Manage Sites

Configure Intersite Dialing—Mesh Topology

Before you begin

- Ensure that you have read the architecture and design considerations that are required for configuring Intersite Dialing.

- A minimum of two BE4000 sites in the "Online" state.

Step 1	Log in to the BE4000 portal.
Step 2	Hover the mouse over **Manage Customer** under **Actions** column for the row containing desired customer name and click **Intersite Dialing**.
Step 3	Click **Mesh Topology**.
Step 4	Move the slider to enable Mesh topology.
Step 5	Check the **Enable** check box corresponding to the desired site and enter a two-digit identification number for the site in the **Site ID** field.
	Calls between the sites are routed based on the identification number that is entered here.
Step 6	Enter the maximum number of concurrent outbound intersite calls that can be made from the site in the **Max Calls** field.
Step 7	Click **Save**.

Intersite Dialing - Field Descriptions for Mesh Topology

Field	Description
Enable	Enables the BE4000 Site for Intersite Dialing. "Enable" check box is active only for the BE4000 sites that are in the "Online" state.
Site ID	This unique ID is added to the calling party number for outbound calls so that recipients can use 'Redial' to call back to that caller. Site ID should be a two digit value (Range: 10 to 99).

Field	Description
Site Name	Name of the BE4000 Site. The details are displayed by default and cannot be changed.
Site Status	Status of the BE4000 site.
Max Calls	Limits the maximum number of concurrent outbound inter-site calls for a site. Set this based on available bandwidth, service provider policies, and site codec. Max Calls should be a two digit value (Range: 10 to 99).

Configure Intersite Dialing—Star Topology

Before you begin

- Ensure that you have read the architecture and design considerations that are required for configuring Intersite Dialing.

- A minimum of two BE4000 sites in the "Online" state.

Step 1   Log in to the BE4000 portal.

Step 2   Hover the mouse over **Manage Customer** under **Actions** column for the row containing desired customer name and click **Inter Site Dialing**.

Step 3   Click **Star Topology**.

Step 4   Click **Yes, Let's Begin** in the **First Time Setup** pop-up window to display configuration options.

Step 5   Choose the template that best matches your controller hardware from the **Provider Template** drop-down list on **Service Settings** page and click **Next**.

Step 6   Enter the required information in the **Proxy Server** page. For detailed information on fields, refer Intersite Dialing - Field Descriptions for Star Topology, on page 4. Click **Next**.

Step 7   Click **Add Row** and enter a trusted IP address in the **Security** page to authenticate incoming calls for toll fraud prevention.

Step 8   Verify the configured central call control configurations. If you choose to modify the configurations, click **Replace Configuration**. This allows you to configure star topology from the beginning.

Step 9   Check the **Enable** check box corresponding to the desired site and enter a two-digit identification number for the site in the **Site ID** field.

Calls between the sites are routed based on the identification number that is entered here.

Step 10   Enter the maximum number of concurrent outbound intersite calls that can be made from the site in the **Max Calls** field.

Step 11   Click **Save**.

Intersite Dialing - Field Descriptions for Star Topology

Field	Description
Provider Template	Choose a preconfigured provider template based on your SIP service provider. If you do not have a preconfigured provide template, choose **ISD Custom**.
Proxy Address	Enter an IP address and Port, Fully Qualified Domain Name (FQDN) and Port, or SRV record for your service proxy.
Proxy Port	Enter a port number for your service proxy.
Transport Layer	Choose the protocol used for transport layer by your service provider.
Fax Transmission Protocol	Choose one of the ITU-T T.38 standard Fax Transmission Protocols to be used in your configuration. Available options are: • T.38 • T.38 fall back to G.711u • T.38 fall back to G.711a • Pass Through G711u • Pass Through G711a
Trusted IP Address	Enter a trusted IP address to authenticate incoming SIP trunk calls for toll fraud prevention.
Add Row	Click **Add Row** and enter the trusted IP addresses.
Site ID	This unique ID is added to the calling party number for outbound calls so that recipients can use 'Redial' to call back to that caller. Site ID should be a two digit value (Range: 10 to 99).
Max Calls	Limits the maximum number of concurrent outbound inter-site calls for a site. Set this based on available bandwidth, service provider policies, and site codec. Max Calls should be a two digit value (Range: 10 to 99).
Enable	Enables each site to participate in the central topology configuration.
Connectivity Report	Validates and confirms the connectivity between enabled sites and Central Call Controller.

Manage Customer Administrators

Cisco Partners and Customer Administrators can add additional Customer Administrators. The newly added Customer Administrator can view all the BE4000 sites that are associated with the customer organization.

Add Additional Customer Administrators

Cisco Partners and Customer Administrators can add additional Customer Administrators. The newly added Customer Administrator can view all the BE4000 sites that are associated with the customer organization.

Before you begin

Have the name, email address, and contact number of the new customer administrator ready.

Step 1	Log in to the Cisco Business Edition Selfcare Portal (https://myphone.cisco.com) as customer administrator.
Step 2	Click ![icon] **Admins** on the dashboard.
Step 3	Click **Add Admin**.
Step 4	Enter the **Name**, **Email**, **Contact Number**. **Role** is chosen as "Admin" by default.
Step 5	Click **Save**.
	An email containing the link to register to the Cisco Business Edition Selfcare Portal is sent to the entered email address.

Change Customer Administrator for a BE4000 Site

You can change the customer administrator who is assigned for a BE4000 site.

Before you begin

BE4000 site must be online.

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from the Actions column for the desired site.
Step 3	Click edit (pen icon) under Contact Information.
Step 4	Choose the desired customer administrator's email address from the **Customer Admin Email** drop-down list. Only the administrators belonging to the same customer is listed in the drop-down. The **Contact Name** field gets auto-populated based the chosen email address.
Step 5	Enter the phone number in the **Outside Phone Number** field. Provide the mobile number or a number that can be reached by outside the BE4000 system. Do not enter the extensions within the BE4000 system.

Resend Registration Email for BE4000 Customer Administrators

If a customer administrator has lost the initial registration email that is sent by the BE4000, you can resend it from the **Admins** menu.

Step 1	Log in to the Cisco Business Edition Selfcare Portal (https://myphone.cisco.com) as customer administrator.
Step 2	Click ![icon] **Admins** from the dashboard.

Step 3   Click **Resend Registration Mail** from the **Action** column corresponding to the desired name.

A pop-up appears after the email is successfully sent.

Remove Customer Administrator

A Cisco Partner or Customer Administrator can remove Customer Administrator. When a Customer Administrator is removed, the admin permissions to view and modify BE4000 sites is also removed.

Step 1   Log in to the Cisco Business Edition Selfcare Portal (https://myphone.cisco.com) as customer administrator.

Step 2   Click Admins on the dashboard.

Step 3   Click **Remove Admin** from the "Action" column corresponding to the desired Customer Administrator.

Step 4   Click **Remove**.

If the Customer Administrator who was removed had a BE4000 site assigned, then the portal automatically assigns the Customer Administrator who performed the "Remove Admin" action as the new Customer Administrator.

Log In to the Cisco Business Edition Selfcare Portal Using Recovery Code

Before you begin

Have the recovery code that you received when you registered to the Cisco Business Edition Selfcare Portal.

Step 1   Enter the URL https://myphone.cisco.com.

Step 2   Click **Sign in**.

Step 3   Click **Use Recovery Code**.

Step 4   Enter the Recovery code in the **Recovery Code** field.

Using the recovery code permanently deletes all additional OTP accounts that you have created.

Step 5   Click **Continue**.

Step 6   Accept the Terms of Service.

Step 7   Click **Continue**.

Step 8   Enter the device name. The name entered here is displayed on the OTP application that is configured on your smart phone. Click **Continue**.

Step 9   Use the OTP application to scan the QR code. Click **Continue**.

Step 10   Enter the security code that is generated by your OTP application. Click **Continue**.

Step 11   Save the **Rescue Code** displayed on the screen.

Step 12   Click **Continue**.

Share the Configuration Details of a BE4000 Site

You can share the configuration details of a BE4000 site with others.

Before you begin

Ensure that the BE4000 site is in the "Online" state.

Step 1   Log in to the BE4000 portal.

Step 2   Click **...** under **Actions** column corresponding to the desired site from the dashboard.

Step 3   Click **Configuration Summary**.

Step 4   Click **Export**. A PDF version of the configuration summary is downloaded.

Overview of Auto Attendant

Auto Attendant service (also referred to as a virtual receptionist), is a phone system that enables your callers to be automatically transferred to an extension, eliminating the need for a receptionist and avoiding extended waiting period. BE4000 provides you an automated phone answering facility to communicate effectively with customers and improve your business operations. An auto attendant answers all incoming calls with an audio greeting and options menu (different for open and closed hours). A maximum of five submenus with a maximum depth of 3 levels can be configured. The caller can select a menu option to reach to the desired extension.

You can define the number of times the menu options is played to the caller before the call reaches the drop through destination. You can also define where the call lands if no action is performed by the caller even after the defined number menu repetitions.

Cisco partner, customer administrator, and any end user with PromptAdministrators privileges on their personal voicemail box can update auto attendant greetings and prompts.

Add Auto Attendant

You can add or update Auto Attendant service post site deployment.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from "Actions" menu for the desired site on the dashboard.

Step 3   Click **Auto Attendant**.

Step 4   Enter the information in the fields.

Step 5   Click the arrows at the bottom of the screen to move to next screen or to go back to the previous screen.

Step 6   Click **Yes**.

Auto Attendant - Field Descriptions

Field	Description
Pilot Number	Enter the number that callers dial to reach auto attendant. A minimum of four digits is required. Range is from 1000 to 9999.
How many times do you want message to be played	Number of times the audio file is played to the caller before the call reaches the drop through destination. Value range is from 1 to 9. Default value is 4.
Drop Through Destination	Defines where the call lands if no action is performed by the caller even after playing the menu for the defined number of repeats. You can configure one of the following as the drop through destination: • **Route to Extension**—All extensions are listed in the drop-down. • **Route to Voicemail Box**—All extensions that have "Voicemail" enabled are listed in the drop-down. • **Route to Hunt Group**—All Hunt Groups are listed in the drop-down. • **Route to Group Voicemail Box**—All group voicemail boxes are listed in the drop-down. **Note** During the initial site deployment (in the Setup Assistant), the drop-down shows an option, only if you create a group voicemail box on the **Hunt Groups** page, by choosing "Route to Group Voicemail Box" from the **When No Member is Available** drop-down list. • **Disconnect Call**—The call gets disconnected.
Audio Prompt (Welcome Message)	Add an audio prompt for welcome Message. The BE4000 provides a default audio file. This audio message is played first when a call is answered by the auto attendant. You can also upload a new .wav file. To select a new file, click Upload. **Note** BE4000 supports only .wav audio file with G.711 u-law, 8kHz, 8 bit, Mono format. The file cannot be larger than 1 MB (about 2 minutes). The filename cannot have space and special characters. The system default message is a silent recording. You need to replace the message with a custom recording.

Field	Description
Audio Prompt (Open Message)	Add an audio prompt for open message. The BE4000 provides a default audio file. This audio message is played when a call is answered during the open business hours. You can also upload a new .wav file. To select a new file, click Upload. **Note** BE4000 supports only .wav audio file with G.711 u-law, 8kHz, 8 bit, Mono format. The file cannot be larger than 1 MB (about 2 minutes). The filename cannot have space and special characters. The system default message is a silent recording. You need to replace the message with a custom recording.
Add Menu Option	Add customized menu options. You can add 0-9 menu options in addition to a * menu. Each menu option can be labeled in a meaningful way to help identify locations or users in your system using any one of the following: Dial by Name, Pilot Number, Dial by Number, Call Hunt Group, Return to Main Menu, or Submenu.
Audio Prompt (Closed Message)	Displays the default audio file that is played for all calls received during closed hours. You can play the existing file or upload a new .wav file. To select a new audio file, click **Upload**. **Note** BE4000 supports only .wav audio file with G.711 u-law, 8kHz, 8 bit, Mono format. The file cannot be larger than 1 MB (about 2 minutes). The filename cannot have space and special characters. The system default message is a silent recording. You need to replace the message with a custom recording.
Add Menu Option	Add customized menu options. You can add 0-9 menu options in addition to a * menu. Each menu option can be labeled in a meaningful way to help identify locations or users in your system using any one of the following: Dial by Name, Pilot Number, Dial by Number, Call Hunt Group, Return to Main Menu, or Submenu.

Enable Users to Update Auto Attendant Greeting and Prompts

An end user with "PromptAdministrators" privileges can update auto attendant greeting and prompts from a phone. To assign "PromptAdministrators" privilege for an end user, perform the following:

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Personal Voicemail Box**.

Step 4 Click **Modify** from the Actions menu for the desired user.

Step 5 Navigate to **Groups** page and click **Groups**.

Step 6   Check **PrompAdministrators**. (Hover the mouse on **PromptAdm** to view the complete text)

Step 7   Click **OK**.

Auto Attendant Prompts

BE4000 provides default audio files for Welcome, Open, and Closed prompts. You can also upload custom audio files to the prompts according to your business need.

- The BE4000 portal displays *AA_aPrompt1.wav* as the default for welcome prompt and contains the message "*Prompt number 1, Auto Attendant, Prompt Number 1*"

- The BE4000 portal displays *AA_bPrompt2.wav* as the default for open prompt and contains the message "*Prompt number 2, Auto Attendant, Prompt Number 2*"

- The BE4000 portal displays *AA_cPrompt3.wav* as the default prompt for closed prompt and contains the message "*Prompt number 3, Auto Attendant, Prompt Number 3*"

Modify Auto Attendant Prompts from a Phone

Before you begin

If you are an end user, you should have "PromptAdministrators" privileges. Contact your customer administrator to know if you have "PromptAdministrators" privileges.

Step 1   Dial **70397**.

Step 2   Enter your extension followed by the pound (#) key.

Step 3   Enter the PIN followed by the pound (#) key.

Enter the PIN that you use to access your voicemail. If you have forgotten the PIN, log in to the BE4000 Selfcare Portal and reset the PIN. Refer End User Guides for detailed procedure to reset the PIN.

You hear the prompts as follows:

Welcome to administration via telephone system.

- *To administer the Automated Attendant Alternate Greeting, press 1*

- *To administer custom prompts, press 2*

Step 4   Press **2**.

You hear the following:

- *Press 1 record a new prompt*

- *Press 2 to edit previously recorded prompt*

- *Press * to go back previous menu*

Step 5   Press **2**.

You hear the total number of prompts available for your network and the message recorded for each prompt. You hear the following options for each prompt:

- *Press 2 to rerecord the prompt*

- *Press 3 to delete it*

- *Press # To go to next prompt*

- *Press * to skip playback of prompts*

Step 6   Press **2** when you hear the prompt you want to rerecord.

You hear the following:

Record the new prompt at the tone. Press # to finish recording.

Step 7   Record the new prompt after the beep.

Step 8   Press **#** key to complete the recording.

You hear the newly recorded prompt followed by the following options:

- *Press 1 to save the prompt*

- *Press 2 to delete the prompt*

Step 9   Press **1** to save the newly recorded prompt.

The consecutive prompt is played.

Step 10   Repeat steps 6 to 9 for the prompts that you want to edit.

Step 11   End the call.

Alternate Greeting

Alternate greeting is a voice message recorded from a phone by the customer administrator or a phone user with the "PromptAdministrators" privileges. Alternate greeting is used if there is a disaster or a sudden business need to replace the existing prompts. Alternate greeting is played first followed by the welcome prompt and the open or closed prompts. Alternate greeting is played irrespective of open and closed business hours.

Note   Alternate Greeting does not show up in the BE4000 portal and all administration must be done via the Prompt Administrator Phone Menu 70397.

Add Auto Attendant Alternate Greeting from a Phone

Before you begin

If you are an end user, you should have "PromptAdministrators" privileges. Contact your customer administrator to know if you have "PromptAdministrators" privileges.

Step 1   Dial **70397**.

Step 2   Enter your extension followed by the pound (#) key.

Step 3   Enter the PIN followed by the pound (#) key.

Enter the PIN that you use to access your voicemail. If you have forgotten the PIN, log in to the BE4000 Selfcare Portal and reset the PIN. Refer End User Guides for detailed procedure to reset the PIN.

You hear the prompts as follows:

Welcome to administration via telephone system.

- *To administer the Automated Attendant Alternate Greeting, press 1*

- *To administer custom prompts, press 2*

Step 4   Press **1**.

You hear the following prompts:

- *Press 1 to record a new prompt*

Step 5   Press **1**.

You hear the following:

Record the new prompt at the tone. Press # to finish recording.

Step 6   Record a new prompt after the beep.

Step 7   Press the **#** key to complete the recording.

You hear the newly recorded prompt followed by the following options:

- *Press 2 to save the prompt*

- *Press 3 to delete the prompt*

Step 8   Press **2** to save the prompt.

Modify Auto Attendant Alternate Greeting from a Phone

Before you begin

- If you are an end user, you should have "PromptAdministrators" privileges. Contact your customer administrator to know if you have "PromptAdministrators" privileges

- Alternate Greeting must be active

Step 1   Dial **70397**.

Step 2   Enter your extension followed by the pound (#) key.

Step 3   Enter the PIN followed by the pound (#) key.

Enter the PIN that you use to access your voicemail. If you have forgotten the PIN, log in to the BE4000 Selfcare Portal and reset the PIN. Refer End User Guides for detailed procedure to reset the PIN.

You hear the prompts as follows:

Welcome to administration via telephone system.

- *To administer the Automated Attendant Alternate Greeting, press 1*

- *To administer custom prompts, press 2*

Step 4   Press **1**.

You hear the following prompts:

Your alternate greeting is currently active.

- *Press 1 to hear the alternate greeting*

- *Press 2 to rerecord the alternate greeting*

- *Press 3 to deactivate this alternate greeting. Deactivating the alternate greeting will delete it from the system*

Step 5   Press **2**.

You hear the following:

Record the new prompt at the tone. Press # to finish recording.

Step 6   Record a new prompt after the beep.

Step 7   Press the **#** key to complete the recording.

You hear the newly recorded prompt followed by the following options:

- *Press 1 to save and activate this alternate greeting*

- *Press 2 to hear the alternate greeting*

- *Press 3 to rerecord the alternate greeting*

Step 8   Press **1** to save the modified alternate greeting.

Delete Auto Attendant Alternate Greeting from a Phone

Before you begin

- If you are an end user, you should have "PromptAdministrators" privileges. Contact your customer administrator to know if you have "PromptAdministrators" privileges

- Alternate Greeting must be active

Step 1   Dial **70397**.

Step 2   Enter your extension followed by the pound (#) key.

Step 3   Enter the PIN followed by the pound (#) key.

Enter the PIN that you use to access your voicemail. If you have forgotten the PIN, log in to the BE4000 Selfcare Portal and reset the PIN. Refer End User Guides for detailed procedure to reset the PIN.

You hear the prompts as follows:

Welcome to administration via telephone system.

- *To administer the Automated Attendant Alternate Greeting, press 1*

- *To administer custom prompts, press 2*

Step 4   Press **2**.

You hear the following prompts:

Your alternate greeting is currently active.

- *Press 1 to hear the alternate greeting*

- *Press 2 to rerecord the alternate greeting*

- *Press 3 to deactivate this alternate greeting. Deactivating the alternate greeting will delete it from the system*

Step 5   Press **3**.

You hear the following:

This operate is not reversable. Are you sure you wish to continue?

- *Press 1 for yes*

- *Press 2 for no*

Step 6   Press the **1** to deactivate the alternate greeting.

You hear the following:

The alternate greeting has been deleted.

Change Business Hours

A Cisco partner configures business hours for a site during the initial site deployment in the "Setup Assistant". After the site is successfully deployed, a Cisco partner or a customer administrator can change the business hours.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Managed Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Business Hours**.

Step 4   Click one of the following options:

- **24/7 (No Closed Hours)**—Business is functional on 24 hours on all days.

- **Dual Hours (Open and Closed)**—Set open and closed hours for each day based on the organization requirements.

Note	• Auto Attendant menu is played based on the Open and Closed hours
	• You must enter time in 24-hour format only (17:00 for example). Time must be either full (:00) or half hours (:30).

Step 5 (Optional) Click **Add New Holiday** to add the holidays for the organization.

Note	• You can add holidays only for the current year and a year ahead.
	• Date should not be less than the present date.

Step 6 Click the arrow at the bottom-right corner of the screen and click **Yes** to apply the changes made.

Phone Branding

BE4000 allows a Cisco partner or customer administrator to apply customer branding to phones with customized backgrounds and ringtones.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Managed Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Phone Branding**.

Step 4 In the **Customize Phones** page, click **Upload image file** in the **Phone Background** section. Click **Save**.

Step 5 Click **Add Background** and **Upload image file** to upload the next background image. Click **Save**.

Step 6 In the **Customize Phones** page, click **Upload ringtone file** in the **Phone Ringtones** section. Click **Save**.

Step 7 Click **Add Ringtone** and **Upload ringtone file** to upload the next ringtone. Click **Save**.

Step 8 Click **Save**.

The background or ringtone file upload details appear on the **Progress** page.

Step 9 Click **Close**.

SIP Trunk

SIP Trunk can be configured on the BE4000 during the initial deployment. However, in case, if the SIP Trunk is not added during the initial deployment, it can be added post site deployment.

If SIP trunk is added during initial deployment, you can remove it from BE4000 post site deployment.

Note	• You cannot modify the SIP Trunk configurations. You can only add or remove the SIP Trunks.
	• You can add only one SIP Trunk per site.

Add SIP Trunk

Before you begin

Ensure that you have:

- The DID numbers provided by your SIP service provider.

- The inbound call mapping and outbound caller ID display preference ready.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site > SIP Trunk** for the desired site from the Actions menu.

Step 3 Click **Add SIP Trunk**.

Step 4 Add DID Numbers on the **Direct Dial Numbers** page.

Step 5 Add SIP Trunk details on the **SIP Trunk** page.

> **Note** Choose your provider from the "Provider Template" or use "Custom". To have a new provider added as template, or to request assistance, contact the dedicated SIP Support Team. Do not open a TAC case as they will not be able to assist you.

Step 6 Choose a default target for each service provider from the **Default Target** drop-down list on the **Inbound Call Mapping** page. Based on your preference, choose a target type (auto attendant, extension, or hunt group) for each extension from **Target Type** drop-down and choose a corresponding target number from **Target Number** drop-down list.

Step 7 Choose a default outbound DID for each service provider from the **Default Outbound DID** drop-down list on the **Outbound Caller ID** page. Based on your preference, choose a DID number as Caller ID for each extension from the **Caller ID** drop-down list.

Step 8 Click **Yes** to save the changes.

Remove SIP Trunk

Removing the SIP Trunks from the BE4000 removes all the configurations that are related to SIP trunk.

Before you begin

Ensure that you have a SIP Trunk already configured.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site > SIP Trunk** for the desired site from the Actions menu.

Step 3 Click **Delete SIP** from the Actions menu.

You get an option to save or purge the DID numbers that are associated with the SIP Trunk. If you choose to save, the DID numbers are stored in the system and provides an option to load the same while adding a new SIP Trunk in future. If you choose to purge, all the DID numbers get deleted from the system.

Step 4 Click **Save** or **Purge**, based on your preference for saving DID numbers.

Manage Trusted IP Address List

Before you begin

Have the list of IP addresses to be added, modified, or deleted.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from the "Actions" column for the desired site.

Step 3 Click **SIP Trunk**.

Step 4 Click **Trusted List** at the top of the page. "Modify Trusted List" page appears.

Step 5 Click **Add Row** and enter the IP address in the newly added row.

Step 6 (Optional) You can modify or delete the IP addresses, if necessary.

Step 7 Click **Save**.

Supported Codecs

BE4000 supports the following codecs:

- g711ulaw

- g711alaw

- g729r8

- ilbc

- g722-64

Priority Order of Codecs

g711ulaw and g711alaw are enabled by default for sites using custom SIP service provider template during the initial deployment. Out of the two, g711ulaw is given the highest priority. You can choose to enable other codecs and also change the priority based on your SIP service provider needs. If you are using the predefined SIP service provider template, then based on the SIP service provider preference the codecs are enabled by default.

The order of codecs in the list affects the order of the codecs presented in the Session Description Protocol (SDP) for outbound calls on the SIP trunk. The choice of codec used is defined by the recipient of the codec list, and may or may not be affected by the order of the codecs. The order of the codecs is mainly for use when a service provider mandates that codecs are listed in a specific order.

A Cisco Partner can enable the codecs and set the priority during initial deployment in the Setup Assistant wizard. Post site deployment, both Cisco Partner and Customer Administrator can modify the list of codecs that are enabled and also change priority order.

Modify Codec List and Set Priority

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site > SIP Trunk** for the desired site from the Actions menu.

Step 3   Click **Modify Codec** from Actions menu corresponding to the SIP trunk row.

Step 4   Check the check boxes corresponding to the codecs that you want to enable.

Step 5   Drag and drop the codec names row to set the priority order.

Step 6   Click the arrow at the bottom-right corner of the screen and click **Yes** to apply the changes made.

Troubleshooting BE4000 Issues

BE4000 allows you to run show and debug commands from the BE4000 portal. You can download a local copy of show command output and get the debug command output sent to your preferred email address.

The following are the categories available for running show commands:

- SIP

- DID Translation

- Phone

- TDM

- Hunt Group

- Dial Plan

The following are the categories available for running debug commands:

- SIP Signaling

- Shared Line

- Voip

- Vpm Signal

- Voice Register Errors

- Call Routing

- Presence

- Isdn Q31

- Voice Register Events

- Voice Register Session Servers

Run Show Commands

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from the "Actions" menu for the desired site.

Step 3   Click **Troubleshooting**.

Step 4   Click **Run** corresponding to the "Show Command" section.

Step 5   Choose a method based on your preference from the **Choose Method** drop-down list. The following are the methods available:

- SIP
- DID Translation
- Phone
- TDM
- Hunt Group
- Dial Plan

Step 6   (Optional) Hover the mouse pointer over the text **View the chosen show commands**. Based on the chosen method, you can see the list of show commands that are selected for execution.

Step 7   Click **Run**. The output is generated and downloaded on to your system. Close the pop-up window after the output is successfully downloaded.

Run Debug Commands

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from the "Actions" menu for the desired site.

Step 3   Click **Troubleshooting**.

Step 4   Click **Run** corresponding to the "Run Debug" Command" section.

Step 5   Choose a method based on your preference from the **Choose Method** drop-down list. The following are the methods available:

- SIP Signaling
- Shared Line
- Voip
- Vpm Signal
- Voice Register Errors
- Call Routing
- Presence

- Isdn Q31

- Voice Register Events

- Voice Register Session Servers

Step 6   Enter the duration for executing debug commands in the **Enter Duration 5-30 Minutes** field. You can run debug commands for a minimum of 5 minutes and a maximum of 30 minutes.

Step 7   Enter the email address for receiving the debug command output in the **Enter Email to receive report (separated by commas)** field.

Step 8   Click **Run**. An email containing the debug command output is sent to the entered email address.

Night Service

Night service allows you to transfer the incoming calls to a designated set of extensions during closed hours. During the night service hours (also known as closed hours), calls coming in to the designated extension, known as night service extensions, sends a special "burst" ring to night-service phones (phones that receive the calls coming from the night service extension) that have been specified to receive the special ring. Phone users at the night-service phones can then answer the incoming calls for the night-service extensions.

Note   You can configure only one night-service phone per night-service extension.

Example

Nancy is a receptionist with extension 1234 at ABC organization having business hours from 0900 to 1700. When night service is configured, the calls received on extension 1234 between 17:00 and 08:59 is transferred to extension 5678 that is designated to receive calls during closed hours.

Night Service Manual Activation Code

Manual Activation Code allows a user to manually activate the night service mode on the phone.

Manual Activation Code must start with an asterisk (*) followed by a minimum of 3 digits. A maximum of 15 digits can be configured excluding the asterisk (*). Default value for manual activation code is *1234.

Example

ABC organization has set the night service activation code as *1234. Business hours are from Monday to Friday 09:00 to 17:00. If Nancy at the reception wants to log out for a couple of hours during the business hours, she can dial *1234 to trigger the night service mode on the phone. The incoming calls to her extension are transferred to the designated night service phone when she is away. When she is back, she can dial *1234 to log out of the night service mode and enter the business hours mode to receive the incoming calls.

Configure Night Service

Table 1: Configure Night Service—Task Flow

Step	Task	Purpose	Prerequisite	References
Step 1	Activate Night Service	Activate and enter the active hours for Night Service.	—	Activate Night Service, on page 21
Step 2	Designate an Extension for Night Service Hours	Configure the extension that you want to designate as the Night Service phone.	—	Designate an Extension for Night Service Hours, on page 21
Step 3	Designate Extension to Receive Calls During Night Service Hours	Configure the extension that you want to designate to receive Night Service calls.	—	Designate an Extension to Receive Night Service Calls, on page 22

Activate Night Service

Step 1   Click **Actions** corresponding to your site from the Dashboard.

Step 2   Click **Night Service** under **Manage Site**.

Step 3   Turn on the slider to enable Night Service.

Step 4   (Optional) Enter the **Manual Activation Code**.

> **Note**   Manual Activation code is used to manually trigger night service mode during business hours.
>
> Manual Activation Code must start with an asterisk (*) followed by a minimum of 3 digits. A maximum of 15 digits can be configured excluding the asterisk (*). Default value for manual activation code is *1234.

Step 5   Enter the active hours for Night Service.

You cannot overlap the end time of Night Service and the opening time of the business hours. For example, if your business closes at 17:00 and opens next day at 09:00, enter the Night Service hours as 17:00 and 08:59. You cannot enter 09:00 as it overlaps with the open business hours. Night Service hours must be entered in 24-hour time format.

Step 6   Add the business holidays when you want the Night Service to be active.

Step 7   Click **Yes** to save the changes.

Designate an Extension for Night Service Hours

Step 1   Click **Actions** corresponding to your site from the Dashboard.

Step 2   Click **Extensions** under **Manage Site**.

Step 3   Click **Modify Extension** from **Actions** column for the extension that you want to designate as Night Service phone.

Step 4   Enter the number to which calls must be forwarded during night service hours in the **Night Service** field on **Call Forward** page.

Step 5   Check the **Night Service Bell** check box on the **Additional Settings** page.

> **Note**   You must configure Night Service Bell for all the Night Service phones designated to receive Night Service calls. The Night Service bell ensures that all the phones designated for Night Service receive the burst ring. For more information, refer to .

Step 6   Click **Yes** to save the changes.

Designate an Extension to Receive Night Service Calls

Step 1   Click **Actions** corresponding to your site from the Dashboard.

Step 2   Click **Extensions** under **Manage Site**.

Step 3   Click **Modify Extension** from **Actions** column for the extension that you want to designate to receive Night Service calls.

Step 4   Check the **Night Service Bell** check box on the **Additional Settings** page.

Step 5   Click **Yes** to save the changes.

Specify Maintenance Schedule

Maintenance Schedule allows you to designate a two hour block of time each day when it is safe for the system to install software updates. The system may be offline and unable to make or receive phone calls during the maintenance schedule.

Before you begin

• Customer site is successfully deployed and available Online.

Step 1   Click **Manage Site** from "Actions "menu for the desired site on the dashboard.

Step 2   Click **Settings > Maintenance Schedule**.

Step 3   Enter the Maintenance Schedule **Beginning** time for each day of the week.

> **Note**   The two hour duration of maintenance schedule is auto adjusted in the **Ending** field based on the value you entered in the **Beginning** field.

Step 4   Click **Save**.

Modify SMTP Server Details

Before you begin

- For Office 365, have the MX FQDN Record ready.

- For SMTP Relay other than Office 365, have the SMTP server IP address or domain name ready.

- If you are configuring a custom SMTP server certificate, ensure that you have the certificate in *.crt* file format.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Settings > System Settings.**

Step 4   Enter the SMTP server details under "SMTP Server – Enables Voicemail to Email Functionality" section.

     a)   In the "IP Address or Domain Name" field, enter one of the following:

        Note   You can configure Fully Qualified Domain Name (FQDN) of public SMTP servers only. BE4000 handles all Domain Name System (DNS) resolutions through the internet and thus only public FQDNs are accepted. For example, smtp.office365.com, smtp.gmail.com.

- If the customer has Office 365, enter the MX FQDN record that is copied from Office 365 portal.

- If the customer has SMTP relay other than Office 365, enter the SMTP server IP Address or Domain Name.

     b)   In the "Port" field, enter one of the following:

- If the "Security Mode" is "None, enter **25**

- If the "Security Mode" is "SSL", enter **465**

- If the "Security Mode" is "STARTTLS", enter **587**

     c)   From the "Security Mode" drop-down list, enter one of the following:

- If the customer has Office 365, choose **None**. "SSL" and "STARTTLS" are not supported for Office 365.

- If the customer has SMTP relay that is trusted by the BE4000, choose **SSL** or **STARTTLS** as per customer's network.

     d)   In the "Sender's Email Address", enter the email address that is used as "From" address while sending the emails containing the voicemail attachments.

     e)   Click **Choose File** in the "Upload SMTP Server Certificate" section to upload the certificate.

        Note   - The "Upload SMTP Server Certificate" section is visible only when you choose "SSL" or "STARTTLS" from the **Security** drop-down list.

- BE4000 accepts SMTP server certificates only in *.crt* file format.

     f)   (Optional) Check the **Authenticate** check box and enter the username and password.

The authenticate username and password are the credentials required by the voicemail application to authenticate itself with the SMTP relay server before sending the voicemail file.

g) (Optional) Click **Validate SMTP**. Enter the email address to receive the test email containing the voicemail. Click **Save & Send**.

Note • Before validating the SMTP server address, the portal saves the changes made to the SMTP server settings and the previously configured settings cannot be restored.

• The portal only checks if the SMTP server address is valid and does not check the validity of the email address entered. You see success message displayed on the portal based on the voicemail application's ability to authenticate and send email using the SMTP Relay. If the email address entered is invalid, you may not receive the email. Hence, you may see the success message on the portal even when you have not received the email.

Step 5 Click **Save**.

Modify the Digit to Dial an Outside Line

Before you begin

Ensure that you have a digit freely available and is not assigned as a prefix for making intercom calls and sending calls to voicemail automatically. Also, the digit is not the starting digit of Hunt Group pilot number, Auto Attendant Pilot Number, and Paging Group Pilot Number.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** corresponding to the desired site.

Step 3 Click **Settings > System Settings**.

Step 4 Choose a digit from the **Dial an Outside Line** drop-down list.

Step 5 Click **Save**.

Modify the Digit to Send Calls to Voicemail Automatically

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Settings > System Settings**.

Step 4 Enter a digit between 1 to 6 in the **Send to voicemail automatically** field.

Step 5 Click **Save**.

Modify Time Zone

After the initial site deployment, you can modify the time zone to which your BE4000 belongs.

Before you begin

Ensure that:

- BE4000 site is in the Online state.

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Settings > System Settings**.

Step 4 Choose the time zone from the **Time Zone** drop-down list under Region Settings.

Changing the time zone reboots the Auto Attendant and Voicemail services and therefore will be unavailable for up to 3 minutes.

Step 5 Click **Save**.

Smart License

BE4000 mandates its customers to update their smart license token in the cloud portal. You can get the smart license token from the Cisco Smart Software License (CSSM) portal. For more information on getting the smart license token from CSSM, see Smart Licensing Guide.

Adding smart license token is optional during the initial site deployment (in the Setup Assistant). If you don't have smart license token during the initial deployment, BE4000 enters the evaluation mode. You can configure the BE4000 in the evaluation mode for 90 days. Before the evaluation period is completed, enter the smart license token in the BE4000 cloud portal.

Cisco Partners and Customer Administrators can view the status of the smart license account and also update the smart license token under **Manage Site > Settings > Licensing**. The **Smart Account** field displays if your BE4000 site is in Registered or Unregistered state. Also, you can view the corresponding **License Status** for the **Smart Account** status. Click **Refresh Status** to get the latest license status of your site. For more information, see Table 2: Smart License Account Status, on page 25.

The following table shows the various registration statuses and its associated description:

Table 2: Smart License Account Status

Smart Account Status	License Status	Description
Registered	Authorized	You have valid license.
Registered	Out of Compliance	You're using more number of licenses than purchased.

Smart Account Status	License Status	Description
Registered	Authorization Expired	After you are registered and authorized, BE4000 automatically renews the authorization for every 30 days. If the BE4000 is unable to reach the CSSM for 90 days, then you see "Authorization Expired" state.
Unregistered	Evaluation Mode	You don't have smart license token or you have entered an invalid smart license token.
Unregistered	Evaluation Expired	Your evaluation period of 90 days is completed.

Add Smart License Token

You can associate the site with the Cisco Smart Account by providing the smart license token. The **Smart License Token** field is a placeholder for entering the smart license token ID. Currently, the BE4000 does not register to the Cisco Smart Account. If you enter a smart license token ID, we recommend setting a validity date of at least 180 days when the token is created.

Before you begin

Ensure that you have noted down the smart license token provided by Cisco.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Settings > Licensing**.

Step 4   Enter the smart license token ID received from Cisco in the **Smart License Token** field.

Step 5   Click **Save**.

Configure Back Up Schedule for a Site

BE4000 allows you to back up your site configurations on a regular interval. You can set the day and time during which the system automatically backs up your site configuration. The previous five consecutive backups are stored in the BE4000 portal.

Note

- You cannot schedule more than one backup per day.

- You must configure at least one day in a week to back up your site configuration. If you have not scheduled a backup for your site, by default, the site is backed up every Saturday during the maintenance schedule.

Before you begin

Ensure that BE4000 site is available in the Online state

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from Actions menu for the desired site on the dashboard.
Step 3	Click **Settings > Maintenance Schedule**.
Step 4	Check the **Back Up** check box for the days when you want the site to back up the configurations.
Step 5	Click **Save**.

Note   If the scheduled back-up fails for a site in "Online" state, an email is sent to the customer administrators to notify the failure.

What to do next

Restore the backed-up site configuration. Refer to, for more details.

Restore the Backed-Up Site Configuration

BE4000 supports restoring the backed-up site configuration. You can restore any one of the previous five consecutive backed-up site configurations.

Before you begin

• Ensure that the BE4000 site is available in the Online state

• A minimum of one backed-up configuration is available to restore

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from Actions menu for the desired site on the dashboard.
Step 3	Click **Settings > System Restore**.
Step 4	Choose a backed-up configuration from the **Please select a backup** drop-down list.

The previous five consecutive back-ups are available to restore.

Step 5	Choose the configurations that you want to restore by clicking one from the following:

• **All Data**—Restores the complete site configuration that includes, phone, extension, dial plan, voicemail, and auto attendant data.

• **Phone, Extensions, Dial Plan**—Restores the phone, extensions, and dial plan data.

• **Voicemail & Auto Attendant**—Restores the voicemail and auto attendant data.

Note   We recommend you to click **All Data** and restore the complete site configuration.

Step 6   Click **Restore**.

Note   • The phone system goes offline and is made unavailable for use until the restoration is complete.

• The backup of the current settings is made first before the restoration begins.

Direct Inward Dial Numbers

Direct Inward Dial (DID) numbers are the registered numbers that you get from your service provider. A Cisco partner can add the DID numbers during initial site deployment in the Setup Assistant. A customer administrator and Cisco partners can add, edit, or delete the DID numbers post-site deployment. DID number should be entered in the national number format. You can add Direct Inward Dial (DID) numbers for your SIP, BRI, and PRI connections.

Note   Do not add FXO line numbers while adding DID numbers as FXO line numbers can be added when line cards are added.

Inbound Call Mapping

BE4000 allows you to map the incoming calls of a DID number to one of the following:

• **Auto Attendant**—You can map the incoming calls on the DID number to the Auto Attendant. The Auto Attendant menu options is played when the callers dial the DID number. You choose one of the following as the target number:

• **Auto Attendant Pilot Number**—Choose pilot number of the Auto Attendant.

• **Greeting Admin**—You can assign a Direct Inward Dial (DID) number as the number to log in the Auto Attendant Alternate admin settings. The customer administrator (or an end user with "PromptAdministrators" privileges), can dial the DID number instead of 70397 to reach the Auto Attendant admin settings on the phone.

• **Extension**—You can select any one of the existing extensions. The incoming calls on the DID number ring on the specified extension.

• **Hunt Group**—You can map the incoming calls on the DID number to an existing hunt group. The incoming calls on the DID numbers ring on the extensions belonging to the specified hunt group.

• **Meet Me**—You can map the incoming calls on the DID number to a Meet Me conference number. The incoming calls on the DID numbers ring on the Meet Me conference numbers that you configure.

Note   If you disable hardware conference, BE4000 reverts to the default option of Built-In Bridge conference. Also, the mapping of incoming calls on the DID number to a Meet Me conference number is reset. Hence, the incoming calls on the DID number is directed to the default target. The default **Select** option is displayed against the DID number in the BE4000 portal.

You can also set a default target for all the DID numbers belonging to a service provider. The default target can be auto attendant, extension, or hunt group. If there are any DID numbers that are not mapped to auto attendant, extension, or hunt group, the call is directed to the default target set for the service provider.

Outbound Caller ID

BE4000 allows you to configure a specific DID number to be displayed on the called phone when an outbound call is placed from an extension within the organization.

You can also set a default outbound DID number for a service provider. In such a case, an extension without a specified DID number displays the default outbound DID number configured for the service provider.

The format of the number in **Outbound Caller ID** need not be same as the Direct Inward Dial numbers (DID) in **Inbound Call Mapping**. Based on requirements, you can configure the incoming DID and outbound number to different formats. For example, consider a scenario where area code is prefixed to your inbound DID number (1168117777). However, your preferred outbound call format does not include the area code. Then, BE4000 portal allows you to edit the format of the outbound number to 68117777. Check the check box **Outbound Caller IDs do not match Direct Inward Dial numbers** under **Manage Site>Direct Inward Dial Numbers>Outbound Caller ID**.

Custom Outbound Call ID

You can define the name that appears as caller ID for outbound calls made through the SIP trunk. BE4000 portal allows you to enter the outbound caller ID name in the field **Custom Outbound Caller ID**. The field is available under **Manage Site>Direct Inward Dial Numbers>Outbound Caller ID**. **Custom Outbound Caller ID** is an optional field. If you leave the field blank, the default caller ID is used.

Example

ABC organization has 12345 and 67890 as DID numbers from service provider XYZ. The default outbound DID number set for XYZ service provider is 12345. ABC organization has two extensions 4501 and 4502. Extension 4501 is mapped to 67890 as Outbound DID. Extension 4502 is mapped to default outbound DID.

When an outbound call is placed from extension 4501, the called phone displays the incoming call number as 67890 without revealing the extension number within the organization ABC.

When a call is placed from extension 4502, the called phone displays the incoming call number as 12345 which is the default outbound DID number set for the service provider XYZ.

Manage DID Numbers

Before you begin

- Ensure that you have the new DID numbers to be added, or the changes to be made to the existing DID numbers ready

- Ensure that you have the inbound call mapping and outbound caller ID display preference ready

Step 1   Click **Actions** corresponding to your site from the Dashboard.

Step 2   Click **Direct Inward Dial Numbers** under **Manage Site**.

Step 3   Add, modify, or delete the DID numbers on the **Manage Numbers** page.

The DID numbers set on the **Direct Dial Number** page under **Call Routing** in the Setup Assistant are auto-populated on this page.

Step 4   Choose a default target for each service provider from the **Default Target** drop-down list on the **Inbound Call Mapping** page. Based on your preference, choose a target type (auto attendant, extension, or hunt group) for each registered number from **Target Type** drop-down and choose a corresponding target number from **Target Number** drop-down list.

Step 5   Choose a default outbound DID for each service provider from the **Default Outbound DID** drop-down list on the **Outbound Caller ID** page. Based on your preference, choose a DID number as Caller ID for each extension from the **Caller ID** drop-down list.

Step 6   Click **Yes** to save the changes.

Inbound Call Blocking

Inbound call blocking helps you in blocking calls from telemarketers. A Cisco Partner or Customer Administrator can add a list of numbers from which the calls are blocked. When an incoming call is received from such numbers, the call gets disconnected immediately.

Note   You can add numbers to inbound call blocking list only after the initial deployment.

Add Numbers to Inbound Call Blocking List

Before you begin

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from "Actions" menu on the dashboard.

Step 3   Click **Inbound Call Blocking**.

Step 4   Click **Add Row**.

Step 5   Enter the number.

Note
- You can add a minimum of 3 digits and a maximum of 32 digits.

- You can add a maximum of xx numbers.

- Do not include international or national access digits when adding numbers. Following are a couple of examples:
  - US Domestic—5105551234 (no leading 1)

  - US International—441234567890 (no leading 011)

  - UK Domestic—1234567890 (no leading 0)

  - UK International—15105551234 (no leading 00)

Reach System Operator from a Personal Voicemail Box

Every call is important for the organization and any unattended call can be business critical. BE4000 provides the ability where none of the calls are unattended, even when callers reach a user's personal voicemail box. You can allow a caller to dial zero (0) after reaching a user's personal voicemail box and reach the system operator.

Example—John calls his client Edward, working in the ABC organization for a query. Edward is out of office and hence John hears Edward's personal voicemail box greeting, such as "Edward is out of office. Please dial zero to reach the system operator". John dials 0 and the system operator, Lisa's phone rings. Lisa answers John's call and responds to his query.

A Cisco partner can configure the system operator number during the initial site deployment. If the system operator number is not configured during the initial site deployment, you can configure it post site deployment. You can configure one of the following as system operator number:

- Route to Extension—Any extension that is configured for the site.

- Route to Voicemail Box—Voicemail of any extension.

- Route to Hunt Group—Any hunt group that is configured for the site.

- Route to Group Voicemail Box—Any group voicemail box that is configured for the site.

- Route to Pilot Number—Any pilot number, such as Auto Attendant pilot number.

Configuring system operator for the site, enables all the personal voicemail boxes to allow its callers to dial zero for reaching the system operator.

Configure System Operator Number

BE4000 allows callers to dial zero after listening to a personal voicemail box greeting and reach the system operator.

Before you begin

Ensure that you have the details of the number to be configured for the system operator.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** for the desired site from the Actions menu.

Step 3   Click **Setting > System Settings**.

Step 4   Choose the one of the following as the system operator target type from the **Target Type** drop-down list:

• Route to Extension—Any extension that is configured for the site.

• Route to Voicemail Box—Voicemail of any extension.

• Route to Hunt Group—Any hunt group that is configured for the site.

• Route to Group Voicemail Box—Any group voicemail box that is configured for the site.

• Route to Pilot Number—Any pilot number, such as Auto Attendant pilot number.

Step 5   Choose the number corresponding to the target type selected from the **Target Number** drop-down list.

Note   The **Allow caller to dial 0** check box on the **Voicemail Box** page is checked by default for all the extensions that have voicemail enabled (or have personal voicemail box configured). Hence, all the personal voicemail boxes allow its callers to dial zero for reaching the system operator. However, ensure that the **Allow caller to dial 0** check box is not disabled for the personal voicemail boxes.

What to do next

Cisco Partner or Customer Administrator communicates to all the users who have voicemail enabled (or have personal voicemail box configured) to record their personal greeting to include instructions on dialing zero to reach system operator. For example, the user, Edward records his personal greeting as "Edward is out of office. Please dial zero to reach the system operator".

Conference

Conference feature allows end users to involve three or more parties in a call. BE4000 supports both Built-in Bridge and Hardware Conference.

Built-In Bridge

Built-in Bridge is enabled by default in the BE4000 and allows the end users to host conference calls involving a maximum of three parties. For conference calls using the built-in bridge, the DSP resources that are available in the phone hardware is used.

End users can start a conference by pressing the **Transfer** button on the phone.

Hardware Conference

Hardware conference uses the DSP resources that are available in the BE4000 appliance. Hardware conference can be either Ad-hoc or Meet-me. Cisco Partners or Customer Administrators can enable hardware conference post site deployment under **Manage Site > Conferencing**.

Note   When hardware conference is enabled, Built-in bridge gets disabled automatically.

Ad-Hoc Conference

Ad-Hoc conference allows BE4000 to host conference calls involving a maximum of 8 parties. End users can press the **Transfer** button on phone to start an ad-hoc conference call.

Cisco Partners and Customer Administrators can enable Ad-hoc conferencing.

Meet-me Conference

Meet-Me conference requires the end users to dial a predetermined Meet-me conference number. Cisco Partners or Customer Administrators can configure the meet-me conference numbers. Meet-me conference can involve a maximum of 8 parties.

Meet-Me Conference Dependency on Codecs

The number of Meet-me conference bridges that can be configured depends on the codecs that are configured for the site. The following are the number of Meet-me conference bridges that are allowed based on the codecs:

Codec Enabled	Maximum Number of Simultaneous Meet-me Conferences Allowed
g711ulaw	5
g711alaw	5
g729r8	3
g722-64	3
ilbc	—

Enable Hardware Conference

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site > Conferencing**.

Step 3   Turn on the slider to enable hardware conferencing.

Note   The slider is unavailable if "ilbc" is selected as the SIP codec.

Step 4   Enter the Meet-me conference numbers in the "Extensions" field. The number of Meet-me conferences that can be enabled depends on the codecs selected. For more information, refer .

Note	• Meet-me conference meeting number must have same number of digits as that of an extension.
	• The first digit of the Meet-me conference number cannot be the same as the digit used for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.

Step 5 Click the arrow at the bottom of the screen.

Step 6 Click **Yes** to save the changes.

Hunt Groups

Hunt Groups allow incoming calls to a specific number (pilot number) to be directed to a defined group of extension numbers. Incoming calls are redirected from the pilot number to the first extension number as defined in the configuration. If the first number is busy or does not answer, the call is redirected to the next phone in the list. A call remains redirected on busy or no answer from number to number in the list until it is answered or until the call reaches the number that is defined as the final number.

A Hunt Group can have static and dynamic members.

- **Static Members**—Permanent members belonging to the Hunt Group.

- **Dynamic Members**—Not the permanent members, but they can join or unjoin a Hunt Group on a need basis using the softkeys available on the phone.

Note	• You can add a maximum of 20 Hunt Groups.
	• The total number of members in a Hunt Group, including static and dynamic members cannot exceed 32.
	• If you check the "Allow dynamic members" check box on the **Hunt Groups** page, ensure that you check the "Hunt Group Login" check box for each dynamic member extension under **Manage Site > Extensions > Basic Info** page.
	• Once a Hunt Group is created, you cannot modify the Hunt Method. You have to delete and readd the Hunt Group to modify the Hunt Method.

A Cisco Partner can configure the hunt groups using the Setup Assistant in Cisco Business Edition 4000 Management Portal during the initial site deployment. If the Cisco Partner has already created hunt groups during the initial deployment, a customer administrator can see them on Hunt Groups screen. A customer administrator can also create a new hunt group and also modify the existing hunt groups.

If a phone belongs to multiple Hunt Groups, and an incoming call is received from one of the Hunt Groups, the name of the Hunt Group is displayed on the phone display.

Call Queuing

BE4000 supports call queuing in Hunt Groups. When all agents in the hunt group are busy, incoming calls wait in a queue until one of the agents become free. Thus, the calls are not disconnected but made to wait. A

pre-recorded greeting is played while the callers wait in the queue. This is called "Comfort Greeting". You can set the frequency in which the comfort greeting is repeated to the incoming callers. Frequency ranges from 30 to 120 seconds and the default is 60 seconds. Thus, call queue enhances the experience of the incoming callers.

Check the **Enable Call Queuing** check box on the "Hunt Groups" page to enable call queueing. Specify the desired values in the **Comfort Greeting frequency** and **Max Waiting Time** fields.

BE4000 plays the following default comfort greeting:

All agents are currently busy assisting other customers. Continue to hold for assistance. Someone will be with you shortly.

Note You cannot upload custom comfort greetings.

Add a New Hunt Group

Step 1 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 2 Click **Hunt Groups**.

Step 3 Click **Add Hunt Groups**.

Step 4 Enter the information in the fields. For field descriptions, refer Hunt Groups - Field Descriptions, on page 35.

Step 5 Click the arrows at the bottom of the screen to move to next screen or to go back to the previous screen.

Step 6 Click **Yes**.

Hunt Groups - Field Descriptions

Suggestions for Setting Up Hunt Group Timers

The Hunt Group page does not load any default values. You must set the values for the following fields based on the selected Hunt Method.

- Member Timeout

- Comfort Greeting Frequency

- Max Waiting Time

For effective inbound call handling and providing a positive experience for inbound callers, we recommend you to set the timers as per the following calculations:

Example for Parallel Hunt Method

Set the Comfort Greeting Frequency as two times the value of Member Timeout and the Max Waiting Time as the value equal to that of Member Timeout.

Member Timeout = 30 seconds

Comfort Greeting Frequency = 2 x Member Timeout = 2 x 30 = 60 seconds

Max Waiting Time = Member Timeout = 30 seconds

Example for Peer, Sequential, and Longest Idle Hunt Methods

Set the Comfort Greeting Frequency as two times the value of Member Timeout and the Max Waiting Time as Member Timeout value multiplied by the number of members in the Hunt Group.

Members in the Hunt Group = 5

Member Timeout = 30 seconds

Comfort Greeting Frequency = 2 x Member Timeout = 2 x 30 = 60 seconds

Max Waiting Time = Number of Members x Member Timeout = 5 x 30 = 150 seconds

Field	Description
Group Name	Enter a unique name for the hunt list. To easily identify the hunt list, consider appending the pilot extension to the name; for example, hl5001. Group name must contain a minimum of 2 and a maximum of 12 characters. **Note** Only the following characters are allowed: a-z, A-Z, 0-9, space, hyphen(-), and underscore (_)

Pilot Number	Enter a number to access the Hunt Group that serves as the pilot for the hunt list. This number serves as the trigger for hunting to begin. Pilot number can be any number equal to the length of extension. For example:
	
	   • If the extension length is 3, pilot number can be any number from 000 to 999
	
	   • If the extension length is 4, pilot number can be any number from 0000 to 9999
	
	   • If the extension length is 5, pilot number can be any number from 00000 to 99999
	
	Note • The Pilot Number of the Hunt Groups cannot be the same as any existing extension and cannot start with the digit that is used for sending calls to voicemail automatically and for placing intercom calls.
	
	            • The Pilot Number of the Hunt Groups cannot be the same as any existing Pickup Group number.

Add Members	Click **Add** to add members to this hunt group from the Stations page (Show Member Directory). You can also search for the users by entering member name or extension. All extensions that are assigned to users or departments can be included as members of a Hunt Group. You must add a minimum of two members (can include dynamic members) for a hunt group.

Field	Description
Show Member Directory	From the list of extensions that display, select which extensions must be included in the hunt list. Click **Show Member Directory**, to select the list of extensions for the hunt list. Check the respective member's name and click **OK**.
Allow dynamic members	Allows members that are not part of the Hunt Group to join and unjoin the Hunt Group on a need basis using the softkeys displayed on the screen.
Max dynamic members	**Note** This field is visible only when "Allow dynamic members" is checked. Number of dynamic members allowed for the hunt group. A minimum of one dynamic member needs to be added. **Note** • You can add a maximum of 20 Hunt Groups. • The total number of members in a Hunt Group, including static and dynamic members cannot exceed 32. • If you check the "Allow dynamic members" check box on the **Hunt Groups** page, ensure that you check the "Hunt Group Login" check box for each dynamic member extension under **Manage Site > Extensions > Basic Info** page. • Once a Hunt Group is created, you cannot modify the Hunt Method. You have to delete and readd the Hunt Group to modify the Hunt Method.

Field	Description
Hunt Method	Select how BE4000 distributes the calls to members of the hunt list based on one of the following hunt methods: • Longest-idle—BE4000 only distributes a call to idle members, starting from the longest idle member to the least idle member of a hunt list. • Parallel—Calls ring all numbers in that hunt group simultaneously. The extension to first answer the call is connected. • Sequential—Call hunting always starts with the first member in the hunt group. Continues to each number in the group in the order in which they are listed, from top to bottom, in the hunt group. • Peer—Call hunting starts with the extension immediately after the one that last rang. Ringing proceeds in a circular manner, that is from left to right. That is, BE4000 distributes a call to idle or available members starting from the (n+1)th member of a hunt list, where the nth member is the member to which BE4000 most recently extended a call. If the nth member is the last member of a hunt list, BE4000 distributes a call starting from the top of the hunt list. **Note** Once a Hunt Group is created, you cannot modify the Hunt Method. You have to delete and readd the Hunt Group to modify the Hunt Method.

Field	Description
When No Member is Available	If no members of the hunt list are available to answer a call, you can choose to perform one of the following: • Disconnect Call—the call is disconnected. • Route to Group Voicemail Box—the call is forwarded to a group voicemail box. Enter the email address and extension associated with the group voicemail box. Group Voicemail boxes are created with the first member of the hunt group as the user owner of the group. **Note** • During initial deployment, a group voicemail box is created including the members who are available in the Hunt Group. • When adding Hunt Group post site deployment, ensure that you create a group voicemail box first. Else, the drop-down list does not show "Route to Group Voicemail Box" option. • Route to Hunt Groups—the call is forward to another Hunt Group. Select the Hunt Group from the drop-down list. • Route to Other—the call is forwarded to any other number, such as extensions. • Route to Voicemail Box—the call is forwarded to the voicemail box of a user.
Timer Settings	
Member Timeout (3-60000 Sec)	Enter the maximum time for a member's phone to wait before passing the call to the other member's phone. This happens only for longest-idle, sequential, and peer hunt methods. The range is from 3 to 60,000 seconds.
Enable Call Queuing	Enables call queueing.
Comfort Greeting frequency (30-120 Sec)	**Note** This field is visible only when **Enable Call Queuing** check box is checked. Set the frequency in which the pre-recorded voice message (comfort greeting) is played while the callers wait in the queue. The range is from 30 to 120 seconds.
Max Waiting Time (20-3600 Sec)	**Note** This field is visible only when **Enable Call Queuing** check box is checked. Enter the maximum time to wait when the queue is busy or full before routing the call to "When No Member is Available " destination. The range is from 20 to 3600 seconds.

Field	Description
Save	Save your hunt group configuration settings.

Add Users to Hunt Groups

Step 1   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 2   Click **Hunt Groups**.

Step 3   Click **Modify** from the Actions menu for the desired Hunt Group.

Step 4   Add the extension associated with the users in **Extension List** field. Use commas to separate extensions.

Step 5   Click **Yes**.

Intercom

An intercom call is a dedicated two-way audio path between two intercom lines. BE4000 supports intercom functionality for one-way and press-to-answer voice connections using a dedicated pair of intercom directory numbers on two phones.

BE4000 creates an intercom line for all the extensions by default. A digit is configured as intercom prefix by the Cisco partner while configuring the BE4000 site. You can place an intercom call by dialing the intercom prefix and the extension. When an intercom call is placed, one-way audio is established. Once the called number answers the intercom call by pressing the line key, the call becomes two-way audio.

When an intercom line is deleted, you can add it back using the **Type** drop-down list under **Manage Site > Phones > Modify Phone > Buttons** page.

Note   You can add only the primary extension as an intercom line.

For details on how to place an intercom call, see End-User Guides.

Paging

Paging provides one-way voice path to the phones that have been designated to receive paging. The paged phone automatically answers the page in speakerphone mode with "Mute" activated.

By default, the BE4000 system creates a "PageAll" group that consists of all phones available in the site.

For details on how to send a page, see End-User Guides.

Whisper Paging

When the paged phone is busy with a connected call, the user of the paged phone can hear both the active conversation and the paging message. This is called Whisper paging.

> **Note**   Whisper paging is enabled on all phones by default and cannot be disabled.

Paging Group

You can create paging groups with the desired set of phones and designate a specific number to relay audio messages. The number that is specified to relay audio messages is called as the Pilot Number.

When a caller dials the pilot number, each phone available in the paging group automatically answers the page in speaker mode. There is no need to press any button on the phone to answer the page.

The paged phone can receive the page when it is idle or busy on an active call. When the phone is busy on an active call, the user of the paged phone can hear both the conversation of the active call and the whisper page at a time.

By default, the BE4000 system creates the "PageAll" group consisting of all phones available in the site. To add members to the newly created paging group, you must first remove the members from "PageAll" group or any other paging group where the phone is already a member.

When a new paging group is created, by default, it is not part of the "PageAll" group. You can choose to add it to the "PageAll" group or to exclude.

> **Note**
> - A phone can be part of only one paging group, including the "PageAll" group.
>
> - You can add a maximum of 10 paging groups, including the "PageAll" group.
>
> - A paging group cannot contain other paging groups in it. Only "PageAll" group can contain other paging groups in it.

When paging an extension that is shared on two phones (shared line), each phone on the shared line is treated separately. A phone that can be paged by two paging groups receives the page from the first paging group and ignores the page from the second paging group. When the page from the first paging group is disconnected, the phone can receive the page from the second paging group.

Paging is not considered a call and hence, a paging phone that is in a connected state can press another line to make a call using the phone's softkeys.

A Cisco Partner can create paging groups during the initial site deployment. A Cisco Partner and the customer administrator can create or modify the paging group post site deployment.

> **Note**   Paging is not supported on Cisco Unified IP Conference Phone 8831 and Cisco ATA 190 Analog Telephone Adapter.

Add a Paging Group

Before you begin

Ensure that you have the list of extensions that you want in the paging group.

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site > Paging**.
Step 3	Click **Add New Paging Group**.
Step 4	Enter the information in the fields. For descriptions of the fields, see Paging—Field Descriptions, on page 42.
Step 5	Click **Save**.

Paging—Field Descriptions

Field	Description
Group Name	Name of the paging group. Enter a minimum of 2 and a maximum of 30 characters. **Note** Only 0-9 a-z A-Z !#%,-./=_? are allowed.
Pilot Number	Number that is designated to relay audio pages. Pilot Number must contain same number of digits as that of the extension length. **Note** The first digit cannot be the same as the digit used for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.
Add Members	List of members in the paging group. Search for the members using the username or extension. You can also search for the members using the "Show Directory" option.
Save	Creates a paging group or saves the changes that are made to an existing paging group.

Modify a Paging Group

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site > Paging**.
Step 3	Click **Edit** from "Action" column, corresponding to the paging group that you want to modify.
Step 4	Modify the details of the paging group as required. For description of the fields, see Paging—Field Descriptions, on page 42.
Step 5	Click **Save**.

Delete a Paging Group

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site > Paging**.

Step 3   Click **Delete** from "Action" column, corresponding to the paging group that you want to modify.

Step 4   Click **Yes**.

Privacy and Privacy Button

Note   Privacy and Privacy Button features must be enabled on shared lines only.

Privacy

Privacy feature blocks other users in the same shared line from viewing or handling any activities on the shared line. Privacy can be enabled on per phone basis. If one phone has multiple shared lines, Privacy is enabled for all the shared lines.

Privacy is disabled on all phones by default. A Cisco Partner or a Customer Administrator can enable Privacy on desired phones post-site deployment.

Privacy can be enabled by checking the **Privacy** check box on the "Additional Settings" page while adding and modifying phone under **Manage Site > Phones > Manage Phone**.

Privacy Button

Note   Privacy Button can be enabled only when Privacy is enabled.

Privacy Button feature provides an option to toggle the privacy mode on and off by the end user on the phone.

When Privacy Button is enabled, a line button on the phone gets labeled as "Privacy". Before enabling the Privacy button, ensure that at least one line button is available for assigning the privacy button. If line button is not available on the phone, then the end user does not get an option to toggle the privacy mode on and off.

A Cisco Partner or a Customer Administrator can enable Privacy Button by checking the **Privacy Button** check box on the "Additional Settings" page while adding and modifying phone under **Manage Site > Phones > Manage Phone**.

When a phone receives an incoming call on a shared line, the end user can make the call private by pressing the Privacy line button.

Once Privacy Button is enabled, the "Privacy" mode is on by default for all shared lines on the phone. To toggle the Privacy mode on and off, end users can press the Privacy line button.

If Privacy is enabled and Privacy Button is disabled, then the end user does not get an option to toggle the Privacy mode on and off.

Enterprise Line Key

> **Note** Enterprise Line Key (ELK) is supported only on the Cisco IP Phone 8800 Series for BE4000.

Enterprise Line Key allows you to use the buttons on both sides of the phone screen. For example, Enterprise Line key supports the use of all the ten buttons on a Cisco IP Phone 8865.

You can see all the ten buttons in the **Buttons** page under **Manage Site > Phones > Modify Phone** if ELK is enabled. When ELK is not enabled, the page displays only the five buttons.

The phone buttons can be configured for Extension, Intercom, Speed Dial, or BLF Speed Dial. The buttons in the phone are configured in the following preference order:

1. Extension
2. Intercom
3. Speed Dial
4. BLF Speed Dial

The Enterprise Line Key is not enabled by default.

Cisco IP Phone 8800 Series with Enterprise Line Key

The Cisco IP Phone 8800 Series phone displays the following lights to indicate the state:

- Flashing Amber—Incoming call
- Steady Green—Active call
- Flashing Green—Call on Hold

For the Enterprise Line Key (ELK) support offered on BE4000, the Cisco IP Phone 8800 Series behavior is as follows:

> **Note** If ELK is disabled, directory numbers aren't assigned to the line keys on the right side of the phone screen.

Table 3: Cisco IP Phone 8800 Series Phone Behavior with ELK

Feature	ELK Enabled	ELK Disabled
Call Answering	For an incoming call, the line key on the phone screen displays Flashing Amber. Press the **Answer** softkey. Once the call is answered, steady green is displayed on the line key.	For an incoming call, the line key on the right side of phone screen displays Flashing Amber. Press the **Answer** softkey. Once the call is answered, steady green is displayed on the line key.

Feature	ELK Enabled	ELK Disabled
Call Waiting or Call Queue Indication	For a second or subsequent incoming call, the line key with active call displays flashing amber.	For a second or subsequent incoming call, the line key on the right side of the phone screen displays flashing amber. The line key flashing amber is below the line key with active call.
Call Resume	If your second call is active and you need to resume the first call, press the same line key with active call to display the menu. Use four-way navigation button to select the number corresponding to the first call. Press the resume softkey to resume the first call.	If your second call is active, press the line key (on the right side of phone screen) corresponding to the first call. Press the resume softkey to resume the first call.
Caller ID Display	Press the line button with active call to display the menu. Use the four-way navigation button to select the number and perform the required action.	The incoming call is displayed on the phone screen, with the subsequent calls listed one below the other. Press the corresponding line key on the right with flashing amber to answer a call or perform required action.
Call Handling with Multiple Active Calls	If there's a second incoming call, the line key with active call itself displays flashing amber. Press the line key that displays flashing amber to answer the incoming call. A menu is displayed. Scroll down using the four-way navigation button to select the second incoming call. Press the **Answer** softkey. Now the second call is active, and the first call is put on hold. The line key displays flashing green.	If there's a second incoming call, line key on the right displays Flashing Amber. Press the line key that displays Flashing Amber to answer the second call. When the second incoming call is answered, the first call is put on hold and the corresponding line key displays flashing green. The line key with the new active call displays steady green.

Note   All the other features supported in BE4000 retain their behavior, irrespective of whether ELK is enabled or not.

Enable Enterprise Line Key

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** for the desired site from the Actions menu.

Step 3   Click **Setting > System Settings**.

Step 4   Check the **Enable Enterprise Line Key Support** check box.

Step 5   Click **Save**.

Barge

Note   Barge is enabled for shared lines only.

The Barge feature enables phone users who share a directory number to join an active call on the shared line. Barge uses the built-in conference bridge on the target phone (the phone that is being barged) which limits the number of users allowed to barge. A barge conference supports up to three parties. When the initiator barges into a call, a conference is created between the barge initiator, the target party, and the other party connected in the call. Parties see the call information on their phones. If a phone that is using the shared line has Privacy enabled, call information does not appear on the other phones that share the line and the call cannot be barged.

Connected parties hear the barge tone (single beep) after the conference is set up. When a party leaves the conference, a barge leave tone is played to the remaining parties.

Cisco Partners or Customer Administrators can enable "Barge" softkey on end users phone. Check the **Enable Barge Softkey** under "Shared Line Settings" section of the **Additional Settings** page under **Manage Site > Phone > Add Phone** and **Manage Site > Phones > Modify Phone**.

Call Park

The Call Park feature allows you to put a call on hold, so it can be retrieved from another phone.

For example, a customer calls a retail store and asks for the sporting goods department. The operator who answers the call, transfers the call to one of the extensions that is associated with the sporting goods department and informs them to retrieve the call. Any phone in the sporting goods department can answer the call by retrieving the parked call.

Only End Users can park and retrieve the calls.

For details on how to park and retrieve a call, see End-User Guides.

Group Voicemail Box

BE4000 allows you to create a voicemail box specific to a group of users. This is called group voicemail box. All users belonging to the group can access the messages in the group voicemail box. When a caller leaves a message in a group voicemail box, users' phones do not show message waiting indicator (MWI). To listen to the messages in the group voicemail box, users must log in to their personal voicemail box (by pressing **Messages** on the phone). Only one user can access the group voicemail box at a time. After the first user saves or deletes a message in the group voicemail box, the message is no longer played as "new" for any subsequent users.

Add a Group Voicemail Box

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from the "Actions" menu for the desired site on the dashboard.
Step 3	Click **Group Voicemail Box**.
Step 4	Click **Add Voicemail**.
Step 5	Enter the information in the fields. For field description, refer .
Step 6	Click the arrows at the bottom of the screen to move to next screen or to go back to the previous screen.
Step 7	Click **Yes**.

Group Voicemail Box - Field Descriptions

Table 4: Group Profile

Field	Description
Group Name	Name of the group. You can enter a minimum of 2 and a maximum of 12 characters.
Full Name	Long name of the group as it appears on other phone display.
Description	Description for the group.
Primary Extension	Extension associated with the group. **Note** The Primary Extension of the Group Voicemail Box cannot start with the digit used for dialing an outside line, placing an intercom call, and sending calls to voicemail automatically. The primary extension cannot overlap with any existing extension, Auto Attendant Pilot Number, Hunt Group Pilot Number, and Paging Number.
E.164 Number	Phone number (including country and area code) associated with the group.
Privileges/Capabilities	Privilege associated with the group. Search for the privilege and click **Add**.

Table 5: Owners/Members

Field	Description
Add Owners	Users who own the group. Search for a user and click **Add**. **Note** You must have at least one user as owner of the group.
Add Members	Users who belong to the group. **Note** You must have at least one user as member of the group.

Table 6: Voicemail Box

Field	Description
Description	Description of the group.
Mailbox Size in secs	Size of the voicemail box.
Message Expiry Time in days	Maximum number of days up to which messages are stored in the voicemail box.
Message Size in secs	Maximum message storage size, in seconds.
Play Tutorial	• **Yes**—Voicemail tutorial is played when the user enters the voicemail box for the first time. • **No**—Voicemail tutorial is not played when the user enters the voicemail box for the first time.
Recording Type	• **None**—Do not use a recording • **System-default**—Use the system default recording • **User-Recording**—Use the user recording
Greeting Type	• **Standard**—Enable, record, or download an active greeting. • **Alternate**—Enable, record, or download an alternate greeting. • **Meeting**—Enable, record, or download a meeting greeting • **Vacation**—Enable, record, or download a vacation greeting • **Extended Absence**—Enable, record, or download an extended-absence greeting • **Busy**—Enable, record, or download a busy greeting • **Closed**—Enable, record, or download a closed greeting • **Internal**—Enable, record, or download an internal greeting

Add Users to Group VoiceMail Box

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Group Voicemail Box**.

Step 4 Click **Modify Group** from the Actions menu for the desired group voicemail box.

Step 5 Navigate to **Owners/Members** page.

Step 6 Search for a user in the Add Members field and click **Add**.

Step 7 Click the arrows at the bottom of the screen to move to next screen or to go back to the previous screen.

Step 8   Click **Yes**.

Modify Owners of a Group Voicemail Box

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Group Voicemail Box**.

Step 4   Click **Modify Group** from the Actions menu for the desired group voicemail box.

Step 5   Navigate to **Owners/Members** page.

Step 6   Search for a user in the Modify Owners field and click **Add**.

Step 7   Click the arrows at the bottom of the screen to move to next screen or to go back to the previous screen.

Step 8   Click **Yes**.

Enable Voicemail Box Space Notification Emails

Cisco Partners and Customer Administrators can configure the BE4000 to trigger an email notification when the users' voicemail box is nearly full.

When the voicemail box is more than 80 percent full, an email notification **Voicemail Box [XXXX] is Nearly Full** is sent to the registered email address. When the voicemail box is 100 percent full, an email notification **Voicemail Box [XXXX] is Full** is sent to the registered email address.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** for the desired site from the Actions menu.

Step 3   Click **Setting > System Settings**.

Step 4   Check the **Enable mailbox space notification email** check box.

Step 5   Click **Save**.

What to do next

Delete obsolete voicemail messages from your voicemail box so that you can receive new voicemail messages.

Configure Voicemail Description Style

Cisco Partners and Customer Administrators can change the length of information played when end users check for voicemail messages on their phone.

The default option contains a set of information such as sender information, date and time of when the message was sent, reception date and time. You can choose to include or exclude some information with the following options:

- **Default**—Contains sender information, date and time of when the message was sent, reception date and time (if the message was received later than 30 minutes after it was sent), and other details such as the message number, type, priority.

- **Include Day of Week**—Includes information on which day of week the message was received along with all the information available for default option.

- **Brief**—Shortens information that is read before messages in voicemail by omitting data about who sent the messages. No longer reads the entire caller number or the recording time.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** for the desired site from the Actions menu.

Step 3   Click **Setting > System Settings**.

Step 4   Choose a voicemail application narrator option from the **Voicemail Description Style** drop-down list.

Step 5   Click **Save**.

Pickup Group

Pickup Group is a group of extensions, where the phone users can answer an incoming call on any of the extensions belonging to the same pickup group. By default, BE4000 creates **1** as the pickup group for all the extensions. You can change the pickup group based on your preference. You can assign only one pickup group per extension.

Create a Pickup Group

Before you begin

Have the list of extensions belonging to the pickup group ready

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Sites** from the Actions menu for the desired site.

Step 3   Click **Modify Extension** from the Actions menu for the desired Extension.

Step 4   Navigate to the **Additional Settings** page.

Step 5   Enter a number for the pickup group in the **Pickup Group** field.

Note   Pickup Group number cannot overlap with any existing hunt group pilot number.

Step 6   Click **Yes** to save the changes.

Step 7   Repeat steps 3 to 6 for all the extensions belonging to the group.

Answer a Pickup Group Call

Answer an Incoming Call from the Same Pickup Group

A phone user can answer an incoming call on any of the extensions belonging to the same pickup group in the following ways:

• Press the **Pickup** softkey

• Press the **GPickup** softkey followed by *****

Answer an Incoming Call from a different Pickup Group

A phone user can answer an incoming call on any of the extensions belonging to different pickup groups by pressing the **GPickup** softkey followed by pickup group number.

If **Pickup Call** check box is enabled for an extension in the BE4000 Portal (Refer Enable "Pickup Call" Check Box for an Extension, on page 51), the phone user can answer an incoming call in any of the pickup groups by pressing **GPickup** softkey followed *****.

Example

Pickup Group 2—Extensions 4501, 4502, and 4503

Pickup Group 3—Extensions 4504, 4505, and 4506

Default Pickup Group 1—Extensions 4507 and 4508

Table 7: Answering a Pickup Group Call

Pickup Call check box	Incoming Call On	Call Answered By	Press
Disabled	4501 (Pickup Group 2)	4502 or 4503 (any extension in the same group)	• **Pickup** softkey • **GPickup** softkey followed by *****
Disabled	4501 (Pickup Group 2)	4504 (any extension in a different Pickup Group)	**GPickup** softkey followed by **2** (Pickup Group number)
Enabled for 4507	4501 (Pickup Group 2)	4507	**GPickup** softkey followed by *****
	4504 (Pickup Group 3)	4507	

Enable "Pickup Call" Check Box for an Extension

You can configure an extension to answer an incoming call on an extension belonging to any other group by pressing **GPickup** softkey followed by *****. Enabling **Pickup Call** check box eliminates the need for pressing pickup group number while answering the call.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Sites** from the Actions menu for the desired site.

Step 3   Click **Modify Extension** from the Actions menu for the desired Extension.

Step 4   Navigate to the **Additional Settings** page.

Step 5   Check the **Pickup Call** check box.

Step 6   Click **Yes** to save the changes.

Call Detail Records

BE4000 allows you to generate the call detail records (CDR) reports after the site is successfully deployed. The CDR report is generated based on the successful incoming and outgoing external calls that are made per customer site for a specific date range. The report includes details such as the calling and called numbers, location, time, and duration of each call.

Note
- You cannot configure CDR during initial site deployment.
- The calls that are made within the organization is not captured in the CDR reports.

You can generate the following types of reports:

- Call History
- Busy Hours
- Concurrent Calls

You can download a local copy of the Call History CDR reports.

Currently, all CDR reports are retained indefinitely, unless CDR collection is disabled. However, the CDR report retention period is subjected to change in the future.

Note   If CDR collection is disabled, all previously collected call detail records get deleted and cannot be recovered.

Enable Collection of Call Detail Records

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Reports > CDR Collection**.

Step 4   Turn on **Collect Call Detail Records**.

Step 5   (Optional) Click **Add Row** and enter the desired numbers in the **Dialed Number(s)** column. When a user in the BE4000 system dials the number(s) that are entered here, an email notification is triggered.

Step 6   (Optional) Click **Add Row** and enter the email address(es) in the **Email address(es)** column. The email address that is entered here receives the notification when a user dials the numbers that are specified under **Dialed Number(s)** column.

Step 7   Click **Save**.

View Call History Report

Call History CDR report contains the details of all successful incoming and outgoing external calls that are made per site for a specific date range. The call history report includes the following details:

- Date and Time of the call

- Site Name

- Calling Number

- Called Number

- Duration of the Call

- Reason for Call Disconnect

- Trunk type

Before you begin

Ensure that the CDR Collection is enabled for the required site. For more details, see Enable Collection of Call Detail Records , on page 52.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Reports > Call Detail Records**. The Manage Express Reports page is launched in a new tab.

Step 4   Choose a location from the **Location** drop-down list.

Step 5   Choose **Call History** from **Call History** drop-down list.

Step 6   Select the required date range. The report is displayed on the screen.

Step 7   View the call history details. You can also perform a search based on the calling and called numbers.

What to do next

(Optional) Download Call History Report .

Download Call History Report

Call History CDR report contains the details of all successful incoming and outgoing external calls that are made per site for a specific date range. The call history report includes the following details:

- Date and Time of the call

- Site Name

- Calling Number

- Called Number

- Duration of the Call

- Reason for Call Disconnect

- Trunk type

Before you begin

Ensure that the CDR Collection is enabled for the required site. For more details, see Enable Collection of Call Detail Records , on page 52.

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from Actions menu for the desired site on the dashboard.
Step 3	Click **Reports > Call Detail Records**. The Manage Express Reports page is launched in a new tab.
Step 4	Choose a location from the **Location** drop-down list.
Step 5	Choose **Call History** from **Call History** drop-down list.
Step 6	Select the required date range. The report is displayed on the screen.
Step 7	View the call history details. You can also perform a search based on the calling and called numbers.
Step 8	Click **Export**. Wait for a few seconds until the report is downloaded. Close the **Export** screen by clicking **X** below your username.

View Busy Hours Report

Busy hours report provides you the total number of incoming and outgoing external calls that are made per day. The report contains the day or hour during which the calls are made, the total number of calls, and the average duration per call. The busy hours report also provides a graphical representation of the generated data.

Before you begin

Ensure that the CDR Collection is enabled for the required site. For more details, see Enable Collection of Call Detail Records , on page 52.

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from Actions menu for the desired site on the dashboard.
Step 3	Click **Reports > Call Detail Records**. The Manage Express Reports page is launched in a new tab.
Step 4	Choose a location from the **Location** drop-down list.
Step 5	Choose **Busy Hours** from the **Voice** drop-down list.
Step 6	Select the required date range. The report is displayed on the screen.
Step 7	View the busy hours CDR report. Hover the mouse pointer over the graph to view the total number of calls, Peak Hour, and Peak Value.

Step 8   (Optional) Click the graph to view the total number of calls that are made for day on an hourly basis.

View the Concurrent Calls Report

Concurrent calls CDR report provides the details of the number of internal and external outgoing calls that are active at the same time on an hourly basis. The report also contains the peak time and the total number of active calls during the peak time.

Before you begin

Ensure that the CDR Collection is enabled for the required site. For more details, see Enable Collection of Call Detail Records , on page 52.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Reports > Call Detail Records**. The Manage Express Reports page is launched in a new tab.

Step 4   Choose a location from the **Location** drop-down list.

Step 5   Choose **Concurrent Calls** from the **Voice** drop-down list.

Step 6   Select the required date range. The report is displayed on the screen.

Step 7   View the concurrent calls CDR report.

Step 8   (Optional) Click **Grid View** to view the number of concurrent calls on an hourly basis, peak time, and peak count.

View Operator Report

The Operator report contains the details of all call-related statistics including call queueing statistics, agent statistics, and service level statistics. The reports are recorded at an operator level for a specific date range. The Operator report is updated daily once. The Operator report includes the following details:

• **Agent Activity**–Provides information on the Calls Answered by agents per site.

• **Agent Statistics**—Provides information on call-related statistics:

  • **Site Name**

  • **Start Time**

  • **End Time**

  • **Total Calls Ans (From Queue)**

  • **Avg Time in Call (From Queue)**

  • **Longest Time in Call (From Queue)**

• **Call Queueing**—Provides information on the call queue:

  • **Date/Time**

- **To Queue**

- **Answered**

- **Avg Ans Time**

- **Long Ans Time**

- **Abandoned**

- **Average Abandon Time**

- **Details**

- **Queue Statistics**—Provides information on call-queue related statistics:

  - **Site Name**

  - **Start Time**

  - **End Time**

  - **Max Agents**

  - **Min Agents**

  - **Avg Time in Call (secs)**

  - **Longest Time in Call (secs)**

- **Service Level**—For more information on the Service Level report, see Service Level Report, on page 56.

Before you begin

Ensure that the CDR Collection is enabled for the required site. For more details, see Enable Collection of Call Detail Records , on page 52.

Step 1	Log in to the BE4000 portal.
Step 2	Click **Manage Site** from Actions menu for the desired site on the dashboard.
Step 3	Click **Reports > Call Detail Records**. The Manage Express Reports page is launched in a new tab.
Step 4	Choose the required report type from **Operator** drop-down list.
Step 5	Select the required date range. The report is displayed on the screen.

Service Level Report

Service Level report is an interactive graphical representation of call waiting period and abandoned calls. Service Level report has two modes:

- **Grouped**

- **Stacked**

Note All the call waiting ranges are enabled by default and appears in the graph. You can turn off a call waiting range by clicking on the legend definition. The statistics related to that call waiting period is not displayed on the graph.

Click the graph for a tabular representation of the generated data, including:

- **Day**
- **Calls Waited Less Than 12 s**
- **Calls Waited Btwn 12–24 s**
- **Calls Waited Btwn 25–36 s**
- **Calls Waited Btwn 37–60 s**
- **Calls Waited More Than 60 s**
- **Calls Abdnd Less Than 12**
- **Calls Abdnd Btwn 12–24 s**
- **Calls Abdnd Btwn 25–36 s**
- **Calls Abdnd Btwn 37–60 s**
- **Calls Abdnd After 60 s**

Disable Collection of Call Detail Records

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3 Click **Reports > CDR Collection**.

Step 4 Turn off **Collect Call Detail Records**.

Step 5 Click **Save**.

Task Log

You can view the history of all the tasks that are performed on a BE4000 site in the past 62 days.

By default, the portal shows the tasks that are performed from 12AM on the current date. You can select the preferred date range from the calender available at the top right corner of the page. The options available for selecting the date range are:

- **Today**—Tasks that are performed from 12AM on the current date.
- **Yesterday**—Tasks that were performed yesterday.

- **Last 7 Days**—Task that were performed in the past 7 days.

- **Last 30 Days**—Tasks that were performed in the past 30 days.

- **This Month**—Tasks that were performed from the first of the current month until the current date.

- **Custom Range**—Choose start date and end date from the calender and click "Apply". If you want to select a single date (as both start and end date), double-click the date and click "Apply".

The task log provides the following details:

- Task Name—Name of the task.

- Username—User who performed the task.

- Start Time—Time when the task started.

- Last Updated Time—Time when the task ended.

- Status—Result of the task.

- Action—View trace file for the task.

You can view the trace file of each task. Trace file helps in troubleshooting the issues with the task performed. You can also get the trace file sent to your preferred email address.

Note

- Task Log is enabled by default and cannot be disabled.

- Task Log contains the list of tasks that are performed only after the initial site deployment.

View Task Log

Step 1 Log in to the BE4000 portal.

Step 2 Click **Manage Site** from "Actions" column corresponding to the desired site.

Step 3 Click **Reports > Task Log**.

Step 4 Select the duration for which you want to view the task log. You can choose one of the following options:

- **Today**—Tasks that are performed from 12AM on the current date.

- **Yesterday**—Tasks that were performed yesterday.

- **Last 7 Days**—Task that were performed in the past 7 days.

- **Last 30 Days**—Tasks that were performed in the past 30 days.

- **This Month**—Tasks that were performed from the first of the current month until the current date.

- **Custom Range**—Choose start date and end date from the calender and click "Apply". If you want to select a single date (as both start and end date), double-click the date and click "Apply".

Step 5 View the task log displayed on the screen. You can filter the log based on "Task Name" or "Username".

Step 6   (Optional) Click **View Trace** from the "Actions" column corresponding to the task and view the trace file. You can also get the trace file of the task sent to your email address.

Emergency Alerting

BE4000 allows you to monitor the calls that are made from within the organization to specific external numbers, such as (911). You can enter the numbers to monitor and the email address to receive the notification when the specified numbers are dialed.

Add a Number for Emergency Alert Notification

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Reports > CDR Collection**.

Step 4   Turn on **Collect Call Detail Records**.

Step 5   Click **Add Row** and enter the desired numbers under **Dialed Number(s)**. When a user in the BE4000 system dials the number(s) entered here, an email is triggered.

Step 6   Enter the **Add Row** and enter the email address(es) in under **Email address(es)**. The email address entered here receives a notification when a user dials the numbers specified under **Dialed Number(s)**.

Step 7   Click **Save**.

Remove a Number from Emergency Alert Notification

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Reports > CDR Collection**.

Step 4   Turn on **Collect Call Detail Records**.

Step 5   Click **Delete** corresponding to the required number under **Emergency Alerting** section.

Step 6   Click **Save**.

Cisco ATA Analog Telephone Adapters

Cisco ATA Analog Telephone Adapters are the handset-to-Ethernet adapters that enable you to turn analog devices, such as phones, fax machines, into IP devices. The Cisco ATA Analog Telephone Adapter supports two voice ports, each with an independent phone number and has an RJ-45 10/100BASE-T data port.

Note
- Cisco ATA Analog Telephone Adapters can be added only post site deployment. You cannot add during initial site deployment.
- Cisco ATA Analog Telephone Adapter can be added only under **Manage Site > Phones**. You cannot add Cisco ATA Analog Telephone Adapter under **Manage Site > Extensions** and **Manage Site > User Management**.
- A fax call does not fall back to voice only call after the fax is sent.
- The following features are not supported:
    - Paging
    - Single Number Reach (SNR)
    - Speed dial

Configure Cisco ATA Analog Telephone Adapter

Before you begin

Ensure that you have:
- Added an extension that is freely available for provisioning. While adding an extension, you do not select any phone type. For detailed information on adding an extension, see Add an Extension.
- Added Cisco ATA Analog Telephone Adapter in the BE4000 portal. For detailed information on, see Add a New Phone.
- Phone devices ready
- TFTP Address
- IP Address and subnet mask for Cisco ATA Analog Telephone Adapter

Step 1 Connect the phones to Line 1 and Line 2.

Step 2 Access Interactive Voice Response (IVR).

a) Go off hook on the phone and press ********. The IVR prompts for password

b) Enter the default password, **24726** followed by **#**.

Note The Cisco ATA Analog Telephone Adapter allows you to enter only numerical values for the password.

Step 3 Set the static IP mode.

a) Press **101** followed by **#** to enter the LAN configuration mode.

b) Press **1** followed by **#** to set the static IP mode.

c) Press **1** to save the changes.

Note After setting the IP mode to static IP, you must go on-hook to configure the IP address, subnet mask, and default gateway. To set static IP, network cable must be connected.

Step 4 Configure the IP Address.

    a) Press **111** followed by **#** to enter the IP address configuration mode.

    b) Enter the **IP address** followed by **#**. Enter "*" for in place of "." while entering the IP address. For example, if your IP address is 198.51.100.20, enter 198*51*100*20#.

    c) Press **1** to save the changes.

Step 5 Configure the subnet mask.

    a) Press **121** followed by **#** to enter the subnet configuration mode.

    b) Enter the **subnet mask** followed by **#**. Enter "*" for in place of "." while entering the subnet mask. For example, if your IP address is 255.255.0.0, enter 255*255*0*0#.

    c) Press **1** to save the changes.

Step 6 Configure the default gateway.

    a) Press **131** followed by **#** to enter the default gateway configuration mode.

    b) Enter the **default gateway** followed by **#**. Enter "*" for in place of "." while entering the subnet mask. For example, if your default gateway address is 198.51.100.1, enter 198*51*100*1#.

    c) Press **1** to save the changes.

Step 7 Configure the TFTP address.

    a) Press **221** followed by # to enter the TFTP configuration mode.

    b) Enter the **TFTP address** followed by **#**. Enter "*" for in place of "." while entering the subnet mask. For example, if your TFTP address is 203.0.113.20, enter 203*0*113*20#.

    c) Press **1** to save the changes.

Step 8 Restart Cisco ATA Analog Telephone Adapter by unplugging and re-plugging the Ethernet cable from phone to the Cisco ATA Analog Telephone Adapter.

Line Cards

BE4000 allows you to connect to traditional telephony services and devices by adding Network Interface Modules (NIM). You can add a maximum of two NIM cards (also known as Line cards). For the list of supported NIM cards, refer Supported Line Cards.

On the portal, you can add a new NIM card or edit an existing NIM card. Currently, removal and replacement of NIM cards are not supported on the BE4000 portal.

Line Cards and their Usage

- NIM-2FXS and/or NIM-4FXS: You can use two or four ports FXS to connect directly to phones, fax machines, and key systems. These line cards are not recommended for off-premises use. Voicemail is automatically set up when extensions are added.

- NIM-2FXO and/or NIM-4FXO: You can use two or four ports FXO to provide off-premises connections to the public switched phone network (PSTN). FXO ports are placed in either of the two trunk groups—Inbound Only or In and Outbound.

- NIM-2FXS and/or NIM-4FXO: Allows FXS and FXO connections in a single card.

- NIM-2BRI-NT/TE and/or NIM-4BRI-NT/TE: You can use to connect a client-side (TE) physical interface for connection to a network termination (NT) interface with phantom power terminating at an ISDN telephone network. BRI is configured for a single port.

- NIM-1MFT-T1/E1: Use this card to connect Primary rate ISDN services to the BE4000 appliance.

Add a Line Card

Before you begin

- A line card must be installed into the BE4000 appliance

- Have the DID numbers for the line cards ready

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Line Cards (NIM)**.

Step 4   Click **Add NIM Card**.

Step 5   Add the DID numbers on the **Direct-Inward-Dial** page.

| Note | Do not add FXO line numbers. FXO line numbers can be added as port configurations on the upcoming Line Card page. |

Step 6   Enter the configuration details based on the type of card inserted. For field descriptions, refer Line Card - Field Descriptions, on page 63.

| Note | If you already have a line card installed into your BE4000 appliance, then it automatically gets listed. However, you cannot edit the configurations of an installed line card while adding a new line card. Refer to Edit a Line Card , on page 71 for details on editing a line card configuration. |

Step 7   Enter the details in the **Inbound Call Mapping** and **Outbound Caller ID** pages based on your requirements.

| Note | You can map the incoming calls on the DID number to a **Meet Me** conference number, apart from **Auto Attendants**, **Hunt Group**, and **Extensions**. |

Step 8   Click **Yes** to save the changes.

| Note | The phone systems go offline and you cannot make or receive calls until the updates are successfully completed. |

Step 9   Click **Done**.

| Note | Ensure that emergency services, DID numbers, and card features are tested and verified. |

Line Card - Field Descriptions

NIM-2FXS or NIM-4FXS

Field	Description
If you are using **NIM-2FXS** or **NIM-4FXS**, input the following details:	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Analog Phone**—The line is connected to an analog phone. • **Paging**—The line is connected to an external paging system. • **Automatic Ringdown**—The line is connected to a phone that has a dedicated connection with another phone. When the phone connected to this line goes off hook, the destination phone rings. There is no need to dial any number.
Label	Enter the label to identify each line.
Extension	Enter the extension number.
Destination	**Note** This field is visible only when you choose "Automatic Ringdown" from the **Type** drop-down list. Enter the extension of the destination phone that rings when the phone goes off hook.
Restrictions	Choose one of the following Class of Restriction (COR): • **internal** • **local** • **local-plus** • **national** • **national-plus** • **international** Class of Restriction decides the type of calls that can be placed from the FXS phone line. **Note** • You must select the Class of Restriction (COR) for every line while adding FXS cards in the Setup Assistant. • This field is available only when you choose **Analog Phone** from the "Type" drop-down list.

Field	Description
Law	Choose the type of algorithm used for modifying an input signal for digitization: • u-law • a-law
Input Gain (db)	The amount of amplification or deamplification of sound in terms of decibels. Range is from -6 to 14 db. Default value for FXO line is 0 db and for FXS line is 6 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
Output Attenuation (db)	The value that is configured for minimizing the signal loss. Range is from -6 to 14 db. Default value for FXO and FXS line is 0 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).

NIM-2FXO or NIM-4FXO

Field	Description
If you are using **NIM-2FXO** or **NIM-4FXO**, input the following details:	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network. • **Paging**—The line is connected to an external paging system.
Label	Enter the label name.
Name	**Note** This field appears only when "Trunk" is chosen from the "Type" drop-down list. Enter a name for the trunk line.
Extension	Enter the number for analog phone line.

Field	Description
Number	**Note** This field appears only when "Trunk" is chosen from the "Type" drop-down list. Enter the number for the trunk line.
Start Type	Choose one of the following: • Ground • Loop Start
Direction	Mark the line as incoming only or bidirectional. The system builds the trunk groups based on what you select. • In + Out—Allows the phone line to receive and make calls. • Inbound Only—Allows the phone line to receive the calls.
Law	Choose the type of algorithm used for modifying an input signal for digitization: • u-law • a-law
Input Gain (db)	The amount of amplification or deamplification of sound in terms of decibels. Range is from -6 to 14 db. Default value for FXO line is 0 db and for FXS line is 6 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
Output Attenuation (db)	The value that is configured for minimizing the signal loss. Range is from -6 to 14 db. Default value for FXO and FXS line is 0 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
Show Advanced	Displays the advanced fields.
Configuration Voice Port Commands **Note** The following fields are displayed only when the **Show Advanced** check box is checked.	
Cable Detect	Enables or disables cable polling on an analog Foreign Exchange Office (FXO) port.

Field	Description
Bearer Capability	**Note** Configuring "Bearer Capability" drop-down list does not have any affect on the FXO port. Specifies the information transfer capability of the bearer capability information element (IE) in the outgoing ISDN SETUP message for Session Initiation Protocol (SIP) early-media calls. You can choose one of the following: • Speech—Specifies speech as the information transfer capability. • 3100hz—Specifies 3.1 kHz audio as the information transfer capability.
Supervisory Disconnect	Configures the type of supervisory disconnect signaling available on the FXO port. When the supervisory disconnect tone is detected on the FXO port, the system interprets this as a disconnect indication from the switch and clears the call. Choose one of the following: • dualtone—Disconnects calls when the router detects call-progress tones from a PBX or the PSTN. • anytone—Disconnects the call if the PBX or PSTN does not provide a supervisory tone. Examples of tones that trigger a disconnect include busy tone, fast busy tone, and dial tone. • signal—Enables a disconnect indication by detecting the power denial which uses the LCFO signal on the remote end.
Time-outs	
Call Disconnect	Specifies when to disconnect the call. You can choose one of the following: • timer—Specifies the duration to wait while the phone is ringing before disconnecting the call. Range is from 0 to 120 seconds. • infinity—Disables disconnect supervision. The voice port does not disconnect when a disconnect tone is detected.
Initial Timeout	Specifies the number of seconds for which the system waits for the caller to input the first digit of the dialed digits. Range is from 0 to 120 seconds.
Power Denial	**Note** Configuring Power Denial drop-down list does not have affect on the FXO port. Sets the duration of the power denial that the voice gateway applies to the FXS port when a call disconnects. Range is from 0 to 2500 seconds.
Wait Release	Limits the time a voice port can be held in a call failure state. After the timeout, the release sequence is enabled. You can choose one of the following: • timer—Range is from 1 to 3600 seconds. • infinity—The voice port is never released as long as the call-failure state remains.

NIM-2FXS/4FXO

Field	Description
If you are using **NIM-2FXS/4FXO**, input the following details:	
FXS	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Analog Phone**—The line is connected to an analog phone. • **Paging**—The line is connected to an external paging system. • **Automatic Ringdown**—The line is connected to a phone that has a dedicated connection with another phone. When the phone connected to this line goes off hook, the destination phone rings. There is no need to dial any number.
Label	Enter the label to identify each line.
Extension	Enter the extension number.
Destination	**Note** This field is visible only when you choose "Automatic Ringdown" from the **Type** drop-down list. Enter the extension of the destination phone that rings when the phone goes off hook.
Restrictions	Choose one of the following Class of Restriction (COR): • **internal** • **local** • **local-plus** • **national** • **national-plus** • **international** Class of Restriction decides the type of calls that can be placed from the FXS phone line. **Note** • You must select the Class of Restriction (COR) for every line while adding FXS cards in the Setup Assistant. • This field is available when you choose **Analog Phone** from the "Type" drop-down list.

Field	Description
Law	Choose the type of algorithm used for modifying an input signal for digitization: • u-law • a-law
Input Gain (db)	The amount of amplification or deamplification of sound in terms of decibels. Range is from -6 to 14 db. Default value for FXO line is 0 db and for FXS line is 6 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
Output Attenuation (db)	The value that is configured for minimizing the signal loss. Range is from -6 to 14 db. Default value for FXO and FXS line is 0 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
FXO	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network. • **Paging**—The line is connected to an external paging system.
Label	Enter the label name.
Name	**Note** This field appears only when "Trunk" is chosen from the "Type" drop-down list. Enter a name for the trunk line.
Extension	Enter the number for analog phone line.
Number	**Note** This field appears only when "Trunk" is chosen from the "Type" drop-down list. Enter the number for the trunk line.

Field	Description
Start Type	Choose one of the following: • Ground • Loop Start
Direction	Mark the line as incoming only or bidirectional. The system builds the trunk groups based on what you select. • In + Out—Allows the phone line to receive and make calls. • Inbound Only—Allows the phone line to receive the calls.
Law	Choose the type of algorithm used for modifying an input signal for digitization: • u-law • a-law
Input Gain (db)	The amount of amplification or deamplification of sound in terms of decibels. Range is from -6 to 14 db. Default value for FXO line is 0 db and for FXS line is 6 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).
Output Attenuation (db)	The value that is configured for minimizing the signal loss. Range is from -6 to 14 db. Default value for FXO and FXS line is 0 db. **Note** This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).

NIM-2BRI-NT/TE or NIM-4BRI-NT/TE

Field	Description
If you are using **NIM-2BRI-NT/TE** or **NIM-4BRI-NT/TE**, input the following details:	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network.

Field	Description
Service Name	Choose a service name from the drop-down list. The drop-down list contains the list of service providers that you added in the DID page. Service name entered here is shared across all the lines or ports for the selected NIM type. **Note** You cannot choose the same service provider for SIP and Line Cards.
Static TEI	If your service provider requires that your line use a static Terminal Endpoint Identifier, enter the value between 0 and 63. If the field is left blank, the line attempt to negotiate a TEI.
Overlap Receiving	Choose whether you want your call setup to work based on overlap receiving. You can enable or disable this option. If your service provider does not use "enbloc" signaling, this option allows BE4000 to wait for additional digits to be received before the call is routed.
Send Redirecting IE Number	Check the "Send Redirecting IE Number" check box to include the Redirecting Number Information Element in the outbound Setup messages. Leave unchecked if you are not sure about your service provider supporting this feature.
ISDN SPID	Enter the ISDN SPID. Some service providers use service profile identifiers (SPIDs) to define the services subscribed to by the ISDN device that is accessing the ISDN service provider. A SPID is usually a seven-digit phone number with some optional numbers.
TEI Negotiation Method	Choose a method for TEI negotiation based on your service provider requirements. Setting a static TEI overrides TEI negotiation. The default behavior is TEI to be negotiated on power-up. The following options are provided to preserve or remove a negotiated TEI when the interface is reset: • Power Up and Remove • Power Up and Preserve • First Call and Remove • First Call and Preserve

NIM-1MFT-T1/E1, NIM-2MFT-T1/E1, or NIM-4MFT-T1/E1

Field	Description
If you choose **NIM-1MFT-T1/E1**, **NIM-2MFT-T1/E1**, or **NIM-4MFT-T1/E1**, enter the following details:	
Card Type	Choose the card type based on your customer network requirement. E1 PRI is chosen by default. The available options are: • T1 PR1 • E1 PRI

Field	Description
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network.
Service Name	Choose a service name from the drop-down list. The drop-down list contains the list of service providers that you added in the DID page. Service name entered here is shared across all the lines or ports for the selected NIM type. **Note** You cannot choose the same service provider for SIP and Line Cards.
ISDN Switch Type	Choose one of the following ISDN Service Provider PRI Switch Types: • primary-4ess • primary-5ess • dms100 • primary-net5 • primary-ni "primary-4ess" is chosen by default.
Controller Setup	Defines the controller setup for configuring channelized T1 or E1 controllers. Choose either Full PRI or partial PRI.
Line Code	Choose a line code. By default, the line code for E1 PRI is high-density bipolar 3 (hdb3).
Framing	Choose the framing from the drop-down list. This option defines the framing characteristics.
Send Redirecting IE Number	Check the "Send Redirecting IE Number" check box to include the Redirecting Number Information Element in the outbound Setup messages. Leave unchecked if you are not sure about your service provider supporting this feature.

Edit a Line Card

Before you begin

• A line card must be installed into the BE4000 appliance.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site** from Actions menu for the desired site on the dashboard.

Step 3   Click **Line Cards (NIM)**.

Step 4   Click **Edit Configuration**.

Step 5   Modify the required details for the installed line cards. For field descriptions, refer Line Card - Field Descriptions, on page 63.

Step 6   Click **Yes** to save the changes.

> **Note**   The phone systems go offline and you cannot make or receive calls until the updates are successfully completed.

Step 7   Click **Done**.

> **Note**   Ensure that emergency services, DID numbers, and card features are tested and verified.

Private Line Automatic Ringdown

BE4000 supports configuring Private Line Automatic Ringdown (PLAR) on FXS ports available on the NIM cards. PLAR is also known as "Automatic Ringdown".

PLAR creates a dedicated connection with between two phones. When a phone configured with PLAR goes off-hook, the destination phone rings automatically without dialing any number.

You can configure PLAR while adding or modifying Line cards containing FXS ports.

> **Note**   PLAR is not supported on Analog Telephone Adapter (ATA) FXS ports.

Configure Private Line Automatic Ringdown

Before you begin

Line Card (NIM) with FXS port is inserted into the BE4000 appliance.

Step 1   Log in to the BE4000 portal.

Step 2   Click **Manage Site > Line Cards (NIM)** corresponding to the desired site.

Step 3   Click **Add NIM Card** if you are adding line card for the first time or **Edit Configuration** if you are modifying the existing line card configuration.

If you are adding line card for the first time, add DID numbers (except for FXO numbers).

Step 4   Click the **Line** corresponding to which you wish to configure Private Line Automatic Ringdown (PLAR).

Step 5   Choose **Automatic Ringdown** from the **Type** drop-down list.

Step 6   Enter a text in the **Label** field which makes you easy to identify the line.

Step 7   Enter an extension that is associated with the PLAR line in the **Extension** field.

Step 8   Enter an extension in the **Destination** field. The extension that is configured here rings automatically when the phone configured with PLAR goes off-hook.

Step 9   Choose an option from the **Law** drop-down list based on your network preferences.

Step 10   Save the configurations.

Troubleshoot the Status of Line Cards

Click **Manage Site > Line Cards (NIM)** to check the status of the line cards configured on your BE4000 appliance. The **Installed Hardware** section provides the detailed status of the line cards including the timestamp when the line card status was fetched. You can click **Refresh** to fetch the latest status of the line cards.

✎

Note It takes around 30 seconds to fetch the status of the line cards.

Figure 1: Line Card Status Summary

Table 8: Line Card Status Analysis

Column	Description
Slot	Two line cards slots available in the BE4000 appliance. NIM 1 and NIM 2 are always displayed irrespective of the card inserted on to the BE4000 appliance.
Card Type	Type of card inserted on to the BE4000 appliance. If you have not inserted a line card, "No card found" is displayed.

Column	Description	
Status	Based on your line card configuration, one of the following is displayed:	
	Good	Line card is in service. DSP resources and all the ports are in service.
	Not Configured	The line card is inserted on the BE4000 appliance, but not configured on the BE4000 portal. Click **Add NIM Card** under "Manage Card Settings" section to configure the line card.
	Needs Attention	There is an error related to the line card service. Either DSP resources or one of the ports is not in service. Click **Needs Attention** to see the detailed status.
DSP	DSP (PVDM4) resources available for the line card. DSP resources are required for configuring the ports.	
	⚠ (Needs Attention)	Either DSP resources are not available or DSP resources are not in service. Click **Needs Attention** to see the detailed status.
	✔	DSP is available and in service.
Port	The number of ports available for the line card type is listed with its status. The status of the ports can be one of the following:	
	⚠ (Needs Attention)	Either port is not configured or not in service. Click **Needs Attention** to see the detailed status.
	✔	Port is in service.

Click **Needs Attention** for detailed status.

Figure 2: Line Card Detailed Status

Click **Show** corresponding to the L1 Status for error report.

Figure 3: L1 Status

Click **Cycle Port** to disconnect and reconnect the port. When you recycle the port, all active calls on the port are disconnected.

Figure 4: Cycle Port
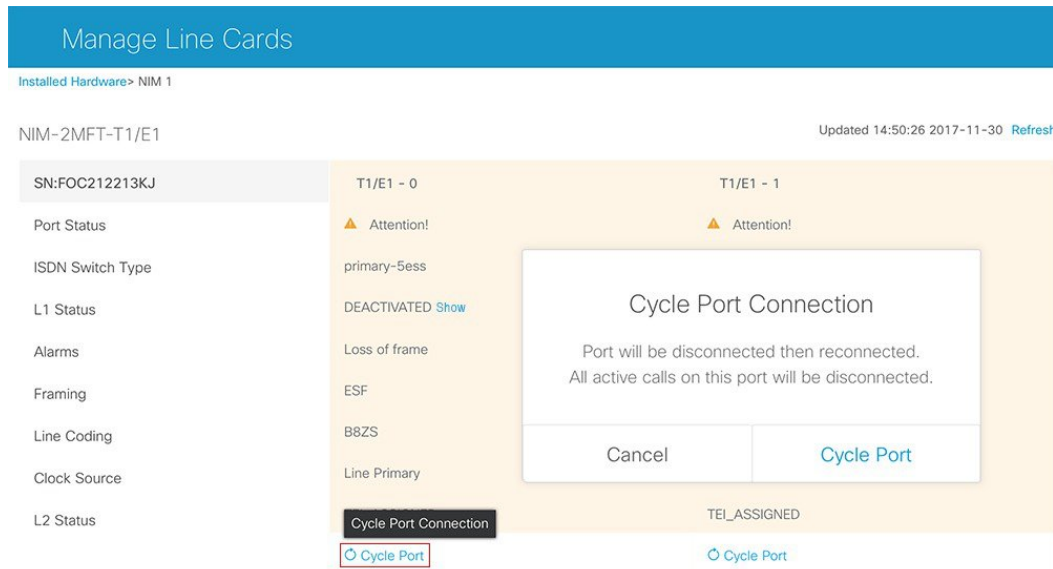
Private IP Address Ranges

IP addresses used for provisioning the phones must be allowed in RFC 1918 (10.0.0.8 /16, 172.16.0.0 /12, 192.168.0.0 /16). If the IP address that is used for provisioning phone is other than the ones allowed in RFC 1918, the phone provisioning fails. To avoid such phone provisioning issues, you must enter the IP addresses in the **Private IP Ranges** page. You can add individual IP address (host address) in the x.x.x.x format or an IP address range (subnet) in the x.x.x.x /n format.

Cisco Partners can add IP addresses during the initial site deployment on the **Private IP Ranges** page. Post site deployment, both Cisco Partners and Customer Administrators can add private IP addresses, under **Manage Sites > Settings > Private IP Ranges**.

Modify Music on Hold

By default, the standard audio file is played during call hold. You may replace the standard audio file by uploading your own custom audio file.

Step 1 Click **Actions** corresponding to your site from the Dashboard.

Step 2 Click **Music on Hold** under Manage Site.

Step 3 Click **Upload**.

Note BE4000 supports only .au and .wav audio file with G.711; ITU-T a-law or u-law, 8kHz, 8 bit, Mono format. The file cannot be larger than 1 MB (about 2 minutes). The filename cannot have space and special characters.

Step 4 Click the arrow at the bottom of the screen and click **Yes** to save.