430HD, 420HD, 405	2.2.16.142.12	2.2.16 or later supported
AudioCodes 440HD, 420HD, 405	2.2.12	2.2.12 or later supported

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 (see Section 2.1, above) within a Genesys configuration environment.

Features can be configured in the SIP Server Switch on a DN object of 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 Name 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>Caller ID</td><td>No configuration is required.</td></tr>
<tr><td>Call Forward</td><td>No configuration is required.</td></tr>
<tr><td>Do Not Disturb</td><td>No configuration is required.</td></tr>
</table>

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=1502, where 1502 is a mailbox number.
Shared Call Appearance (SCA), in SIP Server standalone deployments	**Note:** Only AC 440HD supports full SCA functionality. 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 the **shared-line-capacity** option. • 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 the **shared-line-number** option.
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:** The string must match the phone setting in **Configuration -> Voice Over IP ->Line Settings -> Authentication User Name** and Authentication **Password**.
TLS/SRTP	See the Transport Layer Security for SIP Traffic chapter in the Genesys 8.1 SIP Server Deployment Guide for details.

Secure SIP (SIPS) support, in accordance with RFC 5630	No configuration required.
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 value that SIP Server will add in the Alert-Info header of the INVITE message, which it sends to the SIP Endpoint. In the TServer section of the DN object, configure: **sip-alert-info=info=alert-autoanswer**
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 or REFER 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 ->** to use **INVITE** method or **refer-enabled=true ->** to use **REFER** method
Remote DTMF tones generation	Configure SIP Server to remotely control DTMF generation on the SIP phone. In the TServer section of the DN object, configure: **sip-cti-control=dtmf**
Genesys Business Continuity (Simultaneous, dual-registration mode)	Configure SIP Server to forward an incoming call to the second SIP Server peer if SIP Server determines that there is no agent logged into the DN. In the TServer section of the DN object, configure: **dr-forward=no-agent**

Genesys Business Continuity (Primary-Fallback, single-registration mode)	Configure SIP Server to forward an incoming call to the second SIP Server peer if SIP Server determines that there is SIP registration. In the TServer section of the DN object, configure: **dr-forward=oos** **Note:** Agent State Control from the Phone functionality only supported in this mode.

Example of the DN .cfg file:
[TServer]
authenticate-requests=invite,register
blind-transfer-enabled=true
contact=sip:1502@172.21.82. 86:2048
dual-dialog-enabled=true
enable-agentlogin-subscribe=true
make-call-rfc3725-flow=1
refer-enabled=false
sip-alert-info= info=alert-autoanswer
sip-cti-control=talk,hold

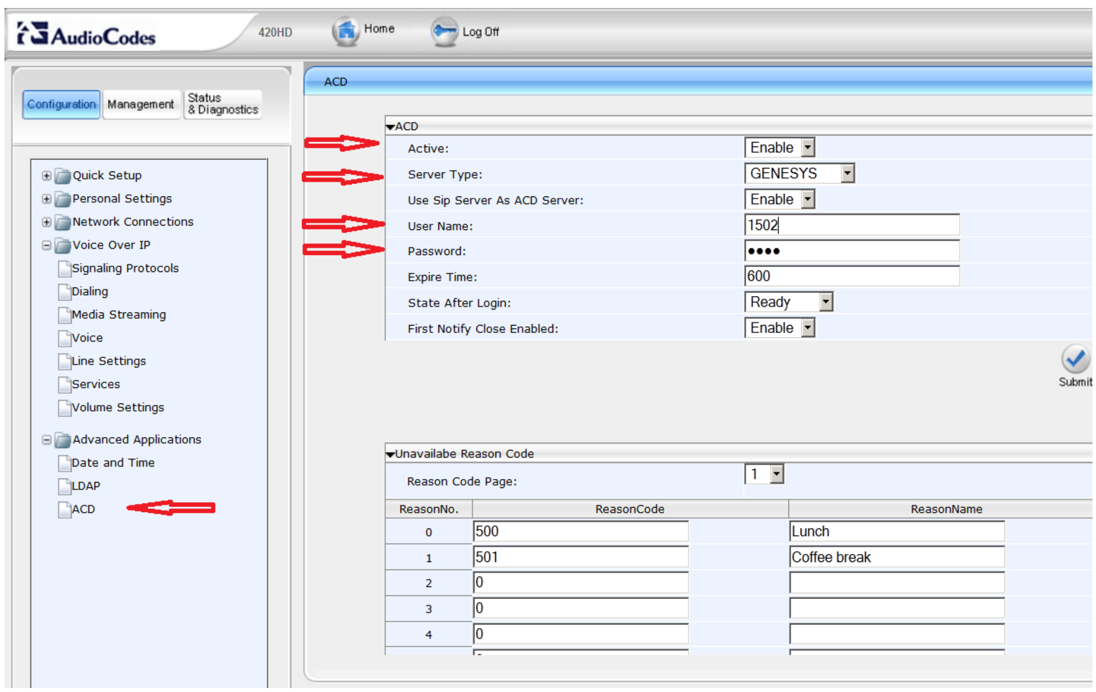
5 AudioCodes Phone Configuration

This section describes how to configure features represented in the Feature Chart (see Section 2.1) using the phone Web interface.

The following table displays screenshots of the Web interface of the AudioCodes 420HD.

<table>
<tr><th colspan="2">AudioCodes Phone Configuration</th></tr>
<tr><th colspan="2">General Features Supported By Phone</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Agent Login from the Phone</td><td>

1. Using the Web interface, **Configuration -> Advanced Applications -> ACD,** set the ACD feature:
   a. Set **Active** to **Enable**.
   b. Set **Server Type** to **Genesys**
   c. Enter the User **Name** and **Password**.



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

</td></tr>
<tr><td>Agent State Control from the Phone</td><td>

1. Using the Web interface, **Configuration -> Advanced Applications -> ACD,** set the ACD feature:
   a. Set **Active** to **Enable**.
   b. Set **Server Type** to **BROADSOFT.**

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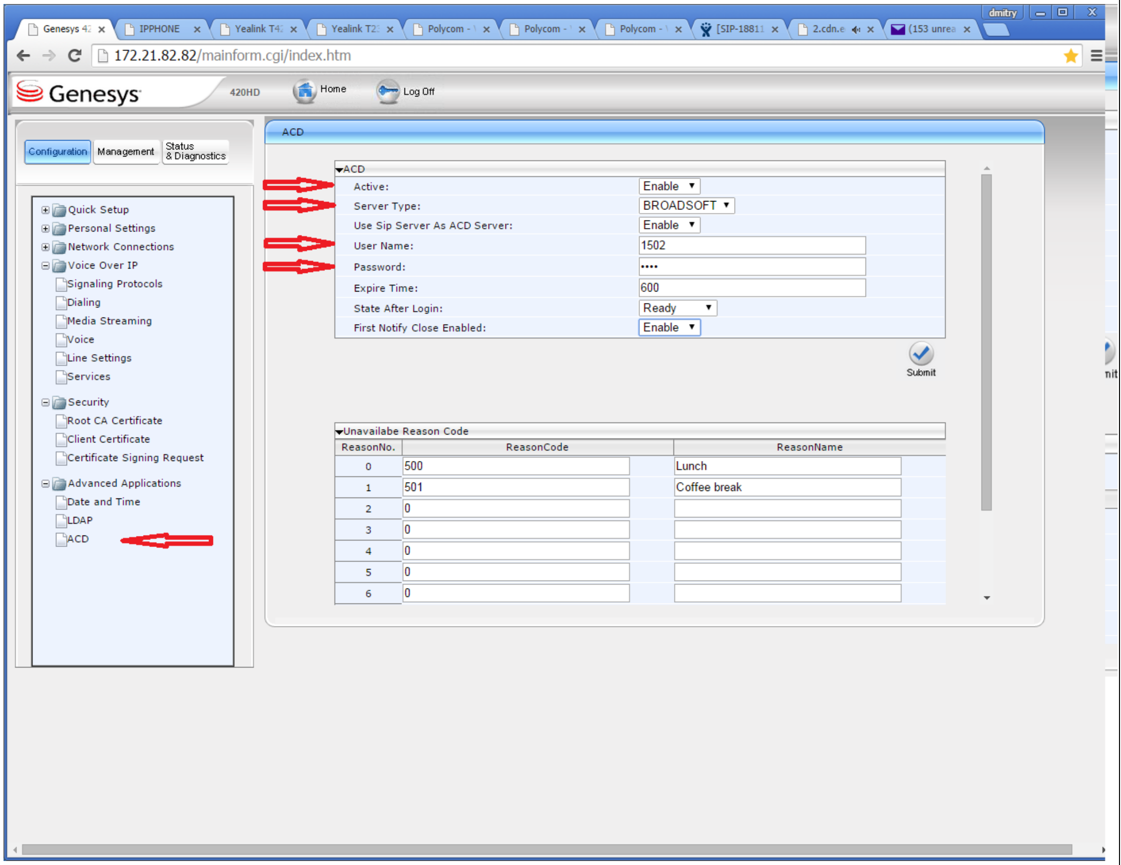
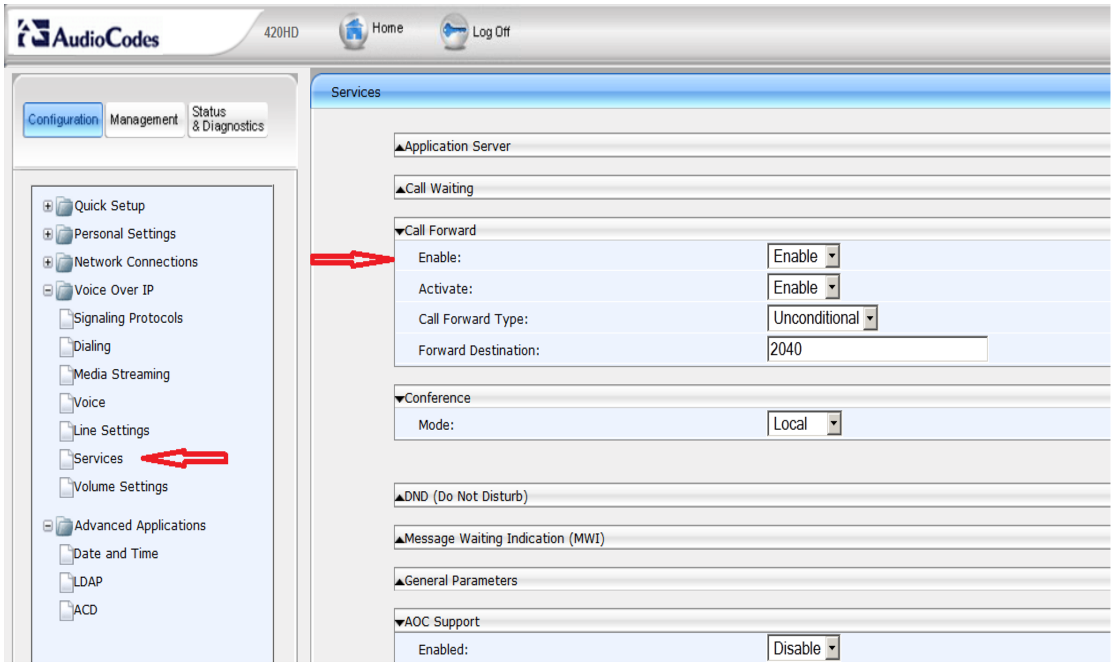
</table>

	c. Enter the **User Name** and **Password**.  2. For Login/Logout/Available/Unavailable status, press **Login** and enter Login Credentials from the phone touch screen: **User Name** and **Password**.
Caller ID	No configuration is required.

<table>
<tr>
<td>Call Forward</td>
<td>Using the Web interface, Configuration -> Voice Over IP -> Services -> Call Forward, enable call forward by selecting Enable.


OR:
Using the phone, enable call forward from the phone by pressing the Fwd button.</td>
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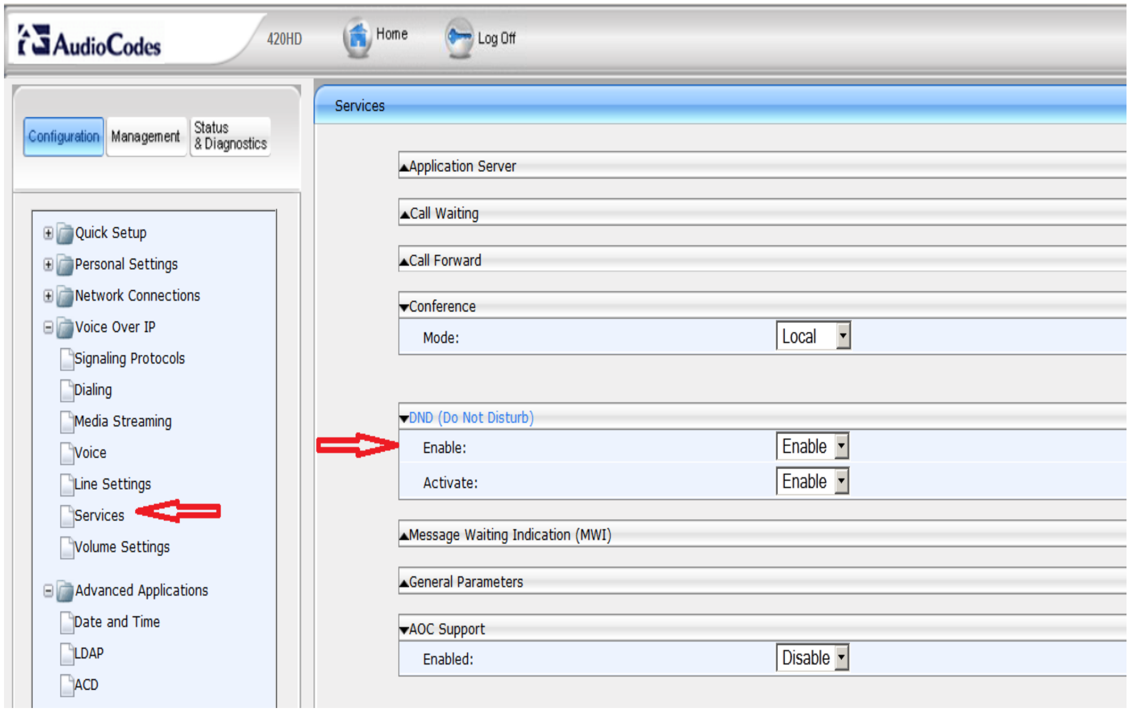
<table>
<tr><td>Do Not Disturb</td><td>Using the Web interface, Configuration -> Voice Over IP -> Services -> DND (Do Not Disturb), enable DND by selecting Enable.



OR:

Using the phone, enable DND by pressing the DND button.</td></tr>
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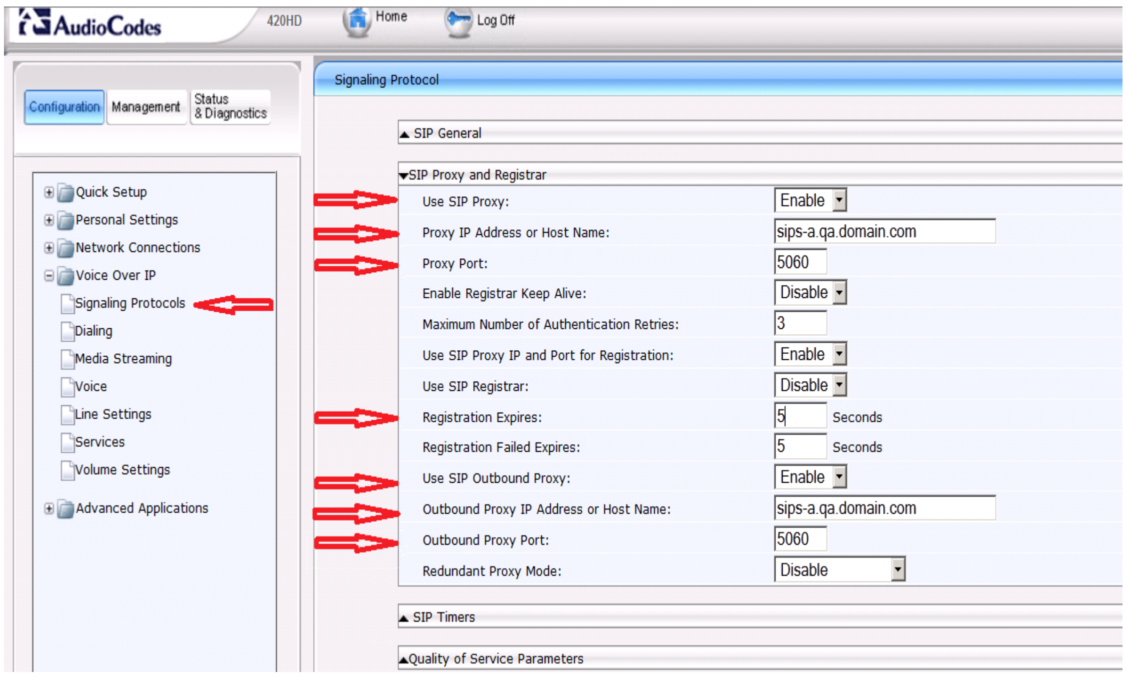
<td>DNS-based redundancy (using SIP Proxy)</td>
<td>

Using the Web interface, **Configuration -> Voice Over IP-> Signaling Protocols -> SIP Proxy and Registrar:**

1. Set **Use SIP Proxy** and **Use SIP Outbound Proxy** to **Enable**.
2. Specify the IP address (FQDN) of the SIP Proxy pool in the **Proxy IP Address or Host Name** and **Outbound Proxy IP Address or Host name** fields.
3. Specify the SIP Proxy port in the **Proxy Port** and **Outbound Proxy Port** fields.
4. Set **Registration Expires** to **5** seconds.



Notes:

- The IP Address fields have 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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<tr>
<td>DTMF tones generation</td>
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Using the phone's configuration file, modify the line to specify the method for DTMF tone generation, as follows:

- **voip/media/out_of_band_dtmf=RFC2833**

or:

- **voip/media/out_of_band_dtmf=VIA_SIP**

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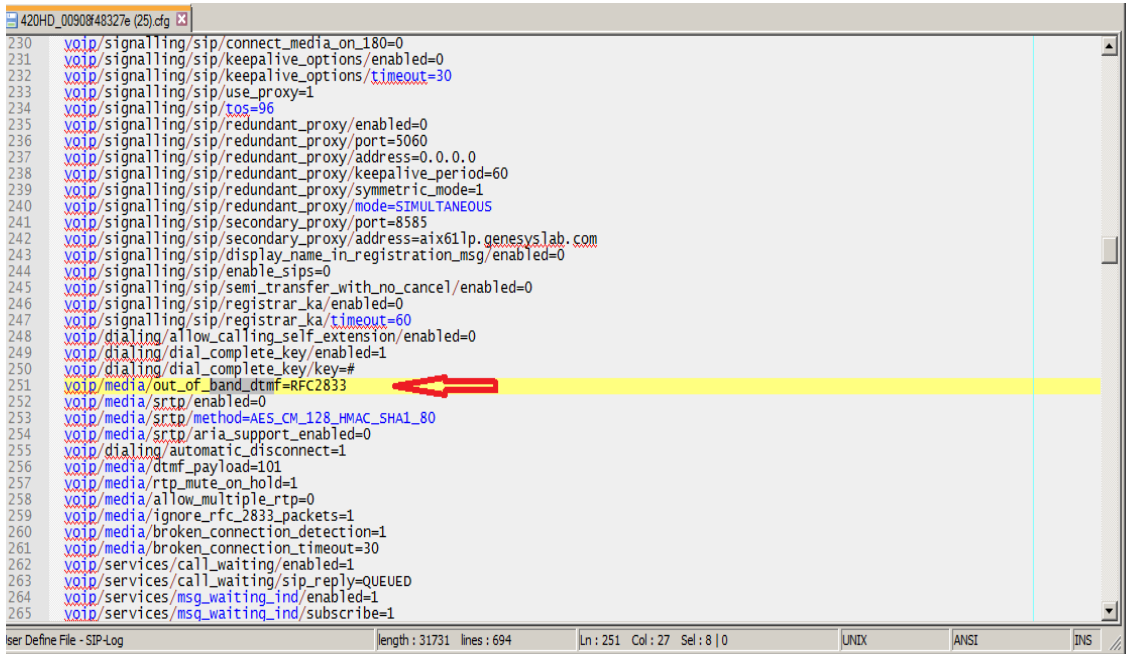
</table>

	 420HD_00908f48327e (25).cfg voip/signalling/sip/connect_media_on_180=0 voip/signalling/sip/keepalive_options/enabled=0 voip/signalling/sip/keepalive_options/timeout=30 voip/signalling/sip/use_proxy=1 voip/signalling/sip/tos=96 voip/signalling/sip/redundant_proxy/enabled=0 voip/signalling/sip/redundant_proxy/port=5060 voip/signalling/sip/redundant_proxy/address=0.0.0.0 voip/signalling/sip/redundant_proxy/keepalive_period=60 voip/signalling/sip/redundant_proxy/symmetric_mode=1 voip/signalling/sip/redundant_proxy/mode=SIMULTANEOUS voip/signalling/sip/secondary_proxy/port=8585 voip/signalling/sip/secondary_proxy/address=aix61lp.genesyslab.com voip/signalling/sip/display_name_in_registration_msg/enabled=0 voip/signalling/sip/enable_sips=0 voip/signalling/sip/semi_transfer_with_no_cancel/enabled=0 voip/signalling/sip/registrar_ka/enabled=0 voip/signalling/sip/registrar_ka/timeout=60 voip/dialing/allow_calling_self_extension/enabled=0 voip/dialing/dial_complete_key/enabled=1 voip/dialing/dial_complete_key/key=# voip/media/out_of_band_dtmf=RFC2833 voip/media/srtp/enabled=0 voip/media/srtp/method=AES_CM_128_HMAC_SHA1_80 voip/media/srtp/aria_support_enabled=0 voip/dialing/automatic_disconnect=1 voip/media/dtmf_payload=101 voip/media/rtp_mute_on_hold=1 voip/media/allow_multiple_rtp=0 voip/media/ignore_rfc_2833_packets=1 voip/media/broken_connection_detection=1 voip/media/broken_connection_timeout=30 voip/services/call_waiting/enabled=1 voip/services/call_waiting/sip_reply=QUEUED voip/services/msg_waiting_ind/enabled=1 voip/services/msg_waiting_ind/subscribe=1 ser Define File - SIP-Log length : 31731 lines : 694 Ln : 251 Col : 27 Sel : 8 | 0 UNIX ANSI INS
Multiple calls on one extension	Genesys recommends having no more than two concurrent calls.
Message Waiting Indicator	Using the Web interface, **Configuration -> Voice Over IP -> Services -> Message Waiting Indication (MWI):** 1. Specify the number to call a Voice Mail System in the **Voice Mail Number** field. 2. Enable the Voice Mail System by setting **Activate** to **Enable**. 3. Set **Subscribe To MWI** to **Enable**. 4. Specify the **MWI Server IP Address or Host Name**. 5. Specify the **MWI Server Port**. 6. If required, set **MWI Subscribe Expiry Time** in seconds (default 3600 sec).

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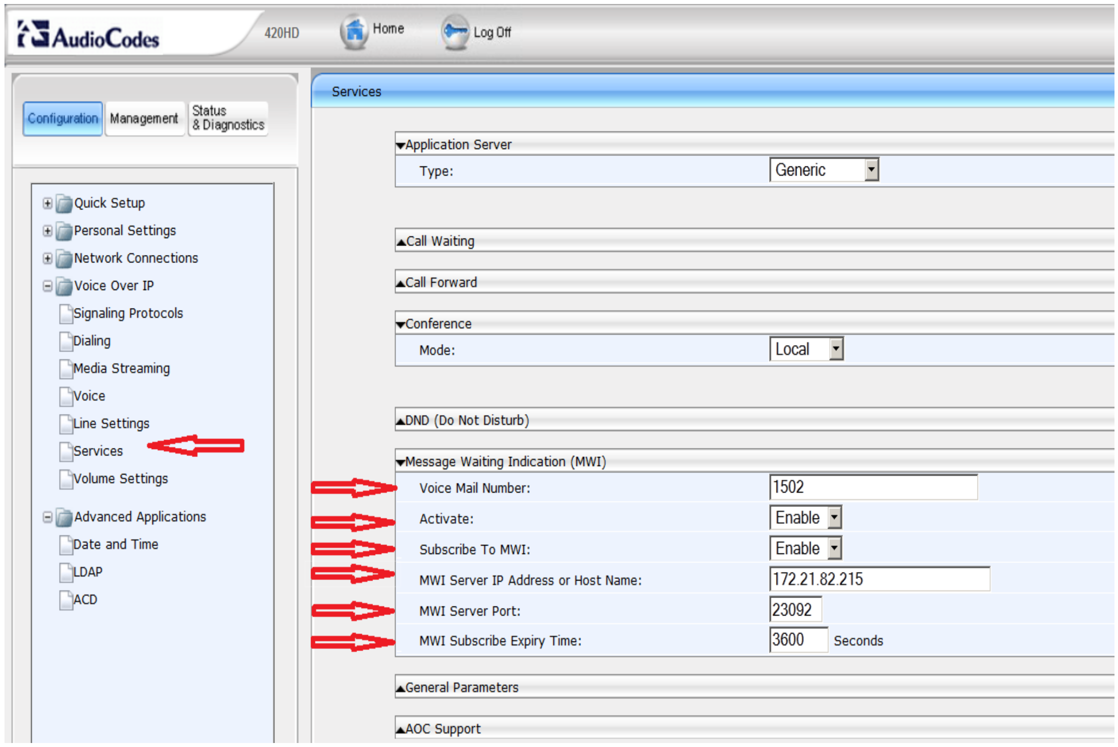

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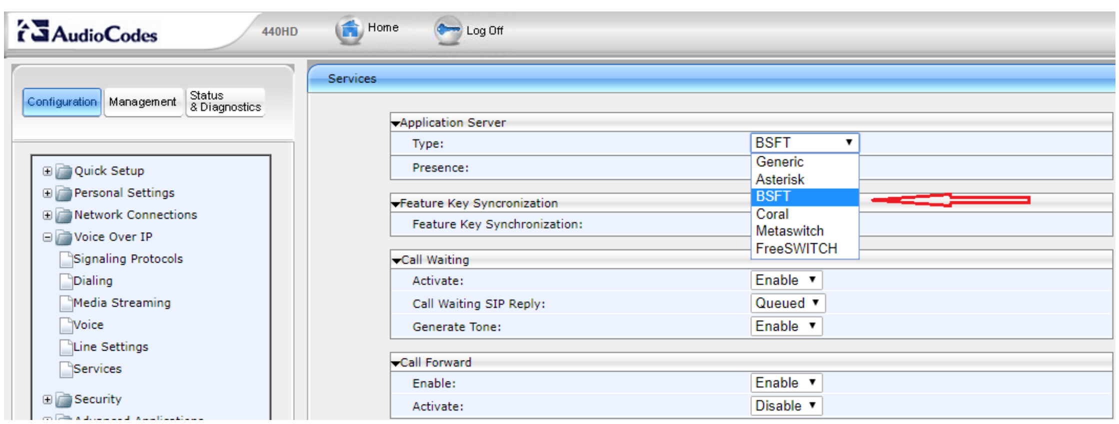
<tr>
<td>Shared Call Appearance (SCA)</td>
<td>
Important: Only AC 440HD supports SCA and only in SIP Server standalone deployments. AC 440HD does not support SCA in Business Continuity deployments.

1. Using the Web interface, Configuration -> Voice Over IP -> Services -> Application Server, select BSFT as the type.


2. Using the Web interface, Configuration -> Voice Over IP -> Line Settings, configure the Primary Shared Line for Shared Call Appearance:
a. Select Shared as the line type.
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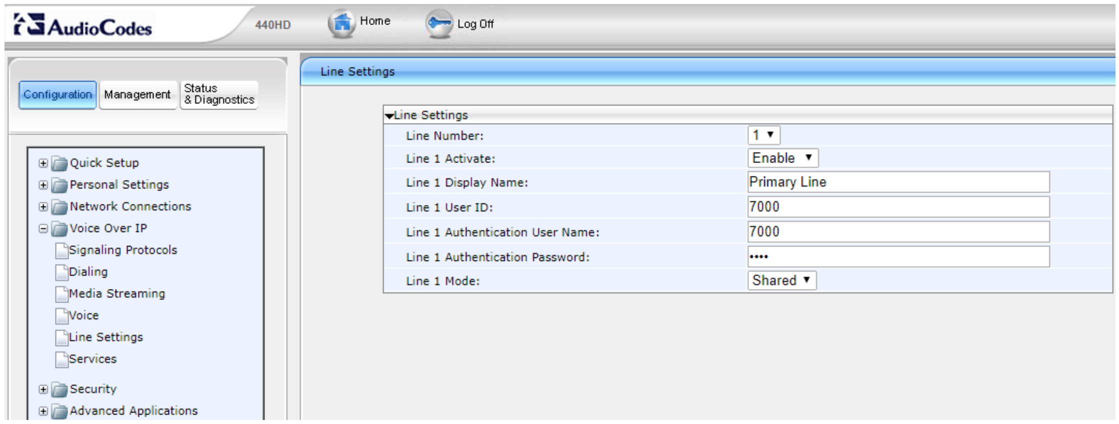
</table>



b. If required, configure SIP authentication for the Primary Shared Line. (See SIP authentication in this table.)

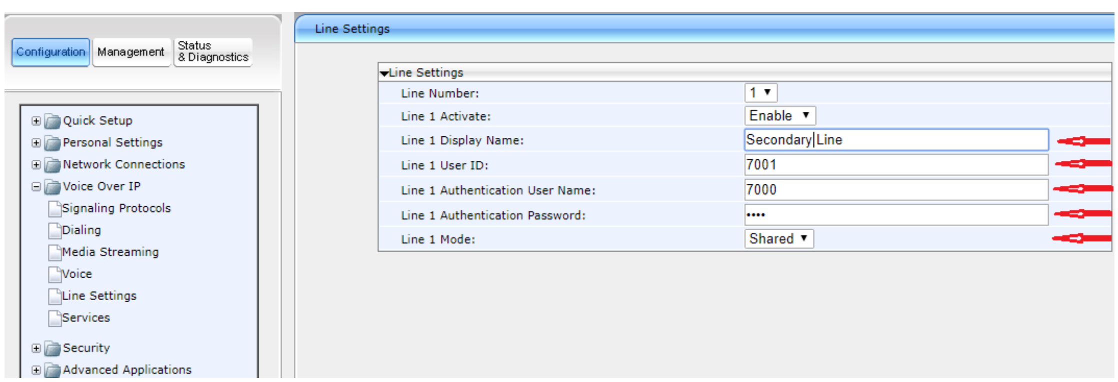
c. Configure basic calling for the Primary Shared Line. See Call Control Using Phone -> Basic calling (incoming and outgoing calls).

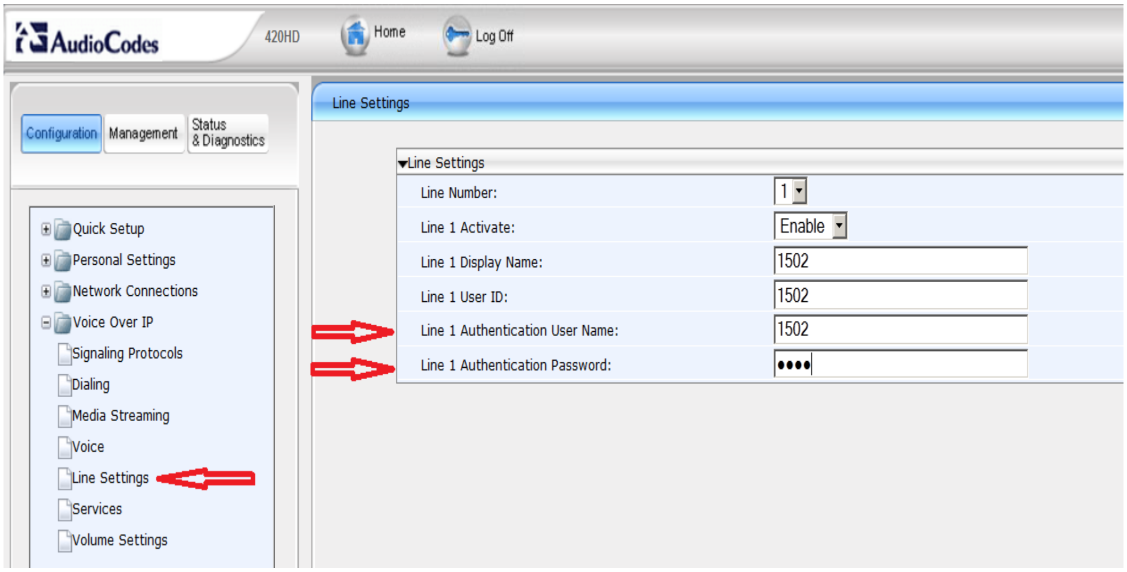
3. Using the Web interface, configure another phone as the Secondary Shared Line for Shared Call Appearance:

a. Select **Shared** as the line type.

b. If required, configure SIP authentication for the Secondary Shared Line. (See SIP authentication in this table.)
Note: The Primary User Name must be used in the Authentication when you configure the Secondary Shared Line.

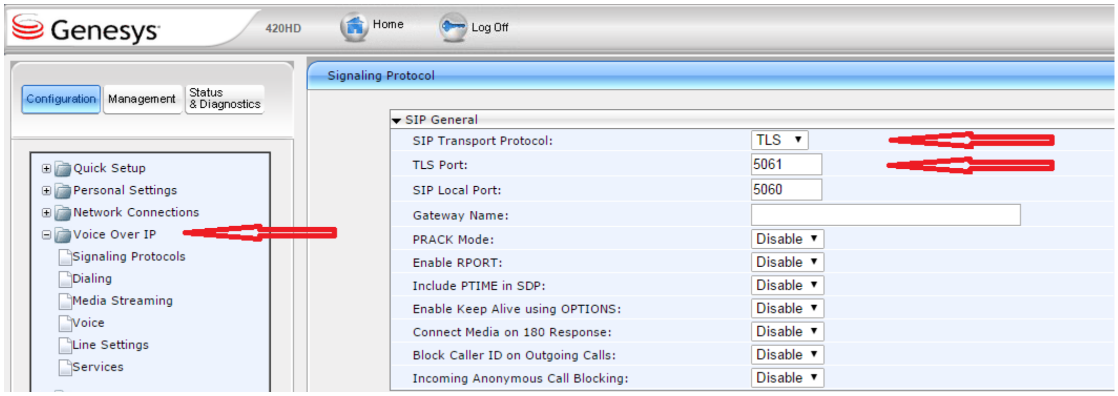
c. Configure basic calling for the Secondary Shared Line. See Call Control Using Phone -> Basic calling (incoming and outgoing calls).

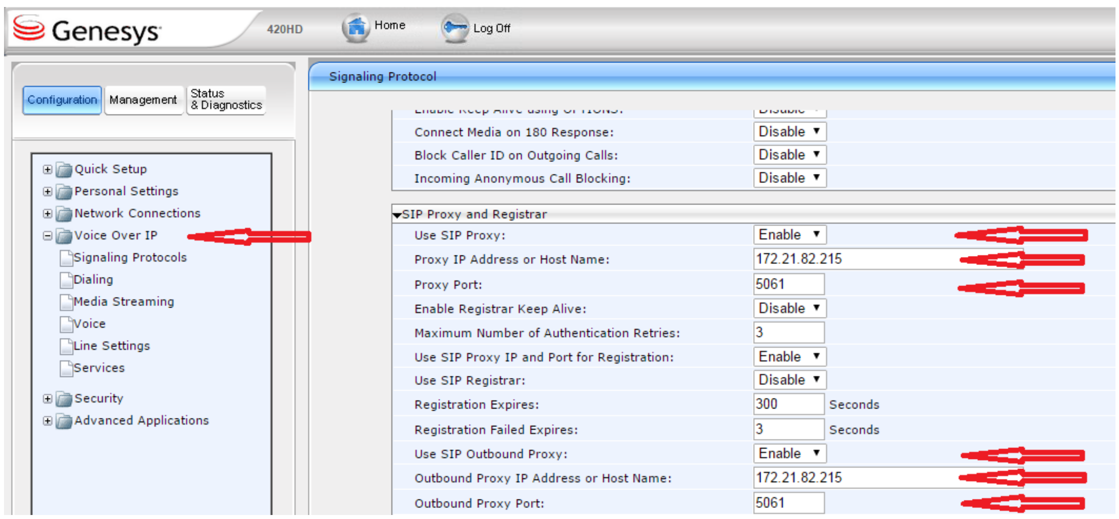


SIP authentication	Using the Web interface, **Configuration -> Voice Over IP-> Line Settings,** specify login credentials for SIP authentication in the **Authentication User Name** and **Authentication Password** fields. **Note:** The Password parameter must have the same value as the **password** option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

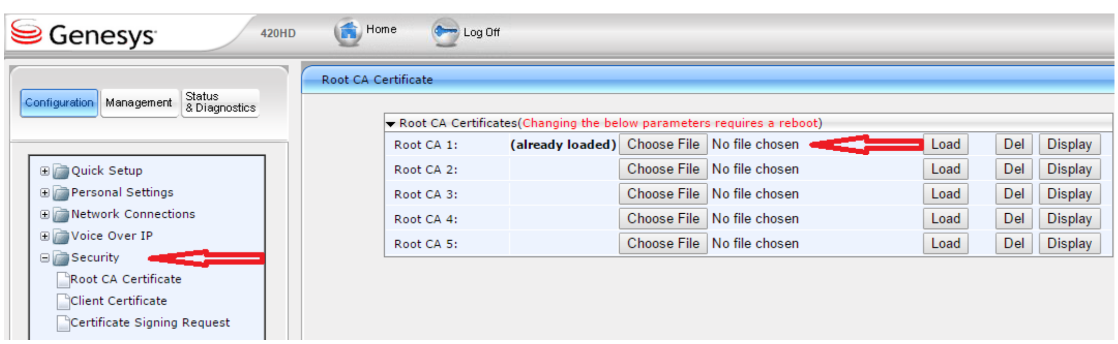
TLS/SRTP

1. Using the Web interface, specify the SIP Server **IP Address, SIP Transport Protocol** set to **TLS**, and **TLS Port**.





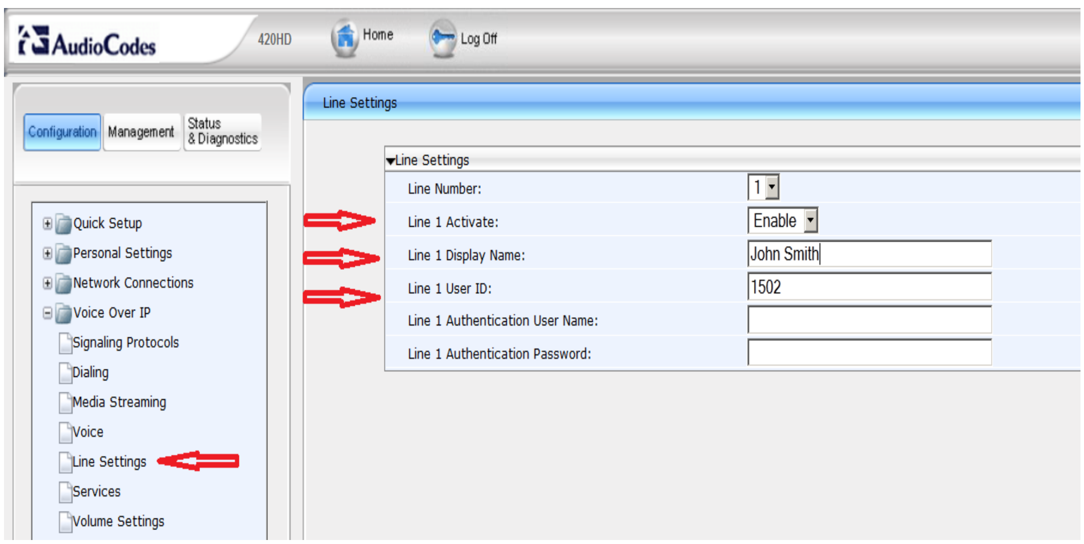
2. Using the Web interface, specify a custom CA Certificate (.pem file) at **Home -> Security -> Advanced -> Root CA Certificate -> Root CA 1 -> Choose File.**



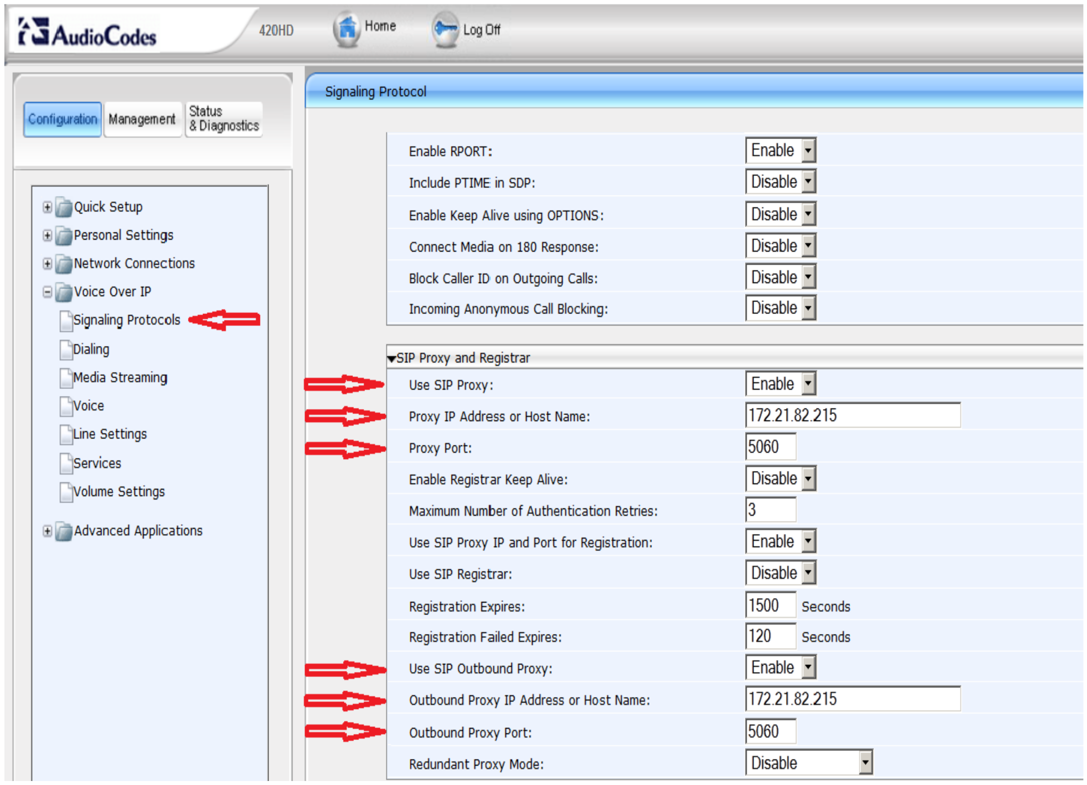
TLS integration of SIP Server deployed on UNIX and AudioCodes 420HD v.2.2.8.xx requires Genesys Security Pack 8.5.1 or later to support TLS 1.2 protocol. AudioCodes 420HD v.2.2.2.79 supports TLS 1.0 protocol.

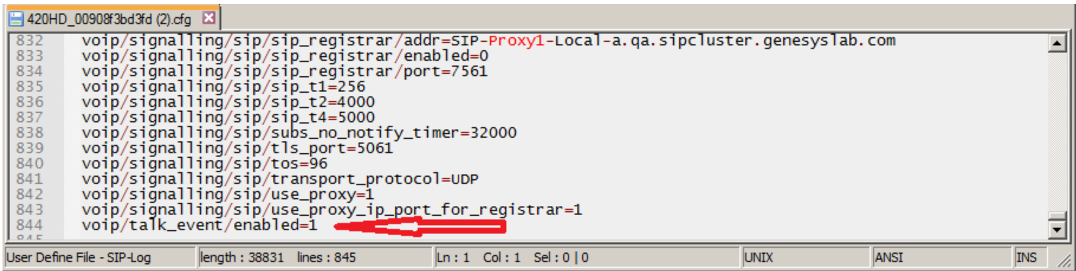
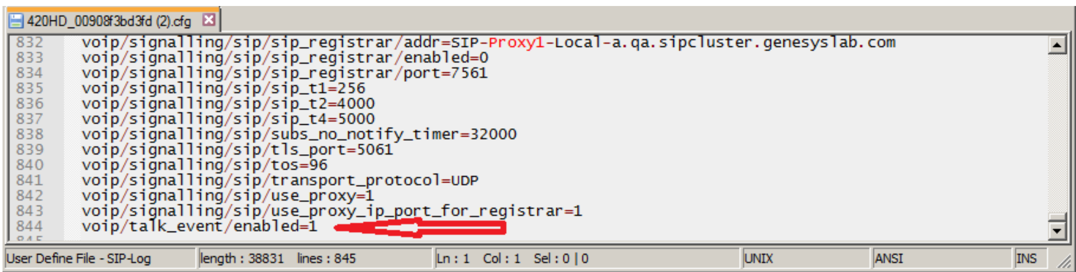
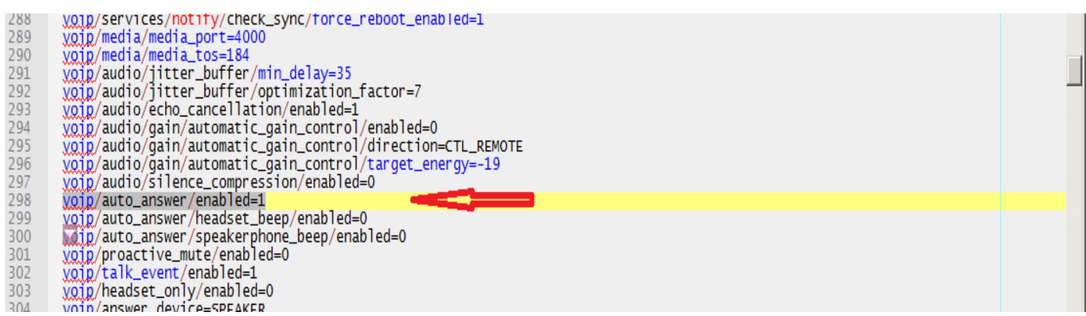
<table>
<tr><td>Secure SIP (SIPS) support, in accordance with RFC 5630</td><td>To enable SIPS support, set the following option in the phone’s configuration file:
voip/signalling/sip/enable_sips=1

voip/signalling/sip/display_name_in_registration_msg/enabled=0
voip/signalling/sip/enable_sips=1
voip/signalling/sip/ext_error_codes=
voip/signalling/sip/failback_retry_timeout=0
voip/signalling/sip/hk_blind_transfer/enable=0
voip/signalling/sip/keepalive_options/enabled=0
voip/signalling/sip/keepalive_options/timeout=300
voip/signalling/sip/lync_type_number_rules=0</td></tr>
<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>1. Using the Web interface, Configuration -> Voice Over IP -> Line Settings:
a. Activate the line by setting Activate to Enable.
b. Specify the Display Name and User ID.



2. Using the Web interface, Configuration -> Voice Over IP -> Signaling Protocols -> SIP Proxy and Registrar:
a. Set Use SIP Proxy and Use SIP Outbound Proxy to Enable.
b. Specify the SIP Server IP address in the Proxy IP Address or Host Name and Outbound Proxy IP Address or Host Name fields.
c. Specify the SIP Server port in the Proxy Port and Outbound Proxy Port fields.</td></tr>
</table>

<table>
<tr><td></td><td>
</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, and 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>
<tr><td colspan="2">Call Control Using Desktop Client</td></tr>
<tr><td>Feature</td><td>Key Actions and Procedures</td></tr>
<tr><td>Answer Incoming Call</td><td>Using the phone’s configuration file, modify the line to enable call control using the Desktop client:
• voip/talk_event/enabled=1</td></tr>
</table>

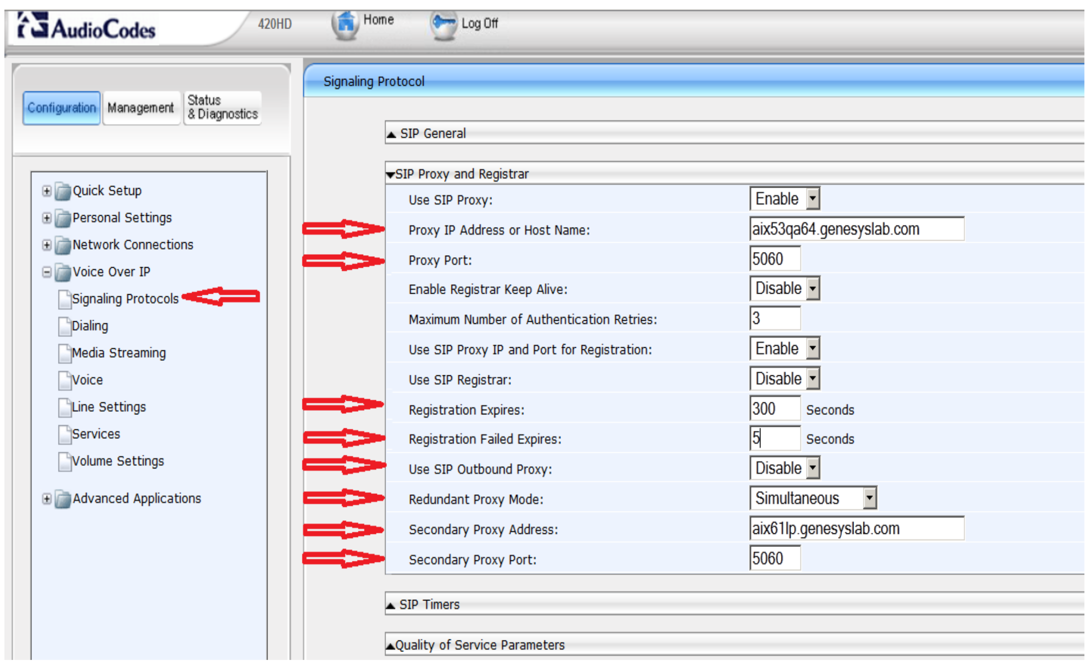
	 420HD_00908f3bd3fd (2).cfg voip/signalling/sip/sip_registrar/addr=SIP-Proxy1-Local-a.qa.sipcluster.genesyslab.com voip/signalling/sip/sip_registrar/enabled=0 voip/signalling/sip/sip_registrar/port=7561 voip/signalling/sip/sip_t1=256 voip/signalling/sip/sip_t2=4000 voip/signalling/sip/sip_t4=5000 voip/signalling/sip/subs_no_notify_timer=32000 voip/signalling/sip/tls_port=5061 voip/signalling/sip/tos=96 voip/signalling/sip/transport_protocol=UDP voip/signalling/sip/use_proxy=1 voip/signalling/sip/use_proxy_ip_port_for_registrar=1 voip/talk_event/enabled=1 User Define File - SIP-Log length : 38831 lines : 845 Ln : 1 Col : 1 Sel : 0 | 0 UNIX ANSI INS
Conference	No configuration is required.
Hold/Retrieve	Using the phone's configuration file, modify the line to enable call control using the Desktop client: • **voip/talk_event/enabled=1**  420HD_00908f3bd3fd (2).cfg voip/signalling/sip/sip_registrar/addr=SIP-Proxy1-Local-a.qa.sipcluster.genesyslab.com voip/signalling/sip/sip_registrar/enabled=0 voip/signalling/sip/sip_registrar/port=7561 voip/signalling/sip/sip_t1=256 voip/signalling/sip/sip_t2=4000 voip/signalling/sip/sip_t4=5000 voip/signalling/sip/subs_no_notify_timer=32000 voip/signalling/sip/tls_port=5061 voip/signalling/sip/tos=96 voip/signalling/sip/transport_protocol=UDP voip/signalling/sip/use_proxy=1 voip/signalling/sip/use_proxy_ip_port_for_registrar=1 voip/talk_event/enabled=1 User Define File - SIP-Log length : 38831 lines : 845 Ln : 1 Col : 1 Sel : 0 | 0 UNIX ANSI INS
Remote DTMF tones generation	No configuration is required.
Make Outgoing Call	See the Basic calling (incoming and outgoing calls) feature.
Remote Auto-Answer (based on SIP header)	Using the phone's configuration file, modify the line to enable the phone's Auto-Answer functionality: • **voip/auto_answer/enabled=1**  voip/services/notify/check_sync/force_reboot_enabled=1 voip/media/media_port=4000 voip/media/media_tos=184 voip/audio/jitter_buffer/min_delay=35 voip/audio/jitter_buffer/optimization_factor=7 voip/audio/echo_cancellation/enabled=1 voip/audio/gain/automatic_gain_control/enabled=0 voip/audio/gain/automatic_gain_control/direction=CTL_REMOTE voip/audio/gain/automatic_gain_control/target_energy=-19 voip/audio/silence_compression/enabled=0 voip/auto_answer/enabled=1 voip/auto_answer/headset_beep/enabled=0 voip/auto_answer/speakerphone_beep/enabled=0 voip/proactive_mute/enabled=0 voip/talk_event/enabled=1 voip/headset_only/enabled=0 voip/answer_device=SPEAKER
Unattended transfer	No configuration is required.
Semi-attended transfer	No configuration is required.

<table>
<tr>
<td>Attended (consultative) transfer</td>
<td>No configuration is required.</td>
</tr>
<tr>
<td>Genesys Business Continuity (Simultaneous, dual-registration mode)</td>
<td>

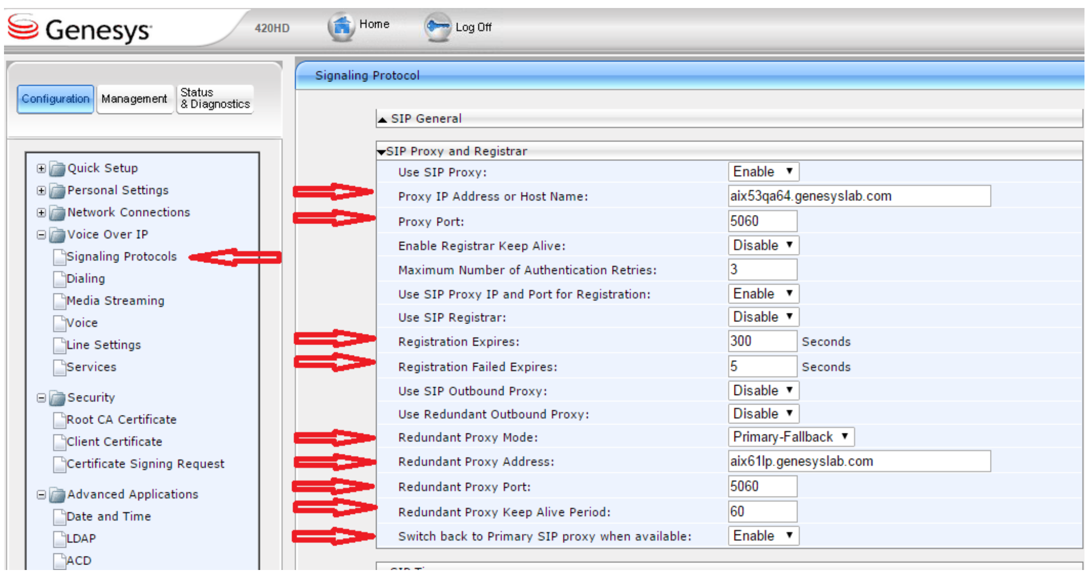
Using the Web interface, **Configuration -> Voice Over IP -> Signaling Protocols -> SIP Proxy and Registrar**:

1. Specify the IP address (FQDN) of SIP Server peers in the **Proxy IP Address or Host Name** and **Secondary Proxy Address** fields.
2. Specify the port used by SIP Server peers in the **Proxy Port** and **Secondary Proxy Port** fields.
3. Set **Registration Expires** to **300** (seconds).
4. Set **Registration Failed Expires** to **5** (seconds).
5. Set **Use SIP Outbound Proxy** to **Disable**.
6. Set **Redundant Proxy Mode** to **Simultaneous**.

For Genesys Business Continuity deployment, the AudioCodes phone registers (SIP REGISTER) with both SIP Server peers.



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

Genesys Business Continuity (Primary-Fallback, single-registration mode)	Using the Web interface, **Configuration -> Voice Over IP-> Signaling Protocols -> SIP Proxy and Registrar**: 1. Specify the IP address (FQDN) of SIP Server peers in the **Proxy IP Address or Host Name** and **Redundant Proxy Address** fields. 2. Specify the port used by SIP Server peers in the **Proxy Port** and **Redundant Proxy Address** fields. 3. Set **Registration Expires** to **300** (seconds). 4. Set **Registration Failed Expires** to **5** (seconds). 5. Set **Use SIP Outbound Proxy** to **Disable**. 6. Set **Redundant Proxy Mode** to **Primary-Fallback**. 7. Set **Switch back to Primary SIP Proxy when available** to **Enable**. Agent State Control from the Phone must be configured with the Primary-Fallback mode. 

6 Known Issues and Limitations

6.1 Issues and Limitations Identified with Genesys Products

- Three-way conferences initiated on any SIP Phone will not be reported as a conference.
- The phone sometimes can merge a consultation leg into a conference prematurely.
- Shared Call Appearance is not supported when Genesys SIP Server is deployed in Business Continuity mode.