Setup Assistant

Add Customer Field Descriptions

Add Customer Field Descriptions

Field	Description
Customer Name	Enter the name of the customer. You can enter a maximum of 15 characters. **Note** One customer can have multiple sites. Each site needs one BE4000 appliance. Each site is configured separately.
Location	Enter the location of the customer. You can enter a maximum of 15 characters.
Customer Admin Email	Enter the email address of the customer administrator.
Contact Name	Enter the name of the customer administrator.
Phone Number	Enter the phone number of the customer administrator.
Template Name	Select an existing customer site template. Template saves the site configurations (connectivity details, dial plans, stations, call routing and feature details). Using template, you can avoid rekeying the configuration details while creating a new site similar to an existing site configuration. **Note** If you are creating a site for the first time, you may not find any existing templates listed in the drop-down.

Connectivity Field Descriptions

Note The following IP Networks are not supported:

- 10.0.1.x/24
- 10.0.2.x/24
- 10.0.3.x/24
- 10.1.1.x/24
- 10.1.2.x/24
- 10.1.3.x/24
- 10.2.x.x/16
- 10.3.x.x/16

LAN Connection Field Descriptions

Field	Description
Network Details	
BE4000 IP Address	Enter IP address of the BE4000 appliance. Ensure that the IP address matches with that of the customer subnet address and ports (UDP 500, UDP 4500, and ESP 50) are connected, active, and reachable to the internet. Check the status of the ports using Cisco Business Edition 4000 Port Check Tool.
Voicemail IP Address	Enter the IP address of the voicemail server. Voicemail IP address cannot be the same as BE4000 IP address.
Subnet Mask	Enter the IP subnet mask. For example, 255.255.255.0 is an IP subnet mask.
Gateway Address	Enter the default gateway address.
Internet Service Provider (ISP)	Enter the ISP label. In case, there are issues in deploying the BE4000, the ISP label helps technicians to check if the fault is with the service provider. ISP label is an arbitrary identifier.
SMTP Server - Enables Voicemail to Email Functionality The SMTP Server configuration is required for voicemail to Email functionality, wherein voicemail received by an extension is sent as an email attachment to the registered email address. • If the customer has Office 365, Configure Office 365 first. • If the customer has their own SMTP relay (other than Office 365), ensure that the proprietary certificates of the SMTP relay are trusted by BE4000.	

Field	Description
IP Address or Domain Name	Enter the IP Address or Domain Name of the SMTP server. You can enter a maximum of 100 characters for the domain name. **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.
Port	Enter the port number. Default value is 25.
Security Mode	Choose one of the secure modes: • None • SSL—Provide ways to encrypt a communication channel between two computers. • STARTTLS—An extension to the Simple Mail Transfer Protocol (SMTP) service that allows an SMTP server and client to use Transport Layer Security (TLS) to provide private, authenticated communication over the internet.
Sender's Email Address	Enter an email address that is used as “From” address to send emails containing voicemail as an attachment. Ensure that the SMTP server allows receiving emails from the entered email address.
Authenticate	Check the "Authenticate" check box to ensure that the voicemail to email functionality is secure. Enter a username and password that needs to be filled by the users while accessing the voicemail from email.

System Settings Field Descriptions

Note Choose unique digits for dialing an outside line, sending a call to voicemail automatically, and dialing an intercom extension.

Field	Description
Dial an Outside Line	Choose a digit to make a call to an outside phone number. You can set any digit between 0 to 6, 8, and 9. You cannot set * and 7. Default is 9. Users should dial this digit before dialing an external phone number.
Extension length	Choose the total number of digits in an extension. You can set your extension to contain 3, 4, or 5 digits. Default is 4.
Interdigit Timeout	Choose the number of seconds to wait after each digit is entered, before assuming the caller has finished entering digits. Range is from 0 to 9. Default is 5.
Send to Voicemail Automatically	Choose a digit to dial for sending a call to voicemail automatically. Range is from 1 to 6. Default is 2.

Field	Description
Intercom	Choose a digit to dial for making an intercom call. Range is from 1 to 6. Default is 4.
Advanced Options	
Forwarding Local	Choose to enable or disable forwarding local. This decides if internal (local) calls can be forwarded.
Phone Redirect Limit	Set the phone redirect limit. Limits the maximum number of 3XX responses that can be accepted for a single call. Range is from 5 to 20. By default, 5 is entered.

Direct Inward Dial (DID)

Direct Inward Dial (DID) numbers are the registered numbers received from your service provider. Add the DID numbers for the SIP and ISDN (PRI and BRI) connections. Do not add FXO line numbers here. The FXO line numbers can be added while adding Line Cards (NIM) details.

DID numbers can be added in two ways:

- Click "Add Row" and add the DID numbers and service name one by one.
- Download the template provided on the screen. Fill in the details and upload all the DIDs in one go.

Note

- DID numbers must be entered in E.164 format. A minimum of 10 digits is required for the DID number. For example: +14155552671
- Only up to 98 DID numbers are supported for SIP trunks and 98 DID numbers for ISDN trunks (PRI and BRI).

Direct Dial Numbers Field Descriptions

Field	Description
Add Row	Click to add DID numbers manually.
Download Template	Click to download the template provided on the screen.
Choose file	Click to browse and upload the file containing DID numbers.
Service Name	Enter a service name for each DID. Provide any name that is easy to identify the service to which each DID number belongs to.
Registered Numbers	Add the DID numbers received from your service provider for each row.
Delete	You can delete the individual rows of DID numbers.
Replace this list	Click to remove all the DID numbers displayed on the page.

SIP Trunks Field Descriptions

Turn on the slider to configure SIP trunks.

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.

Service Settings Field Descriptions

Field	Description
Service Name	Choose a service name from the drop-down list. The service names added in the **Direct Inward Dial (DID) Numbers** page are listed in the drop-down.
Provider Template	Choose a preconfigured provider template based on your SIP service provider. If your SIP service provider is not in the drop-down list, choose "Custom".

Interface Settings Field Descriptions

Field	Descriptions
Use Secondary Interface for Trunk?	Choose the type of interface connectivity for SIP trunk. Primary interface refers to GE 0/0/0 and Secondary interface refers to GE 0/0/1. Primary interface is always connected to the internet service provider. If the internet service provider and SIP trunk service provider are different, use secondary interface for SIP trunk connectivity. • Proxy Server Field Descriptions—SIP trunk and internet connectivity is provided by the same service provider. • Secondary Interface with Static Address—SIP trunk and internet are provided by two separate service providers. Internet service provider is connected using the primary interface and SIP trunk service provider is connected using the secondary interface. The SIP trunk service provider provides static IP address for the network connectivity. • Secondary Interface with Dynamic Address—SIP trunk and internet are provided by two separate service providers. Internet service provider is connected using the primary interface and SIP trunk service provider is connected using the secondary interface. The SIP trunk service provider provides dynamic IP address for the network connectivity.
External Public Address	Enter the public IP address assigned by your internet service provider so that SIP services work across Network Address Translation (NAT).

No Secondary Interface

Proxy Server Field Descriptions

Field	Description
Proxy Address	Enter an IP address and Port, Fully Qualified Domain Name (FQDN) and Port, or SRV record for your service proxy. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Proxy Port	Optional. If you have provided an IP address for the proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Outbound Proxy	
Outbound Proxy Address	Enter an IP Address, fully qualified domain name, or domain SRV for your service outbound proxy if one is used. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Outbound Proxy Port	Optional. If you have provided an IP address for the outbound proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Call Authentication	
Username and Password	Enter the username and password if your service provider requires authentication for every call.
Authentication Realm	Enter the authentication realm for call authentication. Typically, authentication realm is the service domain name.
Include in Invite	Check the "Include in Invite" check box, if your service provider requires authentication details to be sent in the initial invite. If unchecked, authentication is provided in the response to a 407 challenge.

Proxy Server—Advanced Options Field Descriptions

Field	Description
Min-SE	Enter the minimum value for the session expiry parameter sent in the initial invite. Range is from 90 to 86,400 seconds. Unless instructed by your SIP service provider, the default value of 90 seconds must be used.
Session Expires	Enter the maximum duration of a session in seconds. During a call, the session expiry time is periodically refreshed based on the value entered here. Range is from 90 to 86,400 seconds. Unless instructed by your provider, the default value of 1800 seconds should be used.
RTP Port Range	Limit the range of ports used for RTP. Enter even numbers between 8,000 to 48,198.

Field	Description
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 for a specific VoIP dial peer. Available options are: • T.38 • T.38 fall back to G.711 u-law • T.38 fall back to G.711 a-law • Pass Through G711u • Pass Through G711a
DTMF Signaling Protocol	Choose one of the following as the DTMF signaling mechanism based on the protocol offered by your SIP service provider. • RFC2833 • sip-notify
Calling Party Header Selection	Choose one of the following for calling party header selection: • From • Remote Party ID (RPID) • P-AID Pilot DID • P-AID Assigned DIDs
Calling Party Domain	Leave the "Calling Party Domain" field blank to send the BE4000 IP address with calling party headers. Enter a domain name or full qualified domain if you want to replace the BE4000 IP address.
Pilot Number	**Note** "Pilot Number" field is displayed only when "Calling Party Header Selection" drop-down is chosen as "P-AID Pilot DID". Choose the "Pilot Number" from the drop-down list if the service provider requires a specific number to be used for P-Asserted Identity Headers.
CLI Restriction Prefix	Enter the dialing prefix if the service provider allows calling line ID to be withheld on a call by call basis.
RFC3555 Compliant G.729 Annex B	Uncheck the "RFC3555 Compliant G.729 Annex B" check box if the call server is not RFC3555 compliant for G.729 Annex B SDP formatting (Adds g729-annexb override). Check if you are unsure.
Two way media override	Check the "Two way media override" check box to override modification of media stream from send/receive to sendonly or inactive. When checked, two way media is always be requested.

Field	Description
Redirection (Optional)	Check the Redirection option to reset the default processing of 3xx messages. By default, SIP gateways process all incoming 3xx redirect messages according to RFC 2543. However if the Redirection option is disabled, the gateway treats the incoming 3xx responses as 4xx error class responses. Redirection should be selected by default and only unselected if required by the SIP trunk provider.
Options Ping Enable to monitor the SIP service availability allowing traffic to be rerouted, if possible, in the event of failure.	
Service Up Interval	Enter the period between Options packets being sent while the service is considered to be up. Range is from 5 to 1,200 seconds. Default is 60 seconds.
Service Down Interval	Enter the period between Options packets being sent while the service is considered to be down. Range is from 5 to 1,200 seconds. Default is 30 seconds.
Retries	Enter the number of missed responses allowed before a service is considered unavailable. Range is from 1 to 10. Default is 5.

Registrar Server

Registrar server can be configured either through DHCP or by providing IP address and port. Click one of the following options based on your network:

- Configure via DHCP
- Configure address and port

Registrar Server Field Descriptions

Field	Description
Registrar Address	Enter an IP Address, fully qualified domain name, or domain SRV for service registrar. This field appears only when you choose **Configure address and port**. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Registrar Port	If you have provided an IP address for the registrar, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060. This field appears only when you choose **Configure address and port**.
Authentication Realm	Enter the authentication realm used for registration by your service provider. Mandatory if Registrar Address or DHCP is configured.
Registrar with realm	Check the "Registrar with Realm" configuring the registrar with the realm information provided for the proxy. Uncheck to remove the configuration.

Field	Description
Username and Password	Enter the username and password, if the service provider requires per call authentication.
Include DID	Select appropriate DID for each username if the service provider requires a DID to be included with registration authentication.
Add Row	Click to add multiple rows for username, password, and include DID. You can add a maximum of 98 rows.

Registrar Server—Advanced Options Field Descriptions

Field	Description
Registration Timeout	Enter the Registration Timeout period. Determines how frequently the system registers.
Transport Layer	Choose TCP or UDP from the "Transport Layer" drop-down list as the transport protocol used by the service provider.

Security

Add at least one trusted IP Address. The BE4000 accepts incoming VoIP (SIP) calls only if the remote IP address of an incoming VoIP call matches an address in the trusted IP address list. Enter the IP addresses provided for proxy, outbound proxy, and registrar from your service provider. IP addresses must be provided if hostnames are used. Entries can be provided either as a host address (x.x.x.x) or subnet (x.x.x.x /nn)."

Security Field Descriptions

Field	Description
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.

Secondary Interface with Static Address

Interface Settings

Field	Description
Interface Options	Ethernet ports usually use the auto-negotiate protocol settings. If your switch does not support this option by itself, choose from the following interface options: • Auto Negotiate • Gigabit Ethernet • Fast Ethernet Full Duplex • Fast Ethernet Half Duplex • Ethernet Full Duplex • Ethernet Half Duplex **Note** By default, Auto Negotiate is selected.
IP Address and Mask	Enter IP address and subnet mask of the secondary interface. The fields are mandatory.
Default Gateway	Enter the IP address of the default gateway. This field is mandatory.
Name Servers	Enter the IP address of the dedicated, private DNS used by your service provider. Ensure that you enter the name server addresses even if they are provided via DHCP. You can enter a maximum of 6 IP addresses separated by spaces.
External Public Address	Enter To ensure that SIP services work across Network Address Translation, provide the public IP address provided by your service provider.

Proxy Server Field Descriptions

Field	Description
Proxy Address	Enter an IP address and Port, Fully Qualified Domain Name (FQDN) and Port, or SRV record for your service proxy. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Proxy Port	Optional. If you have provided an IP address for the proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Outbound Proxy	
Outbound Proxy Address	Enter an IP Address, fully qualified domain name, or domain SRV for your service outbound proxy if one is used. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.

Field	Description
Outbound Proxy Port	Optional. If you have provided an IP address for the outbound proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Call Authentication	
Username and Password	Enter the username and password if your service provider requires authentication for every call.
Authentication Realm	Enter the authentication realm for call authentication. Typically, authentication realm is the service domain name.
Include in Invite	Check the "Include in Invite" check box, if your service provider requires authentication details to be sent in the initial invite. If unchecked, authentication is provided in the response to a 407 challenge.

Proxy Server—Advanced Options Field Descriptions

Field	Description
Min-SE	Enter the minimum value for the session expiry parameter sent in the initial invite. Range is from 90 to 86,400 seconds. Unless instructed by your SIP service provider, the default value of 90 seconds must be used.
Session Expires	Enter the maximum duration of a session in seconds. During a call, the session expiry time is periodically refreshed based on the value entered here. Range is from 90 to 86,400 seconds. Unless instructed by your provider, the default value of 1800 seconds should be used.
RTP Port Range	Limit the range of ports used for RTP. Enter even numbers between 8,000 to 48,198.
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 for a specific VoIP dial peer. Available options are: • T.38 • T.38 fall back to G.711 u-law • T.38 fall back to G.711 a-law • Pass Through G711u • Pass Through G711a
DTMF Signaling Protocol	Choose one of the following as the DTMF signaling mechanism based on the protocol offered by your SIP service provider. • RFC2833 • sip-notify

Field	Description
Calling Party Header Selection	Choose one of the following for calling party header selection: • From • Remote Party ID (RPID) • P-AID Pilot DID • P-AID Assigned DIDs
Calling Party Domain	Leave the "Calling Party Domain" field blank to send the BE4000 IP address with calling party headers. Enter a domain name or full qualified domain if you want to replace the BE4000 IP address.
Pilot Number	**Note** "Pilot Number" field is displayed only when "Calling Party Header Selection" drop-down is chosen as "P-AID Pilot DID". Choose the "Pilot Number" from the drop-down list if the service provider requires a specific number to be used for P-Asserted Identity Headers.
CLI Restriction Prefix	Enter the dialing prefix if the service provider allows calling line ID to be withheld on a call by call basis.
RFC3555 Compliant G.729 Annex B	Uncheck the "RFC3555 Compliant G.729 Annex B" check box if the call server is not RFC3555 compliant for G.729 Annex B SDP formatting (Adds g729-annexb override). Check if you are unsure.
Two way media override	Check the "Two way media override" check box to override modification of media stream from send/receive to sendonly or inactive. When checked, two way media is always be requested.
Redirection (Optional)	Check the "Redirection (Optional)" to reset the default processing of 3xx messages. By default, SIP gateways process all incoming 3xx redirect messages according to RFC 2543. However if the Redirection option is disabled, the gateway treats the incoming 3xx responses as 4xx error class responses. Redirection should be selected by default and only unselected if required by the SIP trunk provider.
Options Ping Enable to monitor the SIP service availability allowing traffic to be rerouted, if possible, in the event of failure.	
Service Up Interval	Enter the period between Options packets being sent while the service is considered to be up. Range is from 5 to 1,200 seconds. Default is 60 seconds.
Service Down Interval	Enter the period between Options packets being sent while the service is considered to be down. Range is from 5 to 1,200 seconds. Default is 30 seconds.
Retries	Enter the number of missed responses allowed before a service is considered unavailable. Range is from 1 to 10. Default is 5.

Registrar Server

Registrar server can be configured either through DHCP or by providing IP address and port. Click one of the following options based on your network:

- Configure via DHCP
- Configure address and port

Registrar Server Field Descriptions

Field	Description
Registrar Address	Enter an IP Address, fully qualified domain name, or domain SRV for service registrar. This field appears only when you choose **Configure address and port**. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Registrar Port	If you have provided an IP address for the registrar, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060. This field appears only when you choose **Configure address and port**.
Authentication Realm	Enter the authentication realm used for registration by your service provider. Mandatory if Registrar Address or DHCP is configured.
Registrar with realm	Check the "Registrar with Realm" configuring the registrar with the realm information provided for the proxy. Uncheck to remove the configuration.
Username and Password	Enter the username and password, if the service provider requires per call authentication.
Include DID	Choose appropriate DID for each username if the service provider requires a DID to be included with registration authentication.
Add Row	Click to add multiple rows for username, password, and include DID. You can add a maximum of 98 rows.

Registrar Server—Advanced Options Field Descriptions

Field	Description
Registration Timeout	Enter the Registration Timeout period. Determines how frequently the system registers.
Transport Layer	Choose TCP or UDP from the "Transport Layer" drop-down list as the transport protocol used by the service provider.

Security

Add at least one trusted IP Address. The BE4000 accepts incoming VoIP (SIP) calls only if the remote IP address of an incoming VoIP call matches an address in the trusted IP address list. Enter the IP addresses provided for proxy, outbound proxy, and registrar from your service provider. IP addresses must be provided if hostnames are used. Entries can be provided either as a host address (x.x.x.x) or subnet (x.x.x.x /nn)."

Security Field Descriptions

Field	Description
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.

Secondary Interface with Dynamic Address

Interface Settings

Field	Description
Interface Options	Ethernet ports usually use the auto-negotiate protocol settings. If your switch does not support this option by itself, choose from the following interface options: • Auto Negotiate • Gigabit Ethernet • Fast Ethernet Full Duplex • Fast Ethernet Half Duplex • Ethernet Full Duplex • Ethernet Half Duplex **Note** By default, Auto Negotiate is selected.
Name Servers	Enter the IP address of the dedicated, private DNS used by your service provider. Ensure that you enter the name server addresses even if they are provided via DHCP. You can enter a maximum of 6 IP addresses separated by spaces.
External Public Address	Enter To ensure that SIP services work across Network Address Translation, provide the public IP address provided by your service provider.

Proxy Server Field Descriptions

Field	Description
Proxy Address	Enter an IP address and Port, Fully Qualified Domain Name (FQDN) and Port, or SRV record for your service proxy. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Proxy Port	Optional. If you have provided an IP address for the proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Outbound Proxy	
Outbound Proxy Address	Enter an IP Address, fully qualified domain name, or domain SRV for your service outbound proxy if one is used. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Outbound Proxy Port	Optional. If you have provided an IP address for the outbound proxy, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060.
Call Authentication	
Username and Password	Enter the username and password if your service provider requires authentication for every call.
Authentication Realm	Enter the authentication realm for call authentication. Typically, authentication realm is the service domain name.
Include in Invite	Check the "Include in Invite" check box, if your service provider requires authentication details to be sent in the initial invite. If unchecked, authentication is provided in the response to a 407 challenge.

Proxy Server—Advanced Options Field Descriptions

Field	Description
Min-SE	Enter the minimum value for the session expiry parameter sent in the initial invite. Range is from 90 to 86,400 seconds. Unless instructed by your SIP service provider, the default value of 90 seconds must be used.
Session Expires	Enter the maximum duration of a session in seconds. During a call, the session expiry time is periodically refreshed based on the value entered here. Range is from 90 to 86,400 seconds. Unless instructed by your provider, the default value of 1800 seconds should be used.
RTP Port Range	Limit the range of ports used for RTP. Enter even numbers between 8,000 to 48,198.
Transport Layer	Choose the protocol used for transport layer by your service provider.

Field	Description
Fax Transmission Protocol	Choose one of the ITU-T T.38 standard Fax Transmission Protocols to be used for a specific VoIP dial peer. Available options are: • T.38 • T.38 fall back to G.711 u-law • T.38 fall back to G.711 a-law • Pass Through G711u • Pass Through G711a
DTMF Signaling Protocol	Choose one of the following as the DTMF signaling mechanism based on the protocol offered by your SIP service provider. • RFC2833 • sip-notify
Calling Party Header Selection	Choose one of the following for calling party header selection: • From • Remote Party ID (RPID) • P-AID Pilot DID • P-AID Assigned DIDs
Calling Party Domain	Leave the "Calling Party Domain" field blank to send the BE4000 IP address with calling party headers. Enter a domain name or full qualified domain if you want to replace the BE4000 IP address.
Pilot Number	**Note** "Pilot Number" field is displayed only when "Calling Party Header Selection" drop-down is chosen as "P-AID Pilot DID". Choose the "Pilot Number" from the drop-down list if the service provider requires a specific number to be used for P-Asserted Identity Headers.
CLI Restriction Prefix	Enter the dialing prefix if the service provider allows calling line ID to be withheld on a call by call basis.
RFC3555 Compliant G.729 Annex B	Uncheck the "RFC3555 Compliant G.729 Annex B" check box if the call server is not RFC3555 compliant for G.729 Annex B SDP formatting (Adds g729-annexb override). Check if you are unsure.
Two way media override	Check the "Two way media override" check box to override modification of media stream from send/receive to sendonly or inactive. When checked, two way media is always be requested.

Field	Description
Redirection (Optional)	Check the "Redirection (Optional)" to reset the default processing of 3xx messages. By default, SIP gateways process all incoming 3xx redirect messages according to RFC 2543. However if the Redirection option is disabled, the gateway treats the incoming 3xx responses as 4xx error class responses. Redirection should be selected by default and only unselected if required by the SIP trunk provider.
Options Ping Enable to monitor the SIP service availability allowing traffic to be rerouted, if possible, in the event of failure.	
Service Up Interval	Enter the period between Options packets being sent while the service is considered to be up. Range is from 5 to 1,200 seconds. Default is 60 seconds.
Service Down Interval	Enter the period between Options packets being sent while the service is considered to be down. Range is from 5 to 1,200 seconds. Default is 30 seconds.
Retries	Enter the number of missed responses allowed before a service is considered unavailable. Range is from 1 to 10. Default is 5.

Registrar Server

Registrar server can be configured either through DHCP or by providing IP address and port. Click one of the following options based on your network:

- Configure via DHCP
- Configure address and port

Registrar Server Field Descriptions

Field	Description
Registrar Address	Enter an IP Address, fully qualified domain name, or domain SRV for service registrar. This field appears only when you choose **Configure address and port**. **Note** Enter a unique address that does not overlap with the address that is assigned for Gig 0/0/0 interface, Gig 0/0/1 interface, and SMTP server.
Registrar Port	If you have provided an IP address for the registrar, you may also specify a non-standard SIP port if necessary. Leave blank to use port 5060. This field appears only when you choose **Configure address and port**.
Authentication Realm	Enter the authentication realm used for registration by your service provider. Mandatory if Registrar Address or DHCP is configured.
Registrar with realm	Check the "Registrar with Realm" configuring the registrar with the realm information provided for the proxy. Uncheck to remove the configuration.

Field	Description
Username and Password	Enter the username and password, if the service provider requires per call authentication.
Include DID	Choose appropriate DID for each username if the service provider requires a DID to be included with registration authentication.
Add Row	Click to add multiple rows for username, password, and include DID. You can add a maximum of 98 rows.

Registrar Server—Advanced Options Field Descriptions

Field	Description
Registration Timeout	Enter the Registration Timeout period. Determines how frequently the system registers.
Transport Layer	Choose TCP or UDP from the "Transport Layer" drop-down list as the transport protocol used by the service provider.

Security

Add at least one trusted IP Address. The BE4000 accepts incoming VoIP (SIP) calls only if the remote IP address of an incoming VoIP call matches an address in the trusted IP address list. Enter the IP addresses provided for proxy, outbound proxy, and registrar from your service provider. IP addresses must be provided if hostnames are used. Entries can be provided either as a host address (x.x.x.x) or subnet (x.x.x.x /nn)."

Security Field Descriptions

Field	Description
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.

Line Cards Field Descriptions

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. For the list of supported NIM cards, refer Supported Line Cards.

Turn on the slider for the NIM cardslot where you want to insert the NIM card.

Field	Description
NIM 1	Select the NIM card as per your connectivity requirements.
NIM 2	Select the NIM card as per your connectivity requirements.

NIM-2FXS or NIM-4FXS

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

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

NIM-2FXO or NIM-4FXO

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

Setup Assistant

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Direction</td><td>Mark the line as incoming only or bidirectional. The system builds the trunk groups based on what you select.
• In + Out—Allows the phone line to receive and make calls.
• Inbound Only—Allows the phone line to receive the calls.</td></tr>
<tr><td>Law</td><td>Choose the type of algorithm used for modifying an input signal for digitization:
• u-law
• a-law</td></tr>
<tr><td>Input Gain (db)</td><td>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).</td></tr>
<tr><td>Output Attenuation (db)</td><td>The value that is configured for minimizing the signal loss. Range is from -6 to 14 db. Default value for FXO and FXS line is 0 db.
Note This field is available only while editing the line cards containing FXS and FXO lines. This field does not appear while adding the line card during initial site deployment (in the Setup Assistant wizard) or post site deployment (under Manage Site > Line Cards (NIM) > Add NIM Card).</td></tr>
<tr><td>Show Advanced</td><td>Displays the advanced fields.</td></tr>
<tr><td colspan="2">Configuration Voice Port Commands
Note The following fields are displayed only when the Show Advanced check box is checked.</td></tr>
<tr><td>Cable Detect</td><td>Enables or disables cable polling on an analog Foreign Exchange Office (FXO) port.</td></tr>
<tr><td>Bearer Capability</td><td>Note Configuring "Bearer Capability" drop-down list does not have any affect on the FXO port.
Specifies the information transfer capability of the bearer capability information element (IE) in the outgoing ISDN SETUP message for Session Initiation Protocol (SIP) early-media calls. You can choose one of the following:
• Speech—Specifies speech as the information transfer capability.
• 3100hz—Specifies 3.1 kHz audio as the information transfer capability.</td></tr>
</table>

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

NIM-2FXS/4FXO

Field	Description
If you are using **NIM-2FXS/4FXO**, input the following details:	
FXS	

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

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

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

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

Field	Description
If you are using **NIM-2BRI-NT/TE** or **NIM-4BRI-NT/TE**, input the following details:	
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network.
Service Name	Choose a service name from the drop-down list. The drop-down list contains the list of service providers that you added in the DID page. Service name entered here is shared across all the lines or ports for the selected NIM type. **Note** You cannot choose the same service provider for SIP and Line Cards.

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

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

Field	Description
If you choose **NIM-1MFT-T1/E1**, **NIM-2MFT-T1/E1**, or **NIM-4MFT-T1/E1**, enter the following details:	
Card Type	Choose the card type based on your customer network requirement. E1 PRI is chosen by default. The available options are: • T1 PR1 • E1 PRI
Type	Choose one of the following: • **Disabled**—The line is inactive. • **Trunk**—The line is connected to the Public Switched Telephone Network.

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

Private IP 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.

Click **Add Row** and add private IP address ranges one by one.

Dial Plan Field Descriptions

Region Settings Field Descriptions

Field	Description
Telephony Port Tones	Choose your home country. This is used to display the date, time, currency, and other dial plan tones and numbers.

Field	Description
Time Zone	Choose your relevant time zone. Typically, your time zone is linked to the area code of your main company number. For example, for area code 919 (RTP), the time zone defaults to Pacific Time.
Phone Display Language	Choose your phone display language as the default language used for all accounts and notifications from the drop-down list.
Phone Tones	Select the country to define dial tone for your phones.
Voicemail and System Prompt Language	Select the language in which you want to receive your phone greetings.
Selfcare Portal	Select the language preference for your customers self care portal.
Time Format	Select the time format as 12-or 24 hour. For example, the default format for the United States is 12 hours.
Date Format	Select the date format to suit your needs. For example, the default format for the United States is MM/DD/YY.
DST Auto Adjust	Enables or disables the automatic adjustment of daylight saving time on your phones.

Dial Plans Field Descriptions

Field	Description
Country	Choose the country and locale that you want for your system.
Local Dialling Options	Select the option for local dialing as per customer requirement. The local area length value depends on the regulation set up by the service providers in your region.
Local Area Code	Enter a valid “Local Area Code” for your main number. This field appears based on the local dialing option selected.

Field	Description
COR	Choose the class of restriction for the dialing patterns belonging to a dial plan. Choose one of the following options from the drop-down list for COR column: • **internal** • **local** • **local-plus** • **national** • **national-plus** • **international**
Preference	Choose the preference for the selected dialing patterns belonging to a dial plan. Choose one of the following options from the drop-down list for Preference column: • POTSthenSIP • SIPthenPOTS • SIPOnly • POTSOnly The "Preference" is set to "SIPthenPOTS" by default. The "Preference" for call-emergency patterns is set to "POTSthenSIP" by default.

Stations Field Descriptions

Stations Field Descriptions

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.

Setup Assistant

Field	Description
Type	Choose a type of user: • **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	Enter the email address of the user. The top-level domain in the email address can contain up to six characters. **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.
Extension	Enter the extension number assigned to the user. **Note** • 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.
COR	Choose the Class of Restriction (COR) for the extension. COR allows you to choose one of the calling privileges: • **internal** • **local** • **local-plus** • **national** • **national-plus** • **international**

Field	Description
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.

Call Routing Field Descriptions

Business Hours

Business hours are the hours during the day in which business is commonly conducted. BE4000 allows you set weekly schedule for open hours and yearly holidays when the business is closed.

Field	Description
Open Business Hours	Select one of the following options: • **24/7 (No closed hours)** - You can upload one audio file that is played for all calls received during 24/7. • **Dual Hours (Open and Closed)** - You can upload different audio files for the calls received during open and closed hours.

When you select **Dual Hours (Open and Closed)**, the following menu is displayed:

Field	Description
Hours of Operation	Customize your business hours for your various departments. You can specify the open hours for each day of the week. **Note** You must enter time in 24-hour format only (17:00 for example). Time must be either full (:00) or half hours (:30).
Add Open Hours	Custom hours let you add and specify hours for each day of the week.
Holiday	Set up your holiday list.

Field	Description
Add New Holiday	Add the list of 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.

Hunt Group

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.

Hunt Group Field Descriptions

Suggestions for Setting Up Hunt Group Timers

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

- Member Timeout
- Comfort Greeting Frequency
- Max Waiting Time

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

Example for Parallel Hunt Method

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

Member Timeout = 30 seconds

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

Max Waiting Time = Member Timeout = 30 seconds

Example for Peer, Sequential, and Longest Idle Hunt Methods

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

Members in the Hunt Group = 5

Member Timeout = 30 seconds

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

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

Field	Description
Group Name	Enter a unique name for the hunt list. To easily identify the hunt list, consider appending the pilot extension to the name; for example, hl5001. Group name must contain a minimum of 2 and a maximum of 12 characters. **Note** Only the following characters are allowed: a-z, A-Z, 0-9, space, hyphen(-), and underscore (_)
Pilot Number	Enter a number to access the Hunt Group that serves as the pilot for the hunt list. This number serves as the trigger for hunting to begin. Pilot number can be any number equal to the length of extension. For example: • If the extension length is 3, pilot number can be any number from 000 to 999 • If the extension length is 4, pilot number can be any number from 0000 to 9999 • If the extension length is 5, pilot number can be any number from 00000 to 99999 **Note** • The Pilot Number of the Hunt Groups cannot be the same as any existing extension and cannot start with the digit that is used for sending calls to voicemail automatically and for placing intercom calls. • The Pilot Number of the Hunt Groups cannot be the same as any existing Pickup Group number.

Setup Assistant

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

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

Setup Assistant

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

Field	Description
Save	Save your hunt group configuration settings.

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.

Auto Attendant Field Descriptions

Field	Description
Pilot Number	Enter the number that callers dial to reach auto attendant. A minimum of four digits is required. Range is from 1000 to 9999.
How many times do you want message to be played	Number of times the audio file is played to the caller before the call reaches the drop through destination. Value range is from 1 to 9. Default value is 4.
Drop Through Destination	Defines where the call lands if no action is performed by the caller even after playing the menu for the defined number of repeats. You can configure one of the following as the drop through destination: • **Route to Extension**—All extensions are listed in the drop-down. • **Route to Voicemail Box**—All extensions that have "Voicemail" enabled are listed in the drop-down. • **Route to Hunt Group**—All Hunt Groups are listed in the drop-down. • **Route to Group Voicemail Box**—All group voicemail boxes are listed in the drop-down. **Note** During the initial site deployment (in the Setup Assistant), the drop-down shows an option, only if you create a group voicemail box on the **Hunt Groups** page, by choosing "Route to Group Voicemail Box" from the **When No Member is Available** drop-down list. • **Disconnect Call**—The call gets disconnected.

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

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.

Night Service Field Descriptions

Field	Description
Manual Activation Code	Enter a code to trigger Night Service feature manually during open 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.
Active Hours	Enter the hours during which the Night Service must be active. You cannot overlap the end time of Night Service and the start of business open hours. For example, if your business closes at 17:00 and opens next day at 09:00 AM, 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.
Holidays	Add the list of 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.

Note After enabling night service, you must the extensions for night service hours and to receive night service calls. Refer Night Service section for more details.

Inbound Call Mapping

You can map the incoming calls of a DID number to an Auto Attendant, Extension, or Hunt Group. You can also set a default target for all the DID numbers belonging to a service provider.

Inbound Call Mapping Field Descriptions

Field	Descriptions
Service Provider	Names of the service providers are listed by default.
Default Target	Choose a default target for all incoming calls belonging to a service provider. If there is a registered number that is not assigned with any target type, then the incoming calls are place on the default target set for the service provider. Choose one of the following: Hunt Group, Auto Attendant, Extension.
Provider Send Digits	Specify the number of digits sent by the service provider. Range is from 2 to 15 and All Digits.
Registered Number	The DID numbers that are registered to SIP Trunk and Line Cards are listed by default.

Target Type	Choose 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.

Target Number	Choose a number from the drop-down list.

Outbound Caller ID

You can 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.

Outbound Caller ID Field Descriptions

Field	Description
Service Provider	Names of the service providers are listed by default.
Default Outbound DID	Choose a default target for all outgoing calls of the service provider. An extension without an assigned DID number displays the default outbound DID number configured for the service provider.
Mapped Extension	Displays the list of extensions available for the site.
Caller ID	Choose a DID number to be displayed on the called phone when an outbound call is placed from the extension within the organization.

System Operator

You can configure a number to be reached when a caller dials zero after listening to the personal voicemail box greeting.

System Operator Field Descriptions

Field	Description
Target Type	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.
Target Number	Choose the number corresponding to the target type selected from the **Target Number** drop-down list.

Features Field Descriptions

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.

Click **Add New Paging Group** to add custom paging groups.

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

Music on Hold Settings

Field	Description
Music on Hold (MoH)	MoH allows you to play audio for incoming and outgoing calls placed on hold. You can play the default audio file or upload a new audio file. To select a new audio file, 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.

Maintenance Schedule Field Descriptions

Maintenance Schedule allows you to designate a 2 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. Select the day and time according to your organization’s preference.

You can also 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. Check the "Back Up" check box for the days when you want the site to back up the configurations.

Note The previous five consecutive backups are stored in the BE4000 portal.

Licensing

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