AudioCodes CPE & Access Gateway Products

Configuration Guide

AudioCodes 420HD IP Phone with Genesys® SIP Server

Document #: LTRT-21941

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Notice

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Date Published: April-29-2014

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Related Documentation

Document Name
IP Phone Administrator's Manual
420HD User's Manual
400HD Series IP Phone Deployment Guide

1 Introduction

This guide shows system administrators how to configure AudioCodes' 420HD IP Phone to operate with Genesys' SIP Server in a Genesys contact center.

The guide also shows contact center agents how to use their IP phone's ACD (Automatic Call Distribution) feature.

In contact centers, ACD is a key feature of CTI (Computer Telephony Integration). The feature automatically distributes incoming calls to a specific group of terminals that contact center agents use. Most ACD functionality is the SIP Server's responsibility, but the IP phones must publish presence status (LOGIN / LOGOUT / AVAILABLE / UNAVAILABLE) for the SIP Server to make call distribution decisions.

1.1 Distributing Calls Automatically to Available Agents

Automatic Call Distribution (ACD) systems allow companies that handle a large number of incoming phone calls to direct the callers to a company employee who is able to talk at the earliest opportunity.

The ACD feature is typically implemented in contact centers encountering large numbers of incoming customer calls that must be distributed to available agents to provide immediate support to callers. The feature automatically directs incoming calls to agents working in the contact center whose presence status is 'Available' rather than unavailable. The feature's main benefit is therefore to reduce the time customers are kept waiting for service and thereby improve customer service.

AudioCodes' IP phones seamlessly interwork with Genesys' SIP Server to support the ACD feature. Once an agent signs in on their phone to ACD, their status is set to 'Available' and synchronized with Genesys' Server. Incoming calls are directed to an agent whenever their status becomes 'Available'.

1.2 Ensuring SIP Business Continuity for Agents

AudioCodes' 420HD IP phone supports dual registration for integrating into Genesys' SIP Business Continuity architecture.

SIP Business Continuity provides the ability for a group of agents to continue offering critical business functions to customers in the event of a loss of all Genesys components running at a particular site.

The SIP Business Continuity architecture uses a synchronized, two-site deployment, where Genesys switch and server components are mirrored at each site in an active-active configuration, so that any agent can log in to either switch, at any time.

In a standalone SIP Server configuration with Business Continuity mode activated, AudioCodes 420HD IP phone will register on two sites simultaneously (i.e., register on both peer SIP Servers at the same time).

Reader's Notes

2 Provisioning IP Phones Automatically

This section shows system administrators how to quickly set up AudioCodes' IP phones to operate with a Genesys SIP Server in a Genesys contact center.

Note: This section is for system administrators only.
If you're a hotline agent, skip to Section 3.

2.1 Using DHCP to Auto Provision Phones

After connecting the LAN ports of your phones to the IP network and then connecting the phones to the power supply, the phones will *by default* send a request to the Genesys contact center's network server which will then automatically allocate an IP address and send configuration information to each phone.

Make sure that the DHCP (Dynamic Host Configuration Protocol) options in your contact center's DHCP server are correctly configured.

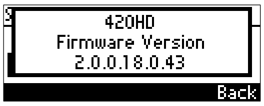
For detailed information on how to set up the DHCP server, see the *Administrator's Manual* or *400HD Series IP Phone Deployment Guide.*

2.2 Verifying Firmware Version

After automatic provisioning, make sure the phone's firmware version is correct.

- **To verify firmware version:**

- Navigate to the Firmware Status screen in the phone's LCD (**MENU** key > **Status** > **Firmware Status**):

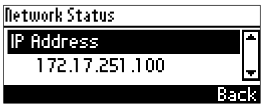


2.3 Accessing a Phone's Web Interface

Use a standard Web browser such as Microsoft® Internet Explorer® to access any phone's Web interface. Use the phone's IP address as the URL.

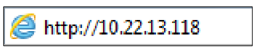
- **To obtain the phone's IP address:**

- Access the Network Status screen in the phone's LCD (**MENU** key > **Status** > **Network Status**) and navigate down to **IP Address**:



- **To access the phone's Web interface:**

1. Open the Web browser and in the URL address field enter the phone's IP address (for example, **http://10.22.13.118** or **https://10.22.13.118**):



The Web login window opens.

> **Note:** The default User Name and Password are **admin** and **1234** respectively.

2. Alternatively, if your DHCP and DNS servers are synchronized, you can access the phone Web browser using the following method:
   http://<Phone Model>-<MAC Address>.<Domain Name>
   E.g. http://440hd-001122334455.corp.YourCompany.com
3. Enter the User Name and Password, and then click **OK**.

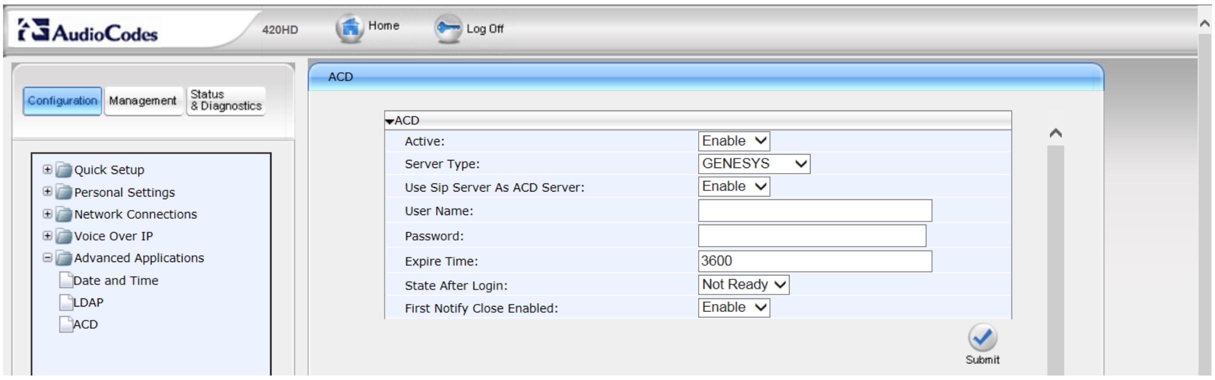
2.4 Enabling the ACD Feature

This section shows how to enable the ACD (Automatic Call Distribution) feature on the phone using either the Web interface or the Configuration File. The feature distributes incoming calls to agents' phones on the basis of agent availability and unavailability.

➢ **To configure the ACD server using the Web interface:**

1. In the Web interface, access the ACD page (**Configuration** tab > **Advanced Applications** > **ACD**):

Figure 2-1: Web Interface - ACD



2. Configure the parameters using Table 2-1 below as reference.

➢ **To configure the ACD server using the Configuration File:**

- Define a path and configure the parameters using the table below as reference.

Table 2-1: ACD Parameters

Parameter	Description
Active **[voip/services/ACD/enabled]**	From the 'Active' drop-down, choose **Enable**. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Server Type **[voip/services/ACD/server_type]**	From the 'Server Type' drop-down, choose **GENESYS**.
Use Sip Server As ACD Server **[voip/services/ACD/server_use_sip_server]**	From the drop-down, choose **Enable**. ▪ **[0]** Disable ▪ **[1]** Enable
Server Address **[voip/services/ACD/server_address]**	Displayed only when 'Use SIP Server As ACD Server' is set to **Disable** (see previous). Defines the IP address of the ACD server. Default: 0.0.0.0

Parameter	Description
Server Port **[voip/services/ACD/server_port]**	Displayed only when 'Use SIP Server As ACD Server' is set to **Disable** (see previous). Defines the port of the ACD server. Default: 80
User Name **[system/user_name]**	Enter the agent's User Name. The agent will use this name when logging in to ACD in order to define or change availability status.
Password **[system/password]**	Enter a password if necessary.
State After Login **[voip/services/ACD/state_after_login]**	The call center's network administrator can select either ▪ **Ready** -OR- ▪ **Not Ready** If set to **Ready**, each phone in the call center will automatically be set to a state of readiness to take incoming calls immediately after the call center's agents log in. If set to **Not Ready**, agents can log in and then manually configure their readiness status in the phone's LCD, giving them time to perform personal tasks before beginning work.

2.5 Configuring Dual Registration for Genesys Business Continuity

This section shows how to configure dual registration for Genesys SIP Business Continuity, using the Web interface or Configuration File.

➢ **To configure using the Web Interface:**

1. In the Web interface, access the SIP Proxy and Registrar section in the Signaling Protocol screen (**Configuration** menu > **Voice Over IP** > **Signaling Protocol**):

Figure 2-2: Web Interface - Signaling Protocol – SIP Proxy and Registrar

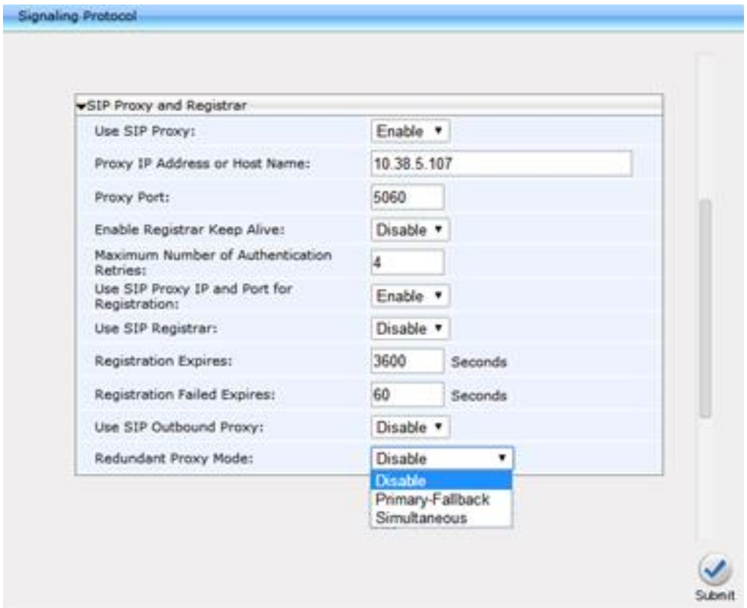


Figure 2-3: Web Interface - Signaling Protocol – SIP Proxy and Registrar – Secondary Proxy

2. Configure the parameters using table below as reference.

➢ **To configure using Configuration File:**

- Use the table below as reference.

Table 2-2: SIP Proxy and Registrar Parameters

Parameter	Description
Use SIP Proxy **[voip/signalling/sip/use_proxy]**	Determines whether to use a SIP Proxy server. Configure **[1] Enable**. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Proxy IP Address or Host Name **[voip/signalling/sip/proxy_address]**	Enter the IP address or host name (for example, **audiocodes.com**) of the SIP proxy server. Default: 0.0.0.0
Proxy Port **[voip/signalling/sip/proxy_port]**	The UDP or TCP port of the SIP proxy server. Range: 1024 to 65535. Default: 5060.
Enable Registrar Keep Alive **[voip/signalling/sip/registrar_ka/enabled]**	Determines whether to use the registration keep-alive mechanism based on SIP OPTION messages. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Notes:** ▪ If there is no response from the server, the timeout for re-registering is automatically reduced to a user-defined value (**voip/signalling/sip/registration_failed_timeout**) ▪ When the phone re-registers, the keep-alive messages are re-sent periodically.

Parameter	Description
Registrar Keep Alive Period **[voip/signalling/sip/registrar_ka/timeout]**	Defines the registration keep-alive time interval (in seconds) between Keep-Alive messages. Range: 40 to 65536. Default: 60.
Maximum Number of Authentication Retries **[voip/signalling/sip/proxy_timeout]**	The SIP proxy server registration timeout (in seconds). Range: 0 to 86400. Default: 3600.
Registration Failed Expires **[voip/signalling/sip/registration_failed_timeout]**	If registration fails, this parameter determines the interval between the register messages periodically sent until successful registration. Range: 1 to 86400. Default: 60.
Use SIP Registrar **[voip/signalling/sip/sip_registrar/enabled]**	Determines whether the phone registers to a separate SIP Registrar server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Use SIP Proxy IP and Port for Registration **[voip/signalling/sip/use_proxy_ip_port_for_registrar]**	Determines whether to use the SIP proxy's IP address and port for registration. When enabled, there is no need to configure the address of the registrar separately. ▪ **[0]** Disable (default) ▪ **[1]** Enable
Registrar IP Address or Host Name **[voip/signalling/sip/sip_registrar/addr]**	The IP address or host name of the Registrar server. Default: 0.0.0.0.
Registrar Port **[voip/signalling/sip/sip_registrar/port]**	The UDP or TCP port of the Registrar server. Range: 1024 to 65535. Default: 5060.
Use SIP Outbound Proxy **[voip/signalling/sip/sip_outbound_proxy/enabled]**	Determines whether an outbound SIP proxy server is used (all SIP messages are sent to this server as the first hop). ▪ **[0]** Disable (default) ▪ **[1]** Enable
Outbound Proxy IP Address or Host Name **[voip/signalling/sip/sip_outbound_proxy/addr]**	The IP address of the outbound proxy (for example, **audiocodes.com**; i.e., the same as that configured for the 'Proxy IP Address or Host Name' parameter above). If this parameter is set, all outgoing messages (including Registration messages) are sent to this Proxy according to the Stack behavior. Default: 0.0.0.0
Outbound Proxy Port **[voip/signalling/sip/sip_outbound_proxy/port]**	The port on which the outbound proxy listens. Range: 1024 to 65535. Default: 5060.

Parameter	Description
Redundant Proxy Mode **[voip/signalling/sip/redundant_proxy/mode]**	The call center's network administrator can select either ▪ **Disable** -OR- ▪ **Primary Fallback** -OR- ▪ **Simultaneous** For the dual-registration feature, select **Simultaneous**; two proxies are registered simultaneously so that at least one should be up and running at any time, preventing the call center from going down.
Secondary Proxy Address **[voip/signalling/sip/secondary_proxy/address]**	Displayed only when **Simultaneous** is selected for 'Redundant Proxy Mode' (see previous parameter). Define the IP address of the secondary proxy that will be up simultaneously with the primary.
Secondary Proxy Port **[voip/signalling/sip/secondary_proxy/port]**	Displayed only when **Simultaneous** is selected for 'Redundant Proxy Mode' (see the parameter before the previous). Define the port of the secondary proxy that will be up simultaneously with the primary.

2.6 Disabling the Web Interface

This feature lets the call center's network administrator block Web interface access to agents employed in the call center.

➢ **To disable access using Configuration File:**

- Use the table below as reference.

Table 2-3: Disabling the Web Interface

Parameter	Description
[system/web/enabled]	Determines whether or not to enable access to the phone's Web interface. ▪ **[0]** Disable (access prohibited) ▪ **[1]** Enable (default) (access enabled) This can avoid a potential scenario in which agents deliberately or by accident disrupt the operation of the call center.

2.7 Forcing a Reboot on Provisioning

This feature lets the call center's network administrator configure a forced reboot on agents' phones after provisioning.

➢ **To force a reboot on provisioning using Configuration File:**

- Use the table below as reference.

Table 2-4: Forcing a Reboot on Provisioning

Parameter	Description
[voip/services/notify/check_sync/force_reboot_enabled]	Determines whether or not to force a reboot on provisioning. ▪ **[0]** Disable (default) ▪ **[1]** Enable

2.8 Provisioning using TFTP / FTP / HTTP / HTTPS in DHCP Options 66/67

This feature lets network administrators enable phones to be automatically provisioned when the phones are plugged in, using TFTP / FTP / HTTP / HTTPS in DHCP Options 66/67.

2.9 Provisioning using DHCP Option 43

This feature lets network administrators enable phones to be automatically provisioned when the phones are plugged in, using DHCP Option 43.

2.10 Enabling Agents to Sign in with Phone Numbers

This feature lets the call center administrator power up all phones without setting a valid SIP account. When an agent then wants to use their phone, they register to the network with their phone number.

➢ **To enable the feature using Configuration File:**

- Use the table below as reference.

Table 2-5: Enabling Agents to Sign in with Phone Numbers

Parameter	Description
[system/login_sk_before_signed_in]	Determines whether or not to enable agents sign in with phone numbers. ▪ **[0]** Disable (default) ▪ **[1]** Enable

2.11 Locking Agents Phones Alphabetical Keys

This feature lets call center network administrators lock agents' phones' alphabetical (non-numerical) keys so that only numerical keys are available to them. This feature provides call centers the option to limit agents to work-specific tasks. The feature reduces private activity on the part of agents. Agents cannot, for example, add contacts to a personal directory.

When this feature is enabled, agents can only use numbers. Only two menus are available in agents phones LCDs:

- Status
- Administration

➢ **To lock alphabetical keys using Configuration File:**

- Use the table below as reference.

Table 2-6: Locking Agents Phones Alphabetical Keys

Parameter	Description
[voip/block_non_numeric_key]	Determines whether or not to lock agents phones alphabetical keys and only allow them to use numerical keys. ▪ **[0]** Disable (default) ▪ **[1]** Enable

2.12 Provisioning using DHCP Option 12

This feature lets call center network administrators enable automatic provisioning using DHCP Option 12. The hostname is replaced by the agent's phone number after SIP registration.

2.13 Playing a Beep on an Incoming Call

This feature lets call center network administrators configure a beep to be played when a call comes in if auto-answer is configured. The beep is played on both speaker and headset. Agents will know from the beep that they have an incoming call to attend to.

➢ **To configure playing a beep using Configuration File:**

- Use the table below as reference.

Table 2-7: Playing a Beep on an Incoming Call

Parameter	Description
[voip/auto_answer/headset_beep/enabled]	Determines whether or not to play a beep on an incoming call, on the headset. ▪ **[0]** Disable (default) ▪ **[1]** Enable
[voip/auto_answer/speakerphone_beep/enabled]	Determines whether or not to play a beep on an incoming call, on the speaker. ▪ **[0]** Disable (default) ▪ **[1]** Enable

2.14 Enabling Proactive Mute

This feature lets call center network administrators enable a proactive mute when calls come in so that when they come in, callers cannot hear the agents until the agents unmute by pressing the **Mute** button. The feature can protect call centers from agent conduct that might be offensive to callers. Agents may for example pass an offensive remark to one another about a caller whose call is coming in, without realizing the caller can hear.

➢ **To enable proactive mute using Configuration File:**

- Use the table below as reference.

Table 2-8: Enabling Proactive Mute

Parameter	Description
[voip/proactive_mute/enabled]	Determines whether or not to enable proactive mute. ▪ **[0]** Disable (default) ▪ **[1]** Enable

2.15 Limiting the Length of Time 'Logged out' is Displayed

This feature lets call center network administrators limit the length of time the 'Logged out' message is displayed after agents log out. When agents log out, the 'Logged out' message will only be displayed in the phone's idle screen for the length of time (in seconds) configured by the call center network administrator, following which it will disappear.

Reader's Notes

3 Using IP Phones in Genesys Contact Centers

This section shows how to use AudioCodes IP phones in Genesys contact centers.

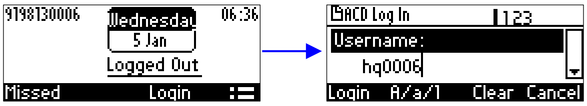
Note: The section is intended mainly for agents / hotline operators.

3.1 Logging in

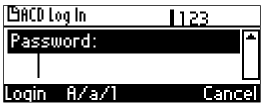
This section shows you how to log in to the phone. Log in immediately after starting a shift.

➢ **To log in to the phone:**

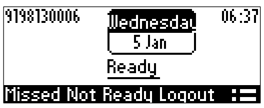
1. When the phone's LCD is in idle mode (Logged Out), press the **Login** softkey; the Log In screen is displayed:



2. Enter your Username. Obtain it from your system administrator. Press the **A/a/1** softkey successively to navigate to and select the alphanumerical mode you require (**abc**, **ABC**, or **Abc**).
3. Scroll down and enter your Password.



4. Press the **Login** softkey; the Ready idle screen is displayed.



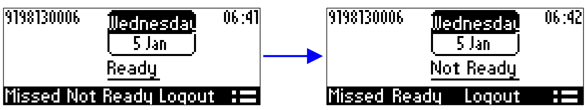
You're now available to take incoming calls. Incoming calls from now on will be directed to your phone.

3.2 Changing your Status from 'Ready' to 'Not Ready'

In the course of a shift, you may need to leave your desk for a break or to attend to other issues. Before leaving your desk, change your status to 'Not Ready' (unavailable) so that calls coming in to the contact center will not be sent to you.

➢ **To change your status to 'Not Ready':**

1. In the idle screen, press the **Not Ready** softkey; the Ready indication changes to Not Ready:

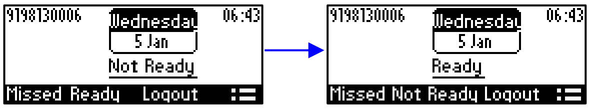


3.3 Restoring your Status from 'Not Ready' to 'Ready'

When you return to your desk after taking a break or after attending to an external issue, it's important to restore your status to 'Ready' and resume work.

➢ **To restore your status to 'Ready':**

- In the idle screen, press the **Ready** softkey; the Not Ready indication changes to Ready.

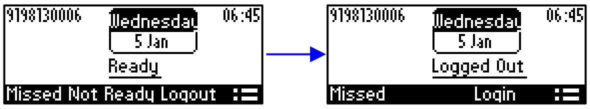


3.4 Logging out

At the end of your shift, log out of the phone.

➢ **To log out of the phone:**

- In the idle screen, press the **Logout** softkey; the Logged Out indication is displayed:

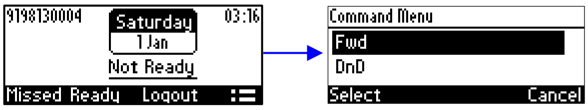


3.5 Configuring Automatic Forwarding

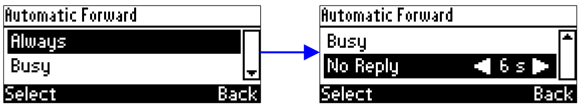
When you leave your workstation you can configure the phone so that any incoming calls will be forwarded.

➢ **To configure automatic forwarding:**

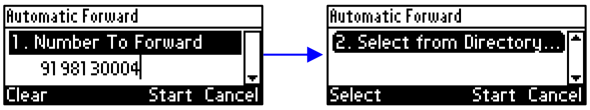
1. In the idle screen, press the **:≡** softkey; the Command Menu opens.



2. Select the **Fwd** option; the Automatic Forward screen opens.



3. Select the **Always** option or scroll down and select the **Busy** or **No Reply** option.
4. Enter the **Number to Forward** to, or scroll down and select **Select from Directory** in which you can choose a contact number to forward calls to.



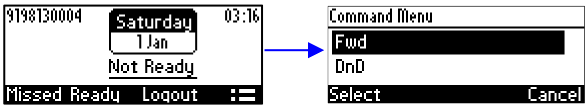
5. In the idle screen to which you're returned, view the foward indication:

3.6 Configuring Do Not Disturb (DnD)

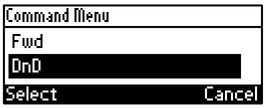
You can configure the phone so that no incoming calls will disturb you.

➢ **To configure DnD:**

1. In the idle screen, press the **:≡** softkey; the Command Menu opens.



2. Scroll down and select the **DnD** option:



3. In the idle screen to which you're returned, view the DnD indication:

3.7 Configuring 3rd Party Call Control (3PCC)

The 3PCC feature lets users control phones remotely from computer applications. 3PCC always supports the following functions:

- MakeCall (call initiation/setup)
- Release
- Hold
- Retrieve
- Transfer
- Consult
- Conference

➢ **To configure 3PCC using Configuration File:**

- Use the table below as reference.

Table 3-1: 3PCC Parameters

Parameter	Description
[voip/talk_event/enabled]	▪ **[0]** Disable (default) ▪ **[1]** Enable

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