



Configuring SIP Trunks between Cisco Unified Communications Manager Express 7.1 and Avaya Meeting Exchange Enterprise Edition 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses a SIP trunk to connect the Cisco Unified Communications Manager Express 7.1 to Avaya Meeting Exchange Enterprise Edition 5.2.

For the sample configuration, Cisco Unified Communications Manager Express 7.1 runs on a Cisco 2811 router while the Avaya Meeting Exchange Application and Media Server run on separate S8510 servers.

1. Introduction

These Application Notes present a sample configuration for a network that uses a SIP trunk to connect the Cisco Unified Communications Manager Express 7.1 to Avaya Meeting Exchange Enterprise Edition 5.2. As shown in **Figure 1**, separate Avaya Meeting Exchange Application and Media Servers are used in the sample configuration. The Application Server is responsible for SIP signaling while the Media Server is used for multiplexing and streaming RTP to the conference participants. The Cisco Unified Communications Manager Express supports a Cisco 7961G IP Phone (SCCP), Cisco 7941G IP Phone (SIP), Cisco IP Communicator Softphone (SCCP) and an analogue PSTN phone (FXS). A SIP trunk is used to connect Avaya Meeting Exchange Enterprise Edition S6200 with Cisco Unified Communications Manager Express (CUCME) over the LAN. Signaling messages are carried over the TCP based SIP trunk while DTMF is transmitted within the RTP stream using RFC2833 compliant messages. While TLS is available on the Avaya Meeting Exchange Enterprise Edition Application Server, it is not supported on Cisco Unified Communications Manager Express in release 7.1. An analogue phone is attached to a PSTN simulator which is connected with an E1 PRI trunk to the sample telephony network. This configuration enables PSTN users to participate in a conference with other enterprise users on the private network. The Telephones are configured in the 500x extension range, while the conference access number (DNIS) on the Avaya Meeting Exchange Enterprise is set to 44444. For the sample configuration, Cisco Unified Communications Manager Express runs on Cisco 2811 router, while Avaya Meeting Exchange Enterprise Application and Media Server run on two separate Avaya S8510 servers.

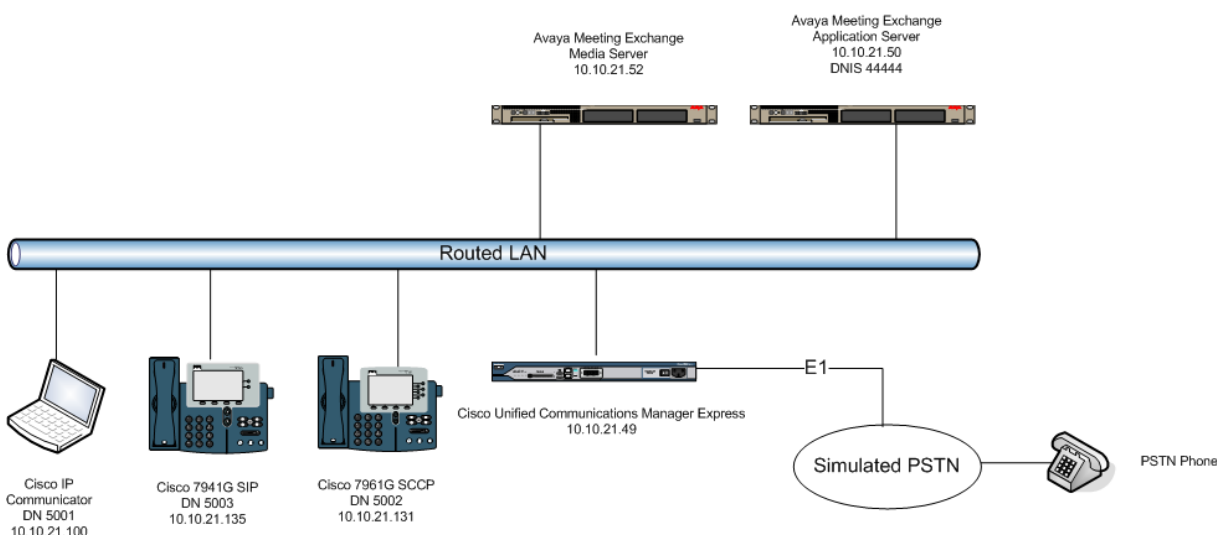


Figure 1: Avaya Meeting Exchange Enterprise with CUCME

These Application Notes will focus on the configuration of the SIP trunk and call routing. Detailed administration of the endpoint telephones and E1 PRI trunk will not be described (see the appropriate documents listed in **Section 8**).

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software Version
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition S6200 R5.2 Build 5.2.0.0.22 (GA)
Cisco 2811 Router	Cisco Unified Communications Manager Express v7.1 IOS version: c2800nm-adventerprisek9_ivs-mz.124-24.T.bin CUCME package version: cme-124-24T.zip
Cisco 7961G IP Phone	SCCP 8-4-1-23
Cisco 7941G IP Phone	SIP 8-4-1-23
Cisco IP Communicator	SCCP 7.0.3.3

Table 1: Version Numbers of Equipment and Software

3. Configure Avaya Meeting Exchange Enterprise Application Server

This section describes the steps for configuring SIP trunks between Avaya Meeting Exchange Enterprise Application Server and Cisco Unified Communications Manager Express (CUCME). This configuration will allow both moderators and participants to access a conference and also for operators to dial participants who have difficulties accessing a conference. It's assumed, that Meeting Exchange is installed, configured and licensed as per [3]. The following steps describe the administrative procedures for configuring Meeting Exchange server:

- Configure SIP Listener
- Configure Dialout
- Configure DNIS Mappings
- Configure Incoming SIP URI Conversion Rules
- Restart the Conference Bridge

The following instructions assume the user is logged in to the Meeting Exchange server Linux console using SSH.

3.1. Configure SIP Listener

The SIP signaling messages in the sample configuration are transmitted between the MX Application Server and a SIP enabled dial-peer of the CUCME over a TCP channel. Configure the following settings to enable SIP connectivity on the MX Application Server:

- Edit **/usr/ipcb/config/system.cfg** using the Linux vi tool or download the file to your local machine using a Secure Copy Protocol (SCP) client (i.e.: WinSCP) for editing.
 - Add the IP address of the MX Application Server: **IPAddress=10.10.21.50** as shown below
 - Add a line to populate the From Header Field in SIP INVITE messages. The following SIP URI will be displayed when the conference operator calls a participant: **MyListener=sip:6000@10.10.21.50:5060;transport=tcp**
 - Add a line to provide a SIP Device Contact address to use for acknowledging SIP messages: **respContact=<sip:6000@10.10.21.50:5060;transport=tcp>**
 - Add the following lines to set the Min-SE timer to **900** seconds in SIP INVITE messages from the MX Application Server:
sessionRefreshTimerValue= 1800
minSETimerValue= 900

Note: The values for the sessionRefreshTimerValue and the minSETimerValue are defined in seconds and should be provisioned to be greater than or equal to the value used by CUCME. This setting is necessary to enable Dial-Out from the MX Application Server to the Cisco IP Phones.

```
# ip address of the server
IPAddress=10.10.21.50

# request we will be listening to
MyListener=sip:6000@10.10.21.50

# if this setting is populated will Overwrite the contact field in responses
respContact=<sip:6000@10.10.21.50:5060;transport=tcp>
MaxChannelCount=3200

# SIP settings
sessionRefreshTimer=900
minSETimerValue=900
```

3.2. Configure Dialout

The FQDN of the CUCME must be configured on MX Application Server for dialout to work. The MX Application Server must be able to resolve the FQDN to an IP address, by either using its' own hosts file or an external DNS server. For the sample configuration an external DNS server was authoritative for the **cucme.avaya.com** zone and had the following entry in it's zone file:

```
cucme.avaya.com      IN      A      10.10.21.49
```

Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor. Add the following line to the file to route outbound calls from the MX Application Server to the extension range administered on the CUCME dial plan.

```
5*                  sip:$0@cucme.avaya.com:5060;transport=tcp  default
```

3.3. Configure DNIS Mappings

The DNIS is the number that the phone users dial to access a conference. To map DNIS entries, run the **cbutil** utility on MX Application Server. Add **44444** as a new DNIS entry using the command shown below.

```
[mx6200-a ~]# cbutil add 44444 0 247 1 N SCAN
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

At the command prompt, enter **cbutil list** to verify the DNIS entries provisioned.

```
[mx6200-a ~]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

DNIS	Grp	Msg	PS	CP	Function	On Failure	Line Name	Company Name	Room	Start	Room End
44444	0	247	1	N	SCAN	ENTER			0		0

3.4. Restart MX Conference Bridge

After the configuration changes are made, restart the MX Application Server. Log in to the MX Application Server using the **dcbmaint** account.

Issue the **dcbmaint** command. The **System Maintenance Main Menu** screen is displayed as shown below.

```
MXAPP2 -- station 257
Avaya, Inc.                      Audio Conferencing System
P/N: S0700502  Revision: 00      Copyright 2007 Avaya, Inc.

System Maintenance Main Menu
Network Configuration
FDAPI Configuration
LAN Configuration
Administrator Menu
Re-Initialization
System Shutdown
Transmission Level
EXIT
```

Navigate to **Re-Initialization**.

```
MXAPP2 -- station 257
Avaya, Inc.                      Audio Conferencing System
P/N: S0700502  Revision: 00      Copyright 2007 Avaya, Inc.

System Maintenance Main Menu
Network Configuration
FDAPI Configuration
LAN Configuration
Administrator Menu
Re-Initialization
System Shutdown
Transmission Level
EXIT
```

Press **Enter** and at the prompt type **yes**.

```
MXAPP2 -- station 257
Avaya, Inc. Audio Conferencing System
P/N: S0700502 Revision: 00 Copyright 2007 Avaya, Inc.

System Re-Initialization
ARE YOU SURE? (yes/NO): ☐
```

The **dcbmaint** utility terminates and the following message is displayed.

```
[dcbmaint@MXAPP2 sroot]$ dcbmaint
System Shutdown in Progress...
Please Stand By.

[dcbmaint@MXAPP2 sroot]$ ☐
```

4. Configure Cisco Unified Communications Manager Express

This section describes the steps for configuring the Cisco SCCP and SIP IP phones to access the conference bridge. The Cisco Unified Communications Manager Express is running on a Cisco 2811 equipped with a PVDM-64 DSP module. This module is required for TDM-to-IP calls as an analogue user is connected to a PSTN simulator over an E1 PRI trunk. The 2811 is configured with a single dial-peer for the SIP trunk to the MX Application Server. The following steps describe the administrative procedures for configuring Cisco Unified Communications Manager Express:

- Verify IP Address and Interface of Cisco Router
- Configure SIP Global Settings
 - Bind Interface to SIP Process
 - Change Transport Protocol to TCP
- Configure Codec Types
- Configure SIP dial-peer

The following instructions assume the phones are configured, registered and the user is logged in to the CUCME console using Telnet or SSH as per [3].

4.1. Verify IP Address and Interface of Cisco Router

Select an IP interface on the router which will be used for SIP signaling and RTP media transmission to the MX Application and Media Servers.

```
interface FastEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-FE 0/0$
ip address 10.10.21.49 255.255.255.0
duplex auto
speed auto
```

4.2. Configure SIP Global Settings

Enable SIP to SIP calls to allow Cisco SIP Phones to make calls through the SIP dial-peer. This functionality is disabled by default. Bind the interface selected in **Section 4.1** to the SIP and RTP processes. Change the transport protocol to **TCP** as UDP is the default. The following settings allow the CUCME to send outgoing SIP and RTP packets from the **FastEthernet0/0** interface (10.10.21.49).

```
voice service voip
allow-connections sip to sip
sip
bind control source-interface FastEthernet0/0
bind media source-interface FastEthernet0/0
session transport tcp
```


4.3. Configure Codec Types

Create a group of supported codec types and assign them to a voice class. When a call is made from the Cisco phones to the MX Application Server the SDP part of the SIP INVITE message will contain all four of the below specified codecs with G711ulaw being the most preferred and G729br8 being the least preferred.

```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729r8
  codec preference 4 g729br8
```

4.4. Configure SIP Dial-peer

Create a new dial-peer for incoming and outgoing calls from/to the MX Application Server. In the sample configuration the conference access number is 44444, while the operator access number is 6000 as per **Section 3.1**. When the conference moderator initiates an outbound call (dialout) using the SIP Request URI from **Section 3.2** the CUCME needs to be setup to accept the call and connect it to one of the registered Cisco SIP or SCCP phones, this is why the same dial-peer is used for both incoming and outgoing calls. The dial-peer is configured to allow the user to call back the operator in case of a missed call from **6000**. RFC2833 needs to be used for DTMF relay which is equivalent to the **rtp-nte** setting in Cisco IOS.

```
dial-peer voice 44444 voip
  destination-pattern [46]...T
  incoming called-number 5...
  session protocol sipv2
  session target ipv4:10.10.21.50
  dtmf-relay rtp-nte
  voice-class codec 1
```

5. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Meeting Exchange Enterprise Edition and Cisco Unified Communications Manager Express.

5.1. Verify Avaya Meeting Exchange Enterprise Edition

Verify that the **sipagent** process is running and that number **1** or number **2** is displayed after **/usr/dcb/bin/sipagent** in the following command output on the MX Application Server:

```
[craft@MXAPP1]$ ps -ef | grep -i sipagent
sroot    11079 10966  0 Feb18 ?          00:00:21 /usr/dcb/bin/sipagent 1
craft    25946 25926  0 14:00 pts/1    00:00:00 grep -i sipagent
```

Verify that the MX Application Server is listening for SIP requests on TCP port 5060:

```
[craft@MXAPP1]$ netstat -antu | grep 506
tcp        0      0 0.0.0.0:5060          0.0.0.0:*            LISTEN
```

5.2. Verify Cisco Unified Communications Manager Express

Dial the conference bridge from a Cisco IP (SCCP) Phone. Look for the extension of the phone and the status of the line appearance, that was used to make the call in the output of the following command.

number 5002	The extension of the test phone
CH1	The line appearance that was used to make the call
CONNECTED	The current status of the first line appearance

```
2811-ent#show ephone registered
```

```
ephone-1[0] Mac:0024.E825.96B5 TCP socket:[1] activeLine:0 whisperLine:0 REGISTERED
in SCCP ver 15/12 max_streams=5
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0
reset_sent:0 paging 0 debug:0 caps:10
IP:10.10.21.100 50276 CIPC keepalive 5182 max_line 8 available_line 8
button 1: dn 1 number 5001 CH1 IDLE
Preferred Codec: g711ulaw
```

```
ephone-2[1] Mac:6416.8D51.337B TCP socket:[2] activeLine:1 whisperLine:0 REGISTERED
in SCCP ver 17/12 max_streams=5
mediaActive:1 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0
reset_sent:0 paging 0 debug:0 caps:8
IP:10.10.21.136 25579 7941 keepalive 5036 max_line 2 available_line 2
button 1: dn 2 number 5002 CH1 CONNECTED
Preferred Codec: g711ulaw
Active Call on DN 2 chan 1 :5002 10.10.21.136 20712 to 10.10.21.49 2000 via
10.10.21.136
G711Ulaw64k 160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1 (media path callID 3952 srcCallID 3953)
```

Verify the status of the dial-peer which was used to route the call to the MX Application Server based on the dialed number.

PeerAddress	The number that was dialed to reach the conference bridge
RemoteIPAddress	The IP address of the MX Application Server
RemoteMediaIPAddress	The IP address of the MX Media Server
tx_DtmfRelay	The DTMF relay method

```
2811-ent#sh call active voice called-number 44444
Telephony call-legs: 0
SIP call-legs: 1
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1

GENERIC:
SetupTime=2319544340 ms
Index=1
PeerAddress=44444
PeerSubAddress=
PeerId=44444
PeerIfIndex=36
LogicalIfIndex=0
ConnectTime=2319544560 ms
CallDuration=00:11:46 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=35339
TransmitBytes=5646596
ReceivePackets=35258
ReceiveBytes=5641280
VOIP:
ConnectionId[0xE8BCEAD7 0x25E011DF 0x9105CF6A 0x285767C1]
IncomingConnectionId[0xE8BCEAD7 0x25E011DF 0x9105CF6A 0x285767C1]
CallID=3953
GlobalCallId=[0xE8BCEAD7 0x25E011DF 0x9105CF6A 0x285767C1]
RemoteIPAddress=10.10.21.50
RemoteUDPPort=42208
RemoteSignallingIPAddress=10.10.21.50
RemoteSignallingPort=5060
RemoteMediaIPAddress=10.10.21.52
RemoteMediaPort=42208
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=rtp-nte
FastConnect=FALSE
```

6. Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Conference calls between various telephones on the Cisco Unified Communications Manager Express can be using G.711 Ulaw.
- Proper display of the calling and called party name and number information was verified for all telephones.
- Dialout from the Operator phone to conference participants.
- Invoke conference features through DTMF codes.

7. Conclusion

As illustrated in these Application Notes, Cisco Unified Communications Manager Express 7.1 can interoperate with Avaya Meeting Exchange Enterprise Edition using SIP trunks. The following is a list of interoperability items to note:

- Due to a mismatch between the RTP codec payload types on the CUCME and the MX Application and Media Servers for all variants of G726, iLBC and G722 codecs, conference calls connected successfully, but were dropped immediately by CUCME with a SIP BYE request.

8. Additional References

Avaya Meeting Exchange Support Documents:

- [1] *Administering Meeting Exchange™ 5.2 Servers*, Doc ID 04-603419, 16-Nov-2009, available at <http://support.avaya.com/css/P8/documents/100068644>
- [2] *Administering Meeting Exchange™ 5.2 Applications*, Doc ID 04-603420, 16-Nov-2009, available at <http://support.avaya.com/css/P8/documents/100068646>

CUCME Support Documents:

- [3] *Cisco Call Manager Express (CME) SIP Trunking Configuration Example* - Document ID: 91535, available from <http://support.cisco.com>

©2010 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com