Analog FXS port SIP Registration with CUCM

Version: 10th February 2020

Introduction:

This document helps configure FXS ports as SIP endpoints registered to Cisco Unified Communications Manager (CUCM) in order to support supplementary services on SIP Endpoints.

Prerequisites:

Cisco recommends to have knowledge of these subjects

SIP protocol
Foreign Exchange Station (FXS) ports
Cisco Unified Communications Manager (CUCM)
Cisco Analog Voice Gateway (VG Series)

Requirements:

The FXS Ports for Supplementary Services feature is supported on Cisco VG450 Voice Gateway and Cisco 4461 ISR. The FXS ports for Supplementary Services supports CUCM version 12.5.1 SU1 or later with IOS XE 16.12.1 and above.

Supported Features:

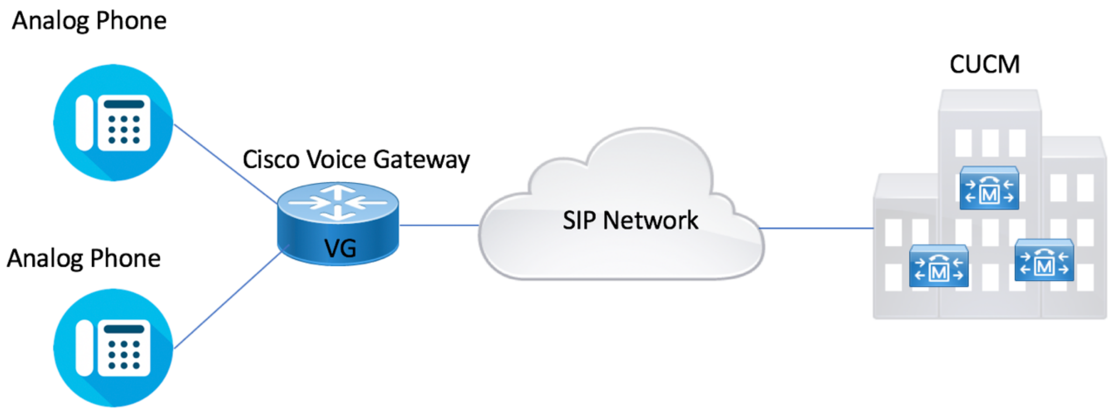
The following supplementary services are supported.

Call Hold
Call Waiting
Call Transfer (unattended/attended)
Call Forward no Answer
Audio Message Waiting Indicator
Call Forwarding Unrestricted
Call Forwarding Busy
Directed Call Park
Directed Call Pickup
Call Pickup Group
Three way Conference

Configure:

To implement supplementary services for Foreign Exchange Station (FXS) ports the call control server (CUCM) should be able to subscribe to the hookflash or onhook events .This requires FXS ports to be registered to CUCM as SIP endpoints. The use of SIP over SCCP facilitates features such as SIP Header modification , endpoint based call routing and enables new features such as directed call retrieval.

Network Diagram:



Configuration steps:

This section describes the configuration required for this feature to work:

Configuring the Device Control Session Application

DSAPP (Device control Session Application) is the application that drives these Hook Flash features. It can be configured globally or on a dial-peer basis.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **application**
4. **global**
5. **service default dsapp**
6. **param dialpeer** *number*
7. **param callWaiting** *string*
8. **param callConference** *string*

9. **param callTransfer** *string*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** Router# configure terminal	Enters global configuration mode.
Step 3	**application global service default dsapp** **Example:** router(config)#application router(config-app)#global router(app-global)#service default dsapp	(**Optional**) Enables the new hookflash functionality globally. Device Control Session Application (DSAPP) 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** Only required if all calls on the gateway are to be controlled by DSAPP else it may lead to call failure.
Step 4	**param dialpeer** *number* **Example:** router(config)#application router(config-app)#service dsapp router(app-global)#param dialpeer 100	If multiple dial-peer matches are made for the destination-pattern, **dialpeer** 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. When multiple matches are possible on hookflash, enable **peer parameters callXXXX TRUE** for DSAPP to interpret hookflash to SIP supplementary service messages.
Step 5	**param callWaiting** *string* **Example:** router(config)#application router(config-app)#service dsapp router(app-global)#param dialpeer 100 router(app-global)#param callWaiting TRUE	Enables call waiting feature.
Step 6	**param callConference** *string*	Enables call conference feature.

	Command or Action	Purpose
	Example: router(config)#application router(config-app)#service dsapp router(app-global)#param dialpeer 100 router(app-global)#param callWaiting TRUE router(app-global)#param callConference TRUE	
Step 7	**param callTransfer** *string* **Example:** router(config)#application router(config-app)#service dsapp router(app-global)#param dialpeer 100 router(app-global)#param callWaiting TRUE router(app-global)#param callConference TRUE router(app-global)#param callTransfer TRUE	Enables 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 this dial-peer will be controlled by dsapp application
- **session transport tcp** – specifies only TCP signaling is supported now
- **voice-class sip extension gw-ana** – used to interop with CUCM
- **voice-class sip bind control source-interface GigabitEthernet0/0/1** – need this interface's mac as the base mac

dial-peer voice 714281111 voip

```
 service dsapp
 destination-pattern .+
 session protocol sipv2
 session transport tcp
 session target ipv4:172.16.10.10
 incoming called-number 7141116...
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 codec g711ulaw
 no shut
```

Note- G711 is the only codec supported for conference calls. It is recommended to add this command.

Configuring Pots Dial-peer

You can configure the pots dial-peer like a regular pots dial-peer for FXS. In addition to the parameters required, you have to configure the following command under pots dial-peer to interpret hook flash correctly and interop with CUCM:

- **service dsapp**—specifies this dial-peer to be controlled by DSAPP application.
- **voice-class sip extension gw-ana**– this parameter is used to interop with CUCM.

```
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 and the hairpin calls are not supported. You have to configure no local-bypass command for the voice-card that have FXS SIP endpoints.

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

```
voice service voip
sip
bind control source-interface GigabitEthernet0/0/0
session transport tcp
no shut
!
voice-card 3/0
 no local-bypass
 no watchdog
!
!
sip-ua
 registrar ipv4:172.16.10.10 expires 3600 tcp
 protocol mode dual-stack
!
```

Note- SIP service should be shut down before configuring the protocol mode. After configuring the protocol mode as dual-stack, SIP service should be reenabled.

Enabling Device Control Session Application Line features

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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dsapp line**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dsapp line Example: router(config)# router(config)#dsapp line router(config)#	Specifies the format of each call feature. Note If you do not configure the **dsapp line** command, the gateway will act like a SIP trunk and the analog phones may not register as SIP endpoints. Also, you cannot configure the FAC. Ensure to configure the **dsapp line** command to enable the FXS for SIP supplementary services.

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, prefix of the FAC is '**' and it can also be changed using the CLI command.

```
router(config)#dsapp line feature access-code
router(config-dsappline-fac)#prefix *#
router(config-dsappline-fac)#cancel-call-waiting **4
router(config-dsappline-fac)#exit
router# 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 the **dsapp line feature access-code** is not configured, 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.

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

 dsappline feature access-code disabled


router(config)#dsapp line feature access-code

router(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


router(config-dsappline-fac)#do show run | b dsapp line

dsapp line

!

dsapp line feature access-code
```


Auto Configuration

To enable the auto-configuration, use the **ccm-manager sipana auto-config local** command. To get the XML configuration file, use the **ccm-manager config server** command to download the configuration file from the CUCM TFTP server. Configurations are needed on both CUCM and voice gateway. CUCM needs to be configured first, then those configurations can be pushed to the voice gateway.

!

ccm-manager sipana auto-config local GigabitEthernetx/y/z

!

ccm-manager config server 172.xx.0.0

Note- Auto-Config only adds the dialpeers for each endpoint configured on CUCM. All other required SIP CLI commands need to be added.

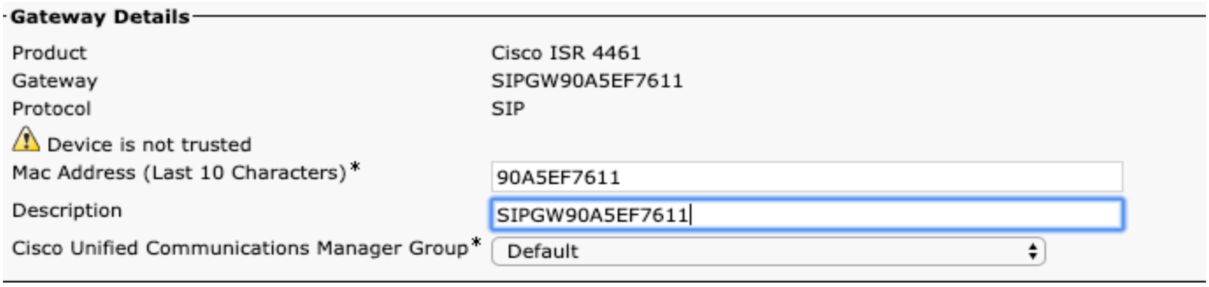
CUCM Configuration

1. Navigate to **Device> Gateway>Add New>Gateway Type**.

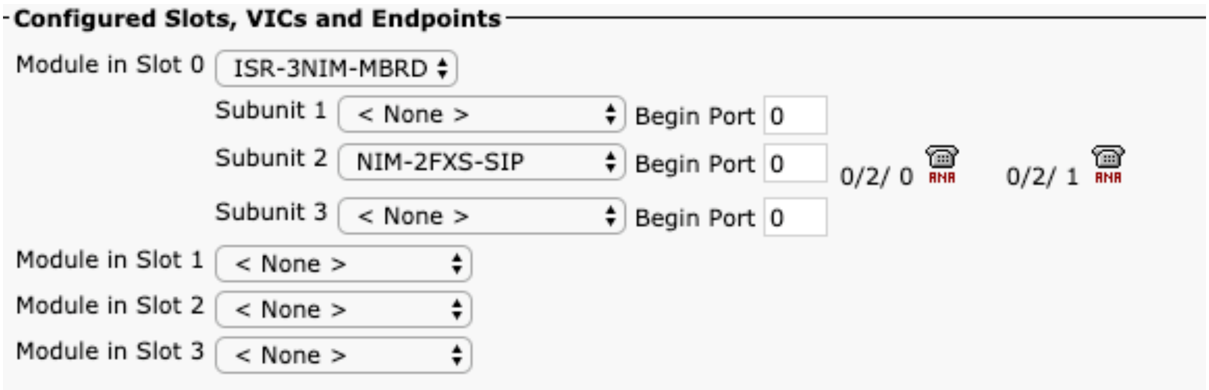
VG450 is used in this example.

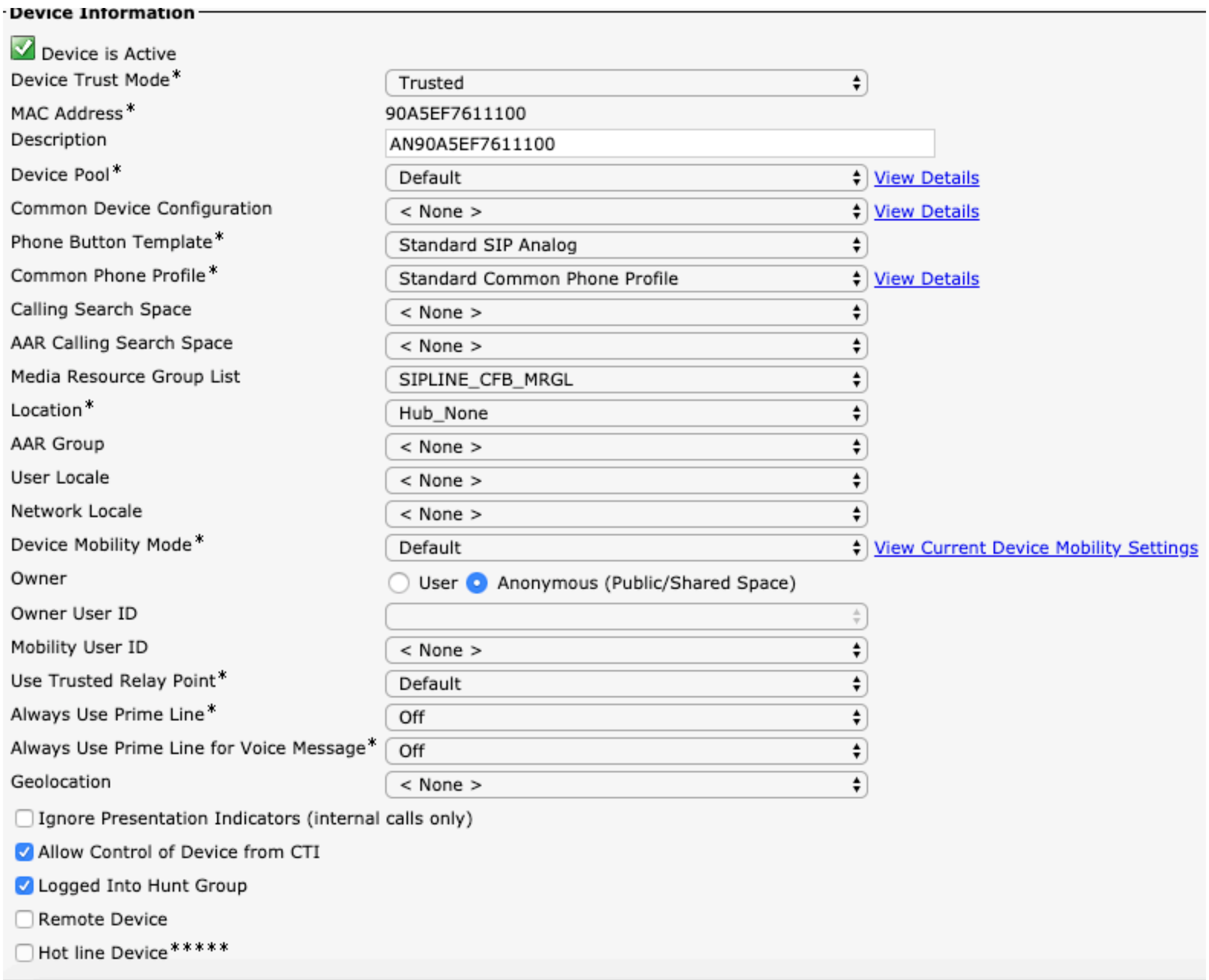
Select **SIP** as the protocol and click **Next**

2. Enter the mac address of the interface used in sip bind control.

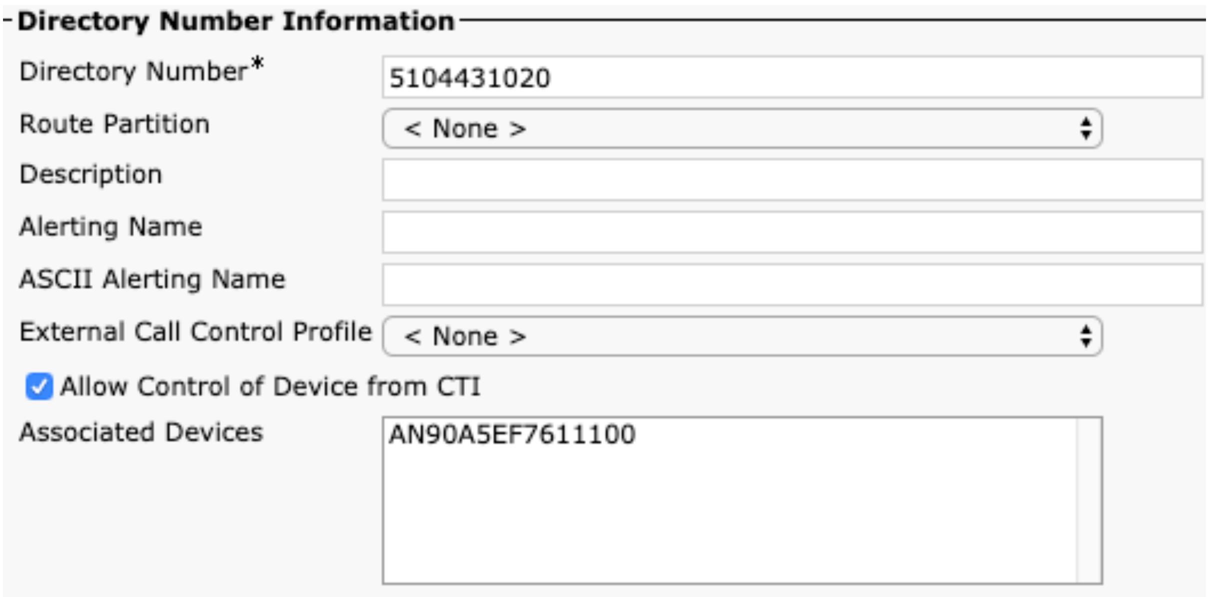


3. Configure the voice module and indivudal voice ports.

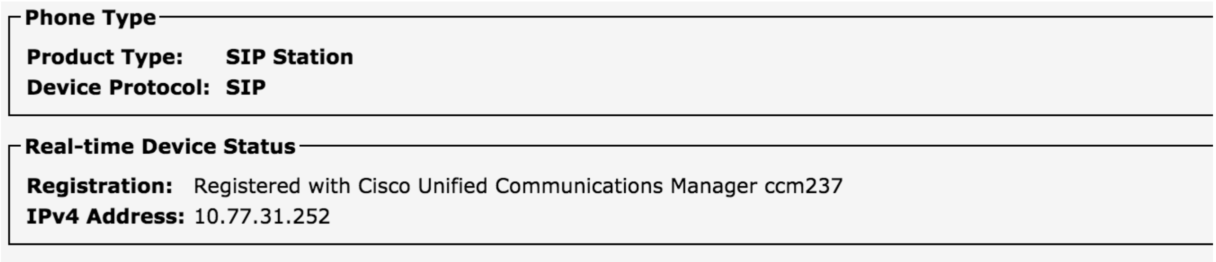




4 Add a **directory Number > Save>Apply config**.



The Analog port would now show as registered



Limitations:

1. Shared line on SIP registered FXS port is not supported.
2. Line side SIP endpoints can be controlled by one CUCM.
3. CUCM failover for analog SIP phones is not supported.
4. Music on hold is not supported.
5. Third-party call agents are not supported.
6. Only non-secure calls are supported.

Verify:

Use the following commands to verify 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.

```
router#show dsapp line device summary
Total Devices:        3
Port       Device          Registration  Dev     Directory   Last Number
Identifier Name            State         Type    Number      Dialed
---------- --------------- ------------- ------- ----------- -----------
3/0/0      ANDD309DD761600 REGISTERED    ALG     2124506300  Not Avail
3/0/1      ANDD309DD761601 REGISTERED    ALG     2124506301  Not Avail
```

```
3/0/2      ANDD309DD761602 UNREGISTERED  ALG     2124506302  Not Avail
router#
```

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

router#**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.

Art_Utah_73#**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 Oct 31 2018

Gateway downloads succeeded: 2

Gateway download attempts: 2

Last gateway download attempt:  16:47:40 UTC Oct 31 2018

Last successful gateway download: 16:47:40 UTC Oct 31 2018

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 Oct 31 2018

Last successful endpoint download: 16:47:40 UTC Oct 31 2018

Endpoint resets: 0

Endpoint restarts: 0

Configuration Error History:

Troubleshoot:

For registration issues capture CUCM SDL/SDI traces and run the following debugs on the gateway

debug voip application session
debug ccsip messages
debug ccsip states
debug ccsip error

For auto-configuration issues run the debug

debug ccm-manager config-download all

Related Information
Configuring the Cisco Fourth-Generation T1/E1 Voice and WAN Network Interface Module