Genesys

Genesys 420HD IP Phone with Genesys SIP Server

Configuration Guide

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Table of Contents

Preface

Welcome to the *420HD IP Phone with Genesys SIP Server Configuration Guide*

This guide shows users how to integrate the Genesys 420 HD IP Phone with Genesys SIP server**.**

About the 420HD IP Phone

Genesys IP phones are based on proprietary High Definition (HD) voice technology, providing clarity and a rich audio experience in Voice-over-IP (VoIP) calls. The phones are fully-featured telephones that provide voice communication over an IP network, allowing you to place and receive phone calls, put calls on hold, transfer calls, make conference calls, and so on.

The phone offers a wide variety of management and configuration tools:

- **Phone's LCD display user interface -** easy-to-use, menu-driven display screen, providing basic phone configuration and status capabilities
- **Web interface -** provides a user-friendly Web interface that runs on a Web browser (Microsoft® Internet Explorer is the recommended browser).
- **Configuration file -** text-based file (created using any plain text editor such as Microsoft's Notepad) containing configuration parameters and which is loaded to the phone using the Web interface or a TFTP, FTP, HTTP or HTTPS server.
- **TR-069** for remote configuration and management
- **CLI over Telnet**

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Document Version Number

A version number appears at the bottom of the inside front cover of this document. Version numbers change as new information is added to this document. Here is a sample version number:

420HD_IP_Phone_Guide_06-2014_v8.1.10100

You will need this number when you are talking with Genesys Customer Care about this product.

Chapter

1 Configuring Genesys 420HD IP Phone

Introduction

This guide describes how to configure Genesys 420HD IP Phone to operate with Genesys SIP Server in a Genesys contact center.

The guide also describes how to use the 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.

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.

Genesys 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'.

Ensuring SIP Business Continuity for Agents

- Genesys 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, Genesys 420HD IP phone will register on two sites simultaneously (i.e., register on both peer SIP Servers at the same time).

SIP Endpoint Features

Genesys 420HD IP 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 configuration guide details the supported features of 420HD with 2.0.0.18.0.45 version of firmware, and includes reference configuration examples.

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).

Feature Chart

Feature Name	
General Features Supported By Phone (1PCC)	**Supported**
Agent Login 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
SIP authentication	Yes
TLS/SRTP	No
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
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 known limitations

Provisioning IP Phones Automatically

This section shows system administrators how to quickly set up Genesys' 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, please see Using IP Phones in Genesys Contact Centers.on page 26.

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 Guide.

Verifying Firmware Version

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

➢ **To verify firmware version:**

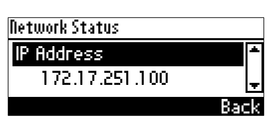
- Navigate to the Firmware Status screen in the phone's LCD (**MENU** key > **Status** > **Firmware Status**).

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.

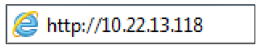
➢ **To obtain the phone's IP address:**

- 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.

Enabling the ACD Feature

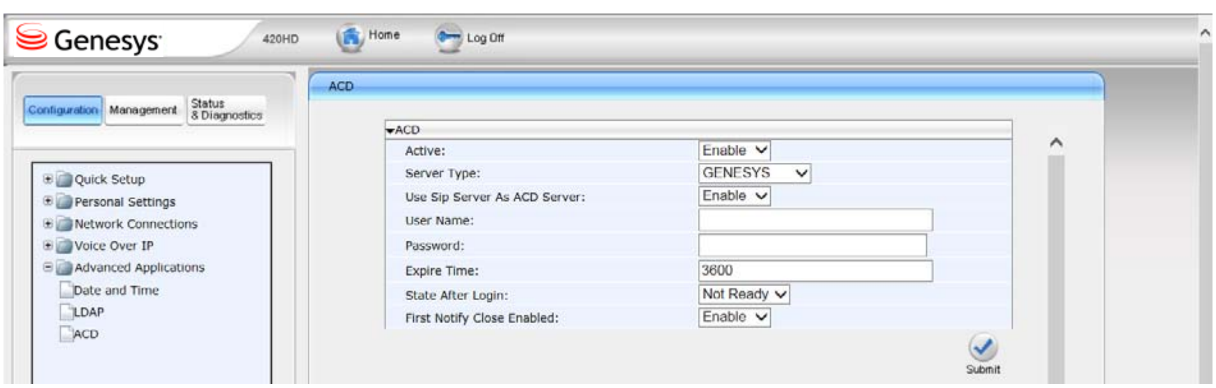
This section describes how to enable the ACD (Automatic Call Distribution) feature on the phone using either the Web interface or the configuration file. The feature distributes incoming calls to agents' phones on the basis of agent availability and unavailability.

An 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 the agent logs in from the phone.

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

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

Figure 1: Web Interface - ACD



2. Configure the parameters using **Table** 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 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 Genesys SIP Server. Default: 0.0.0.0
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 Genesys SIP 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.

Parameter	Description
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.
Expire Time [**voip/services/ACD/expire_time=3600]**	Specify the Expire Time parameter.

Configuring Dual Registration for Genesys Business Continuity

This section describes how to configure dual registration for Genesys Business Continuity, using the Web interface or configuration file.

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

1. In 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**, where **User ID** is the Genesys DN.
2. 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: Web Interface - Signaling Protocol – SIP Proxy and Registrar

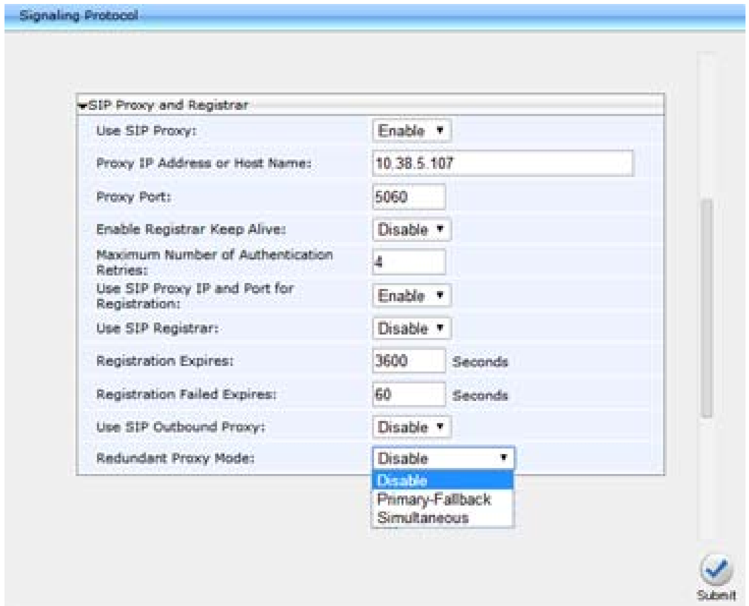


Figure 3: Web Interface - Signaling Protocol – SIP Proxy and Registrar – Secondary Proxy

3. Configure the parameters using the table below as reference.

➢ To configure using the configuration file:

- Use the table below as reference.

Table 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, **genesys.com**) of the Genesys SIP Server. Default: 0.0.0.0
Proxy Port **[voip/signalling/sip/proxy_port]**	The UDP or TCP port of the Genesys SIP 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.
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. Recommended value is 300 seconds.

Parameter	Description
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. Configure **[1] Enable**. • **[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). Configure **[0] Disable**. • **[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**, genesys.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. For the Genesys Business Continuity deployment, select **Simultaneous**; the phone registers with both SIP Server nodes 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 or FQDN of the secondary SIP Server node.
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 SIP Server node.

Parameter	Description
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. **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.

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 the configuration file:**

- Use the table below as reference.

Table 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.

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 the configuration file:**

- Use the table below as reference.

Table 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

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.

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.

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 the configuration file:**

- Use the table below as reference.

Table 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

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 the configuration file:**

- Use the table below as reference.

Table 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

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.

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 the configuration file:**

- Use the table below as reference.

Table 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

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 the configuration file:**

- Use the table below as reference.

Table 8: Enabling Proactive Mute

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

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.

Using IP Phones in Genesys Contact Centers

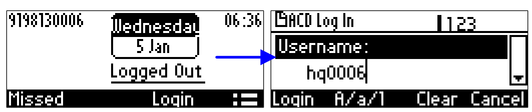
This section describes how to use Genesys IP phones to configure Agent Availability, Forward, and DND features.

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

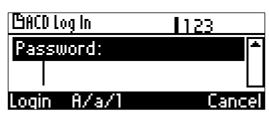
Logging in

➢ **To log in to the phone:**

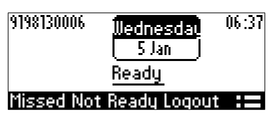
1. When the phone's LCD is in idle mode (Logged Out), press the **Login** softkey; the Log In screen is displayed:



2. Enter your username provided by the 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.



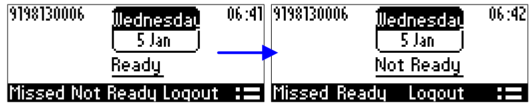
You're now available to take incoming calls. Incoming calls from now on will be directed to your phone.

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

When required, 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':**

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

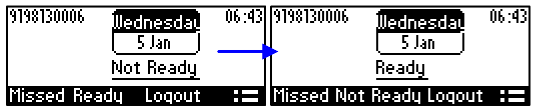


Restoring your Status from 'Not Ready' to 'Ready'

When required, restore your status to 'Ready' and resume work.

➢ To restore your status to 'Ready':

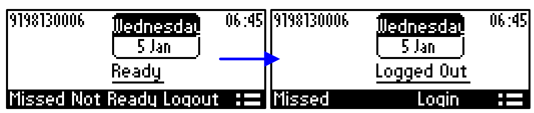
- In the idle screen, press the **Ready** softkey; the Not Ready indication changes to Ready.



Logging out

➢ To log out of the phone:

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

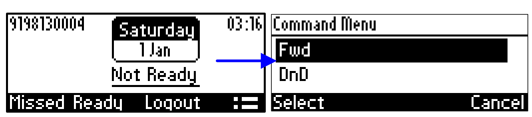


Configuring Automatic Forwarding

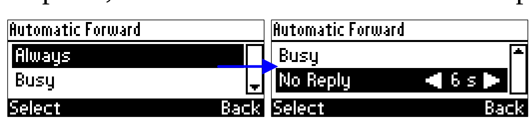
You can configure the phone so that any incoming calls will be forwarded.

➢ To configure automatic forwarding:

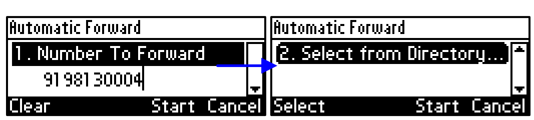
1. In the idle screen, press the **:=** softkey; the Command Menu opens.



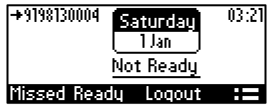
2. Select the **Fwd** option; the Automatic Forward screen opens.



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 forward indication:

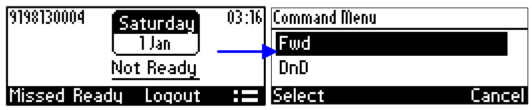


Configuring Do Not Disturb (DnD)

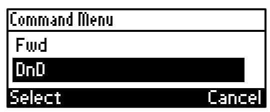
You can configure the phone so that no incoming calls will disturb you.

➢ **To configure DnD:**

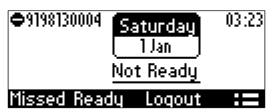
1. In the idle screen, press the ≔ softkey; the Command Menu opens.



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



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



Configuring 3rd Party Call Control (3PCC)

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

➢ **To configure 3PCC support using the configuration file:**

Use the table below as reference.

Table 9: PCC Parameters

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

Genesys

Chapter

2 Configuring SIP Server

Configuring SIP Server

This chapter describes how to configure features represented in the Feature Chart within a Genesys configuration environment.

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

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

Table 10: General Features Supported By Phone (1PCC)

General Features Supported By Phone (1PCC)	
Feature	**Key Actions and Procedures**
Agent Login from the Phone	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 ACD login operation on the phone. In the TServer section of the Agent Login object, configure: **password=<Any alphanumerical string>** **Notes:** • Name of the Agent Login object must match the User Name value entered from the phone when you enter Login credentials. • Value of the **password** option must match the password value entered on the phone when you enter Login credentials.
Caller ID	No configuration is required.
Call Forward	No configuration is required.
Do Not Disturb	No configuration is required.
DNS-based redundancy (using SIP Proxy)	Requires HA deployment using SIP Proxy deployment. SIP Proxy can be used in SIP Server standalone deployment or Genesys Business Continuity with SIP Proxy deployment. Refer to the *Genesys SIP Proxy Deployment Guide* and *Genesys SIP Server High-Availability Deployment Guide.*
DTMF tones generation	No configuration is required.
Multiple calls on one extension	See Call Control using desktop client -> Attended transfer feature.
Message Waiting Indicator	Configure a voice mailbox for an Extension. In the TServer section of the DN object, configure: **gvm_mailbox=<voice mail box number>** For example: gvm_mailbox=1502, where 1502 is a mailbox number.

General Features Supported By Phone (1PCC)	
Feature	**Key Actions and Procedures**
SIP authentication	1. Specify SIP requests (REGISTER, INVITE), which are sent by the phone to be authenticated by SIP Server. In the TServer section of the DN object, configure: **authenticate-requests=register,invite** 2. If required, configure the password used for authentication of incoming REGISTER or INVITE messages to SIP Server. In the TServer section of the DN object, configure: **password=<Any alphanumerical string>** **Note:** String must match the phone setting in **Configuration -> Voice Over IP -> Line Settings -> Authentication User Name** and Authentication **Password**.

Table 11: General Features Using Phone (1PCC)

General Features 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.

Table 12: General Features Using Desktop Client (3PCC)

General Features 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 Call-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.

General Features Using Desktop Client (3PCC)	
Feature	**Key Actions and Procedures**
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 the INVITE method or **refer-enabled=true** -> to use the REFER method
Genesys Business Continuity	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**

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
```

Known Issues and Limitations

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.