## 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			
	Cisco CallManager	Activated			
	Cisco Messaging Interface	Activated			
	Cisco Unified Mobile Voice Access Service	Activated			
	Cisco IP Voice Media Streaming App	Activated			
	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.

System 👻		Call Routing 🔻	Media Reso	urces	• •	Advanc	ed Featur	es 🔻	Device		SIP Trunk Security Profile Informa	
	Server									Name*		
	Cisco Unified CM Cisco Unified CM Group Phone NTP Reference Date/Time Group						Description					
			ar Ai	ar All Delete Selected					Device Security Mode			
										Incoming Transport Type*		
										Incoming mansport rype		
	Presen	ce Group									Outgoing Transport Type	
	Region										Enable Digest Authentication	
	Device	Pool									Nonce Validity Time (mins)*	
	Device	Mobility	•		🔹 begins with 🛟				X.509 Subject Name			
	DHCP		+				-				Incoming Port*	
	LDAP		+						Descriptio			
	Locatio	n				to P	rivateSe	rver			Enable Application level authorizat	
	Physical Location SRST		elet	elete Selected						Accept presence subscription		
			F							Accept out-of-dialog refer**		
	MLPP		•								Accept unsolicited notification	
	Enterprise Parameters									Accept replaces header		
	Enterpr	ise Phone Configur	ation								Transmit security status	
	Service	Parameters									Allow charging header	
	Securit	у	•	Ľ	Cer	tificate					SIP V.150 Outbound SDP Offer Filterir	
	Applica	tion Server			Pho	ne Secur	rity Profile					
	Licensi	ng	•		SIP	Trunk Se	ecurity Pro	ofile		1	Also check the <b>Accept presence subscription</b> op tion.	
	Geoloc	ation Configuration			CU	MA Serve	er Security	Profile	)			
	Geoloc	ation Filter		T						ľ	1.2.2 SIP I FUNK 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.

	Cisco Unified Reporting	
Navigation	<ul> <li>Cisco Unified CM Administration</li> </ul>	Go
ciscoadmin	Disaster Recovery System	ogout
ciscouumin	Cisco Unified Serviceability	ogout
	Cisco Unity Connection Serviceability	
	Cisco Unified OS Administration	
	Cisco Unity Connection Administration	



First select the **Trunk Type** to *SIP Trunk* and the **Device Protocol** to *SIP*, as in fig. X. The **Trunk Service Type** ca n 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	1	
Next		
Status		
U Status: Ready		SIP Information
Trunk Information – Trunk Type*	SIP Trunk	Destination Destination Address is an SRV Destination Address
Device Protocol* Trunk Service Type*	SIP   None(Default)	1*     192.168.11.50       MTP Preferred Originating Codec*     71       Presence Group*     71
- (Next)		SIP Trunk Security Profile*
		Out-Of-Dialog Refer Calling Search Space
		DTMF Signaling Method*
		Normalization Script Normalization Script < None > Enable Trace Parameter Name

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.