Dolby Conference Phone
Configuration guide for Microsoft Skype for Business

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1 Introduction

This documentation provides instructions for configuring your Dolby Conference Phone for Audiocodes SIP Phone Support Server, which provides support for Microsoft Skype for Business third-party SIP device integration. It is compatible with Dolby Conference Phone 3.2.

- About this guide
- Related documentation

1.1 About this guide

This guide presumes an enterprise network environment and does not cover provisioning of the service.

Requirements and prerequisite information on page 5 describes the network and firewall prerequisites and requirements, and lists the key Dolby Conference Phone configuration parameters.

Configuration procedures explains how to deploy a phone by using the Audiocodes SIP Phone Server web administration portal and the phone web interface.

1.2 Related documentation

Review all of the Dolby Conference Phone documentation to ensure that the phone is set up correctly so that you have the best conferencing experience possible.

These documents are available:

- Dolby Conference Phone administrator’s guide
  This guide explains how to set up and provision the Dolby Conference Phone.

- Dolby Conference Phone quick start guide
  This guide describes the contents of the phone package, how to assemble the phone, and how to connect the phone to the network. The quick start guide is included in the phone package. It is also available from the Dolby Conference Phone support pages.

- Dolby Conference Phone open source software guide
  This guide describes the open source software used in the Dolby Conference Phone software.

- Dolby Satellite Microphones quick start guide
  This guide describes the contents of the microphone package and how to connect them to the Dolby Conference Phone. The quick start guide is included in the microphone package. It is also available from the Dolby Conference Phone support pages.

- Dolby Voice compatibility guide
  This guide describes the compatibility relationships between the various Dolby Voice products.
2 Requirements and prerequisite information

Before deploying Dolby Conference Phones, gather the prerequisite network information, ensure that PoE is available for the phones and that the local network complies with the firewall requirements, and become familiar with the phone configuration parameters.

- Prerequisite information
- Network firewall requirements
- Configuration parameters

2.1 Prerequisite information

Have the information described in this section available before you start. Read the appropriate Audiocodes SIP Phone Support Server documentation for information on specific network requirements.

Note: Your local system administrator can provide these details.

You need this information before you can complete the steps outlined in this documentation:

2.2 Network firewall requirements

Network Address Translation (NAT) is not supported by the Dolby Conference Phone. Therefore, the Audiocodes SIP Phone Support Server servers should be on the same side of the organization firewall as the Dolby Conference Phones.

2.3 Configuration parameters

Audiocodes SIP Phone Support Server configuration options have corresponding Dolby Conference Phone configuration parameters.

<table>
<thead>
<tr>
<th>Audiocodes SIP Phone Support Server configuration option</th>
<th>Dolby Conference Phone configuration parameter</th>
<th>Required/optional</th>
</tr>
</thead>
<tbody>
<tr>
<td>User name</td>
<td>Sip.Credential.Name</td>
<td>Required</td>
</tr>
<tr>
<td>Password</td>
<td>Sip.Credential.Password</td>
<td>Required</td>
</tr>
<tr>
<td>Directory number</td>
<td>Sip.Account.UserName</td>
<td>Required</td>
</tr>
<tr>
<td>Primary Audiocodes SIP Phone Support Server IP address (or outbound proxy)</td>
<td>Sip.Pbx.OutboundProxy1</td>
<td>Required</td>
</tr>
<tr>
<td>Secondary Audiocodes SIP Phone Support Server IP address (or outbound proxy)</td>
<td>Sip.Pbx.OutboundProxy2</td>
<td>Optional</td>
</tr>
</tbody>
</table>

For information about using the Dolby Conferencing Console to set configuration parameters, see the Dolby Conference Phone administrator's guide.
3 Configuring a new user and the phone

Deploying a Dolby Conference Phone with Audiocodes SIP Phone Support Server requires adding a new user in the Audiocodes SIP Phone Server, configuring the Dolby Conference Phone by using its touch screen and web interface, and then registering the phone to Audiocodes SIP Phone Support Server with the phone web interface.

• Adding a new user
• Plugging in and setting up the phone automatically
• Configuring 802.1X
• Configuring the operation mode
• Opening the phone web interface
• Configuring NTP
• Registering the phone to Audiocodes SIP Phone Support Server

3.1 Adding a new user

You must add a new user to set up a Session Initiation Protocol (SIP) phone extension/account for the Dolby Conference Phone.

Procedure

1. Log in to the Audiocodes SIP Phone Support Server web administration page with the appropriate credentials.

2. Click Management.

3. In the Navigation Tree, expand Users, and then click Add user.
4. In the **Filter** field, type the Skype for Business user name, or type * for all users, and click **Find**.

5. In the list of Skype for Business users, click **Add** next to an available user, indicated by the **add** icon.

6. Click **Submit**.
7. Add additional users as required.

8. In the **Navigation Tree**, click **Users** to see a list of configured users. Click the Edit User icon for a user that you added.

9. On the **Edit User** page, under **Advanced Settings**, enter the user's allocated Password and Numeric Name, which can be an extension number or any alphanumeric name. This information is used to authenticate the SIP device. The Numeric Name can be up to 16 characters long and cannot include the characters #, @, or %. After entering the Password and Numeric Name, click **Submit**.
3.2 Plugging in and setting up the phone automatically

Each Dolby Conference Phone requires a wired network connection, preferably supporting PoE. Once your phone is plugged in, many settings may be automatically detected.

Procedure

1. Connect the phone to a PoE Ethernet port on the network using an Ethernet cable.
   The setup wizard appears on the phone. The screens vary depending on your network setup.

2. If the phone has Internet access and can connect to the Dolby Voice Console and your conferencing service provider's provisioning service:
   This only happens if you meet the requirements for Plug-and-Play Setup setup.
   Requirements for Plug-and-Play Setup vary based on your conferencing service provider. For a list of requirements, see the configuration guide for your conferencing service provider.
   • If the phone is recognized by the Dolby Voice Console, the phone connects to your conferencing service provider's provisioning service. No additional action is required to provision the phone.
   • If the phone is not recognized by the Dolby Voice Console, select your conferencing service provider from the list. Follow the on-screen prompts to connect the phone to your conferencing service provider's provisioning service.

   If this does not happen, proceed to the next step.

3. If the phone discovers a Dolby Conferencing Console or a provisioning server on your network, follow the on-screen prompts to connect the phone to one of these provisioning servers.
   This only happens if the phone's administrator completed preparations so that the phone can discover the presence of a Dolby Conferencing Console or a provisioning server by means of a Domain Name System (DNS) SRV record.
If this does not happen, proceed to the next step.

4. If you cannot set up the phone with the methods mentioned earlier, you must perform Manual Setup.

   a) Choose **Manual setup** from this screen:

   ![Manual setup screen](image)

   b) Follow the on-screen prompts to connect the phone to your Dolby Conferencing Console or provisioning server.

**Results**

The phone is now connected to either your conferencing service provider's provisioning service, or a Dolby Conferencing Console or a provisioning that is on your network.

**What to do next**

If needed, see the *Dolby Conference Phone administrator's guide* for complete information and instructions for these setup methods:

- Plug-and-Play Setup
- Automatic Local Setup
- Manual Setup

### 3.3 Configuring 802.1X

If your network requires 802.1X authentication, use the touch screen to accept the authentication server certificate and enter credentials.

**About this task**

The network setup wizard detects many of the settings for you, including your virtual LAN (VLAN) and IP network settings. In this figure, the Dolby Conference Phone has detected those settings upon initial setup.

![Network setup screen](image)

If you see this screen, you can configure some 802.1X options, described in this section, directly in the wizard.
Procedure

1. If a PKI-based authentication method is selected by the authentication server, you are prompted to review the server certificate information.

2. Enter the user name and password when the server requires credentials for authentication. Depending on the authentication server configuration, you may be prompted to enter authentication credentials before you accept the server certificate.

3.3.1 Updating 802.1x authentication credentials

You can update 802.1x authentication credentials without rebooting or resetting the Dolby Conference Phone.

Procedure

1. From the phone home screen, tap this sequence:

   ![Sequence of taps]

2. Enter the administrator password, and tap Enter.
3. Select Network Configuration > 802.1x.

3.4 Configuring the operation mode

You can configure the Dolby Conference Phone for either Conferencing mode or Dual mode.
About this task

Conferencing mode
Use if you plan on using your phone with your conferencing service provider only.

Dual mode
Use if you want to have IP private branch exchange (PBX) integration with your conferencing service provider.

If you want to have IP PBX integration with your conferencing service provider, change the configuration mode listed in this table.

<table>
<thead>
<tr>
<th>Configuration parameter description</th>
<th>Configuration parameter name</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selects the operational mode</td>
<td>Features.OperationMode</td>
</tr>
</tbody>
</table>

Procedure

1. Go to **Settings > Features** in the web interface.
2. Choose a value for **Operation Mode**.

3.5 Opening the phone web interface

Enter the phone IP address in your web browser to open the phone web interface.

Prerequisites

You must have a network-connected desktop or laptop computer with a web browser supporting HTML5 installed. (Modern versions of Internet Explorer, Firefox, Chrome, and Safari are supported.)

The web browser will be used to configure the phone, so it must have IP network connectivity to the Dolby Conference Phone.

Procedure

1. From the phone home screen, tap this sequence:
   
   ![Button sequence](image)

   > About > General

2. Scroll down until you see the IP address.

3. Enter this URL in the browser address bar.
   
   If a security warning displays, accept the warning and proceed.
   
   https://your phone IP address

4. Log in to the Dolby Conference Phone using the default credentials:
   
   • **Username** = admin
• **Password** = 1739

### 3.6 Configuring NTP

You must specify Network Time Protocol (NTP) servers in the network settings so that the phone displays the correct date and time.

**Procedure**

1. Click the + to open the network settings parameters.

2. Click the + to open the **NTP** section.
3. Enter the server addresses provided by your local system administrator.

3.7 Registering the phone to Audiocodes SIP Phone Support Server

Use the Settings page of the Dolby Conference Phone web interface to specify the Audiocodes SIP Phone Support Server account settings and register the phone.

Procedure

1. Select Settings > IP PBX Settings > Account.
2. Change the **Account** settings as shown:

- **Display Name**: The name of the phone user or conference room (for example, Sam Smith or Board Room)
- **Extension Number/Address**: Extension number (mandatory)
  
  If desired, you can enter a full phone number instead of just the extension.
- **Display Number**: Extension number
  
  If you do not enter a value, the **Extension Number/Address** value displays.
- **Transport Type**: AUTO (works with Transmission Control Protocol (TCP) and User Datagram Protocol (UDP))
- **Secure Media**: Mandatory
- **Transport Port**: 5060

3. Change the **Server** settings as shown:

- **Primary Call Server/Outbound Proxy**: IP address of your Audiocodes SIP Phone Support Server.
- **Primary Server/Outbound Proxy port**: 5060.
  
  If you have a secondary (failover) server available, enter the details in the **Secondary Call Server/Outbound Proxy** field.
- **Set Transport Type** to match your selection in the connection settings of the previous procedure.
Note: Ensure that you are adding the correct Primary/Secondary account details.

4. Update the credentials.

These settings are mandatory for digest authentication:

a) Enter the **User Credential Name** that you created in Audiocodes SIP Phone Support Server.

b) Enter the **User Credential Password** that you created in Audiocodes SIP Phone Support Server.

c) Enter the server realm you created, or add an * (asterisk) to cover any string for this field.

d) Confirm the password, and click **Save**.

If the information is correct, the phone registers with Audiocodes SIP Phone Support Server.

5. To verify the registration with Audiocodes SIP Phone Support Server, go to **Status > System > Device**.
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### System

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Serial Number</td>
<td>[obfuscated]</td>
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<tr>
<td>Device Setup Completed</td>
<td>true</td>
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<tr>
<td>Configuration Errors</td>
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<tr>
<td>Board Revision</td>
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<tr>
<td>Number of CA Certificates</td>
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<td>Number of Not Valid Yet CA Certificates</td>
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<tr>
<td>Number of Expired CA Certificates</td>
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<td>Number of Expiring Soon CA Certificates</td>
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<td>CA Certificates Details</td>
<td>Click for more details</td>
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<td>Services Memory Usage (KB)</td>
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<td>Services Virtual Memory Usage (KB)</td>
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<tr>
<td>GUI Memory Usage (KB)</td>
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<tr>
<td>GUI Virtual Memory Usage (KB)</td>
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<td>Overall Memory Usage (KB)</td>
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<td>DSP CPU load (%)</td>
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<td>DSP Memory Usage (KB)</td>
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<tr>
<td>Services CPU load (%)</td>
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</tr>
<tr>
<td>CPU load (%)</td>
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<tr>
<td>GUI CPU load (%)</td>
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<tr>
<td>User Data Partition Available Space (KB)</td>
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<td>System Uptime</td>
<td>8 days, 03:12:18</td>
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<tr>
<td>System Boot Time (seconds)</td>
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</tr>
<tr>
<td>MAC Address</td>
<td>[obfuscated]</td>
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<tr>
<td>System Warning</td>
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</tr>
<tr>
<td>Network - Primary VLAN</td>
<td></td>
</tr>
<tr>
<td>Network - Secondary VLAN</td>
<td></td>
</tr>
<tr>
<td><strong>SIP Registered</strong></td>
<td></td>
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<tr>
<td><strong>SIP Registration Response</strong></td>
<td>200 OK</td>
</tr>
<tr>
<td><strong>Active SIP Registrar</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Active SIP Transport</strong></td>
<td>TCP</td>
</tr>
<tr>
<td>Network Time Synchronization</td>
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</tr>
<tr>
<td>BT MeetMe Registration</td>
<td></td>
</tr>
<tr>
<td>LDAP Connected</td>
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</tr>
<tr>
<td>Provision Successful</td>
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<td>Software Update Successful</td>
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<td>Logs Upload Successful</td>
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<td>Logging Impact</td>
<td></td>
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<tr>
<td>Reboot Status</td>
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<td>Call in progress</td>
<td>false</td>
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<td>CA Certificates</td>
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<tr>
<td>Dolby® Satellite Microphones</td>
<td>No Microphones Connected</td>
</tr>
<tr>
<td>Application status</td>
<td></td>
</tr>
<tr>
<td>Application version</td>
<td></td>
</tr>
</tbody>
</table>
Glossary

DHCP
Dynamic Host Configuration Protocol.

DNS
Domain Name System. An Internet service that translates Internet domain and host names to IP addresses and conversely. DNS automatically converts between the name entered in a web browser and the IP addresses of the web server hosting the site whose URL is entered in the web browser.

IP
Internet Protocol.

NAT
Network Address Translation. An Internet standard that enables a local area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic.

NTP

PBX
Private branch exchange. A phone system that is delivered as a hosted service.

PoE
Power over Ethernet. A solution in which an electrical current is run to networking hardware over Ethernet category 5 or higher data cabling.

SIP
Session Initiation Protocol. An application-layer communications protocol used for signaling and controlling communications sessions.

TCP
Transmission Control Protocol. A communications protocol that specifies how data should be formatted, addressed, transmitted, routed, and received at the destination. Part of the Internet protocols communications suite.

TLS
Transport Layer Security. A cryptographic protocol designed to provide communications security over a computer network.

UDP
User Datagram Protocol. A communications protocol that uses no handshaking dialogues to establish a connection with the remote host. UDP is a member of the IP suite.

VLAN
Virtual LAN. Any broadcast domain that is partitioned and isolated in a computer network at the data link layer (OSI layer 2).