




Manage Your Site

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View Status of the BE4000 Appliance

You can view the status of all the BE4000 appliances at various customer locations. Service status is refreshed in real time. Click the  icon to manually refresh the status.

For more information on the various status states available for BE4000, their definitions, and ways to troubleshoot them, see [Troubleshoot the Status of BE4000](#), on page 2.

Troubleshoot the Status of BE4000

Appliance Status	Description	Troubleshooting Information
Online	Service is available.	—

Appliance Status	Description	Troubleshooting Information
Offline	Appliance's management connection to the cloud is down.	<p>Check if:</p> <ul style="list-style-type: none"> • Appliance is disconnected. • Appliance is powered off. • Customer has changed the firewall settings and these ports are blocked: UDP 500, UDP 4500, or ESP 50. • Customer's internet connection is down.
Inactive	Appliance has been inactive for more than a period.	<p>Customer might be:</p> <ul style="list-style-type: none"> • Moving offices. • Switching out internet providers over the weekend. <p>Appliance must be reloaded when it is connected back.</p>
Needs Attention	There is an error related to the service.	<p>Errors can be for the following reasons:</p> <ul style="list-style-type: none"> • More than 15% of the phones are unregistered. • Any issue with the traditional or VoIP trunks. • There is impact to the phone services.
Configuration in Process	Site is being set up by the user.	Ensure that you enter all details in the BE4000 navigation wizard to make the appliance ready to be deployed.
Ready to Deploy	Your configuration is complete and ready to be used for the BE4000 deployment.	—
Deploying	Appliance is already registered with this configuration and is currently being configured.	—
Deployment Failed	Appliance is not deployed.	<p>You must view the navigation on the appliance for more information. Deployment can fail for the following reasons:</p> <ul style="list-style-type: none"> • Most likely the ports are blocked. • Wrong LAN settings. • Connectivity issues.
Voicemail Configuration Failed	Voicemail functionality does not work.	Check if the voicemail appliance is faulty. Contact Cisco TAC for assistance.

Appliance Status	Description	Troubleshooting Information
Inventory Failed	Your appliance inventory failed.	Go to the Deployment status page and click the trace icon to get the deployment failure log details. Redeploy the site again. Contact Cisco TAC for assistance.

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.

Procedure

-
- 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.
-

Modify the Customer Site After Deployment

You can change the settings of your BE4000 appliance after deployment.

Configuration in Process	--	--			
Configuration in Process	--	--			
Online	--	8	(6)	1d	...
Online	--	11	(9)	1d	...
Configuration in Process	--	--		--	...

Manage Site

Configuration Summary

Delete Site

393525

Procedure

-
- Step 1** Select a deployed customer site and click the dotted lines next in the **Actions** column.
 - Step 2** Click **Manage Sites** to display the configuration summary and editing options.

You can edit the following and then save the changes:

- **Auto Attendant**—Modify auto attendant settings.

- **Business Hours**—Modify open and closed hours.
- **Extensions**—Modify or delete extension numbers.
- **Direct Inward Dial Numbers**—Add, edit, or delete the Direct Inward Dial (DID) numbers.
- **Group Voicemail Box**—Create group voicemail box for various departments.
- **Hunt Groups**—Modify hunt groups.
- **Line Cards (NIM)**—Change line card settings.
- **Music on Hold**
- **Night Service**—Enable or disable night service.
- **Personal Voicemail Box**—Create personal voicemail box for users.
- **Phones**—Add or delete phones.
- **SIP Trunk**
- **Troubleshooting**
- **User Management**

Step 3 Save the changes.

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.

Procedure

- 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).
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.

Procedure

-
- 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 7](#). 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: <ul style="list-style-type: none"> • 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).

Field	Description
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.

Configuration Templates

You can save the information that you entered in the Setup Assistant wizard as a configuration template and reuse it while creating other BE4000 sites for the same customer with similar configurations. This avoids entering the configuration details in the Setup Assistant wizard all over again.

You can create configuration templates at both Partner level and Customer level. The Personally Identifiable Information (PII) is not saved as part of the configuration template. The template that is saved at Partner level has limited data compared to that at a customer level.

A customer level configuration template saves all the information that is entered in the Setup Assistant wizard except for the "External Public IP address".

A Partner level configuration template does not save the following information that is entered in the Setup Assistant wizard:

- SMTP Details
- Station Details
- Auto Attendant Audio files (When a saved template is loaded, Auto Attendant is loaded with the default prompts)
- Music On Hold Audio files
- Direct Inward Dial Numbers
- Member list in the Hunt Group
- Auto Attendant Drop Through Destination
- Inbound Call Mapping Details
- Outbound Call Mapping Details
- System Operator Details
- Paging Groups

Create a Configuration Template from an Existing BE4000 Site

You can create a configuration template from an existing site that is deployed and is in the "Online" state.

Procedure

-
- Step 1** Click **Create Template** from the Actions menu corresponding to the desired site.

- You are directed to the "Connectivity" page of the Setup Assistant wizard.
- Step 2** Edit the details on the pages if necessary.
 - Step 3** Click **Save as Template**.
 - Step 4** Choose Cisco Partner or customer name from **Template Level** drop-down list.
 - Step 5** Choose "New" or an existing template from the **Apply to Template** drop-down list.
 - Step 6** Enter a name for the template in the **Template Name** field.
 - Step 7** Enter a description in the **Description** field.
 - Step 8** Click **Save**.
-

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.

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

Procedure

- 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 10](#).
 - 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 19](#) 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: <ul style="list-style-type: none"> • 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	<p>Note This field is visible only when you choose "Automatic Ringdown" from the Type drop-down list.</p> <p>Enter the extension of the destination phone that rings when the phone goes off hook.</p>

Field	Description
Restrictions	<p>Choose one of the following Class of Restriction (COR):</p> <ul style="list-style-type: none"> • internal • local • local-plus • national • national-plus • international <p>Class of Restriction decides the type of calls that can be placed from the FXS phone line.</p> <p>Note</p> <ul style="list-style-type: none"> • 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.
Law	<p>Choose the type of algorithm used for modifying an input signal for digitization:</p> <ul style="list-style-type: none"> • u-law • a-law
Input Gain (db)	<p>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.</p> <p>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).</p>
Output Attenuation (db)	<p>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.</p> <p>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).</p>

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: <ul style="list-style-type: none"> • 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.
Start Type	Choose one of the following: <ul style="list-style-type: none"> • Ground • Loop Start
Direction	Mark the line as incoming only or bidirectional. The system builds the trunk groups based on what you select. <ul style="list-style-type: none"> • 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: <ul style="list-style-type: none"> • 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).

Field	Description
Output Attenuation (db)	<p>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.</p> <p>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).</p>
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.
Bearer Capability	<p>Note Configuring "Bearer Capability" drop-down list does not have any affect on the FXO port.</p> <p>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:</p> <ul style="list-style-type: none"> • Speech—Specifies speech as the information transfer capability. • 3100hz—Specifies 3.1 kHz audio as the information transfer capability.
Supervisory Disconnect	<p>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:</p> <ul style="list-style-type: none"> • 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	<p>Specifies when to disconnect the call. You can choose one of the following:</p> <ul style="list-style-type: none"> • 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.

Field	Description
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	<p>Note Configuring Power Denial drop-down list does not have affect on the FXO port.</p> <p>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.</p>
Wait Release	<p>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:</p> <ul style="list-style-type: none"> • 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	<p>Choose one of the following:</p> <ul style="list-style-type: none"> • 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	<p>Note This field is visible only when you choose "Automatic Ringdown" from the Type drop-down list.</p> <p>Enter the extension of the destination phone that rings when the phone goes off hook.</p>

Field	Description
Restrictions	<p>Choose one of the following Class of Restriction (COR):</p> <ul style="list-style-type: none"> • internal • local • local-plus • national • national-plus • international <p>Class of Restriction decides the type of calls that can be placed from the FXS phone line.</p> <p>Note</p> <ul style="list-style-type: none"> • 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.
Law	<p>Choose the type of algorithm used for modifying an input signal for digitization:</p> <ul style="list-style-type: none"> • u-law • a-law
Input Gain (db)	<p>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.</p> <p>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).</p>
Output Attenuation (db)	<p>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.</p> <p>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).</p>
FXO	

Field	Description
Type	Choose one of the following: <ul style="list-style-type: none"> • 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.
Start Type	Choose one of the following: <ul style="list-style-type: none"> • Ground • Loop Start
Direction	Mark the line as incoming only or bidirectional. The system builds the trunk groups based on what you select. <ul style="list-style-type: none"> • 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: <ul style="list-style-type: none"> • 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).

Field	Description
Output Attenuation (db)	<p>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.</p> <p>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).</p>

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	<p>Choose one of the following:</p> <ul style="list-style-type: none"> • Disabled—The line is inactive. • Trunk—The line is connected to the Public Switched Telephone Network.
Service Name	<p>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.</p> <p>Note You cannot choose the same service provider for SIP and Line Cards.</p>
Static TEI	<p>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.</p>
Overlap Receiving	<p>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.</p>
Send Redirecting IE Number	<p>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.</p>
ISDN SPID	<p>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.</p>

Field	Description
TEI Negotiation Method	<p>Choose a method for TEI negotiation based on your service provider requirements. Setting a static TEI overrides TEI negotiation.</p> <p>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:</p> <ul style="list-style-type: none"> • 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	<p>Choose the card type based on your customer network requirement. E1 PRI is chosen by default. The available options are:</p> <ul style="list-style-type: none"> • T1 PRI • E1 PRI
Type	<p>Choose one of the following:</p> <ul style="list-style-type: none"> • Disabled—The line is inactive. • Trunk—The line is connected to the Public Switched Telephone Network.
Service Name	<p>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.</p> <p>Note You cannot choose the same service provider for SIP and Line Cards.</p>
ISDN Switch Type	<p>Choose one of the following ISDN Service Provider PRI Switch Types:</p> <ul style="list-style-type: none"> • primary-4ess • primary-5ess • dms100 • primary-net5 • primary-ni <p>“primary-4ess” is chosen by default.</p>

Field	Description
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.

Procedure

-
- 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 10](#).
- 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.

Procedure

-
- 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





The screenshot shows a web interface titled "Manage Line Cards" with a sub-section for "Installed Hardware". The hardware table is as follows:

Slot	Card Type	Status	DSP	Port	0	1
NIM 1	NIM-2MFT-T1/E1	Needs Attention	✓	✓	✓	⚠
NIM 2	No card found					

Below the table, there is a "Manage Card Settings" section with two main options: "Add NIM Card" (Install and Configure new NIM cards) and "Edit Configuration" (Change the settings of installed NIM cards). A "Close" button is located at the bottom right of the interface.

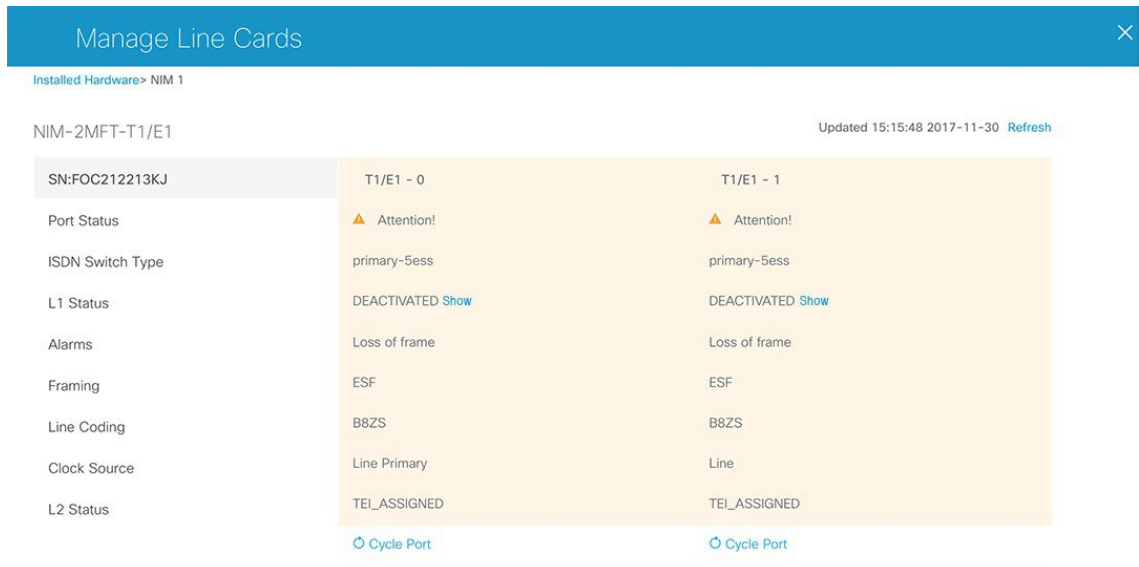
Table 1: Line Card Status Analysis

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.	
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.

Column	Description	
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

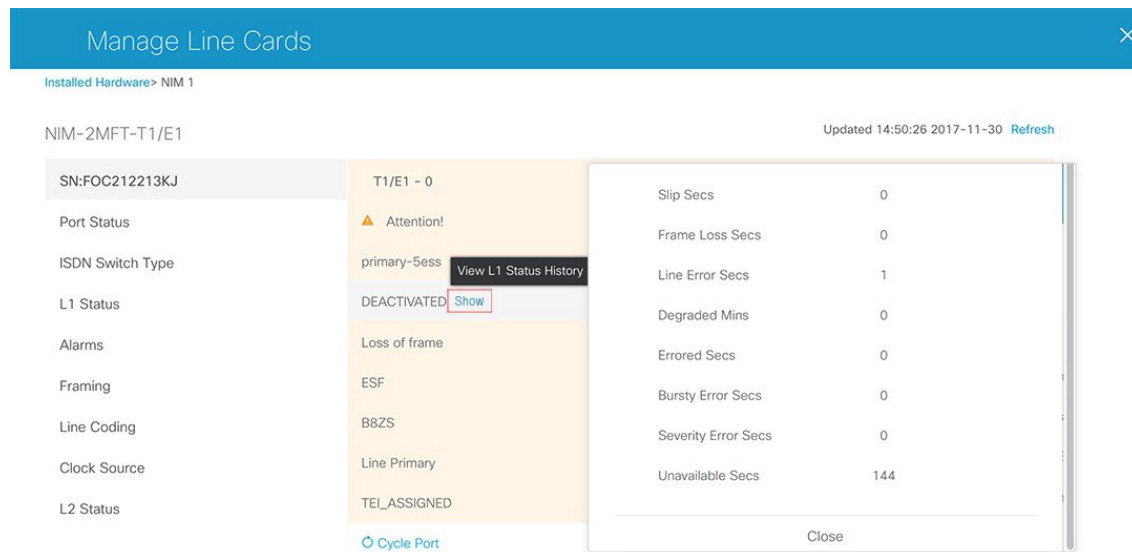


The screenshot shows the 'Manage Line Cards' interface. At the top, there is a blue header with the title 'Manage Line Cards' and a close button. Below the header, the breadcrumb 'Installed Hardware > NIM 1' is visible. The main content area displays details for a line card with SN: FOC212213KJ. The card is identified as 'NIM-2MFT-T1/E1' and was last updated on 15:15:48 2017-11-30. The interface shows two ports, T1/E1 - 0 and T1/E1 - 1. Both ports have a status of 'Attention!' indicated by a yellow warning triangle icon. The 'L1 Status' for both ports is 'DEACTIVATED', with a 'Show' link next to it. Other configuration details include ISDN Switch Type (primary-5ess), Alarms (Loss of frame), Framing (ESF), Line Coding (B8ZS), Clock Source (Line Primary), and L2 Status (TEL_ASSIGNED). At the bottom of each port's configuration, there is a 'Cycle Port' button.

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

393714

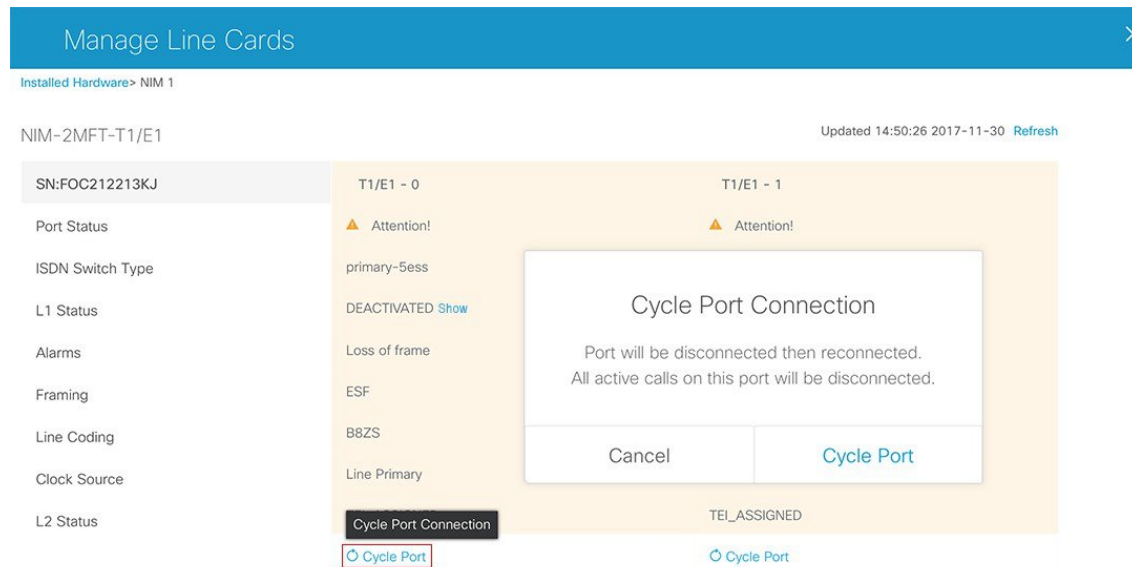
Figure 3: L1 Status



393715

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



393716

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.

Procedure

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.

Procedure

- 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.

Procedure

- 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

Procedure

- 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

Procedure

- 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

Procedure

- 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.
-

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.

Procedure

-
- 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.

Field	Description
Drop Through Destination	<p>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:</p> <ul style="list-style-type: none"> • 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. <p>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.</p> <ul style="list-style-type: none"> • Disconnect Call—The call gets disconnected.
Audio Prompt (Welcome Message)	<p>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.</p> <p>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.</p> <p>The system default message is a silent recording. You need to replace the message with a custom recording.</p>
Audio Prompt (Open Message)	<p>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.</p> <p>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.</p> <p>The system default message is a silent recording. You need to replace the message with a custom recording.</p>

Field	Description
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)	<p>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.</p> <p>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.</p> <p>The system default message is a silent recording. You need to replace the message with a custom recording.</p>
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:

Procedure

-
- 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 **PromptAdministrators**. (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.

Procedure

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.

Procedure

- 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

Procedure

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

Procedure

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.

Procedure

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.

Procedure

- 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**.
-

Overview of Users

Cisco partner adds users to a BE4000 site during initial site deployment. A Cisco partner and customer administrator can add and manage users post site deployment. You can view all the existing users' information under **Manage Site > User Management**.

Add a User

Before you begin

Ensure that you have:

- First name, last name, display name, extension, and email address of the user

Procedure

-
- 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 **User Management**. The User Management page is displayed.
- Step 4** Click **Add User**.
- Step 5** Enter the information in the fields. For field descriptions, see [User Management - Field Descriptions, on page 38](#).
- 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**.
-

What to do next

[Provision the Phone](#) for the user.

User Management - Field Descriptions

Field	Description
Extension	<p>Extension number that is associated with the end user.</p> <p>Note</p> <ul style="list-style-type: none"> • Enter a minimum of 3 digits for an extension. Maximum number of digits in an extension can go up to 5. • You cannot create an extension with leading zero. • The first digit of the extension cannot be the same as the digit used for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.
Display Name	Display name for the end user. The name entered here is displayed on the called phone device when a call is received from this extension. You can enter a maximum of 30 characters.
First Name	First name of the end user.
Last Name	Last Name of the end user.
Email Address	Email address of the end user. Cisco Cisco Business Edition Selfcare Portal registration link is sent to this email address.

Field	Description
Enable Voicemail	Voicemail to email feature is enabled for this extension. Any voicemail coming to this extension is sent as an email attachment to the registered email address.
Enable VM to Email	Enables the end user to receive voicemail as an email attachment. Note Enable VM to Email check box is visible only when the Enable Voicemail check box is checked.
Enable MWI	Note This check box is not visible while modifying a user. Message waiting indicator (MWI) is a lamp that is lit on end user's phone to indicate a waiting voicemail message. Enables MWI functionality for an end user. Note Enable MWI check box is available only when the Enable Voicemail check box is checked.
Hunt Group Login	Allows the extension to become dynamic member of a Hunt Group.

Table 2: Call Forward

Field	Description
Note	You cannot forward calls to your own extension. Enter an extension other than your own extension in the following fields.
All	All the incoming calls are forwarded to another number (such as extension, auto attendant). The following options are available in the drop-down list: <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.

Field	Description
Busy	<p>The incoming calls are forwarded to another extension, voicemail box, or an outside number, only when the extension is busy.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
Unregistered	<p>When the extension is unregistered to BE4000, the calls are forward to another extension, voicemail box, or an outside number. An extension can be in unregistered state when the phone device is unplugged or the network between the BE4000 and the extension is not functional.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
No-Answer	<p>The incoming calls are forwarded to to another number (such as extension, auto attendant).</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list. <p>Note If Single Number Reach (SNR) is enabled, the number that is configured for "Call Forward No-Answer" on the "Single Number Reach (SNR)" page takes precedence.</p>

Field	Description
No-Answer Timeout (3-60000 secs)	Time in seconds up to which a call rings on the extension when no one answers. After this time out period, the call gets forwarded to configured extension, voicemail box, or an outside number.

Table 3: Single Number Reach (SNR)

Field	Description
	Single Number Reach (SNR) provides end users the choice of answering an incoming call on their desk phone using a mobile phone (cellular network) or a Public Switched Telephone Network (PSTN) phone. On enabling SNR, an incoming call rings both on the desk phone and the mobile phone (or PSTN phone). End users can answer the call either on their desk phone or from the mobile phone (or PSTN phone) based on their convenience. An active call can be swapped between the desk phone and the mobile phone (PSTN phone) without disconnecting the call.
SNR Number	The number for Single Number Reach (SNR) functionality. Note When entering the SNR number, you must start with the prefix to dial an outside line, followed by Country Code, National Destination Code (Area Code), and Subscriber Number. For example, if 9 is the digit to dial an outside line, 1 is the country code, 555 is the area code, and 9999999 is the subscriber number, you must enter 915559999999.
Call Forward No-Answer	An incoming call is forwarded to the SNR number when the call is not answered on the desk phone.
Calling Number Local	Calling party number displayed on the configured mobile phone is replaced with the SNR extension number.
SNR Delay (0-10 secs)	Number of seconds up to which a phone rings before transferring the call to configured SNR number. Range = 0 to 10 seconds. Default = 1.
SNR Timeout (5-60 secs)	Number of seconds up to which phone rings after the configured SNR delay. When the timeout value is reached, incoming call display is stopped on the desk phone and the call gets forwarded to the configured SNR number. Range = 5 to 60 seconds. Default = 10.
Ring Stop	An incoming call stops ringing on the desk phone after the call is answered from a mobile phone and conversely.
Answer Too Soon (1-5 secs)	Number of seconds up to which Single Number Reach (SNR) calls are prevented from being diverted to the voicemail box of a mobile phone. Range = 1 to 5 seconds.

Table 4: Additional Settings

Field	Description
Not Register Number	Extension is not associated with an external proxy server.

Field	Description
Auto Answer	Extension automatically answers any inbound call.
Pickup Call	Answer an incoming call on the extensions belonging to any pick-up group. Press GPickup followed by * on the phone to answer the pickup group call.
Pickup Group	Associate the extension with a pick-up group. 1 is the default pickup group created by the BE4000.
Allow calls to blocked numbers	Allows extension to make calls that are blocked in the dial plan.
Night Service Bell	<p>The incoming calls that ring during the night service period on the extension sends an alert indication to all extensions that are marked to receive night service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the phone) and a visible display of the extension. The phone users retrieve the call by pressing the Pickup softkey.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
Forwarding Number	<p>The incoming calls are forwarded to another extension, voicemail box, or an outside number during night service hours.</p> <p>Note Only when the Night Service Bell check box is checked, the Forwarding Number text field appears.</p>
Shared Line	<p>One extension can be shared across multiple phone devices.</p> <p>Note If you assign an extension on more than one phone device using the Buttons page under Manage Site > Phones > Modify Phone, the Shared Line check box under Manage Site > Extensions > Modify Extensions > Additional Settings gets checked automatically and cannot be unchecked.</p>

Field	Description
Max Calls (2-16)	<p>Note This field is visible only when you check the Shared Line check box.</p> <p>Set the maximum number of calls that can be answered for the shared extension.</p> <p>Example: Consider that extension 1001 is shared on 3 phones and the maximum number of calls configured for the shared line is 2. At any time, there cannot be more than 2 active calls going on for 1001 on all the 3 phones.</p>

Table 5: Add Phone

Field	Description
Note	Every extension must be associated with a phone type for it to be available for provisioning through the Extension Assigner. Hence, we recommend you to add a phone while adding the extension. You can also add phone, under Manage Site > Phones .
Phone Type	List of supported phone models. By default, Cisco IP Phone 7841 is chosen.
COR	<ul style="list-style-type: none"> • Internal—Privileges to call numbers that are internal to the organization. • International—Privileges to call all numbers including international numbers. • Local—Privileges to call numbers within the same area code. • Local-Plus—Privileges to call numbers within the same area code with plus dialing. • National—Privileges to call numbers within the same nation code. • National-Plus—Privileges to call numbers within the same nation code with plus dialing.

Modify Email Address of a User

You can change only the email address of the user and retain all the other configurations related to the extension and phone as is.

Before you begin

Ensure that you have the new email address of the user ready

Procedure

-
- 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 **User Management**. The User Management page is displayed.
- Step 4** Click **Modify Email** from the Actions menu for the desired user.
- Step 5** Change the email address as required.
- Step 6** Click **Yes**.

The registration email is sent to the newly entered email address.

Resend Cisco Business Edition Selfcare Portal Registration Email to a User

If a user has lost the email that was sent for registering to the Cisco Business Edition Selfcare Portal, Cisco Partner or customer administrator can resend the registration email.

Procedure

- 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 **User Management**. The User Management page is displayed.
- Step 4** Click **Resend Registration Email** from the Actions menu for the desired user.
- Step 5** Click **OK**.

The registration email is sent to the user. User must re-register to log in to the Cisco Business Edition Selfcare Portal.

Resend Cisco Business Edition Selfcare Portal Registration Email to All Users

Procedure

- 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 **User Management**. The User Management page is displayed.
- Step 4** Choose **Resend Registration Email** from the **Bulk Edit** drop-down list.
- Step 5** Click **Resend All**.

The registration email is sent to all the users. Users must re-register to log in to Cisco Business Edition Selfcare Portal.

Reset Cisco Business Edition Selfcare Portal Password for a User

Before you begin

The user must be registered to the Cisco Business Edition Selfcare Portal.

Procedure

- 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 **User Management**. The User Management page is displayed.
- Step 4** Click **Reset Password** from the Actions menu for the desired user.
- Step 5** Click **OK**.

The current password is erased and password reset instruction email is sent. User must change the password to log in to Cisco Business Edition Selfcare Portal.

Reset Cisco Business Edition Selfcare Portal Password for All Users

Before you begin

The user must be registered to the Cisco Business Edition Selfcare Portal.

Procedure

- 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 **User Management**. The User Management page is displayed.
- Step 4** Choose **Reset Password** from the **Bulk Edit** drop-down list.
- Step 5** Click **Reset All**.

All the current passwords are erased and password reset instruction email is sent to all registered users. Users must change their passwords to log in to Cisco Business Edition Selfcare Portal.

Delete a User

BE4000 supports deleting a user. Deleting a user releases the extension and removes the phone settings from the system.

Procedure

- 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 **User Management**. The User Management page is displayed.
 - Step 4** Choose **Delete User** from the Actions menu for the desired user.
 - Step 5** Click **Delete**.
-

Extensions

An extension number is a unique number that is assigned to an employee in an organization. An employee can have more than one extension number.

Dial Plan for your organization defines the number of digits for an extension and the maximum number of extensions that can be configured at a site. Cisco partner configures the dial plan during the site creation. Contact Cisco partner if you need more details on dial plan configured for a site.

Add an Extension

While adding an extension, you can also configure call forwarding, single number reach, voicemail capabilities, and other calling features for an extension.

Procedure

- 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 **Extensions**. The Manage Extensions page is displayed.
 - Step 4** Click **Add Extension**.
 - Step 5** Enter the information in the fields. For field descriptions, refer [Extensions - Field Descriptions, on page 47](#)
 - 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**.
-

Extensions - Field Descriptions

Table 6: Add Extension

Field	Description
Extension	<p>Extension number that is associated with the end user.</p> <p>Note</p> <ul style="list-style-type: none"> • Enter a minimum of 3 digits for an extension. Maximum number of digits in an extension can go up to 5. • You cannot create an extension with leading zero. • The first digit of the extension cannot be the same as the digit used for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.
Display Name	<p>Display name for the end user. The name entered here is displayed on the called phone device when a call is received from this extension. You can enter a maximum of 30 characters.</p>
Extension Label	<p>Note This field appears only while modifying an extension.</p> <p>Extension label is the text that is associated with the extension. The text added for the extension label appears on the line button. By default, the extension label is taken as the combination of display name and the extension number. For example, if you are adding with display name as "John" and the extension as "3456", the default extension label is "John-3456".</p>
User Type	<ul style="list-style-type: none"> • User—An extension that is assigned to the end user. You must configure an email address associated with the end user. • Public—An extension that is assigned to a phone that is meant for general use by many end users. You need not configure an email address. For example, the extension assigned to a phone in the conference room.
Hunt Group Login	<p>Allows the extension to become dynamic member of a Hunt Group.</p>

Table 7: Create End User

Field	Description
First Name	<p>First name of the end user.</p>
Last Name	<p>Last Name of the end user.</p>
Email Address	<p>Email address of the end user. Cisco Cisco Business Edition Selfcare Portal registration link is sent to this email address.</p>
Enable Voicemail	<p>Voicemail to email feature is enabled for this extension. Any voicemail coming to this extension is sent as an email attachment to the registered email address.</p>

Field	Description
Enable VM to Email	<p>Note This check box is not available while modifying an extension.</p> <p>Enables the end user to receive voicemail as an email attachment.</p> <p>Note Enable VM to Email check box is visible only when the Enable Voicemail check box is checked.</p>
Enable MWI	<p>Message waiting indicator (MWI) is a lamp that is lit on end user's phone to indicate a waiting voicemail message. Enables MWI functionality for an end user.</p> <p>Note Enable MWI check box is available only when the Enable Voicemail check box is checked.</p>

Table 8: Single Number Reach (SNR)

Field	Description
	<p>Single Number Reach (SNR) provides end users the choice of answering an incoming call on their desk phone using a mobile phone (cellular network) or a Public Switched Telephone Network (PSTN) phone. On enabling SNR, an incoming call rings both on the desk phone and the mobile phone (or PSTN phone). End users can answer the call either on their desk phone or from the mobile phone (or PSTN phone) based on their convenience. An active call can be swapped between the desk phone and the mobile phone (PSTN phone) without disconnecting the call.</p>
SNR Number	<p>The number for Single Number Reach (SNR) functionality.</p> <p>Note When entering the SNR number, you must start with the prefix to dial an outside line, followed by Country Code, National Destination Code (Area Code), and Subscriber Number. For example, if 9 is the digit to dial an outside line, 1 is the country code, 555 is the area code, and 9999999 is the subscriber number, you must enter 915559999999.</p>
Call Forward No-Answer	An incoming call is forwarded to the SNR number when the call is not answered on the desk phone.
Calling Number Local	Calling party number displayed on the configured mobile phone is replaced with the SNR extension number.
SNR Delay (0-10 secs)	Number of seconds up to which a phone rings before transferring the call to configured SNR number. Range = 0 to 10 seconds. Default = 1.
SNR Timeout (5-60 secs)	Number of seconds up to which phone rings after the configured SNR delay. When the timeout value is reached, incoming call display is stopped on the desk phone and the call gets forwarded to the configured SNR number. Range = 5 to 60 seconds. Default = 10.
Ring Stop	An incoming call stops ringing on the desk phone after the call is answered from a mobile phone and conversely.

Field	Description
Answer Too Soon (1-5 secs)	Number of seconds up to which Single Number Reach (SNR) calls are prevented from being diverted to the voicemail box of a mobile phone. Range = 1 to 5 seconds.

Table 9: Call Forward

Field	Description
Note	You cannot forward calls to your own extension. Enter an extension other than your own extension in the following fields.
All	<p>All the incoming calls are forwarded to another number (such as extension, auto attendant).</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
Busy	<p>The incoming calls are forwarded to another extension, voicemail box, or an outside number, only when the extension is busy.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.

Field	Description
Unregistered	<p>When the extension is unregistered to BE4000, the calls are forward to another extension, voicemail box, or an outside number. An extension can be in unregistered state when the phone device is unplugged or the network between the BE4000 and the extension is not functional.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
No-Answer	<p>The incoming calls are forwarded to to another number (such as extension, auto attendant).</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list. <p>Note If Single Number Reach (SNR) is enabled, the number that is configured for "Call Forward No-Answer" on the "Single Number Reach (SNR)" page takes precedence.</p>
No-Answer Timeout (3-60000 secs)	Time in seconds up to which a call rings on the extension when no one answers. After this time out period, the call gets forwarded to configured extension, voicemail box, or an outside number.

Table 10: Additional Settings

Field	Description
Not Register Number	Extension is not associated with an external proxy server.
Auto Answer	Extension automatically answers any inbound call.
Pickup Call	Answer an incoming call on the extensions belonging to any pick-up group. Press GPickup followed by * on the phone to answer the pickup group call.

Field	Description
Pickup Group	Associate the extension with a pick-up group. 1 is the default pickup group created by the BE4000.
Allow calls to blocked numbers	Allows extension to make calls that are blocked in the dial plan.
Night Service Bell	<p>The incoming calls that ring during the night service period on the extension sends an alert indication to all extensions that are marked to receive night service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the phone) and a visible display of the extension. The phone users retrieve the call by pressing the Pickup softkey.</p> <p>The following options are available in the drop-down list:</p> <ul style="list-style-type: none"> • No Action • Outside Number—Enter the number in the text field. • Extension—Choose the extension from the drop-down list for available extensions. • Voicemail • Auto Attendant—Choose the auto attendant option from the drop-down list.
Forwarding Number	<p>The incoming calls are forwarded to another extension, voicemail box, or an outside number during night service hours.</p> <p>Note Only when the Night Service Bell check box is checked, the Forwarding Number text field appears.</p>
Shared Line	<p>One extension can be shared across multiple phone devices.</p> <p>Note If you assign an extension on more than one phone device using the Buttons page under Manage Site > Phones > Modify Phone, the Shared Line check box under Manage Site > Extensions > Modify Extensions > Additional Settings gets checked automatically and cannot be unchecked.</p>
Max Calls (2-16)	<p>Note This field is visible only when you check the Shared Line check box.</p> <p>Set the maximum number of calls that can be answered for the shared extension.</p> <p>Example: Consider that extension 1001 is shared on 3 phones and the maximum number of calls configured for the shared line is 2. At any time, there cannot be more than 2 active calls going on for 1001 on all the 3 phones.</p>

Table 11: Add Phone

Field	Description
Note	Every extension must be associated with a phone type for it to be available for provisioning through the Extension Assigner. Hence, we recommend you to add a phone while adding the extension. You can also add phone, under Manage Site > Phones .
Phone Type	List of supported phone models. By default, Cisco IP Phone 7841 is chosen.
COR	<ul style="list-style-type: none"> • Internal—Privileges to call numbers that are internal to the organization. • International—Privileges to call all numbers including international numbers. • Local—Privileges to call numbers within the same area code. • Local-Plus—Privileges to call numbers within the same area code with plus dialing. • National—Privileges to call numbers within the same nation code. • National-Plus—Privileges to call numbers within the same nation code with plus dialing.

Add Extensions in Bulk

Post BE4000 site deployment, you can add extensions (including the phone types) in bulk. The total number of extensions cannot exceed 200 at any time. So, the number of extensions that can be added post site deployment depends on the number of existing extensions.

Before you begin

Ensure that the existing number of extensions are less than 200.

Procedure

-
- Step 1** Log in to the BE4000 portal.
 - Step 2** Click **Manage Site** in the "Actions" column corresponding to the desired site.
 - Step 3** Click **Extensions**.
 - Step 4** Click **Import Stations** on the "Manage Extensions" page.

You can enter the user details in two ways:

- Click “Add Row” and add the user details one by one.
- Download the template that is provided on the screen. Fill in the details and upload all users' details in one go.

For field descriptions, refer [Import Stations - Field Descriptions, on page 53](#).

Step 5 Click **Yes** to apply your changes.

Import Stations - Field Descriptions

Field	Description
Type	<p>Choose a type of user:</p> <ul style="list-style-type: none"> • User—An extension assigned to the user. You must configure an email address associated with the user. • Public—An extension assigned to a phone that is meant for general use by many users. You need not configure an email address. For example, the extension assigned to a phone in the conference room.
First Name	Enter the first name of the user.
Last Name	Enter the last name of the user.
Display Name	Enter the display name of the user. The name entered here is displayed on the phone along with the extension number. You can enter a maximum of 12 characters.
Email	<p>Enter the email address of the user. The top-level domain in the email address can contain up to six characters.</p> <p>Note Email address must not be more than 32 characters in length. Only letters, numbers, and the characters underscore (_), dot (.), and dash (-) are allowed in the user ID portion of the email address. Do not use spaces in the email address.</p>
Extension	<p>Enter the extension number assigned to the user.</p> <p>Note</p> <ul style="list-style-type: none"> • Enter a minimum of 3 digits for an extension. Maximum number of digits in an extension can go up to 5. • You cannot create an extension with leading zero. • The first digit of the extension cannot be the same as the digit used for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.
Phone Type	Choose the phone model associated with the extension. For the list of supported phone models, refer to “Supported Phones” section in the Cisco Business Edition 4000 Release Notes .

Field	Description
COR	Choose the Class of Restriction (COR) for the extension. COR allows you to choose one of the calling privileges: <ul style="list-style-type: none"> • internal • local • local-plus • national • national-plus • international
Voicemail	Enable or disable voicemail functionality.
SNR	Enter the Single Number Reach (SNR) number for an extension. SNR allows you to answer the incoming calls on the desk phone or from a mobile phone. You can also swap active calls on a desk phone or at a remote destination without disconnecting the call. You should include the area code and any additional digits that are required to obtain an outside line prefix to your destination number. Example —If 9 is the digit to dial outside line, 1 is the country code, 555 is the area code, and 9999999 is the subscriber number, you must enter 915559999999.
Delete	Deletes an entry.
Replace this list	Replace an exiting list with an entirely new list.
Add Row	Add more rows to populate the stations list.
Download Template	Allows you to download a customized template. Template should be of .csv format.

Modify an Extension

If there is any change in an employee's phone system or calling features, you can modify the extension.

Procedure

-
- 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 **Extensions**. The Manage Extensions page is displayed.
 - Step 4** Click **Modify Extension** from the Actions menu for the desired Extension.
 - Step 5** Modify the information as desired.

Note You can modify all field information except for the Extension.

Step 6 Click **Yes**.

Shared Line

The concept of sharing one extension across multiple phone devices is called Shared Line. When there is an incoming call to the extension, all the phones that have the extension shared, rings at a time. You can answer the call from any preferred phone.

Example: The extension of the Dr. John is shared with the personal assistant, Ms. Nancy. The incoming calls ring on both the Dr. John's phone and Ms. Nancy's phone. Thus, either Dr. John or Ms. Nancy can answer the incoming call.

To enable shared line, check the **Shared Line** check box on the **Additional Settings** page under **Manage Site > Extensions > Modify Extensions**. Then, navigate to **Manage Site > Phones > Modify Phone > Buttons** page corresponding to the phones on which you want the extension to be shared, and select the extension.



Note You cannot uncheck the **Shared Line** check box for an extension that is shared with another phone. Before you uncheck the **Shared Line** check box, remove the extension from the phone that shares the line.

You can set the maximum number of active calls for a shared line in the **Max Calls (2-16)** field on the **Additional Settings** page under **Manage Site > Extensions > Modify Extensions**.



Note SIP shared line allows multiple SIP phones to share a common directory number. For BE4000, you can configure a maximum of 16 phones in one shared line.

Example: Consider that extension 1001 is shared on 3 phones and the maximum number of calls configured for the shared line is 2. At any time, there cannot be more than 2 active calls going on for 1001 on all the 3 phones.



Note If you assign an extension on more than one phone device using the **Buttons** page under **Manage Site > Phones > Modify Phone**, the **Shared Line** check box under **Manage Site > Extensions > Modify Extensions > Additional Settings** gets checked automatically and cannot be unchecked.

Phones

You can add new phones and provision the phones using extension assigner. A maximum of 200 phones can be configured at your site.

Adding a phone at your site involves the following steps:

1. [Add a New Phone](#) , on page 56
2. [Provision the Phone Using Extension Assigner](#), on page 58



Note The line buttons on the phone (including KEM) get configured sequentially. You cannot skip any button or organize the buttons as you desire.

Add a New Phone

You can add a maximum of 200 phones.

Procedure

- 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 **Phones**.
- Step 4** Click **Add Phone**.
- Step 5** Enter the information in the fields. For field description, refer [Phones - Field Descriptions, on page 56](#)
- 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**.

Phones - Field Descriptions

Table 12: Basic Info

Field	Description
Phone Type	List of supported phone models.
Extension	Extension number that is assigned to the end user.
COR	Class of Restriction (COR). Enables the restrictions on the type of calls that are placed from the phone.
Device Name	Name for the phone. You can enter a minimum of 1 and a maximum of 12 characters.
Busy Trigger per Button (1-50)	The maximum number of calls that are allowed on an octo-line directory number before activating Call Forward Busy or a busy tone.
Session Transport	Transport protocol. <ul style="list-style-type: none"> • TCP • UDP By default, TCP is selected.
Video	Video calling is enabled on the phone.

Field	Description
MAC Address	<p>Note This field is applicable only for Cisco ATA 190 Analog Telephone Adapter.</p> <p>The MAC address of ATA-190.</p>

Table 13: Advanced Options

Field	Descriptions
Block Conference Pattern	Phone is prevented from initiating a conference to external numbers.
Overlap signal	Called digits are sent one by one as they are received from the calling device.
Privacy	<p>Blocks other users in the same shared line from viewing or handling any activities on the shared line.</p> <p>Note Privacy is supported for shared-lines only.</p>
Privacy-button	<p>Assigns a line button as "Privacy" on the phone to toggle the Privacy mode on and off.</p> <p>Note</p> <ul style="list-style-type: none"> • Privacy Button can be enabled only when Privacy is enabled. • Before enabling the Privacy button, ensure that at least one line button is available for assigning the privacy button. If no line key is available on the phone, then the Privacy line button is assigned and the user does not get an option to toggle on and off privacy.
Block Transfer-pattern	<p>Transfer restrictions are applied when a call transfer or conference is initiated toward external parties such as a PSTN trunk, SIP trunk.</p> <p>Note Transfer to local extensions are exempted from this restriction.</p>
ATA-IVR-Password	Password to access Interactive Voice Response (IVR) and change the default phone settings on Analog Telephone Adaptors.
Transfer Max-Length	Maximum number of digits that are dialed for a call transfer.
Local Directory	Local directory service on the phones is available. By default, Local Directory check box is checked.
My Phone Apps	Local services on a phone's My Phone Apps interface is made available. By default, My Phone Apps check box is checked.

Table 14: Buttons

Field	Description
	You can assign an extension, speed dial, or BLF speed dial for the buttons on the phone. Click the Actions column corresponding to the desired button to edit or add a task for button.

Field	Description
Type	<ul style="list-style-type: none"> • Extension • Intercom • Speed Dial • BLF Speed Dial <p>Note</p> <ul style="list-style-type: none"> • When an extension is shared between multiple phones, ensure that you check the Shared Line check box on the Modify Extensions > Advanced Settings page. • The buttons on the phone are configured based on the following preference order: <ol style="list-style-type: none"> 1. Extension 2. Intercom 3. Speed Dial 4. BLF Speed Dial
Number	Number to configure as an extension, speed dial, or BLF speed dial.
Name	Name that is displayed for the button on the phone.

Provision the Phone Using Extension Assigner

Extension Assigner allows you to assign an extension to a new or replacement IP phone.

Before you begin

- Note down the extension that needs to be configured for the user from the Cisco Business Edition Selfcare Portal (Manage Site > Extension)
- BE4000 is accessible and your site is in the Online state
- If you are provisioning Cisco Wireless IP Phone 8821, ensure that you have configured the phone with the wireless LAN. For more information, see [Add Cisco Wireless IP Phone 8821 to the Wireless LAN](#).

Procedure

-
- Step 1** Connect the phone to the network. The phone gets assigned with a temporary extension.
- Step 2** Dial the Extension Assigner directory number **70000**; When prompted for password, enter **1234**.
- Step 3** Enter the pound (#) key.
- Step 4** Enter the permanent extension that has been configured on the portal for this user followed by the pound (#) key.

Step 5 Enter **1** followed by the pound (#) key to assign the extension.

Troubleshooting Phone Provisioning

If the provisioned phone is in the unregistered mode, check the following:

- Whether the phone is connected to power and is operational
- Whether the phone has an IP address and the TFTP address. If the MAC address is correct, phones must have lost its connection to BE4000. Unplug and plug the phones to the BE4000 network again
- If the phone is unregistered even if the IP address and TFTP address are configured correctly, check whether BE4000 is reachable from the phone. If the phone is faulty, you have to replace the phone
- Verify whether your site in Cisco Business Edition Selfcare Portal is in the Online state
- Try resetting the phone. On the phone display, choose **Applications > Admin Settings > Reset Settings > All**.

Monitor Phones That Are Unregistered

When the phone attempts to register to BE4000, the phone displays a status message (either registered or unregistered). You can view the Offline or Not Registered phones on the Manage Phones page. Unregistered phones are monitored in real time.

Serial Number	Phones	Last Change
--	7 (4) Offline	1d
--	--	--

Access Admin Settings on Phone

Procedure

- Step 1** On the phone, press **Applications**.
- Step 2** Select **Admin Settings**.
- Step 3** Enter the password:
- If you are accessing Admin Settings before provisioning the phone, enter **adg234**.
 - If you are accessing Admin Settings after provisioning the phone, enter **ptwmjg**.
- Step 4** Press **Sign-In**.

View Phone Firmware Version and IP Address

The columns containing IP address and phone firmware version are hidden by default. Perform the following steps to view the columns.

Procedure

-
- Step 1** Log in to the BE4000 portal.
 - Step 2** Click **Manage Site > Phones** from Actions menu for the desired site on the dashboard.
 - Step 3** Click **Edit Columns**.
 - Step 4** Check the **IP Address** and **Version** check boxes.
 - Step 5** Click **Update**.

Two new columns are displayed with the IP address and phone firmware version details.

Note In some scenarios, if the phone firmware information that is extracted from the BE4000 appliance contains special characters, the firmware version is displayed as "--".

Replace a Phone



Note Do not delete the phone from the Portal. Deleting the phone from the portal removes all user-specific customizations such as speed dials, single number reach, call forward.

You can replace a faulty phone device or upgrade to a new phone model. Replacing the phone involves the following steps:

1. Unplug the existing or faulty phone from the network.
2. Connect the new phone to the network.
3. Provision the new phone using Extension Assigner. For detailed steps, refer [Provision the Phone Using Extension Assigner, on page 58](#).

Reassign Phone to a Different User

When there is a change in the user who is using the phone device, you can reassign the same phone device to the new user.

Procedure

-
- Step 1** Delete the extension that was previously associated with the phone from the BE4000 Partner Portal. Go to Manage Sites > Extensions > Manage Extensions and click **Delete** from the Actions menu corresponding to the desired extension.

- Step 2** Add an extension for the new user who uses the phone device in BE4000 Partner Portal. For details on adding an extension, refer [Add an Extension , on page 46](#).
- Step 3** Use Extension Assigner to provision the phone with the extension associated with the new user. For details on Extension Assigner, refer [Provision the Phone Using Extension Assigner, on page 58](#).
-

Reset Phone

Cisco Unified IP phones must be rebooted (reset) after making any configuration changes so that the changes take effect. Configurations for phones are downloaded afresh when the phone is rebooted.

Reset Phone from the BE4000 Portal

Before you begin

Phone must be in "Registered" state.

Procedure

- Step 1** Log in to the BE4000 portal.
- Step 2** Click **Manage Site > Phones** for the desired site from the Actions menu.
- Step 3** Click **Reset** from "Action" menu on the **Manage Phones** page.
- Step 4** Click **Yes**. The phones go offline temporarily for a few seconds and will not be able to make or receive calls until they come back online.
-

Reset All Phones from the BE4000 Portal

Before you begin

Phones must be in "Registered" state.

Procedure

- Step 1** Log in to the BE4000 portal.
- Step 2** Click **Manage Site > Phones** for the desired site from the "Actions" menu.
- Step 3** Click **Bulk Action**.
- Step 4** Choose **Reset all phones**.
- Step 5** Click **Reset**. The phones go offline temporarily for a few seconds and will not be able to make or receive calls until they come back online.
-

Reset Phone Using Phone Softkeys

Before you begin

Phone must be in "Registered" state.

Procedure

-
- Step 1** Press **Applications**.
- Step 2** Select **My Phone Apps > Reset Phone**. The phone gets unregistered and registered again.
-

Add a Key Expansion Module

Key Expansion Modules add extra buttons to the Cisco IP Phone 8800 Series phone. The buttons can be configured as Extension, Speed Dial, and BLF Speed Dial.



Note Key Expansion Module cannot be added during initial site deployment in the Setup Assistant. You can add only after the site is successfully deployed.

To add a Key Expansion Module to your Cisco IP Phone 8800 Series, perform the following tasks:

Table 15: Add a Key Expansion Module—Task Flow

Step	Task	Purpose	Prerequisite	Reference
Step 1	Add a Key Expansion Module on the BE4000 Portal	Add a Key Expansion Module for the Cisco IP Phone 8800 Series on the BE4000 portal.	Site is in the Online state	Add a Key Expansion Module on the BE4000 Portal, on page 63
Step 2	(Task performed on the site) Connect a Key Expansion Module to a Cisco IP Phone	Connect the Key Expansion Module to the Cisco IP Phone 8800 Series phone. After successfully connected, Key Expansion Module pulls the configurations from the portal.	Key Expansion Module is adding on the BE4000 portal	Connect a Key Expansion Module to a Cisco IP Phone , on page 63

Add a Key Expansion Module on the BE4000 Portal

Procedure

- 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 **Phones**.
 - Step 4** Click **Modify Phone** from the Actions menu for the desired phone.
 - Step 5** Choose the desired number of KEMs from the **Add-on Module** drop-down list on the **Basic Info** page.
 - Step 6** Click the arrows at the bottom of the screen to move to next screens.
 - Step 7** Choose an action for each button from the **Type** drop-down list on the **Buttons** page. Enter the details based on your selection.
 - Step 8** Click **Yes**.
-

Connect a Key Expansion Module to a Cisco IP Phone

Refer to [Cisco IP Phone Key Expansion Module](#) for detailed steps on connecting a Key Expansion Module to a Cisco IP Phone.

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

Procedure

-
- 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 [Meet-Me Conference Dependency on Codecs, on page 64](#).
- 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.

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.

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**.

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 16: 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.
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.

Feature	ELK Enabled	ELK Disabled
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

Procedure

- 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**.

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**.

Personal Voicemail Box

Personal voicemail box is assigned to a specific user and is accessible only by that user. When a caller leaves a message in the voicemail box, the message waiting indicator (MWI) light turns on.

You can specify the maximum quota for the voicemail box for every user using the portal. The default voicemail box size is 10 minutes with maximum message size of 2 minutes.

Set Up Personal Voicemail Box for a User

Procedure

-
- Step 1** Click **Manage Site** from Actions menu for the desired site on the dashboard.
 - Step 2** Click **Personal Voicemail Box**.
 - Step 3** Click **Add Voicemail**.
 - Step 4** Enter the information in the fields. For field descriptions, refer [Personal Voicemail Box - Field Descriptions](#), on page 69.
 - 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**.
-

Personal Voicemail Box - Field Descriptions

Table 17: User Profile

Field	Description
First Name	First name of the end user.
Last Name	Last name of the end user.
Display Name	Display name for the end user. The name entered here is displayed on called phone device when a call is received from this extension.
Pin(Optional) Confirm Pin	Personal Identification Number (PIN). To manage your voicemail box from the phone, enter this PIN.
Primary Extension	Primary extension of the end user.
E.164 Number	Primary extension of the end user in E.164 format.

Table 18: Voicemail Box


Field	Description
Description	Description for the personal voicemail box.

Field	Description
Mailbox Size in sec	Size of the voicemail box in seconds.
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	<ul style="list-style-type: none"> • 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. <p>Default—Yes.</p>
Greeting Type	<ul style="list-style-type: none"> • 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 <p>Default—Standard.</p>
Recording Type	<ul style="list-style-type: none"> • None—Do not use a recording • System-default—Use the system default recording • User-Recording—Use the user recording <p>Default—User-Recording.</p>
Allow caller to dial 0	Callers can dial zero (0) from their phones to reach the system operator.
Enable VM to Email	Enables the end user to receive voicemail as an email attachment.
Enable MWI	<p>Message waiting indicator (MWI) is a lamp that is lit on end user's phone to indicate a waiting voicemail message. Enables MWI functionality for an end user.</p> <p>Note Enable MWI check box is available only when the Enable Voicemail check box is checked.</p>

Table 19: Groups

Field	Description
Add Groups	Groups to which the user belongs. Search for a group and click Add .

Access Personal Voicemail Box

Press **Messages**  on the phone and follow the voice prompts. To check messages for a specific line, press the line button first.

Set Up Personal Voicemail Box for a User Without an Assigned Phone

Procedure

-
- Step 1** Created an extension for the user. For information on how to create an extension, refer [Add an Extension](#) , on page 46.
- Step 2** Create a Personal Voicemail Box for the user. Ensure that you specify PIN on the **User Profile** page. For information on how to create a voicemail box, refer [Set Up Personal Voicemail Box for a User](#), on page 69.
- Note** If you do not enter PIN, the user cannot configure the voicemail box.
- Step 3** Communicate the PIN to the user.
-

What to do next

Users can access the configured voicemail box from any phone. For information on how to access the configured voicemail box, refer [Access Personal Voicemail Box-Users Without an Assigned Phone](#), on page 71.


Access Personal Voicemail Box-Users Without an Assigned Phone

Before you begin

Collect the following information from your Customer Administrator:

- Extension
- PIN configured while creating the Personal Voicemail Box

Procedure

-
- Step 1** Press **Messages**  on the phone.
You are prompted to enter the password.
- Step 2** Ignore the prompt and press *.

You are prompted to enter your ID.

Step 3 Enter your extension and press #.
You are prompted to enter the password.

Step 4 Enter the PIN that is configured in the portal while creating the personal voicemail box.

Step 5 Follow the voice prompts.

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.

Procedure

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”.

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.

Procedure

- 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**.
-

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.


Procedure

- 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.

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

Procedure

-
- 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 [Group Voicemail Box - Field Descriptions, on page 75](#).
- 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 20: 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 21: 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.

Field	Description
Add Members	Users who belong to the group. Note You must have at least one user as member of the group.

Table 22: 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	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • None—Do not use a recording • System-default—Use the system default recording • User-Recording—Use the user recording
Greeting Type	<ul style="list-style-type: none"> • 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

Procedure

- 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

Procedure

- 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**.
-

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 queuing. 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

Procedure

- 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 79](#).
- 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	<p>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.</p> <p>Note Only the following characters are allowed: a-z, A-Z, 0-9, space, hyphen(-), and underscore (_)</p>

Field	Description
Pilot Number	<p>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:</p> <ul style="list-style-type: none"> • 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 <p>Note</p> <ul style="list-style-type: none"> • 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	<p>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.</p>
Show Member Directory	<p>From the list of extensions that display, select which extensions must be included in the hunt list.</p> <p>Click Show Member Directory, to select the list of extensions for the hunt list. Check the respective member's name and click OK.</p>
Allow dynamic members	<p>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.</p>

Field	Description
Max dynamic members	<p>Note This field is visible only when "Allow dynamic members" is checked.</p> <p>Number of dynamic members allowed for the hunt group. A minimum of one dynamic member needs to be added.</p> <p>Note</p> <ul style="list-style-type: none"> • 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.
Hunt Method	<p>Select how BE4000 distributes the calls to members of the hunt list based on one of the following hunt methods:</p> <ul style="list-style-type: none"> • 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. <p>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.</p>

Field	Description
When No Member is Available	<p>If no members of the hunt list are available to answer a call, you can choose to perform one of the following:</p> <ul style="list-style-type: none"> • 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. <p>Note</p> <ul style="list-style-type: none"> • 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. <ul style="list-style-type: none"> • 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)	<p>Note This field is visible only when Enable Call Queuing check box is checked.</p> <p>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.</p>
Max Waiting Time (20-3600 Sec)	<p>Note This field is visible only when Enable Call Queuing check box is checked.</p> <p>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.</p>

Field	Description
Save	Save your hunt group configuration settings.

Add Users to Hunt Groups

Procedure

-
- 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**.
-

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 23: 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 84
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 85
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 85

Activate Night Service

Procedure

-
- 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

Procedure

- 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 [Designate an Extension to Receive Night Service Calls, on page 85](#).
- Step 6** Click **Yes** to save the changes.
-

Designate an Extension to Receive Night Service Calls

Procedure

- 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.
-

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

Procedure

- 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 87](#)), 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 24: 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)	<ul style="list-style-type: none"> • 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.

Procedure

-
- 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 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](#).

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.

Procedure

- 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 89](#).
- 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.

Field	Description
Save	Creates a paging group or saves the changes that are made to an existing paging group.

Modify a Paging Group

Procedure

-
- 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 89](#).
 - Step 5** Click **Save**.
-

Delete a Paging Group

Procedure

-
- 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**.
-

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](#).

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

Procedure

- 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 91.

Procedure

- 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 91.

Procedure

- 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 91.

Procedure

- 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 91.

Procedure

- 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 queuing 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 96](#).

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 91.

Procedure

- 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

Procedure

- 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**.
-

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

Procedure

- 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

Procedure

- 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**.
-

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 calendar 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 calendar 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

Procedure

- 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 calendar 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.

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](#) , on page 46.
- Added Cisco ATA Analog Telephone Adapter in the BE4000 portal. For detailed information on, see [Add a New Phone](#) , on page 56.
- Phone devices ready
- TFTP Address
- IP Address and subnet mask for Cisco ATA Analog Telephone Adapter

Procedure

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.

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

Procedure

-
- 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

Procedure

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)
-

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.

Procedure

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.

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.

Procedure

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**.

Configure Voicemail to Email

BE4000 provides you the “Voicemail to Email” functionality, wherein the voicemail that is received by your extension is sent as an email attachment to your registered email address. To enable this feature, you must configure the SMTP server in the BE4000 portal.

To enable the voicemail to email functionality, configure the IP address or the domain name of SMTP server. You can set the “From” address for sending emails containing voicemail as an email attachment. Ensure that the SMTP server allows receiving emails from the set email address.



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.

You can make the voicemail to email functionality secure by entering a 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.

You can upload custom SMTP server certificates after the initial site deployment under **Manage Site > Settings > System Settings**.



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.

After the site is deployed, you can update SMTP server address. You can also validate the SMTP server address before configuring. An email address must be entered for receiving the voicemail as email. Ensure that the SMTP server allows receiving emails from the entered email address. If the SMTP server address is valid, you see the success message displayed on the portal and an email sent to the configured email address.



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.

The following are the two options to configure the voicemail to email functionality:

- If the SMTP server is Office 365, follow the steps provided in [Configure Office 365, on page 106](#)

- If the customer has their own SMTP relay (other than Office 365), ensure that the proprietary certificate of the SMTP relay is trusted by BE4000. If the proprietary certificates are trusted by the BE4000, follow the steps provided in [Add SMTP Server Details](#) , on page 112

Configure Office 365

Table 25: Office 365 Configuration—Task Flow

Step	Task	Reference
1	Configure a Connector from your organizational server to the Office 365	Configure a Connector, on page 106
2	Add a domain to Office 365	Add a domain and users to Office 365, on page 111
3	Fetch MX FQDN Record	Fetch MX FQDN Record, on page 112
4	Add SMTP Server details in the BE4000 Portal	Add SMTP Server Details , on page 112

Configure a Connector

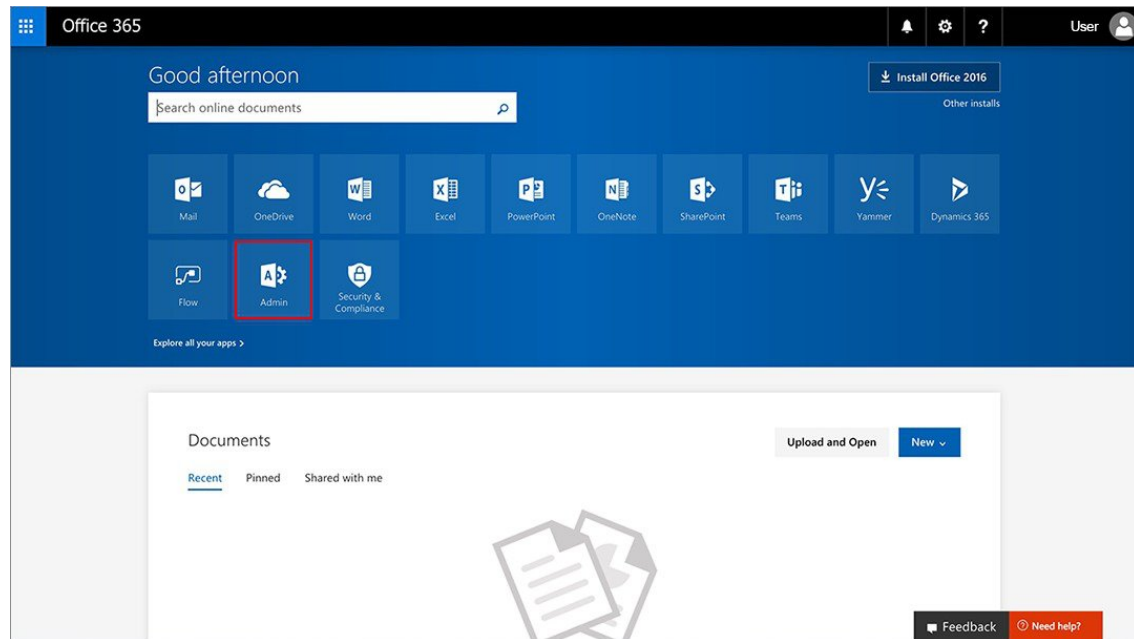
Before you begin

You should have admin privileges

Procedure

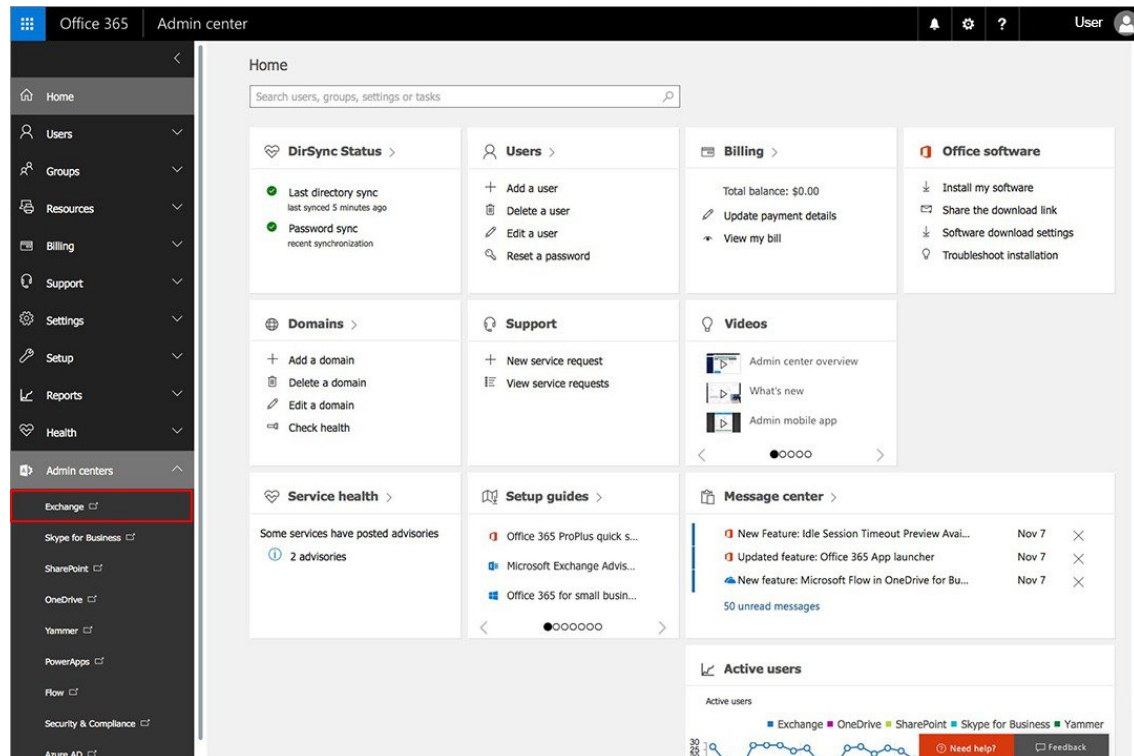
-
- Step 1** Log in to [Office 365 portal](#).
- Step 2** Click **Admin** from the dashboard. The **Admin Centre** page is displayed.

Figure 5: Office 365 Dashboard



Step 3 Click **Admin Centers > Exchange**. The **Exchange admin center** page is displayed.

Figure 6: Admin Center



Step 4 Click **mail flow > Connectors** on the Office 365 portal. Click + to add a new connector. Choose the following for **From** and **To** fields and click **Next**.

- **From:** Your organization's email server
- **To:** Office 365

Figure 7: Exchange Admin Center

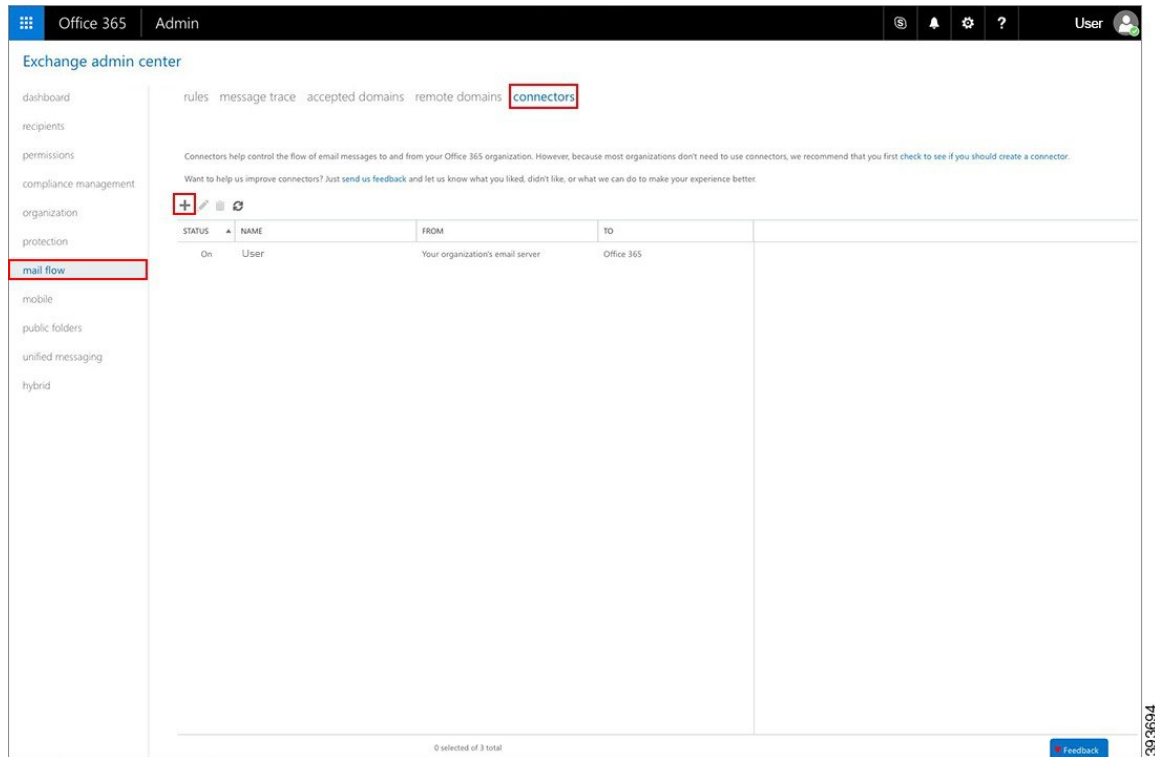
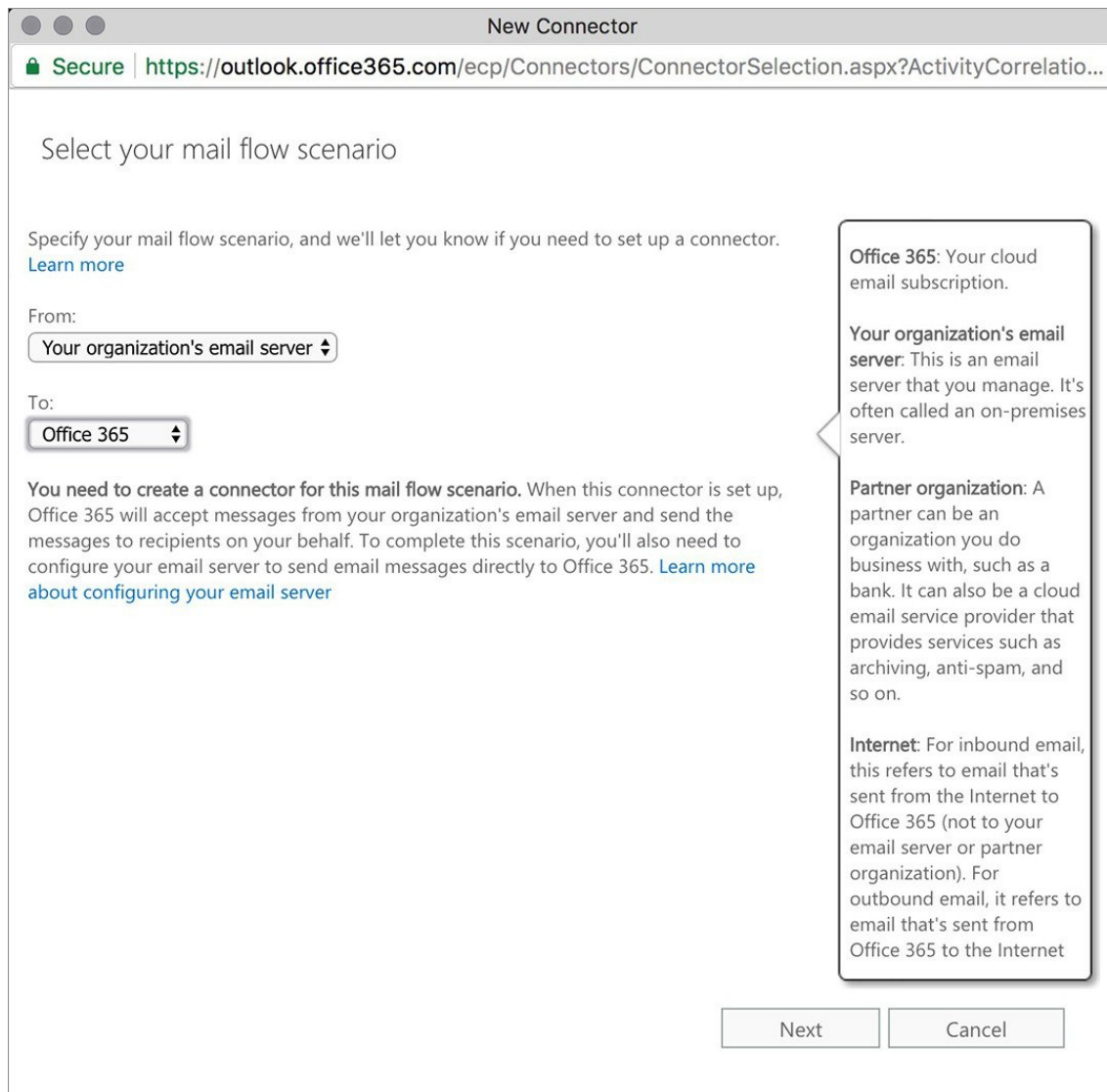


Figure 8: New Connector



Step 5 Provide a name for the connector and click **Next**.

Figure 9: Name for the Connector

New Connector

Secure | <https://outlook.office365.com/ecp/Connectors/InboundOnPremConnector.aspx>

New connector

This connector lets Office 365 accept email messages from your organization's email server (also called an on-premises server).

*Name:
BE4000

Description:
Connector for BE4000

What do you want to do after connector is saved?

- Turn it on
- Retain internal Exchange email headers (recommended)

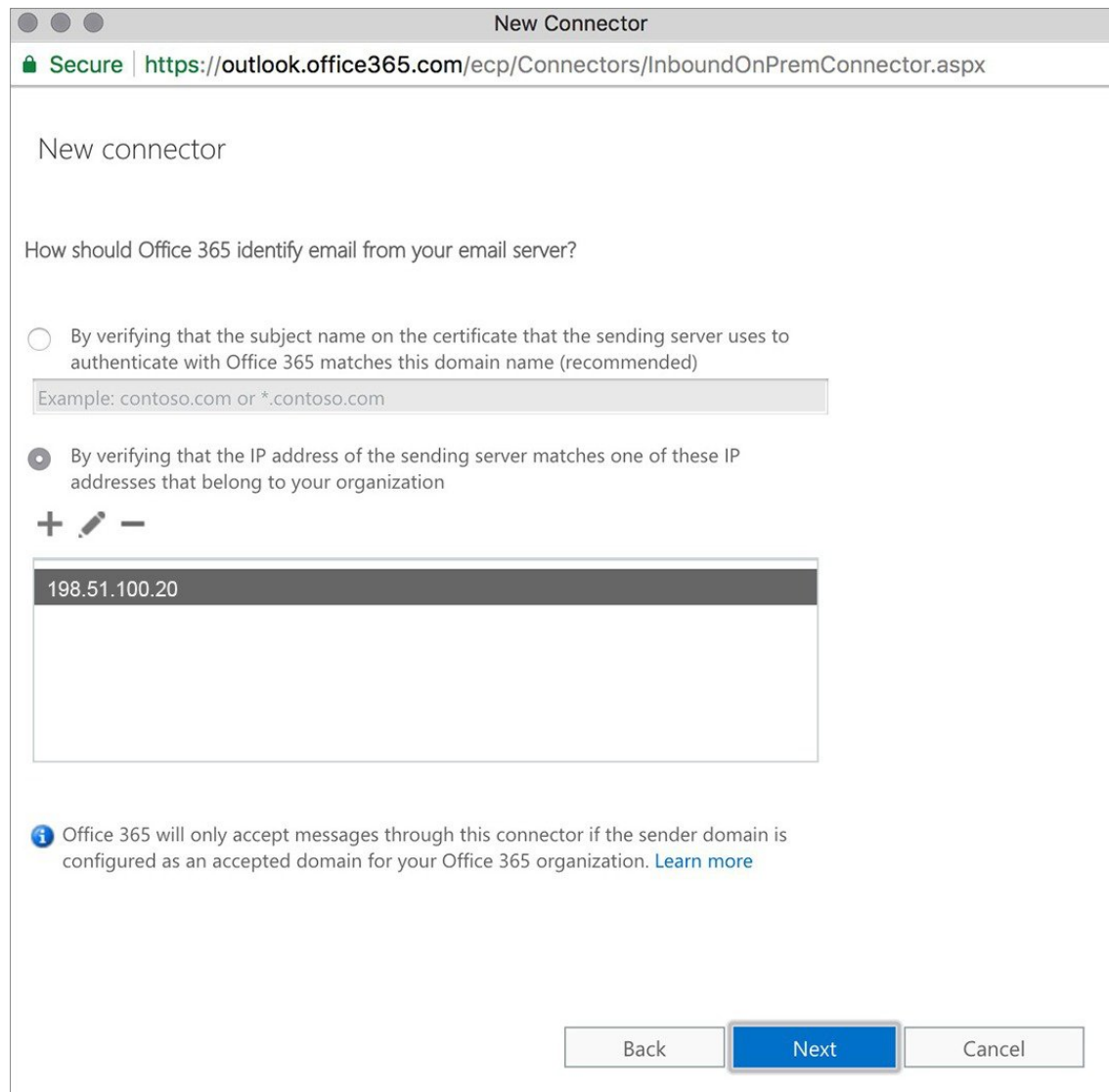
Next Cancel

Optionally include a description for this connector.

399606

Step 6 Click **By** verifying that the IP address of the sending server matches one of these IP addresses that belong to your organization and provide the public IP used for SMTP. Public IP used for SMTP can be got by checking with <http://whatsmyip.com>.

Figure 10: IP Address for Connector



Step 7 Click **Next**. Verify the settings and click **Save**.

Add a domain and users to Office 365

Procedure

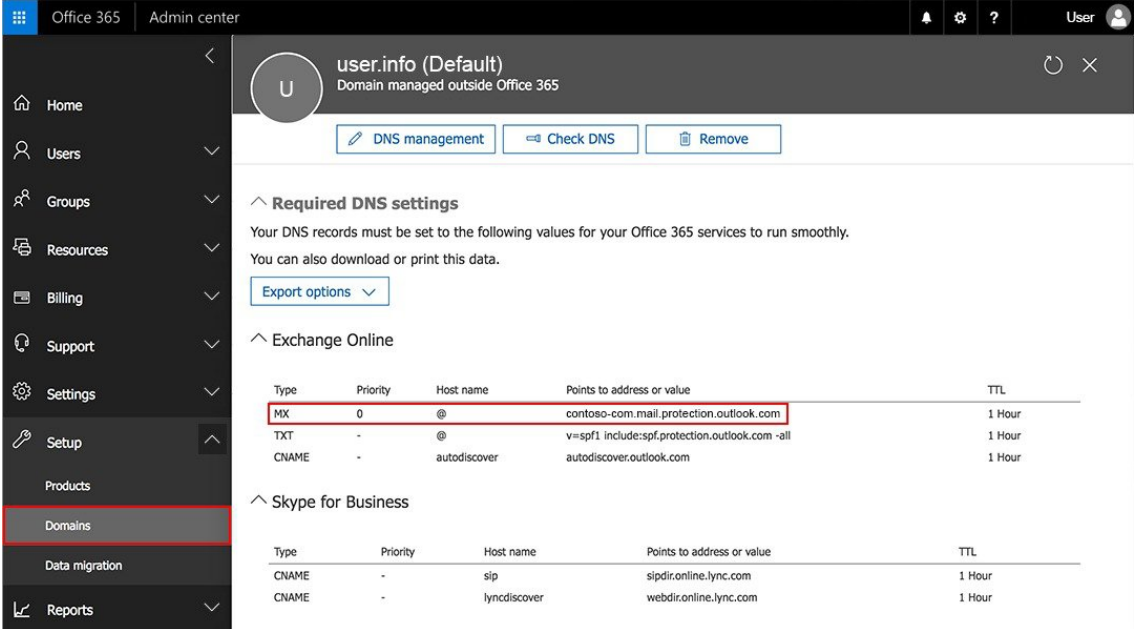
Log in to [Office 365 portal](#) and add a domain and users. For more information, refer [Add a domain and users](#).

Fetch MX FQDN Record

Procedure

- Step 1** Log in to [Office 365 portal](#).
- Step 2** Click **Setup > Domains**.
- Step 3** Copy and save the MX Record value from the “Exchange Online” section. For example, MX Record looks like *contoso-com.mail.protection.outlook.com*.

Figure 11: MX Record



The screenshot shows the Office 365 Admin center interface. The left navigation pane has 'Domains' selected. The main content area shows the 'Required DNS settings' section, which includes an 'Export options' button. Below this, the 'Exchange Online' section is expanded, displaying a table of DNS records. The MX record is highlighted with a red box.

Type	Priority	Host name	Points to address or value	TTL
MX	0	@	contoso-com.mail.protection.outlook.com	1 Hour
TXT	-	@	v=spf1 include:spf.protection.outlook.com -all	1 Hour
CNAME	-	autodiscover	autodiscover.outlook.com	1 Hour

Below the Exchange Online section, the 'Skype for Business' section is also expanded, showing a table of DNS records:

Type	Priority	Host name	Points to address or value	TTL
CNAME	-	sip	sjpdir.online.lync.com	1 Hour
CNAME	-	lyncdiscover	webdir.online.lync.com	1 Hour

Add SMTP Server Details

Before you begin

- For Office 365, have the MX FQDN Record ready. Refer [Fetch MX FQDN Record, on page 112](#)
- For SMTP Relay other than Office 365, have the SMTP server IP address or domain name ready

Procedure

- Step 1** Log in to the Cisco Business Edition 4000 Management Portal—<https://be4000.cisco.com>.
- Step 2** Click **Add Customer** from the dashboard and start the Setup Assistant.
- Step 3** Enter the required details on the initial pages and navigate to **LAN Connection** page.
- 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) (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.

Step 5 Enter the details in rest of the pages and click **Yes** to apply the configuration.

What to do next

Deploy the BE4000 appliance. Refer [Deploy a Customer Site](#) for detailed steps.

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.

Procedure

- 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 Send Calls to Voicemail Automatically

Procedure

- 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 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.

Procedure

- 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 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.

Procedure

-
- 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 26: Smart License Account Status, on page 116](#).

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

Table 26: Smart License Account Status

Smart Account Status	License Status	Description
Registered	Authorized	You have valid license.

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

Procedure

-
- 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

Procedure

- 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, [Restore the Backed-Up Site Configuration, on page 118](#) 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

Procedure

- 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.

Delete a Customer Site

- When you delete a site from the portal, all information related to that site is deleted from the portal. If this is the customer's only site, all customer information is deleted from the portal.
- If the BE4000 is deployed and online when a site is deleted, the BE4000 is reset to factory defaults and stops functioning. Phone calls can no longer be made or received.
- If a deployed site is offline when it is deleted, then BE4000 must be manually reset to factory default condition before it can connect to the portal again.

Procedure

Step 1 Select a site from the partner portal.

Step 2 Right-click and click **Delete Site**.

Step 3 Click **OK** to confirm deletion.

The screenshot shows a table of sites with the following data:

Configuration in Process	--	--	--	...
Offline	--	11 (7)	28d	...
Configuration in Process	--	--		...
Configuration in Process	--	--		...
Deployment Failed	--	--		...
Online	--	--		...

A context menu is open over the 'Offline' site, with the 'Delete Site' option highlighted. A confirmation dialog is displayed in the foreground:

Business Edition 4000

Confirm Deletion of Site :
 Customer - Cisco_TAC
 Location - RTP12-3

Please note that all end user portal users for the selected location will be deleted.

Cancel OK

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Manually Reset to Factory Default

To perform a factory reset manually, the local administration portal allows you to access the BE4000 appliance and perform a factory reset.



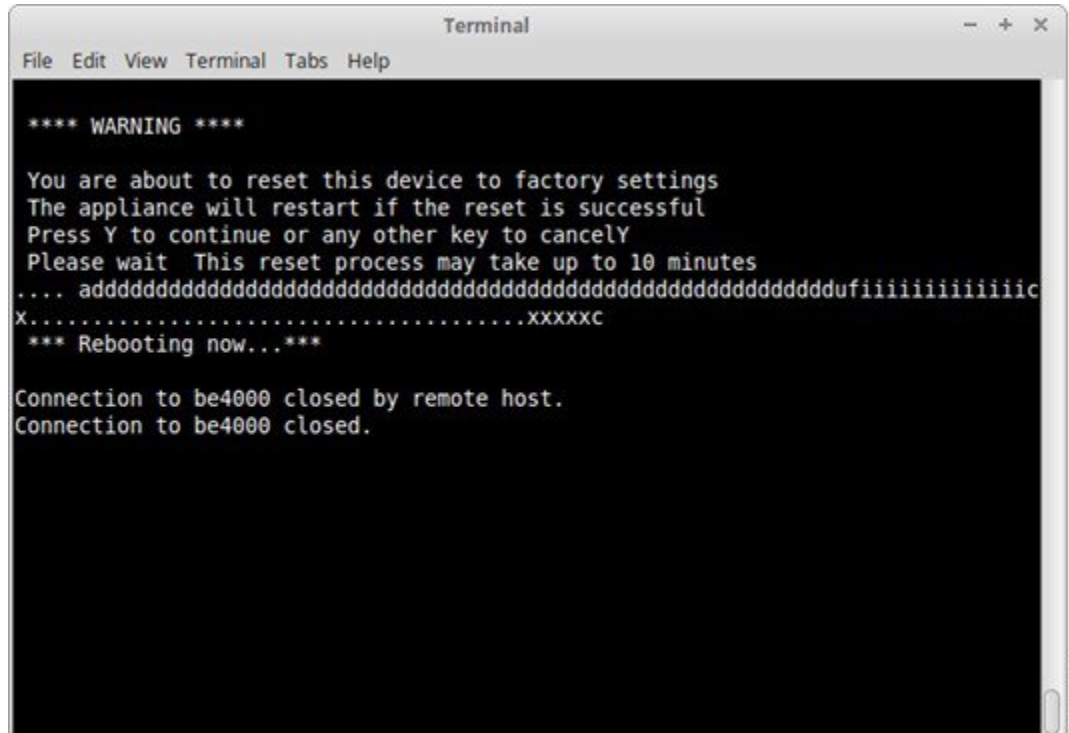
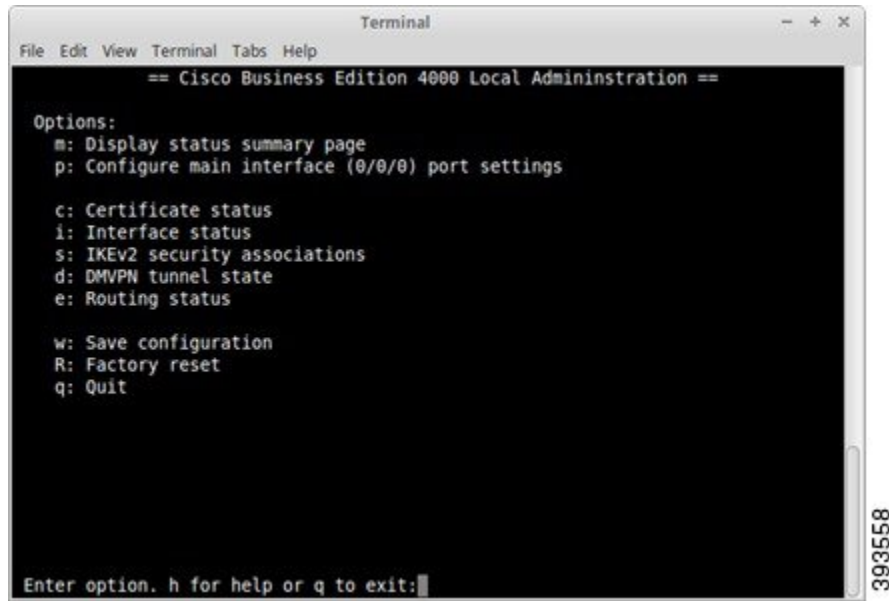
- Note** When the partner administrator performs a **Delete Site** on the BE4000 portal, the BE4000 is automatically reset to factory defaults, and a manual reset is not required.
- If a manual factory reset is desired, perform it only after deleting the site from the BE4000 portal. If the factory reset was performed before deleting a site, you must reload (power cycle) the appliance after deleting the site from BE4000 portal and before it is deployed again.

Choose one of the following two methods to connect to and reset the BE4000:

Console Based Factory Reset

Procedure

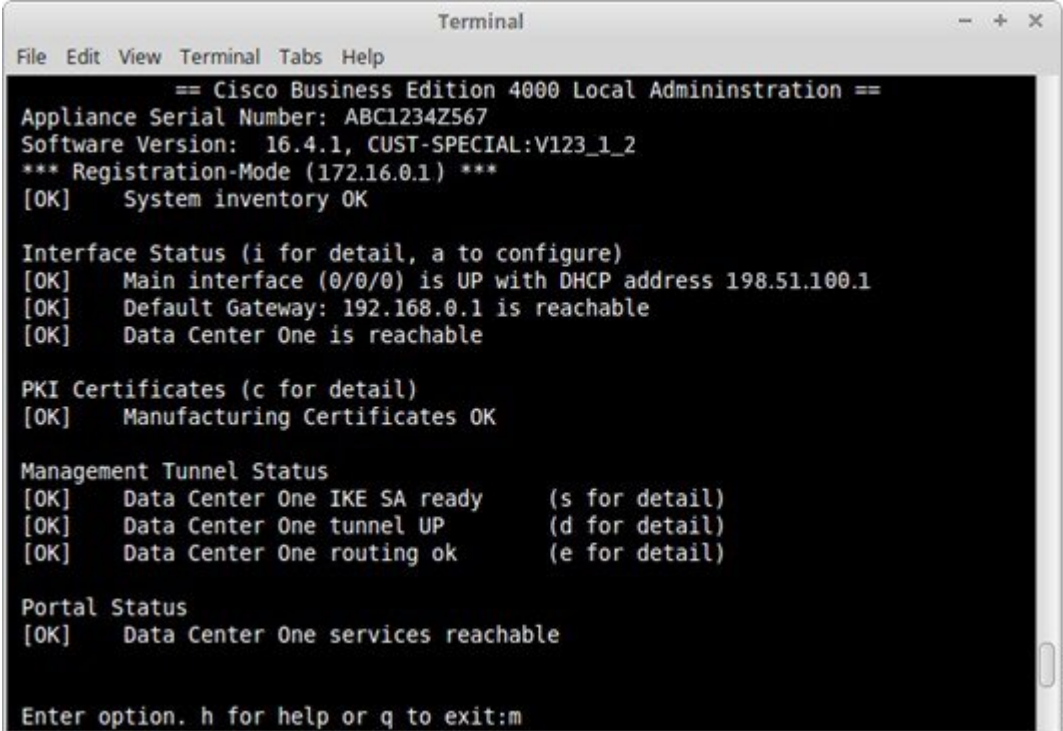
- Step 1** Connect through RJ45 or USB console cable.
- Step 2** Log in with the username “**status**”. Password is not required.
- Step 3** Type **R** and press **Enter** to start the reset process.



Ethernet Based Factory Reset

Procedure

- Step 1** Connect your computer back to back to MGMT port (use a normal Cat5e or Cat6 cable).
- Step 2** Use SSH client to connect to the BE4000 using either the MGMT IP Address (169.254.100.1) or Host Name (status@be4000).
- Step 3** Log in with username `status`. No password is required.
- Step 4** Type `h` to see the available options.



```
Terminal
File Edit View Terminal Tabs Help
== Cisco Business Edition 4000 Local Administration ==
Appliance Serial Number: ABC1234Z567
Software Version: 16.4.1, CUST-SPECIAL:V123_1_2
*** Registration-Mode (172.16.0.1) ***
[OK] System inventory OK

Interface Status (i for detail, a to configure)
[OK] Main interface (0/0/0) is UP with DHCP address 198.51.100.1
[OK] Default Gateway: 192.168.0.1 is reachable
[OK] Data Center One is reachable

PKI Certificates (c for detail)
[OK] Manufacturing Certificates OK

Management Tunnel Status
[OK] Data Center One IKE SA ready (s for detail)
[OK] Data Center One tunnel UP (d for detail)
[OK] Data Center One routing ok (e for detail)

Portal Status
[OK] Data Center One services reachable

Enter option. h for help or q to exit:m
```

- Step 5** To reset the appliance, type `R` to start the process.

```

Terminal
File Edit View Terminal Tabs Help
== Cisco Business Edition 4000 Local Administration ==

Options:
 m: Display status summary page
 p: Configure main interface (0/0/0) port settings

 c: Certificate status
 i: Interface status
 s: IKEv2 security associations
 d: DMVPN tunnel state
 e: Routing status

 w: Save configuration
 R: Factory reset
 q: Quit

Enter option, h for help or q to exit:
    
```

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```

Terminal
File Edit View Terminal Tabs Help

**** WARNING ****

You are about to reset this device to factory settings
The appliance will restart if the reset is successful
Press Y to continue or any other key to cancelY
Please wait This reset process may take up to 10 minutes
... aaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaaauiiiiiiiiiiiiiiic
X.....XXXXXXXXX
*** Rebooting now...***

Connection to be4000 closed by remote host.
Connection to be4000 closed.
    
```

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Reload the BE4000 Appliance

You can switch off and on (power cycle) the BE4000 appliance if you are at the customer location or site using the power switch.

Reset OTP

The Cisco Business Edition 4000 Management Portal allows you to reset the OTP when the recovery code is lost.

The **Reset OTP – From Another Partner User** method requires a minimum of two users to be associated with a partner organization. The second user resets the account of the first user. The **Reset OTP – Yourself** method allows you to reset the OTP account yourself.

Choose one of the following two methods to reset the OTP for Cisco Business Edition 4000 Management Portal:

Reset OTP—Yourself

You can now reset the One Time Password (OTP) account for yourself in the Cisco Business Edition 4000 Management Portal.

Procedure

- Step 1** Enter the URL <https://be4000.cisco.com>.
- Step 2** Click **Sign in**.
- Step 3** Click **Use Recovery Code**.
- Step 4** Click **Don't have a recovery code?**.


The **OTP Account Reset** pop-up window notifies that an email has been sent to your registered email address.

- Step 5** Click the link **Reset OTP** in the email.
-

Reset OTP—From Another Partner User

You can now reset the One Time Password (OTP) account of another user within the same partner organization.

Procedure

- Step 1** From the dashboard, click  to reset the OTP account. My People page is displayed.
 - Step 2** Click **Reset OTP** from Actions menu for the desired user.
-

Update Primary Contact Details

You can update your primary contact details in the Cisco Business Edition 4000 Partner portal.

Procedure

Step 1 From the dashboard, click on your name displayed on the top-right corner.

Step 2 Click **Settings**. The **Profile** page is displayed.

Step 3 Enter your primary contact number in the **Primary Phone** field.

Note The primary phone number must contain a minimum of 10 digits.

Note You cannot edit the email address. The email address is auto populated based on the information provided during partner registration.

Step 4 Click **Save**.

Step 5 Click **OK**.

Add Secondary OTP Devices

You can add more devices to generate the One Time Password (OTP) required to access the BE4000 Partner Portal.

Before you begin

A smart phone or tablet, other than the configured primary device, with One Time Password (OTP) client.

Procedure

Step 1 From the dashboard, click on your name that is displayed on the top-right corner.

Step 2 Click **Settings**. The Settings page is displayed.

Step 3 Click **Add Device**. Two-Step Authentication page is displayed with the list of configured devices.

Note The nicknames for the devices are displayed.

Step 4 Click **Add Device** at the bottom of configured devices list.

Step 5 Enter a nickname for your device. The name entered here is displayed on the OTP application that is configured on your smart phone or tablet.

Step 6 Click the **One Time Password (OTP)** client radio button and click **Next**.

Step 7 Scan the QR code using the OTP application on your smart phone or tablet and click **Next**.

Step 8 Enter the 6-digit security code that is generated by the OTP application on your smart phone or tablet and click **Next**.

Step 9 Save the recovery code that is displayed on the screen and click **Next**. Newly added device is listed in the Two-Step Authentication page.

Note This recovery code is used to log in to the portal if you lose or replace your smart phone or tablet.

Step 10 (Optional) From the drop-down for the secondary device, click **Primary** to make the secondary device primary.

Note You can set a secondary device as your primary device, from the sign-in screen as well.

Log In to the Cisco Business Edition 4000 Management Portal Using Recovery Code

If you have changed or lost your smart phone on which you had the OTP application configured with the BE4000, use the recovery code for logging in to the Cisco Business Edition 4000 Management Portal.

Before you begin

Have the recovery code that you received when you registered to the Cisco Business Edition 4000 Management Portal.

Procedure

Step 1 Enter the URL <https://be4000.cisco.com>.

Step 2 Click **Sign in**.

Step 3 Click **Use Recovery Code**.

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

Note 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 **Recovery Code** displayed on the screen.

Step 12 Click **Continue**.

Sign In to BE4000 Partner Portal Using a Secondary OTP Device

You can login to the BE4000 Partner Portal using any of the smart phones or tablets that you have configured with BE4000.

Before you begin

A smart phone or tablet configured with One Time Password (OTP) application.

Procedure

- Step 1** Go to the URL: <https://be4000.cisco.com/>.
 - Step 2** Sign in using your Cisco.com credentials.
 - Step 3** Click **Change Device**.
 - Step 4** Choose a device from the **Select your TOTP device** drop-down list.
 - Step 5** Click **Continue**.
 - Step 6** Enter the security code generated by your OTP application on your smart phone or tablet and click **Continue**.
-

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

Before you begin

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

Procedure

- Step 1** Log in to the BE4000 portal.
- Step 2** On the dashboard, hover the mouse on the row containing desired customer name and click **Manage Customer > Customer Admin**.
- 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.

Procedure

- 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.
-

Reset Password for a Customer Administrator OTP Account

You can reset the One Time Password (OTP) account of a customer administrator who owns a BE4000 site that is in the Online state.

Procedure

- Step 1** Click **Manage Site** from the Actions column for the desired site.
 - Step 2** Click edit (pen icon) under Contact Information.
 - Step 3** Click **Reset OTP**.
- Note** Resetting the OTP account mandates the customer administrator to create a new OTP application account when logging in to the Cisco Business Edition Selfcare Portal.
-

Modify Customer Administrator's Phone Number

Procedure

- Step 1** Click **Manage Site** from the Actions column for the desired site.
 - Step 2** Click edit (pen icon) under Contact Information.
 - Step 3** Enter the phone number. 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.
 - Step 4** Click **Save**.
-

Remove Customer Administrator

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

Procedure

- Step 1** Log in to the BE4000 portal.
- Step 2** On the dashboard, hover the mouse on the row containing desired customer name and click **Manage Customer > Customer Admin**
- 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.
