



Genesys Application Note

AudioCodes SIP Phones With Genesys SIP Server

Document version 2.0

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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 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	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

3 rd Party Hardware Components		
Model	Version	Notes
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.

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 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.
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>
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:</p> <p>gvm_mailbox=<voice mail box number></p> <p>For example: gvm_mailbox=1502, where 1502 is a mailbox number.</p>
Shared Call Appearance (SCA), in SIP Server standalone deployments	<p>Note: Only AC 440HD supports full SCA functionality.</p> <ol style="list-style-type: none"> 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 the shared-line-capacity option. If required, configure SIP authentication. (See SIP authentication in this table.) Configure Secondary Shared Line DNs: <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 the shared-line-number option.
SIP authentication	<ol style="list-style-type: none"> 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 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> <p>Note: The string must match the phone setting in Configuration -> Voice Over IP -> Line Settings -> Authentication User Name and Authentication Password.</p>
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	<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:</p> <p>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.</p> <p>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:</p> <p>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.

	<p>5. After successful SIP registration the phone is ready for making outgoing calls and receiving incoming calls.</p> <p>6. Run your desktop client to make a test call.</p>
Remote Auto-Answer (based on SIP header)	<p>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:</p> <p>sip-alert-info=info=alert-autoanswer</p>
Unattended transfer (Genesys Single-Step Transfer)	No configuration is required.
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:</p> <p>blind-transfer-enabled=true</p> <p>Note: This option must be set on the DN object that represents a transfer destination party.</p>
Attended transfer (Genesys Two-Step Transfer)	<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 <p>Note: A value of 1 or 2 is sufficient for the phone.</p> <ol style="list-style-type: none"> 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	<p>Configure SIP Server to remotely control DTMF generation on the SIP phone. In the TServer section of the DN object, configure:</p> <p>sip-cti-control=dtmf</p>
Genesys Business Continuity (Simultaneous, dual-registration mode)	<p>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:</p> <p>dr-forward=no-agent</p>

<p>Genesys Business Continuity (Primary-Fallback, single-registration mode)</p>	<p>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</p> <p>Note: Agent State Control from the Phone functionality only supported in this mode.</p>
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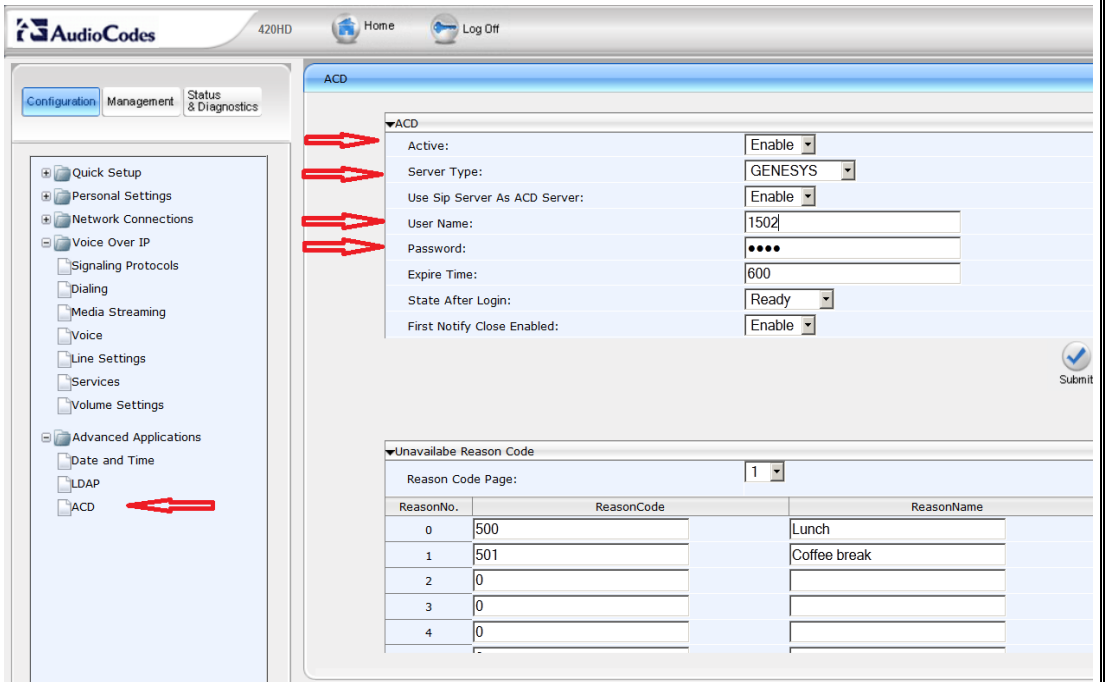
Example of the DN .cfg file:

```
[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.

AudioCodes Phone Configuration	
General Features Supported By Phone	
Feature	Key Actions and Procedures
Agent Login from the Phone	<ol style="list-style-type: none"> Using the Web interface, Configuration -> Advanced Applications -> ACD, set the ACD feature: <ol style="list-style-type: none"> Set Active to Enable. Set Server Type to Genesys Enter the User Name and Password. 
Agent State Control from the Phone	<ol style="list-style-type: none"> Using the Web interface, Configuration -> Advanced Applications -> ACD, set the ACD feature: <ol style="list-style-type: none"> Set Active to Enable. Set Server Type to BROADSOFT.

c. Enter the **User Name** and **Password**.

The screenshot shows the Genesys ACD configuration interface. The left sidebar has a tree view with the following structure:

- Quick Setup
- Personal Settings
- Network Connections
- Voice Over IP
 - Signaling Protocols
 - Dialing
 - Media Streaming
 - Voice
 - Line Settings
 - Services
- Security
 - Root CA Certificate
 - Client Certificate
 - Certificate Signing Request
- Advanced Applications
 - Date and Time
 - LDAP
 - ACD (selected)

The main configuration area for ACD includes the following fields:

- Active: Enable
- Server Type: BROADSOFT
- Use Sip Server As ACD Server: Enable
- User Name: 1502
- Password: ****
- Expire Time: 600
- State After Login: Ready
- First Notify Close Enabled: Enable

Below the form is a table for 'Unavailable Reason Code':

ReasonNo.	ReasonCode	ReasonName
0	500	Lunch
1	501	Coffee break
2	0	
3	0	
4	0	
5	0	
6	0	

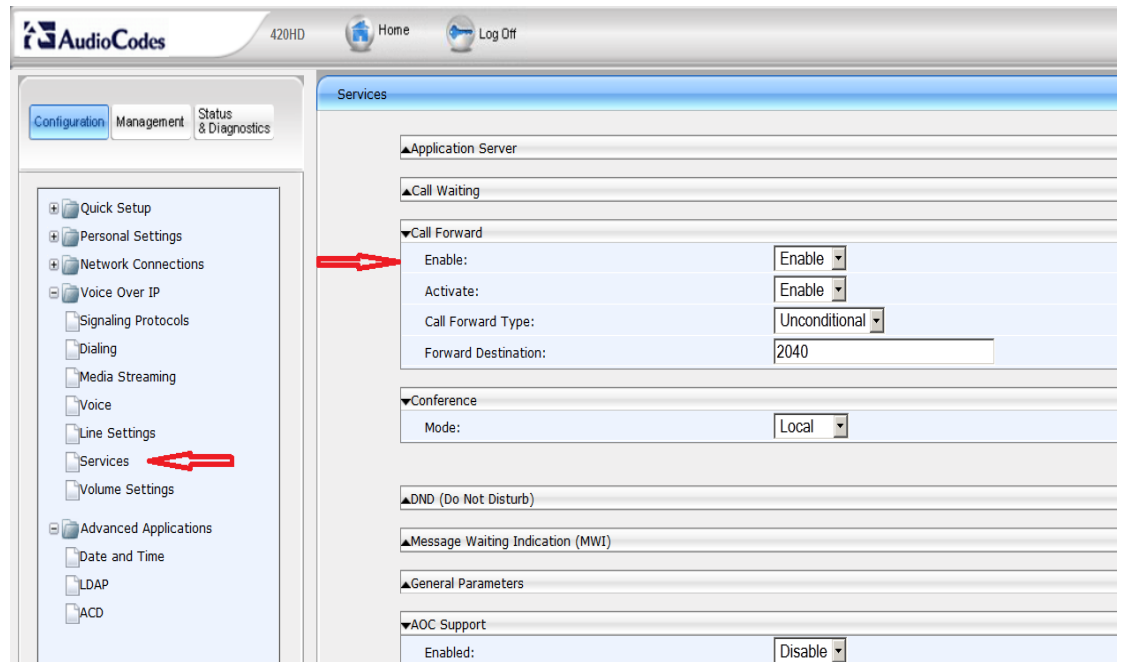
2. For Login/Logout/Available/Unavailable status, press **Login** and enter Login Credentials from the phone touch screen: **User Name** and **Password**.

Caller ID

No configuration is required.

Call Forward

Using the Web interface, **Configuration -> Voice Over IP -> Services -> Call Forward**, enable call forward by selecting **Enable**.

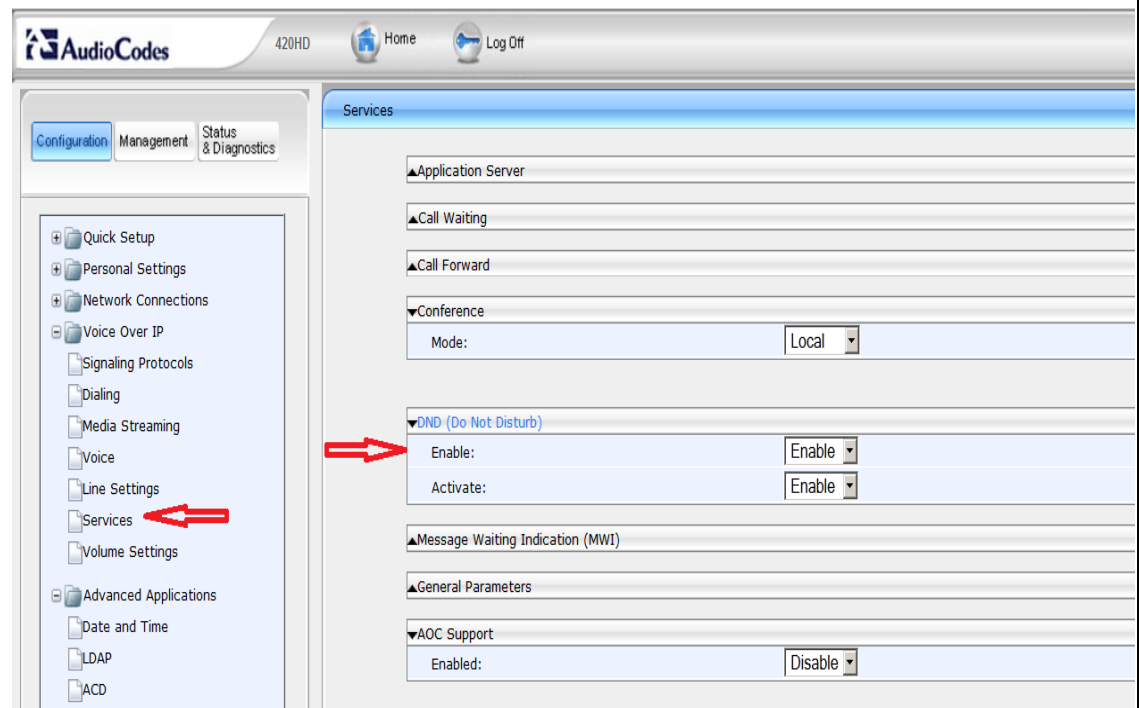


OR:

Using the phone, enable call forward from the phone by pressing the **Fwd** button.

Do Not Disturb

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.

DNS-based redundancy (using SIP Proxy)

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.

The screenshot shows the AudioCodes web interface with the 'Signaling Protocol' configuration page. The left sidebar contains a tree view with 'Signaling Protocols' selected. The main content area shows the 'SIP Proxy and Registrar' section with the following settings:

Field	Value
Use SIP Proxy:	Enable
Proxy IP Address or Host Name:	sips-a.qa.domain.com
Proxy Port:	5060
Enable Registrar Keep Alive:	Disable
Maximum Number of Authentication Retries:	3
Use SIP Proxy IP and Port for Registration:	Enable
Use SIP Registrar:	Disable
Registration Expires:	5 Seconds
Registration Failed Expires:	5 Seconds
Use SIP Outbound Proxy:	Enable
Outbound Proxy IP Address or Host Name:	sips-a.qa.domain.com
Outbound Proxy Port:	5060
Redundant Proxy Mode:	Disable

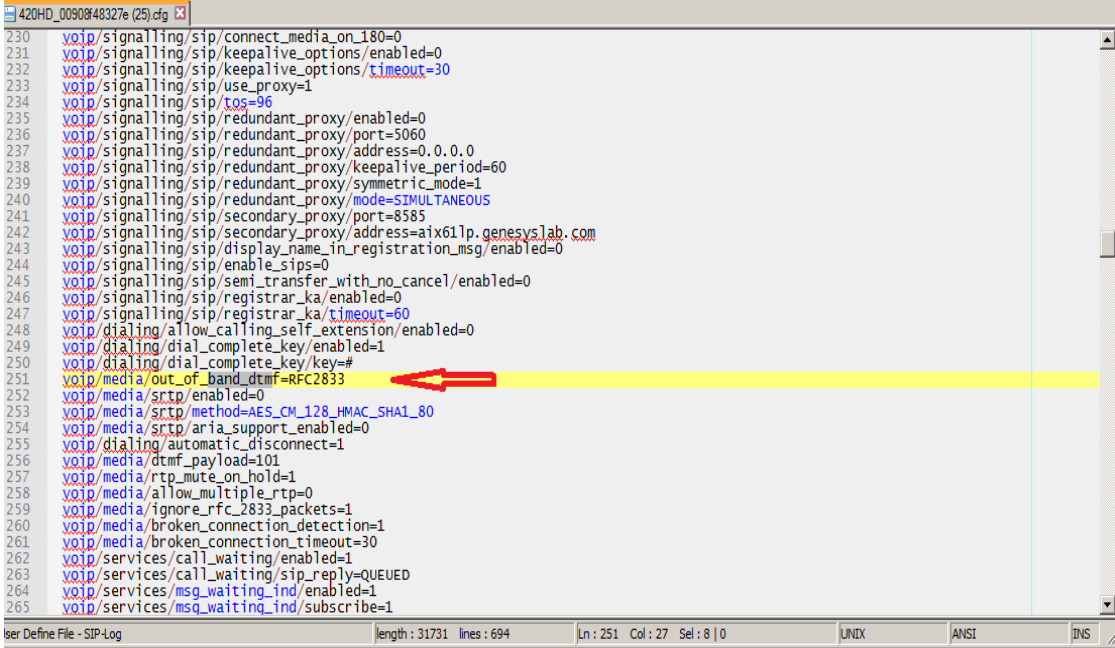
Notes:

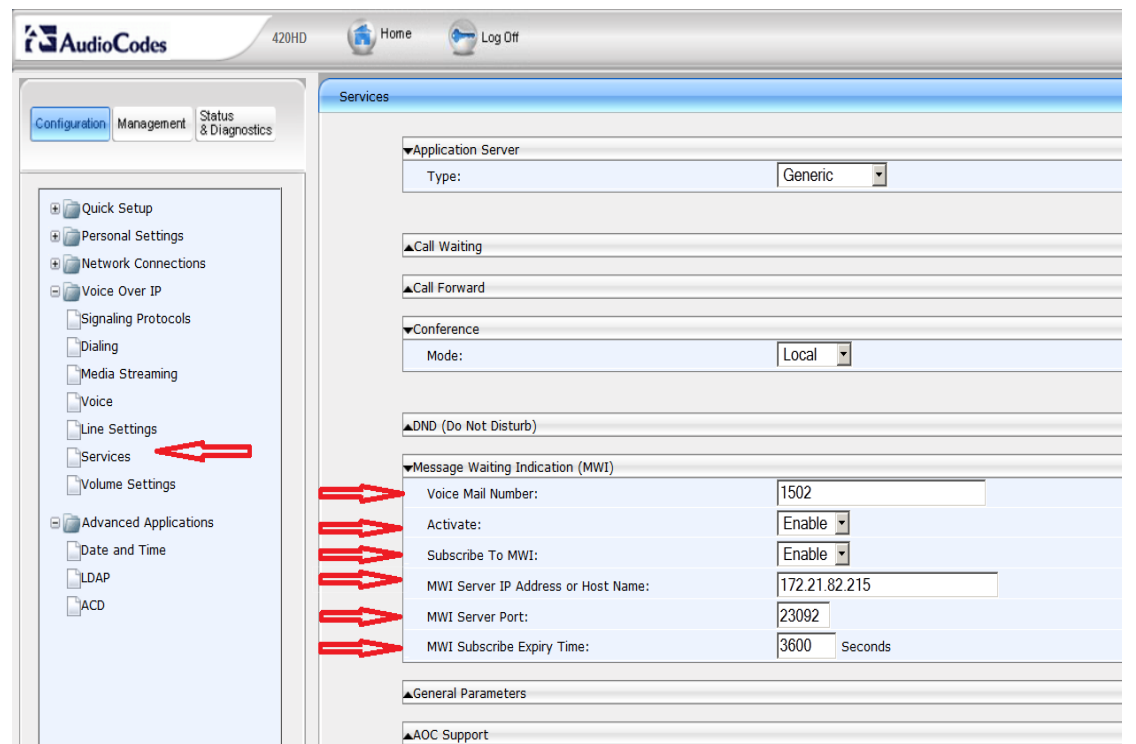
- The IP Address fields have the FQDN (sips-a.qa.domain.com) of the SIP Proxy pool that must be resolved in multiple a-records.
- Each SIP Proxy in the pool has the same SIP port configured in the Genesys configuration environment.

DTMF tones generation

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**

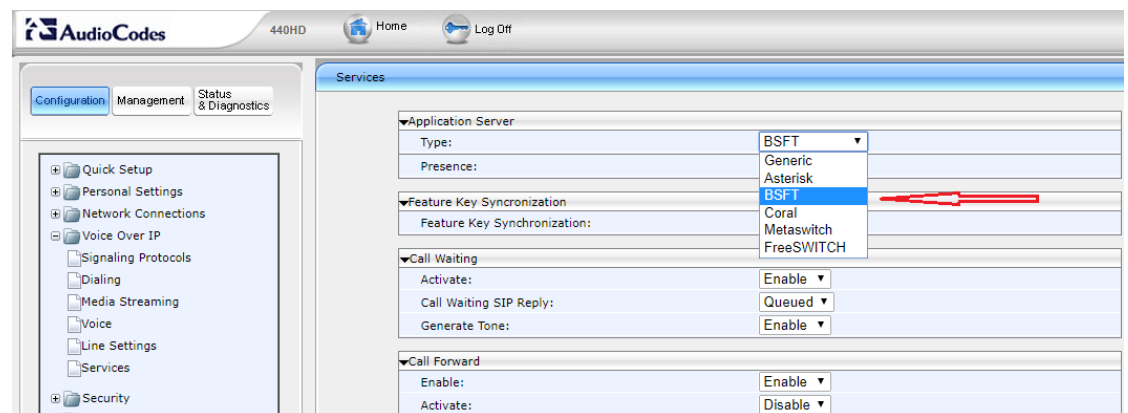
	 <pre> 230 voip/ signalling/sip/connect_media_on_180=0 231 voip/ signalling/sip/keepalive_options/enabled=0 232 voip/ signalling/sip/keepalive_options/timeout=30 233 voip/ signalling/sip/use_proxy=1 234 voip/ signalling/sip/tos=96 235 voip/ signalling/sip/redundant_proxy/enabled=0 236 voip/ signalling/sip/redundant_proxy/port=5060 237 voip/ signalling/sip/redundant_proxy/address=0.0.0.0 238 voip/ signalling/sip/redundant_proxy/keepalive_period=60 239 voip/ signalling/sip/redundant_proxy/symmetric_mode=1 240 voip/ signalling/sip/redundant_proxy/mode=SIMULTANEOUS 241 voip/ signalling/sip/secondary_proxy/port=8585 242 voip/ signalling/sip/secondary_proxy/address=aix61lp.genesyslab.com 243 voip/ signalling/sip/display_name_in_registration_msg/enabled=0 244 voip/ signalling/sip/enable_sips=0 245 voip/ signalling/sip/semi_transfer_with_no_cancel/enabled=0 246 voip/ signalling/sip/registrat_ka/enabled=0 247 voip/ signalling/sip/registrat_ka/timeout=60 248 voip/ dialing/ allow_calling_self_extension/enabled=0 249 voip/ dialing/ dial_complete_key/enabled=1 250 voip/ dialing/ dial_complete_key/key=# 251 voip/ media/out_of_band_dtmf=RFC2833 252 voip/ media/srtp/enabled=0 253 voip/ media/srtp/method=AES_CM_128_HMAC_SHA1_80 254 voip/ media/srtp/aria_support_enabled=0 255 voip/ dialing/ automatic_disconnect=1 256 voip/ media/ dtmf_payload=101 257 voip/ media/ rtp_mute_on_hold=1 258 voip/ media/ allow_multiple_rtp=0 259 voip/ media/ ignore_rfc_2833_packets=1 260 voip/ media/ broken_connection_detection=1 261 voip/ media/ broken_connection_timeout=30 262 voip/ services/ call_waiting/enabled=1 263 voip/ services/ call_waiting/sip_reply=QUEUED 264 voip/ services/ msg_waiting_ind/enabled=1 265 voip/ services/ msg_waiting_ind/subscribe=1 </pre>
Multiple calls on one extension	Genesys recommends having no more than two concurrent calls.
Message Waiting Indicator	<p>Using the Web interface, Configuration -> Voice Over IP -> Services -> Message Waiting Indication (MWI):</p> <ol style="list-style-type: none"> 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).



Shared Call Appearance (SCA)

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.

AudioCodes 440HD Home Log Off

Configuration Management Status & Diagnostics

Line Settings

Line Settings

Line Number: 1

Line 1 Activate: Enable

Line 1 Display Name: Primary Line

Line 1 User ID: 7000

Line 1 Authentication User Name: 7000

Line 1 Authentication Password: ****

Line 1 Mode: Shared

- 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 \(incoming and outgoing calls\)](#).

Configuration Management Status & Diagnostics

Line Settings

Line Settings

Line Number: 1

Line 1 Activate: Enable

Line 1 Display Name: SecondaryLine

Line 1 User ID: 7001

Line 1 Authentication User Name: 7000

Line 1 Authentication Password: ****

Line 1 Mode: Shared

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.

The screenshot displays the AudioCodes 420HD web interface. The top navigation bar includes the AudioCodes logo, the model number '420HD', and links for 'Home' and 'Log Off'. The left sidebar contains a 'Configuration' tab and a tree view of settings categories: Quick Setup, Personal Settings, Network Connections, Voice Over IP, Signaling Protocols, Dialing, Media Streaming, Voice, Line Settings (highlighted with a red arrow), Services, and Volume Settings. The main content area is titled 'Line Settings' and contains a form with the following fields:

Line Settings	
Line Number:	1
Line 1 Activate:	Enable
Line 1 Display Name:	1502
Line 1 User ID:	1502
Line 1 Authentication User Name:	1502
Line 1 Authentication Password:	••••

Red arrows indicate the navigation path: one arrow points to the 'Line Settings' menu item in the sidebar, and two arrows point to the 'Line 1 Authentication User Name' and 'Line 1 Authentication Password' fields in the form.

TLS/SRTP

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

Genesys 420HD Home Log Off

Configuration Management Status & Diagnostics

Quick Setup Personal Settings Network Connections Voice Over IP Signaling Protocols Dialing Media Streaming Voice Line Settings Services

Signaling Protocol

SIP General

SIP Transport Protocol:	TLS
TLS Port:	5061
SIP Local Port:	5060
Gateway Name:	
PRACK Mode:	Disable
Enable RPORT:	Disable
Include PTIME in SDP:	Disable
Enable Keep Alive using OPTIONS:	Disable
Connect Media on 180 Response:	Disable
Block Caller ID on Outgoing Calls:	Disable
Incoming Anonymous Call Blocking:	Disable

Genesys 420HD Home Log Off

Configuration Management Status & Diagnostics

Quick Setup Personal Settings Network Connections Voice Over IP Signaling Protocols Dialing Media Streaming Voice Line Settings Services Security Advanced Applications

Signaling Protocol

SIP Proxy and Registrar

Use SIP Proxy:	Enable
Proxy IP Address or Host Name:	172.21.82.215
Proxy Port:	5061
Enable Registrar Keep Alive:	Disable
Maximum Number of Authentication Retries:	3
Use SIP Proxy IP and Port for Registration:	Enable
Use SIP Registrar:	Disable
Registration Expires:	300 Seconds
Registration Failed Expires:	3 Seconds
Use SIP Outbound Proxy:	Enable
Outbound Proxy IP Address or Host Name:	172.21.82.215
Outbound Proxy Port:	5061

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

Genesys 420HD Home Log Off

Configuration Management Status & Diagnostics

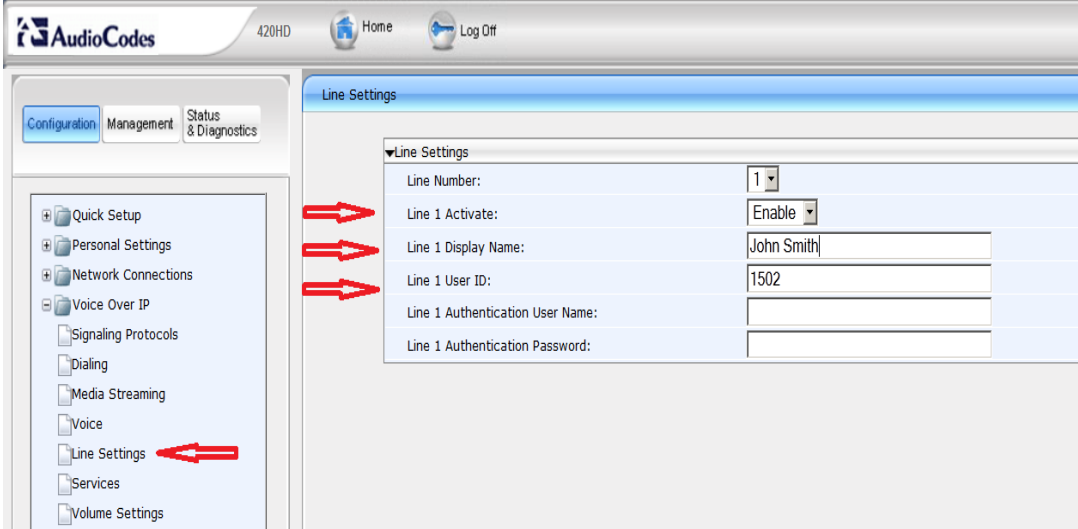
Quick Setup Personal Settings Network Connections Voice Over IP Security Root CA Certificate Client Certificate Certificate Signing Request

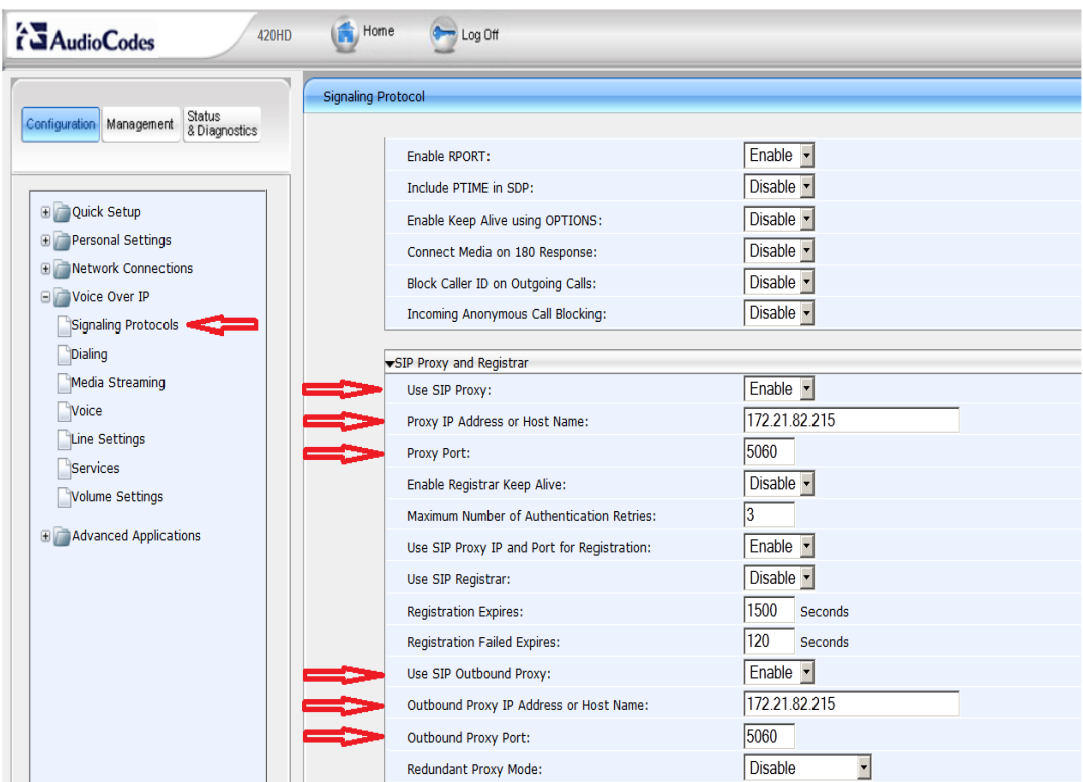
Root CA Certificate

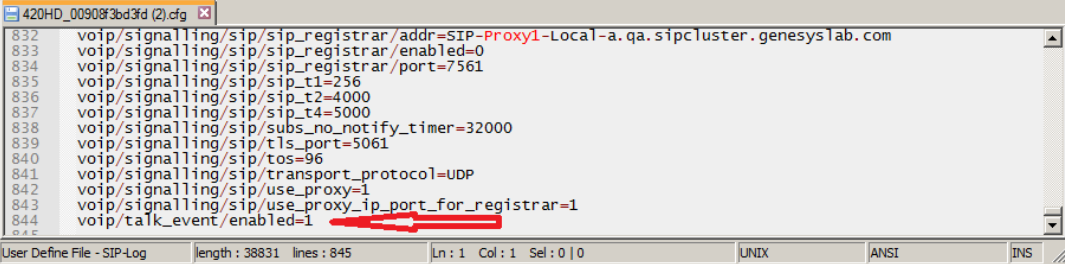
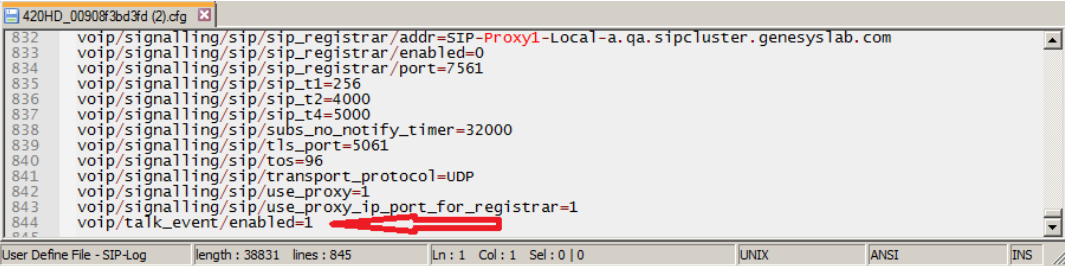
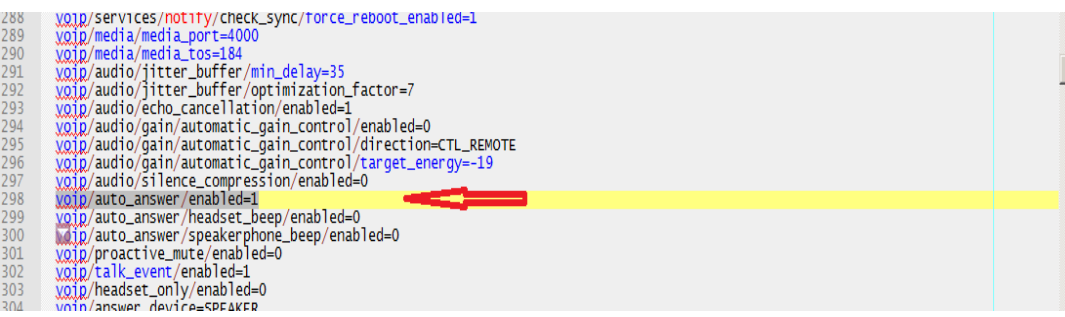
Root CA Certificates (Changing the below parameters requires a reboot)

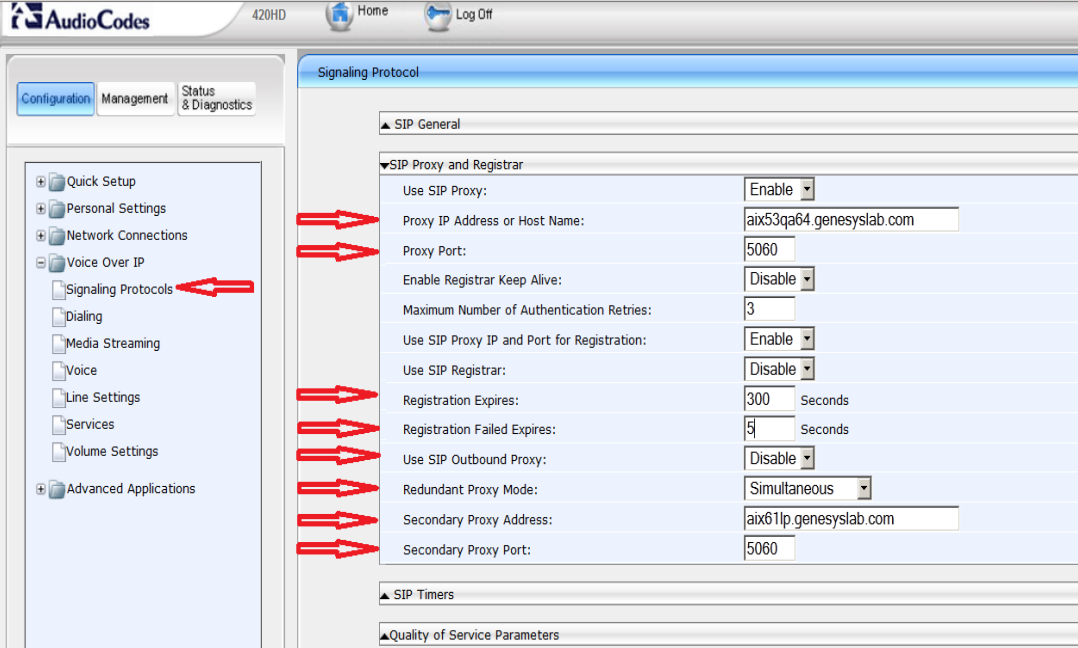
Root CA 1:	(already loaded)	Choose File	No file chosen	Load	Del	Display
Root CA 2:		Choose File	No file chosen	Load	Del	Display
Root CA 3:		Choose File	No file chosen	Load	Del	Display
Root CA 4:		Choose File	No file chosen	Load	Del	Display
Root CA 5:		Choose File	No file chosen	Load	Del	Display

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.

<p>Secure SIP (SIPS) support, in accordance with RFC 5630</p>	<p>To enable SIPS support, set the following option in the phone's configuration file: voip/signalling/sip/enable_sips=1</p> <pre> 1323 voip/signalling/sip/display_name_in_registration_msg/enabled=0 1324 voip/signalling/sip/enable_sips=1 1325 voip/signalling/sip/ext_error_codes= 1326 voip/signalling/sip/failback_retry_timeout=0 1327 voip/signalling/sip/hk_blind_transfer/enabled=0 1328 voip/signalling/sip/keepalive_options/enabled=0 1329 voip/signalling/sip/keepalive_options/timeout=300 1330 voip/signalling/sip/lync_type_number_rules=0 </pre>
Call Control Using Phone	
Feature	Key Actions and Procedures
<p>Basic calling (incoming and outgoing calls)</p>	<ol style="list-style-type: none"> Using the Web interface, Configuration -> Voice Over IP -> Line Settings: <ol style="list-style-type: none"> Activate the line by setting Activate to Enable. Specify the Display Name and User ID.  <ol style="list-style-type: none"> Using the Web interface, Configuration -> Voice Over IP -> Signaling Protocols -> SIP Proxy and Registrar: <ol style="list-style-type: none"> Set Use SIP Proxy and Use SIP Outbound Proxy to Enable. Specify the SIP Server IP address in the Proxy IP Address or Host Name and Outbound Proxy IP Address or Host Name fields. Specify the SIP Server port in the Proxy Port and Outbound Proxy Port fields.

	
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Unattended (blind) transfer	Using the phone, press Transfer , enter the number, and 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	Using the phone's configuration file, modify the line to enable call control using the Desktop client: <ul style="list-style-type: none"> • voip/talk_event/enabled=1

	 <pre> 832 voip/ signalling/sip/sip_registrar/addr=SIP-Proxy1-Local-a.qa.sipcluster.genesyslab.com 833 voip/ signalling/sip/sip_registrar/enabled=0 834 voip/ signalling/sip/sip_registrar/port=7561 835 voip/ signalling/sip/sip_t1=256 836 voip/ signalling/sip/sip_t2=4000 837 voip/ signalling/sip/sip_t4=5000 838 voip/ signalling/sip/subs_no_notify_timer=32000 839 voip/ signalling/sip/tls_port=5061 840 voip/ signalling/sip/tos=96 841 voip/ signalling/sip/transport_protocol=UDP 842 voip/ signalling/sip/use_proxy=1 843 voip/ signalling/sip/use_proxy_ip_port_for_registrar=1 844 voip/ talk_event/enabled=1 </pre>
Conference	No configuration is required.
Hold/Retrieve	<p>Using the phone's configuration file, modify the line to enable call control using the Desktop client:</p> <ul style="list-style-type: none"> voip/talk_event/enabled=1 
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)	<p>Using the phone's configuration file, modify the line to enable the phone's Auto-Answer functionality:</p> <ul style="list-style-type: none"> voip/auto_answer/enabled=1 
Unattended transfer	No configuration is required.
Semi-attended transfer	No configuration is required.

<p>Attended (consultative) transfer</p>	<p>No configuration is required.</p>
<p>Genesys Business Continuity (Simultaneous, dual-registration mode)</p>	<p>Using the Web interface, Configuration -> Voice Over IP -> Signaling Protocols -> SIP Proxy and Registrar:</p> <ol style="list-style-type: none"> 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. <p>For Genesys Business Continuity deployment, the AudioCodes phone registers (SIP REGISTER) with both SIP Server peers.</p>  <p>The screenshot shows the AudioCodes web interface with the 'Signaling Protocol' configuration page. The left sidebar shows a tree view with 'Voice Over IP' expanded and 'Signaling Protocols' selected. The main content area shows the 'SIP Proxy and Registrar' section with the following settings:</p> <ul style="list-style-type: none"> Use SIP Proxy: Enable Proxy IP Address or Host Name: aix53qa64.genesyslab.com Proxy Port: 5060 Enable Registrar Keep Alive: Disable Maximum Number of Authentication Retries: 3 Use SIP Proxy IP and Port for Registration: Enable Use SIP Registrar: Disable Registration Expires: 300 Seconds Registration Failed Expires: 5 Seconds Use SIP Outbound Proxy: Disable Redundant Proxy Mode: Simultaneous Secondary Proxy Address: aix61lp.genesyslab.com Secondary Proxy Port: 5060

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.

Genesys 420HD Home Log Off

Configuration Management Status & Diagnostics

Signaling Protocol

▲ SIP General

▼ SIP Proxy and Registrar

Use SIP Proxy:	Enable
Proxy IP Address or Host Name:	aix53qa64.genesyslab.com
Proxy Port:	5060
Enable Registrar Keep Alive:	Disable
Maximum Number of Authentication Retries:	3
Use SIP Proxy IP and Port for Registration:	Enable
Use SIP Registrar:	Disable
Registration Expires:	300 Seconds
Registration Failed Expires:	5 Seconds
Use SIP Outbound Proxy:	Disable
Use Redundant Outbound Proxy:	Disable
Redundant Proxy Mode:	Primary-Fallback
Redundant Proxy Address:	aix61lp.genesyslab.com
Redundant Proxy Port:	5060
Redundant Proxy Keep Alive Period:	60
Switch back to Primary SIP proxy when available:	Enable

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.