Configuring the Supplementary Features

The following chapter explains how to configure voice gateway SIP Line Side features such as Directed Call Park, Call Pick Up, Call Transfer and so on. To provision these features, you must configure outbound VOIP Dial-peer, Pots Dial-peer, Voice Card, and SIP which is described in this chapter.

Configure FXS Ports for Supplementary Services

To handle supplementary services for Foreign Exchange Station (FXS) ports, the event handler handles the hookflash or onhook events. Additionally, the event handler also sends events to call control and triggers the supplementary service on SIP SPI. However, currently, FXS ports do not register on CUCM as SIP endpoints. To ensure the FXS port are registered as a SIP endpoints, make sure that:

- Each configured FXS ports is registered to CUCM. The CUCM creates the database for proper call routing based on the registered endpoint.
- The SIP stack adds or modifies SIP headers content to a proper interface with CUCM and enables new features such as directed call retrieval, call pick-up, and so on.

The FXS ports for Supplementary Services supports CUCM verions 14SU3, version 15, and later. However, CUCM-controlled endpoints with auto configuration can be enabled only on CUCM version 15 and later.

Note You must use the **no local-bypass** command for all the media in this configuration.

Call Transfer

The call transfer status includes the following concepts:

- Hookflash: A hookflash is a brief interruption in the loop as the system places the active call on hold.
- On hook: This option completes the call transfer.

The following table describes the call transfer action.

Table 1: Supported Call Transfer Action

State	Action	Result	Response on FXS Line
Active call	Controller hookflash	Held call	Second dial tone
Held call and outgoing dialed, alerting, and active call	Controller on hook	Held call and active call transferred	Transfer

Three-Way Conference

A three-way conference call allows three people to participate in a single phone session. The following table describes the three-way conference action.

Table 2: Supported Three-Way Conference Action

State	Action	Result
Active Call	First party hookflash	Held call
First party held and second party active	Active call hookflash	First and second calls are bridged
Three-way conference	Controller on hook	Both call legs torn down
Three-way conference	First called party on hook	Call between controller and first called party terminated. Call between controller and second called party remains active.
Three-way conference	Second called party on hook	Call between controller and second called party terminated. Call between controller and first called party remains active.
Three-way conference	Controller hookflash	Call between controller and second called party terminated, call between controller and first called party remains.

Restrictions for Configuring the Supplementary Services

The following functionalities are not supported for this configuration:

- Only one line number per FXS port is supported, and shared line is not supported.
- The line side SIP endpoints are controlled by one CUCM only. Switch over and switch back are not supported.
- Non-DSAPP-controlled devices are not supported. You must configure 'service dsapp' before you configure pots dial-peer and voip dial-peer.

- You cannot combine IPv4 and IPv6 in this configuration.
- The SIP analog calls go through the CUCM, and hairpin calls are not supported.
- SIP analog ports signaling for failover between the CUCMs is not supported.
- 3-way conference only supports G711 codec.
- Media recording, overlap dialing, secure calls, failover, and fallback are not supported.
- In the CUCM web interface, you can add more than one line under **Phone Configuration**. However, only the first line will be associated with the phone, and the rest of the lines will not be applied

Configuring the Device Control Session Application

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `vg410> enable`	Enables privileged EXEC mode. Enter the password, if prompted.
Step 2	**configure terminal** **Example:** `vg410# configure terminal`	Enters the global configuration mode.

Step 3	**application global service default dsapp**
	Example:
	`vg410(config)# application`
	`vg410(config-app)# global`
	`vg410(app-global)# service default dsapp`
		(Optional) Enables the new hookflash functionality globally. Device Control Session Application (DSAPP) application global service default drives these hookflash features and it must be configured for new bookflash functionality for an application framework module in IOS. DSAPP can be configured globally or on a dial-peer basis.
		Note This is a global configuration command. After you configure this command, all the calls are impacted. Even a FXO call will be controlled by DSAPP application which can lead to a failure. If the gateway is controlled by a DSAPP application, it is not recommended to make DSAPP as the default call controler.

Step 4	**param dial-peer <number>**
	Example:
	`vg410(config)# application`
	`vg410(config-app)# service dsapp`
	`vg410(app-global)# param dial-peer 100`
		If multiple dial-peer matches are made for the destination-pattern, dial-peer 100 command is used.
		Note When you configure DSAPP on a dial-peer basis,specify a VOIP dial-peer for any outbound call. If all outbound calls that use the hookflash functionality are on the same server, it is recommended to use the param dial-peer command.

	Command or Action	Purpose
Step 5	**param callWaiting <string>** **Example:** vg410(config)#application vg410(config-app)# service dsapp vg410(app-global)# param dial-peer 100 vg410(app-global)# param callWaiting TRUE	Enables the call waiting feature.
Step 6	**param callConference <string>** **Example:** vg410(config)# application vg410(config-app)# service dsapp vg410(app-global)# param dial-peer 100 vg410(app-global)# param callWaiting TRUE vg410(app-global)# param callConference TRUE	Enables the call conference feature.
Step 7	**param callTransfer <string>** **Example:** vg410(config)# application vg410(config-app)# service dsapp vg410(app-global)# param dial-peer 100 vg410(app-global)# param callWaiting TRUE vg410(app-global)# param callConference TRUE vg410(app-global)# param callTransfer TRUE	Enables the call transfer feature.

Configuring the Outbound Voip Dial-peer

Outbound dial-peer is configured like regular voip dial-peer for SIP. In addition to the parameters required, the following configurations are required:

- service dsapp: Specifies that the dial-peer is controlled by a DSAPP application
- session transport tcp: Specifies that only TCP signaling is supported
- voice-class sip extension gw-ana: Indicates that this parameter is used to interop with CUCM
- voice-class sip bind control source-interface GigabitEthernetx/y/z: Indicates that this interface's mac address is the base mac.
- dual tone multifrequency (DTMF): Specifies how a Session Initiation Protocol (SIP) gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network. This feature supports **rtp-nte** DTMF relay mechanisms for the SIP dial peers.

Here is a sample outbound voip dial-peer configuration:

```
dial-peer voice 714281111 voip
service dsapp
destination-pattern .+
session protocol sipv2
session target ipv4:172.16.0.0
incoming called-number 7141116...
voice-class sip bind control source-interface GigabitEthernet0/0/0
codec g711ulaw
```

Note G711 is the only codec supported for conference calls. Hence it is recommended that you add this codec for conference calls.

The following is a sample configuration for DTMF relay:

```
dtmf-relay method1 [...[method6]]
dtmf-relay rtp-nte
```

Configuring POTS Dial-peer

Plain Old Telephone Service (POTS) dial peers retain the characteristics of a traditional telephony network connection. POTS dial peers map a dialed string to a specific voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.

You can configure the POTS dial-peer feature by using the **dial-peer voice** command. In addition to the parameters required, you can also configure the following commands under POTS dial-peer to interpret hookflash (HF) and to interop with CUCM:

- service dsapp: Specifies this dial-peer is to be controlled by the DSAPP application
- voice-class sip extension gw-ana: Indicates that this parameter is used to interop with CUCM

See the following sample configuration of the POTS dial-peer feature here:

```
dial-peer voice 19993000 pots
service dsapp
destination-pattern 2124506300
voice-class sip extension gw-ana
port 3/0/0
```

Configuring Voice-card and SIP

When you configure the voice-card, all the traffic should go through the CUCM. Hairpin calls are not supported. You have to execute the **no local-bypass** command for the voice-card that have FXS SIP endpoints.

For FXS SIP endpoints to register, configure the **registrar IP address** command under the sip-ua mode and use the TCP as the transport type. Note that UDP protocal is not supported.

```
!
voice-card 3/0
no local-bypass
no watchdog
!
!
sip-ua
registrar ipv4:172.16.0.0 expires 3600 tcp
protocol mode dual-stack
!
```

Enabling Device Control Session Application Line features

To register to CUCM as a SIP endpoint, and to distinguish line feature from trunk, you should configure the **dsapp line** command.

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `vg410> enable`	Enters the privileged EXEC mode. Enter the password, if prompted.
Step 2	**configure terminal** **Example:** `vg410# configure terminal`	Enters the global configuration mode.

Step 3	**dsapp line**
	Example:
	`vg410(config)#`
	`vg410(config)#dsapp line`
	`vg410(config)#`
		Specifies the format of each call feature.
		Note If you do not configure the **dsapp line** command, the gateway acts like a SIP trunk and the analog phones might not register as SIP endpoints. Further, you cannot configure the Feature Access Code (FAC). You must run the **dsapp line** command to use the SIP line features.

Configuring Feature Access Code

The **dsapp line feature access-code** command invokes the feature to translate the Feature Access Code (FAC) to the format that the CUCM understands. If you do not configure this command, the whole FAC digits are sent to the CUCM and may not invoke features. You can also change the default FAC in the sub-mode.

Analog phones do not have soft keys. The required supplementary service features are invoked through FAC. By default, the FAC has '**' prefix which can be changed using the CLI command.

```
vg410(config)#dsapp line feature access-code
vg410(config-dsappline-fac)#prefix *#
vg410(config-dsappline-fac)#cancel-call-waiting **4
vg410(config-dsappline-fac)#exit
vg410# show dsapp line feature codes
dsapp line feature access-code
prefix *#
call forward all *#1
call forward cancel *#2
pickup local *#5
pickup group *#7
pickup direct *#6
cancel-call-waiting **4
last-redial *#3
```

If you don't configure the **dsapp line feature access-code**, the voice gateway does not translate the FAC to the format that the CUCM understands. The whole FAC digits is sent to the CUCM.

After the FAC is disabled and re-enabled, all the FAC and prefix are rolled back to the default values.

```
vg410(config)#no dsapp line feature access-code
Feature access-code disabled
vg410(config)# do show dsapp line feature codes
dsappline feature access-code disabled
vg410(config)# dsapp line feature access-code
```

```
vg410(config-dsappline-fac)#do show dsapp line feature codes
dsapp line feature access-code
prefix **
call forward all **1
call forward cancel **2
pickup local **5
pickup group **7
pickup direct **6
cancel-call-waiting **9
last-redial **3
vg410(config-dsappline-fac)# do show run | b dsapp line
dsapp line
!
dsapp line feature access-code
!
```

Auto Configuration

Auto configuration of SIP line features allows you to automatically configure the dial peers to set the endpoint to a SIP line. The auto configuration procedure adds the dial peers for each of the endpoint that you have configured on CUCM.

For CUCM-controlled SIP analog endpoints, you must perform configurations on the CUCM as well as the voice gateway. You must first perfrom the configuration on the CUCM, and after this configuration is complete, the voice gateway allows you to perform the configurations on the voice gateway.

For the auto configuration, initiate the configuration from the voice gateway and download the resulting configuration file. The XML configuration file is pushed from the CUCM to the gateway. Subsequently, the gateway parses the XML file and configures the pots dial-peer as per the configuration specified in the file.

Auto configuration is supported on CUCM version 15 and later.

Important Auto configuratioin will automatically be configured once you initiate the configuration, even if there is an active ongoing call. It is highly recommended that you initiate this configuration during non-operating hours.

Enabling the Auto Configuration

For the auto configuration to work, you must first specify the CUCM to the SIP line. Doing so indicates that the CUCM is the configuration server to the SIP line. To perform this step and enable the SIP line auto-configuration feature, run the **ccm-manager sipana auto-config local** command.

Then, run the **ccm-manager config server** command. This command initiates a download request of the configuration file. After the file is downloaded from the CUCM server, the XML file is parsed to determine the number of ports that are configured on the CUCM and the corresponding port IDs. The auto configuration then processes all the port information before configuring the corresponding dial-peers to set the endpoint to a SIP line. The dial-peers are added for each of the endpoints that are configured on CUCM.

Note For DSAPP auto-configuration, only pots dial-peer is auto configured. You must manually configure the outbound dial-peer and the voice card.

```
!
ccm-manager sipana auto-config local GigabitEthernet x/y/z
```

```
!
ccm-manager config server x.x.x.x
```

Here, GigabitEthernet x/y/z is the interface that is used for the SIP signaling.

Sample Configuration

```
!
ccm-manager sipana auto-config local GigabitEthernet0/0/1
!
ccm-manager config server 172.19.156.84

!
```

Verifying the Device Control Session Application Configuration

Use the following commands to verify the the DSAPP configuration:

- show dsapp line device summary
- show dsapp line feature codes
- show ccm-manager config-download

The **show dsapp line device summary** command shows whether the FXS ports are successfully registered to the CUCM as SIP endpoints.

```
vg410# show dsapp line device summary
Total Devices: 3
Port Device Registration Dev Directory Last Number
Identifier Name           State        Type  Number     Dialed
---------- --------------- ------------- ------- ----------- -----------
0/0/xx    ANDD309DD761600 REGISTERED   ALG   2124506300 Not Avail
0/0/xx    ANDD309DD761601 REGISTERED   ALG   2124506301 Not Avail
0/0/xx    ANDD309DD761602 UNREGISTERED ALG   2124506302 Not Avail
router#
```

The **show dsapp line feature codes** command shows whether FAC is enabled and displays the feature codes.

```
vg410# show dsapp line feature codes
dsapp line feature access-code
prefix **
call forward all **1
call forward cancel **2
pickup local **5
pickup group **7
pickup direct **6
cancel-call-waiting **9
last-redial **3
```

The show ccm-manager config-download command provides download status and history of the auto-configuration.

```
vg410# show ccm-manager config-download
SIP Line Side Analog auto-configuration status
===============================================================
Registered with Call Manager: Yes
Local interface: GigabitEthernet0/0/0 (2c5a.0fc8.8b70)
Current version-id: 1541004382-f60b9ac2-ce5b-439e-92e5-02b62e26d15c
Current config applied at: 16:47:40 UTC Aug 18 2023
Gateway downloads succeeded: 2
Gateway download attempts: 2
Last gateway download attempt: 16:47:40 UTC Aug 18 2023
```

```
Last successful gateway download: 16:47:40 UTC Aug 18 2023
Current TFTP server: 172.19.156.84
Gateway resets: 1
Managed endpoints: 3
Endpoint downloads succeeded: 6
Endpoint download attempts: 6
Last endpoint download attempt: 16:47:40 UTC Aug 18 2023
Last successful endpoint download: 16:47:40 UTC Aug 18 2023

Endpoint resets: 0
Endpoint restarts: 0
Configuration Error History:
```