

1. Configuring the CUCMBE

As the CUCMBE is prone to a wide range of configurations and network VoIP setup, we will explain to you the basic configuration and let the CUCMBE managers work out how to integrate it in their own environment. It's out of the scope of the present documentation to teach you how to manage a CUCMBE.

1.1 Prerequisites

Of course you need the CUCMBE appliance, which may come both in VMWare virtualization fashion or in real steel installed one. In either cases we assume that:

- basic services are activated (see the fig. X). Beware: the **Cisco IP Voice Media Streaming App** service **must** be activated in order to create an MTP (Media Termination Point)
- The phones (both SCCP or SIP ones) are correctly configured and functioning
 - You do have a configured **Calling Search Space** (CSS)
 - You do have an accessible **Device Pool**, ready to be connected with PrivateServer
 - You configured a **Route Partition** for making any device accessible
- Your network design let the UDP packets to stream from the PrivateServer to the CUCMBE and vice versa

CM Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco CallManager	Activated
<input checked="" type="checkbox"/>	Cisco Messaging Interface	Activated
<input checked="" type="checkbox"/>	Cisco Unified Mobile Voice Access Service	Activated
<input checked="" type="checkbox"/>	Cisco IP Voice Media Streaming App	Activated
<input checked="" type="checkbox"/>	Cisco CTIManager	Activated

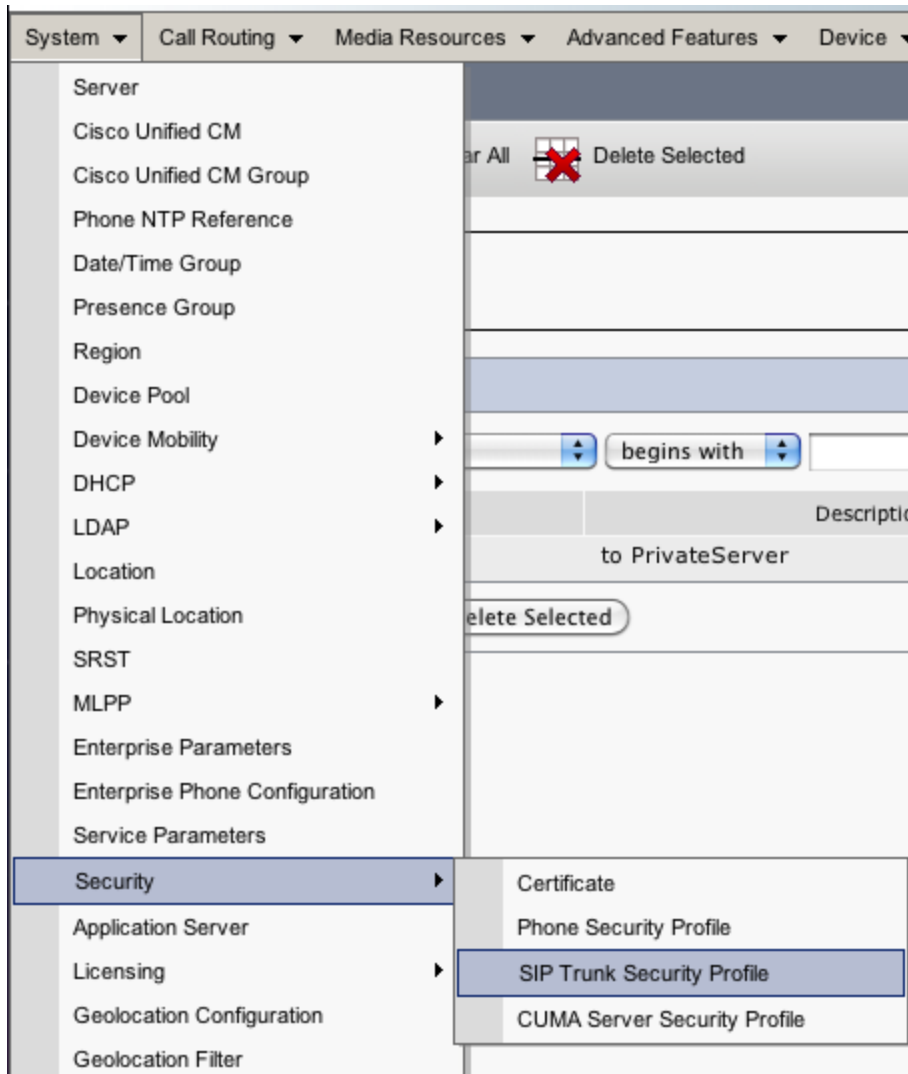
1.2 Basic CUCMBE configuration

Before creating a new SIP Trunk we better define its security profile, that should be *insecure*.

1.2.1 SIP Security Profile

Go to System->Security->SIP Trunk Security Profile.

Make yourself sure that the **Device Security Mode** is set to *Non Secure*, that the **Incoming Transport Type** is *TCP+UDP* and that the **Outgoing Transport Type** is the plain *UDP*.



SIP Trunk Security Profile Informa

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

☐ Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

☐ Enable Application level authorizat

☒ Accept presence subscription

☐ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filterin

Also check the **Accept presence subscription** option.

1.2.2 SIP Trunk Creation

Now we are ready to configure a new SIP Trunk.

Go to the Cisco Unified CM Administration and from its GUI use the **Device->Trunk** menu entry to create a new one.




First select the **Trunk Type** to *SIP Trunk* and the **Device Protocol** to *SIP*, as in fig. X. The **Trunk Service Type** can be *None*.


After you press the **Next** button, you got a long page of configuration to deal with. We are going to give you the essential configuration parameter to make the trunk work with PrivateServer, so you can keep the default options configuration if not else stated.

You can choose your own **Device Name** and **Description**. Please set the **Device Pool** to the one containing the phones you actually want to put in contact with PrivateServer. Also check the **Media Termination Point Required** option. We also check the **Retry Video Call as Audio** for the possible Video phone (PrivateServer doesn't support video calls). Check the **PSTN Access** and the **Run On All Active Unified CM Nodes**. The rest of the options can stay in default values, till the **SIP Information** form (see fig. X), just pay attention to configure the Calling Search Space accordingly with your design where requested.

Trunk Configuration

 Next

Status


 Status: Ready

Trunk Information

Trunk Type* SIP Trunk

Device Protocol* SIP

Trunk Service Type* None(Default)

 Next

SIP Information

Destination

☐ Destination Address is an SRV

Destination Address

1* 192.168.11.50

MTP Preferred Originating Codec* 71

Presence Group* Sti

SIP Trunk Security Profile* Nc

Rerouting Calling Search Space Ge

Out-Of-Dialog Refer Calling Search Space <

SUBSCRIBE Calling Search Space Ge

SIP Profile* Sti

DTMF Signaling Method* RF

Normalization Script

Normalization Script < None >

☐ Enable Trace

Parameter Name

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In the SIP Information form we need to tell the CUCMBE the IP address of our PrivateServer appliance and the Destination Port as well (the latest is 5060). Select the codec 711ulaw and the *Non Secure SIP Trunk Profile* which you should have created before (read above). Use also the *Standard SIP Profile* or just create a suitable one for your need and, most important, set the **DTMF Signaling Method** to the *RFC28323*. Once you're done, just Save it.