

The configuration changes to be done on CUCM prior installation of DGVox as follows:

The screen shots are from CUCM 6.0(1) and there may be slight variation in the screen shots of the latest version CUCM. The step by step configuration remains the same.

1) Add User for Monitoring or Recording Application

Create the application user for monitoring or recording, and the application user must belong to a group with monitoring and recording privileges.

Add an application or end user from Application User Configuration window or the End User

Configuration window.

Use the *User Management > Application User* menu option in CUCM Administration to perform the necessary configuration.

Figure 1 illustrates adding a user for the monitoring or recording application.

Find and List Application Users - Microsoft Internet Explorer		
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System • Call Routing • Media Resources • Voice Mail • Device • Application •	User Management -	Bulk Administration
Find and List Application Users	Credential Policy	Default
- Add New	Credential Policy	
	Application User	
Application User	End User	
Find Application User where User ID begins with 💌	Role	-
No active query. Please enter your search o	User Group	
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	Application User	CAPF Profile
	End User CAPF I	Profile
	SIP Realm	
https://10.89.81.54:8443/ccmadmin/appuserFindList.do	Thernet	



2) Add User to Groups That Allow Monitoring and Recording

Add the user to the user groups:

- Standard CTI Allow Call Monitoring user group
- Standard CTI Allow Call Recording user group.
- Standard CTI Enabled user group.

Use the *User Management > Application User* menu option in CUCM Administration to perform the necessary configuration.

Figure 2 illustrates adding the user to these user groups.







3) Create Recording Profile

Create a recording profile from the Device Setting pull-down menu.

Enter the recording calling search space and recording destination address.

Use the *Device > Device Settings > Recording Profile* menu option in CUCM Administration to perform the necessary configuration.

Figure 3 illustrates creating a recording profile.



Figure 3



4) Create a SIP trunk security profile

Create a SIP trunk security profile for the recorder. Menu as *System > Security profile > SIP Trunk Security Profile*

Non secure SIP trunk security profile can be selected and edit the field *Outgoing transport type as UDP*. Incoming transport type will be TCP + UDP. Save the selection and reset the trunks.

Figure 4 illustrates creating the SIP Trunk security profile

C SIP Trunk Security Profi	le Configuration - Windows Internet Explorer		
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SIP Trunk Security Profil	e Configuration	Related Links: Back To Find/List	🖌 Go
Save 🗶 Delete 🕥	Copy 🚱 Reset 📫 Add New		
(i) Status: Ready			1
- SIP Trunk Security Prof	ile Information		_
Name*	Non Secure SIP Trunk Profile		
Description	Non Secure SIP Trunk Profile authenticated by null Str		
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Incoming Transport Type*	TCP+UDP V		
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Enable Digest Authentica	dion .		~
Done		🏹 🚱 Internet 🔍 1009	5 × .

Figure 4



5) Create a SIP Trunk that points to the Recorder

Create a SIP trunk that points to the recorder.

Enter the recorder DN, which must match a route pattern for the SIP trunk or a route list that includes the recorder.

Use the *Device* > *Trunk* menu option in CUCM Administration to perform the necessary configuration.

Figure 5 illustrates creating a SIP trunk that points to the recorder.

ululu Cisco Unified Call	Manager Administration	Cisco Unified CallManager Admir
CISCO For Cisco	Unified Communications Solutions	CCMAdministrator
System - Call Routing - Modia Resource	s - Voice Mail - Device - Application - User Management	Bulk Administration • Help •
runk Configuration		Related Links: Back To Find/L
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SIP Information Destination Address*	192.168.0.2	Enter re
SIP Information Destination Address* Destination Address is an SRV Destination Port*	5060	Enter re hostnar addr
SIP Information Destination Address* I" Destination Address is an SRV Destination Port* MTP Preferred Originating Codec* Presence Group*	192.168.0.2 5060 711ulaw	Enter re hostnar addr



6) Create a Route Pattern for the Recorder

Create a route pattern for the recorder SIP trunk. The Recording Destination Address in the recording profile must match this pattern.

Select the SIP trunk that points to the recorder, or select a route list of which the recorder is a member.

Use the *Call Routing > Route/Hunt > Route Pattern* menu option in CUCM Administration to perform the necessary configuration.

Route Pattern Co.	nfiguration - Microsoft Internet E	xplorer				
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System · Call Routing	Media Resources • Voice Mail •	Device · App	ication +	User Management +	Bulk Administration +	Help +
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📄 Sava 💥 Deleta	🗋 Copy 🌵 Add New					
Route Pattern*	7777					
Route Partition	< None >					
Description						
Numbering Plan	Not Selected	w.				
Route Filter	< None >	*				
MLPP Precedence*	Default					
Gateway/Route List	DefaultRecorderSIPTrunk			(Edit)		
Route Option	* Route this pattern					
	C Block this pattern No Error	2	¥			
Call Classification*	OffNet					
C Allow Device Ove	rride IP Provide Outside Dial Ton uthorization Code	e E Allow Over	lap Send	ding 🗆 Urgent Prio	rity	
4						8

Figure 6 illustrates creating a route pattern for the recorder.



7) Turn on IP Phone BIB to Allow Monitoring or Recording

The built-in bridge of the agent phone must be set to *On* to allow its calls to be monitored or recorded. You can also set the Built-in Bridge Enable service parameter to *On* and leave the Built-in Bridge in the Phone Configuration window set to *Default*.

Use the *Device > Phone* menu option in CUCM Administration to perform the necessary configuration.

Figure 7 below illustrates turning on the IP phone BIB to allow monitoring or recording.

🚰 Phone Configuration - Microsoft Internet Explorer				
<u>File Edit View Favorites Tools H</u> elp	💿 🔻 🖹 🖹 🏠 🔎 Searc	ch 🔅 Favorites 🐵 🖾 🕶 💺 🔳 👻 👢 🦉		
Address https://ccmcluster-cm1:8443/ccmadmin/deviceEdit.dc	o?key=81799437-29db-1	dda-b2e9-eb4c1a54faf1		
Cisco Unified CallManager Administration Navigation Cisco Unified CallManager Cisco For Cisco Unified Communications Solutions CCMAdministrato				
System - Call Routing - Media Resources - Voice Mail - De	vice - Application - U	ser Management - Bulk Administration - Help -		
Phone Configuration	Related Li	nks: Back To Find/List		
🔚 Save 💥 Delete 🗋 Copy 🎦 Reset 🕂 Add New				
 18 Call PICKUP 19 Conference List 20 Conference 21 Do Not Disturb 22 End Call 23 Forward All 24 Group Call Pickup 25 Hold 26 Hunt Group Logout 27 mintercom [1] - Add a new Intercom 28 Malicious Call Identification 29 Meet Me Conference 30 Mobility 31 New Call 32 Other Pickup 	Audio Source Location* AAR Group User Locale Built In Bridge* Privacy* Device Mobility Mode* Owner User ID Phone Load Name Retry Video Call	Hub_None None > None > None > None > On Off Default View Mobility Settings None > SCCP70.MU-1-0-13DEV 		





8) Enable Recording on the Line Appearance

To enable recording of an agent, set the Recording Option in the line appearance of the agent to *Automatic Call Recording Enabled or Application Invoked Call Recording Enabled.*

Select the pre-created recording profile from the drop-down list box.

Use the *Device* > *Phone*, Select the specified device name and on the left hand side you can see *Line* [1], *Line* [2] menu option in CUCM Administration to perform the necessary configuration.

Figure 8 illustrates enabling recording on the line appearance.

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Ring Setting (Phone Active)	Use System Default has a call in progress.	 Applies to this 	line when any line on the phone	0
Call Pickup Group Audio Alert Setting (Phone Idle)	Use System Default	*		
Call Pickup Group Audio Alext Setting (Phone Active)	Use System Default	•		
Recording Option*	Automatic Call Record	ling Enable		
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10000				Propagate Selected
-Multiple Call/C	all Waiting Cattings	Davice SED000C05E4124		
Note:The range t is: 1-200	o select the Max Numb	er of calls		8



9) Edit the Clusterwide parameters (Device-SIP) "SIP Session Expires Timer" in CUCM.

Menu is *System* > *service parameters* > *Server - Active, Service - Cisco call manager* (Active) > *SIP Session Expires Timer .*

The default value will be 1800 and the recording will get stopped after half of this value, i.e. 900 seconds. Make it 86400 (higher value) and Apply.

Figure 9 illustrates the service parameter option.



System - Call Routing - Media Resources - Voice Mail - Device - Applic	ation Viser Management Bulk Administration Help
Service Parameter Configuration	
🔚 Save 🧬 Set to Default 🍭 Advanced	
Clusterwide Parameters (Device - SIP)	
SIP Interoperability Enabled.*	True
Retry Count for SIP Bye *	10
Retry Count for SIP Cancel *	10
Retry Count for SIP Invite *	6
Retry Count for SIP PRACK *	6
Retry Count for SIP Rel1XX *	10
Retry Count for SIP Publish *	6
Retry Count for SIP Response *	6
SIP Connect Timer *	500
SIP Disconnect Timer *	500
SIP Expires Timer *	180000
SIP PRACK Timer *	500
SIP Rel1XX Timer *	500
SIP Trying Timer *	500
SIP Publish Timer *	500
SIP Rel1XX Enabled *	Disabled 👻
SIP Min-SE Value *	1800
SIPS URI Handling *	Reject 👻
SIP statistics Periodic update Timer *	2
SIP Session Expires Timer *	86400
SIP Trunk TspReg Retry *	2

Figure 9

10) Check the Clusterwide parameters (System- Location and Region) "*G722 Codec Enabled"*

Menu is *System* > *service parameters* > *Server* - *Active*, *Service* - *Cisco call manager* (*Active*) > *G722 Codec Enabled* .

The default value will be **Enabled** for all devices. Make it to "*Disabled"* or "*Enabled for all devices Except Recording-Enabled Devices"* and Apply.

Figure 10 illustrates the service parameter option.



Save 🧬 Set to Default 🍕 Advanced		
Clusterwide Parameters (System - Location and Region) ——		
norce Millisecond Packet Size	True	
ocations Trace Details Enabled	False	
eferred G.711 Millisecond Packet Size *	20	
eferred G.722 Millisecond Packet Size *	20	
eferred G.723 Millisecond Packet Size *	30	
eferred G.729 Millisecond Packet Size *	20	•
ways Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	
eferred GSM EFR Bytes Packet Size *	31	
722 Codec Enabled *	Enabled for All Devices Except Re	cording-Enabled Dev
BC Codec Enabled *	Enabled for All Devices Except Re	cording-Enabled Dev
traregion Audio Codec Default *	G711/G722	
terregion Audio Codec Default	G729	
traregion Video Call Bandwidth Default *	384	
terregion Video Call Bandwidth Default.*	384	
nk Loss Type Default *	Low Loss	
Clear Bandwidth Override *	False	

Figure 10

Download the CISCO TSP from the CUCM and install it on 11) the Recording server.

Menu is *Application > Plugins*.

Refer Figure11 for the screen shot.



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Download	Cisco IP Phone Address Book Synchronizer	Cisco IP Phone Address Book Synchronizer allows users to synchronize Microsoft Windows Address Book with Cisco Personal Address Book. The Synchronizer provides two-way synchronization between the Microsoft and Cisco products. MOS/User/Icod/Microdom/Jakama-tomock/Heappos/folygins/TabSyncIrotal.exel = 4bref191:d1;87/7bi12]771/43;a2;a4/c1:b1:651/46idd	
Download	Cisco JTAPI for Linux	Install this plagin on all computers that host applications that interact with Class Californaper via JTAPI. JTAPI provides the standard programming interface for telephony applications written in the Java programming language. JTAPI reference documentation and sample code are included. This plagin is meant for Linux platforms. VIEX/usr/local/distributions/jtylest-atomat/websplaging/local/JTAPICInterlinuxim] = 354:31:250:438:1058:0158:0159:150:	
Download	Cisco JTAPI for Solaria Spars	Install this plugin on all computers that host applications that interact with Cisco CallManaper via (TAPI), (TAPI) provides the standard programming interface for telephony applications written in the Java programming language. TAPI reference documentation and asrepla code are included. This plug in in sumar for Solaria Spare platforms. MOS/Usr/Solaria-tomcol/webposioluloi/of/Socar/TAPICIent-solaria/Spare (bio)4407/56199(15)17(11)14164(15)6	h
Deveload	Cisco JTAPI for Solaris X85	Install this plagin on all computers that host applications that interact with Class Californager via JTAPI, JTAPI provides the standard programming interface for telephony applications written in the Java programming language. JTAPI reference documentation and sample code are included. This plagin is meet for Solans XBS platforms, JTAPI/Language. JTAPI reference documentation and sample code are included. This plagin is meet for Solans XBS platforms.	1
Download	Cisco JTAPI for Windows	Install this plugin on all computers that host applications that interact with Cisco Califoranger via JTAPI, JTAPI provides the standard programming interface for telephony applications written in the Java programming language. JTAPI reference documentation and sample code are included. This plugin in mark for Windows platforms. MOS/Usr/ViceInterfactore():reference/object/SC/Code/JTAPICIENTER - 018/2168/12168/12168/12168/12168/12168/12	
Download	Cisco TAPS for Windows	Cisco Tool for Auto-Registered Phone Support (TAPS) loads a preconfigured phone sating on a phone. Install this component on a machine with a version of CRS that is compatible with the Osco Unified Califoraper version. USD//unified-infrinteat/visiterat-tomat/vebasco/obuping/ToolforAutoRegisteredPhonesSupport.exe) = es:35:e2:17:10:62:63:e3:06:151:e3:06:15	
Download	Cisco Telephony Service Provider	This product contains the Cisco TAPE service provider (TSP) and the Cisco Wave Drivers, Install the application on the Cisco CalManager server or on any other computer that is running a Microsoft Window operating system that interacts with the Cisco CalManager server via TCP/19. TAPE, a standard programming interfaces for balapheny applications, runs on the Microsoft Microsoft System. The Cisco TAPE beyond software and the Cisco CalManager server via TCP/19. TAPE, a standard programming interfaces for balapheny applications, runs on the Microsoft applications to make and maximum calls on the Cisco IP Telephony Solution. MICS/Usr/Ciscoft Artenian to make service and the Cisco IP Telephony Solution. MICS/Usr/Ciscoft Artenian to make service and the Cisco IP Telephony Solution.	18
Download	<u>Gince Unified CM</u> <u>Real-Time Monitoring</u> <u>Teol - Linus</u>	Cisco Unified Californager Serviceability Real-Time Venitoring Tool, a sizent tool, monitors real-time behavior of the components in a Cisco Unified Californager cluster. NTMT uses HTTP/HTTPS and TCP to monitor device status, system performance, device discovery, and CTI applications. It also connects directly to devices by using HTTP/HTTPS for troubleshooting system problems. Note: To discumbad an Windows, Bight click on Devendead hyperilities and select Saver Target As explain. MDS/usi/Note-into://eita.click.	
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Download	Cisco Unified CallManagar Assistant Console	The Close Unified Californinger Septemts Conside helps assistants to handle calls for their managers more effectively-The assistant can install the assistant console, a diert-server java explication, on a PC that new Windows 200 Windows View. The assistant constales to the Close Offinide California California (California) and directory services. Nulfigle assistant consoles can connect to a single Close Unified California (California) and directory INSC/war/Section - torneal with a services. The California California prevantation Constale.exail – directory INSC/war/Section - torneal webspread/burger/Closed/AmingCalifornia California prevantation Constale.exail – directory insc/war/Section - torneal webspread/burger/Closed/AmingCalifornia prevantation Constale.exail – directory insc/war/Section - torneal webspread/burger/Closed/AmingCalifornia prevantation Constante.exail – directory insc/war/Section - torneal webspread/burger/Closed/AmingCalifornia and AmingCalifornia California (California) insc/war/Section - directory and assistant constale constale for the Closed of the California California (California) insc/war/Section - directory and the constant	¥

Figure 11

- a) Install the TSP application on the Recording Server with the following settings:
- Multiple Instances of CUCM TSP "No" (If you choose "Yes", then choose the value to be 1).
- TFTP Server IP Address Enter IP address of the CUCM.
- Once the installation completes, it will prompt to restart the system. Click "Yes".

b) Configure the CISCO TSP using *Control Panel > Phone and Modem Options > Advances > CiscoTSP001.tsp* as in Figure 12 & 13.





Figure 12

- Provide the Application User Name and Password created in CUCM (Refer step1).
- Enter the CUCM IP address and Apply.

Cisco Unified Communications Manager TSP	Cisco Unified Communications Manager TSP
General User OTI Manager Security Wave Trace Advanced Language Security User Name: Dgvoxtest Password Verify Password	General User CTI Manager Discounty Wave Trace Advanced Language Primary CTI Manager Ducation C None C IP Address: C IPv6 Address: C Host Name:
	Backup CTI Manager Location
	IP Addressing Preference
	Preferred IP Addressing Mode (© IPv6
OK Cancel	OK Cancel



c) Verify the configured phones were displayed in "Dialer" application.



Go to Windows Run > Type *Dialer.exe* and the following window (Figure 14) will appear. Check whether the configured phones are displayed under the Line option.

Reference Dialer	
Number to dial:	Speed dial
Dial	2
1 2 Line	
4 5 (Lisco Line: PRS TI Cisco Line: 7 8 Cisco Line: Cisco Line: Cisco Line: BAS PPBa	[SEP00269E5E7D77] [305) [SEP00269E5E7D77] [305] [SEP1C17D341BE36] (100) [SEPD0574CF6EDE5] (112) ↓ Line _roperties
BAS VPN L SSTP WAN Minip	ine 0 rother programs ort (L2TP) Cancel



DGVox PORT LIST:

The following are the ports used by the DGVox application:

UDP ports: 3001 - 3011, 5060, 5501, 6003, 6005, 6501, 6502, 10029.

For the RTP streams, UDP Port Range: 20040 – 30030

Http port: 80

FUNCTIONAL DESIGN- DGVOX ON CUCM





Now the server is ready to install DGVox Application. Here is the schematic representation of DGVox Active mode recording on CUCM.



SpeechLogix Technologies Pvt Limited