

**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.

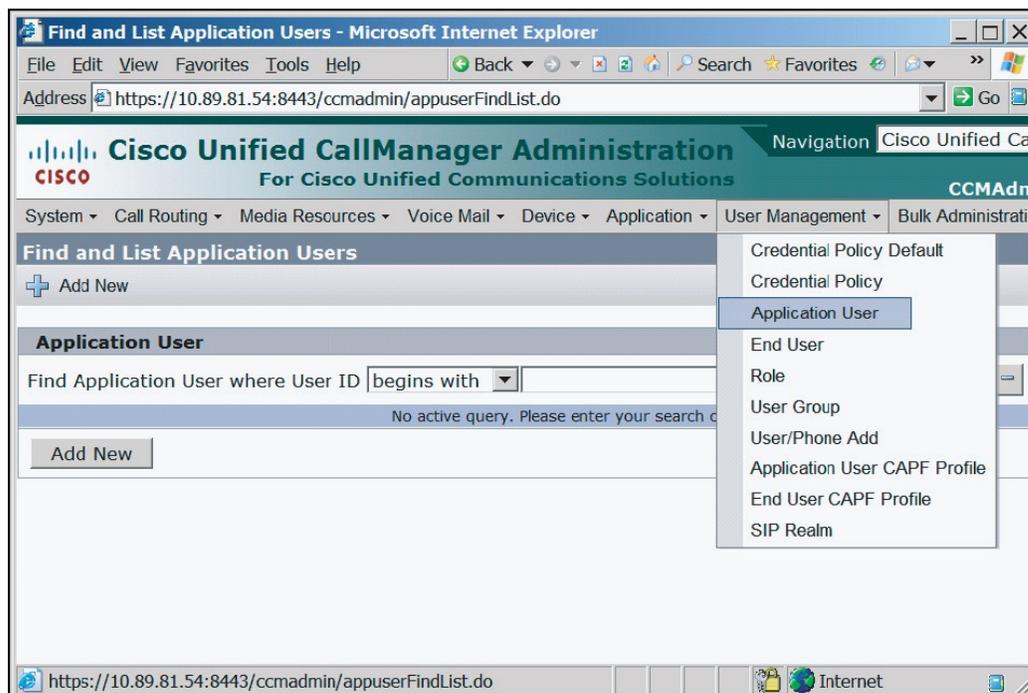


Figure 1

## 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.

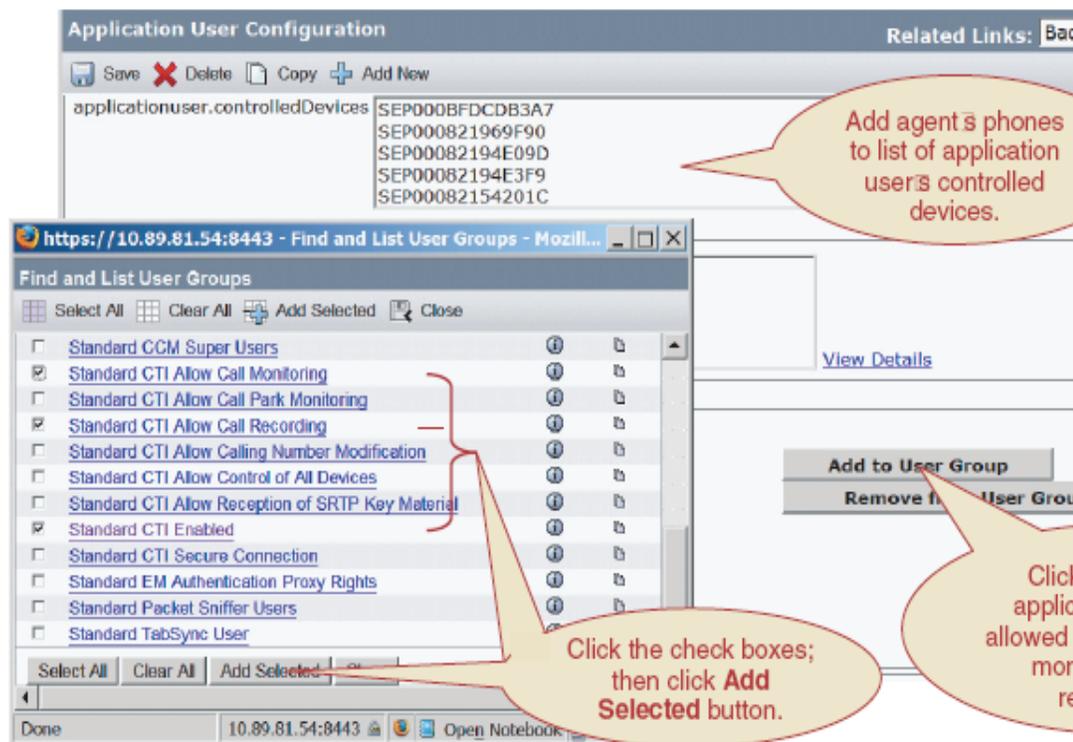


Figure 2

### 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.

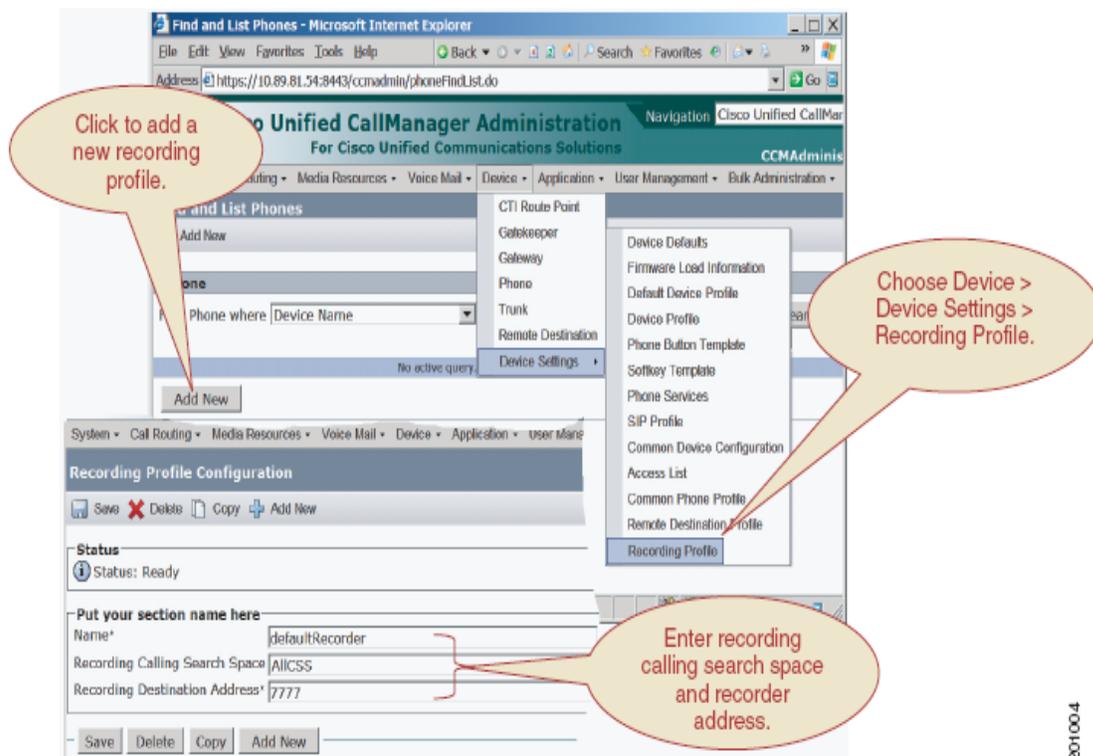


Figure 3

#### 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

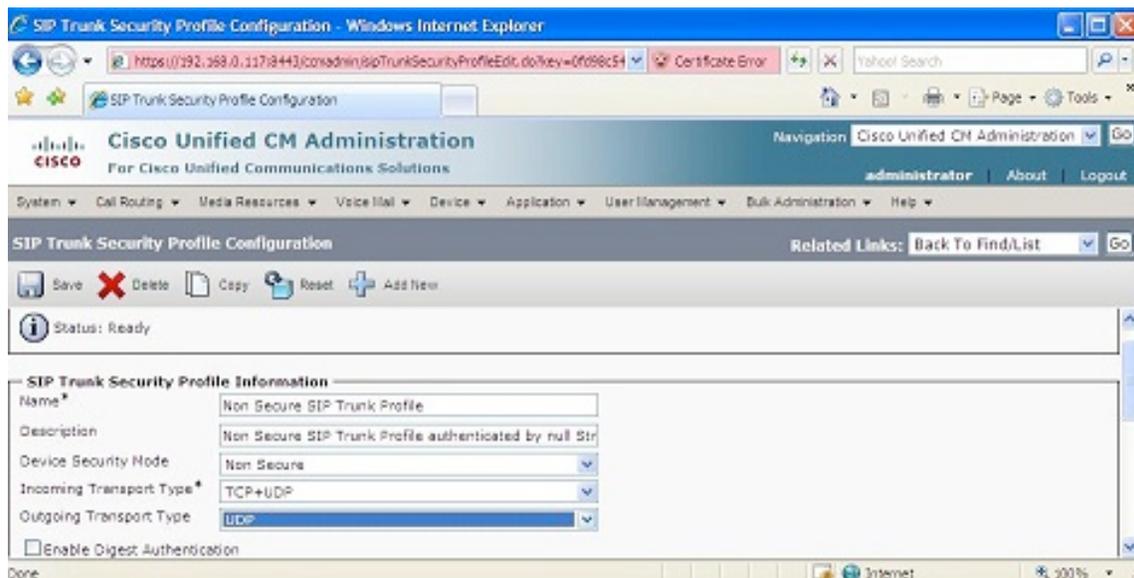


Figure 4

## 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.

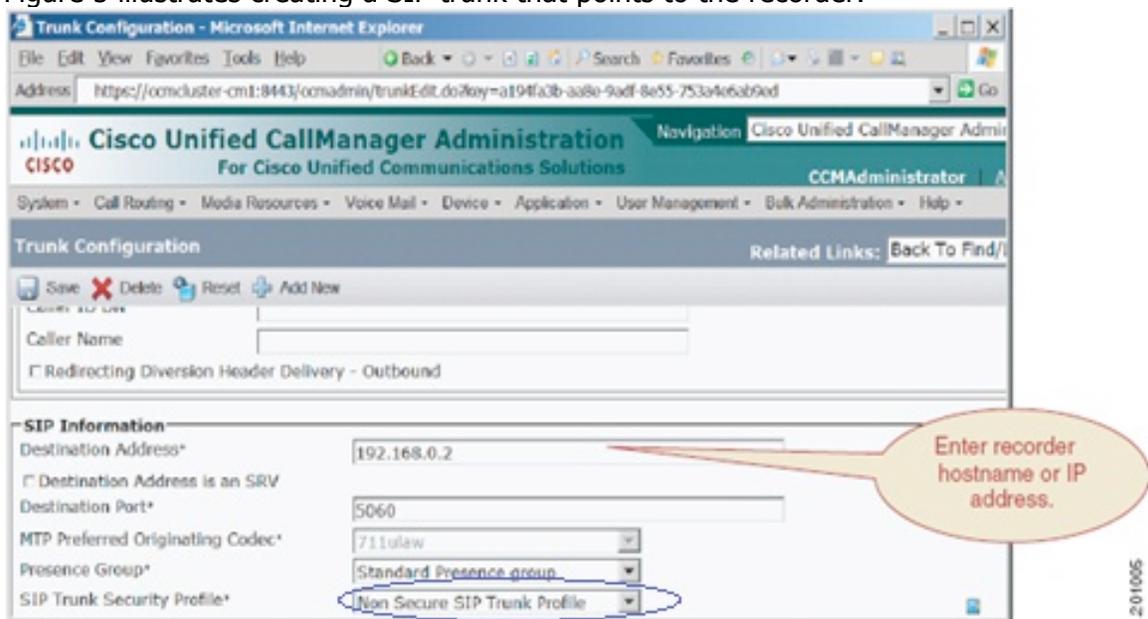


Figure 5

## 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.

Figure 6 illustrates creating a route pattern for the recorder.

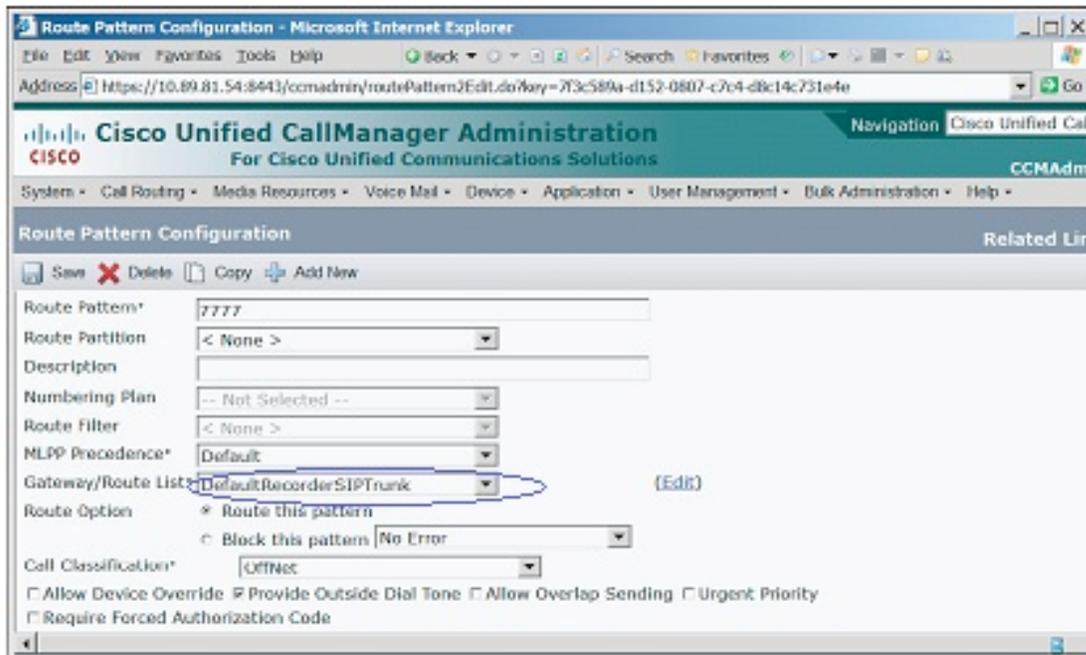


Figure 6

## 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.

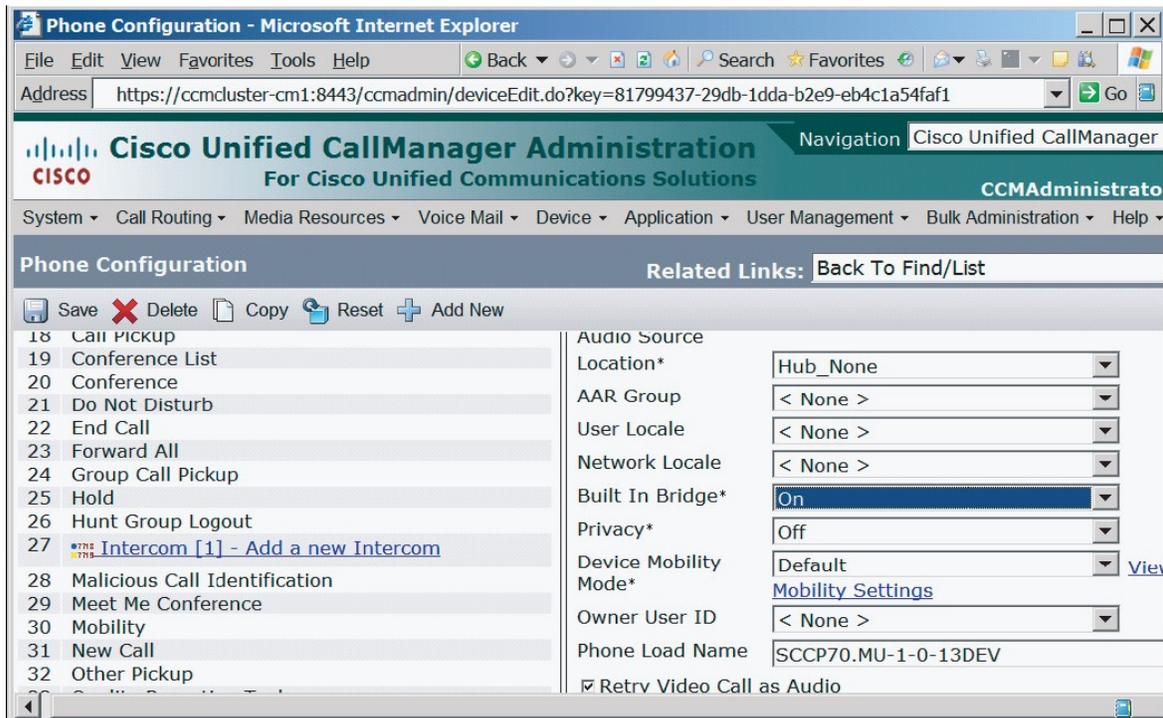


Figure 7

## 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.

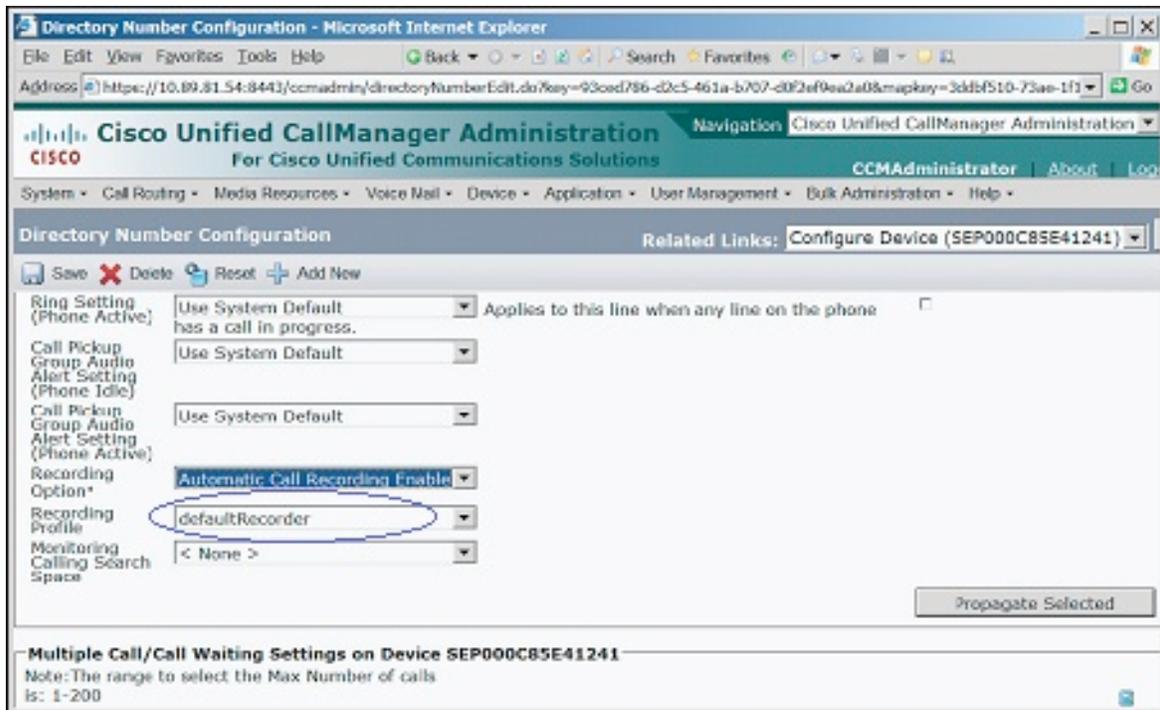


Figure 8

## **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.

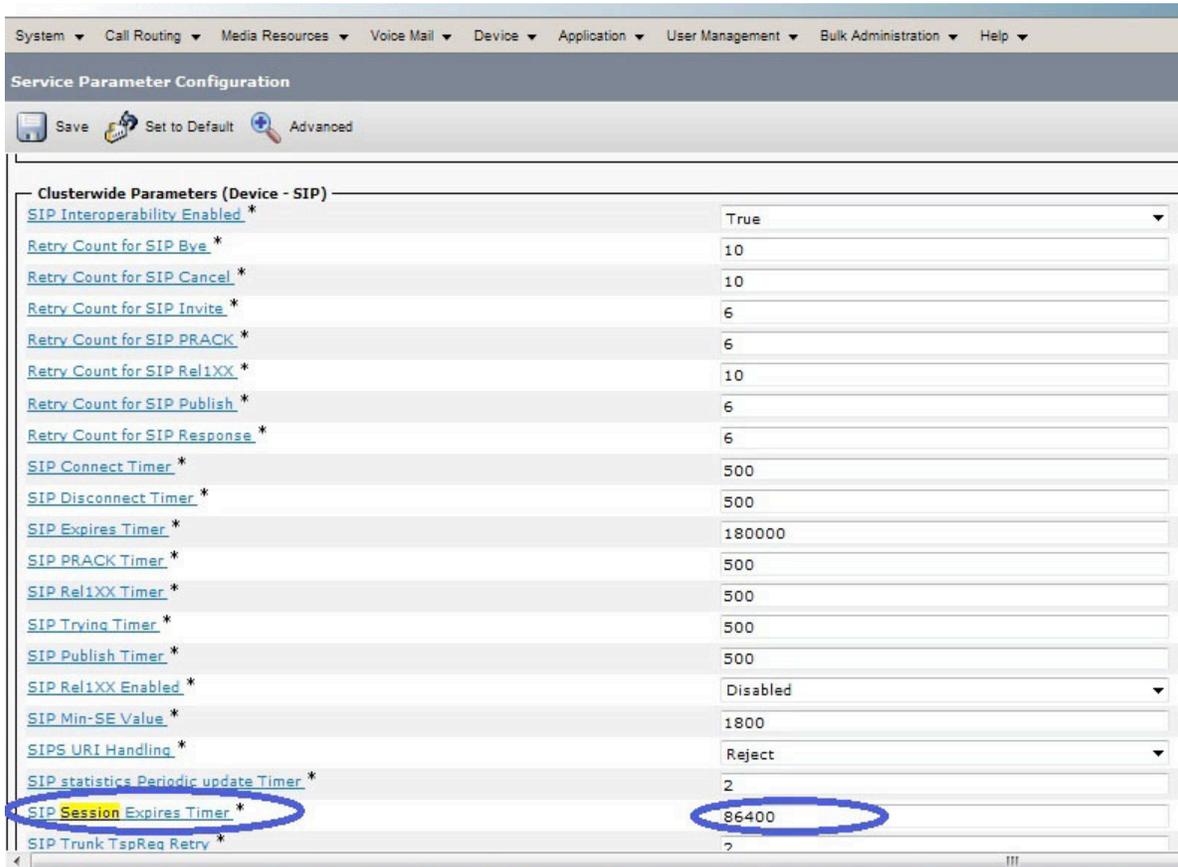


Figure 9

## 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.

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### Service Parameter Configuration

Save Set to Default Advanced

**Clusterwide Parameters (System - Location and Region)**

<a href="#">Enforce Millisecond Packet Size</a> *	True
<a href="#">Locations Trace Details Enabled</a> *	False
<a href="#">Preferred G.711 Millisecond Packet Size</a> *	20
<a href="#">Preferred G.722 Millisecond Packet Size</a> *	20
<a href="#">Preferred G.723 Millisecond Packet Size</a> *	30
<a href="#">Preferred G.729 Millisecond Packet Size</a> *	20
<a href="#">Always Use Preferred G.729 Packet Size For SIP Trunk Answers</a> *	False
<a href="#">Preferred GSM EFR Bytes Packet Size</a> *	31
<a href="#">G722 Codec Enabled</a> *	Enabled for All Devices Except Recording-Enabled Dev
<a href="#">iLBC Codec Enabled</a> *	Enabled for All Devices Except Recording-Enabled Dev
<a href="#">Intraregion Audio Codec Default</a> *	G711/G722
<a href="#">Interregion Audio Codec Default</a> *	G729
<a href="#">Intraregion Video Call Bandwidth Default</a> *	384
<a href="#">Interregion Video Call Bandwidth Default</a> *	384
<a href="#">Link Loss Type Default</a> *	Low Loss
<a href="#">G.Clear Bandwidth Override</a> *	False

**Clusterwide Parameters (System - CCM Automated Alternate Routing)**

<a href="#">Automated Alternate Routing Enable</a> *	False
--	-------

Figure 10

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

Menu is *Application > Plugins*.

Refer [Figure11](#) for the screen shot.

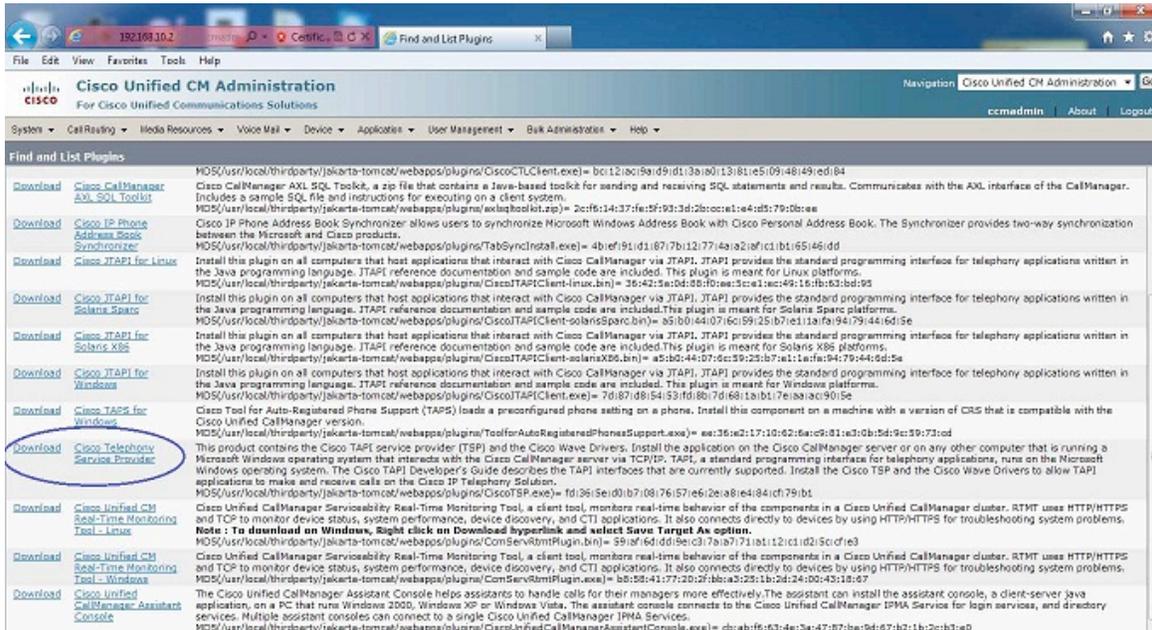


Figure 11

**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.**

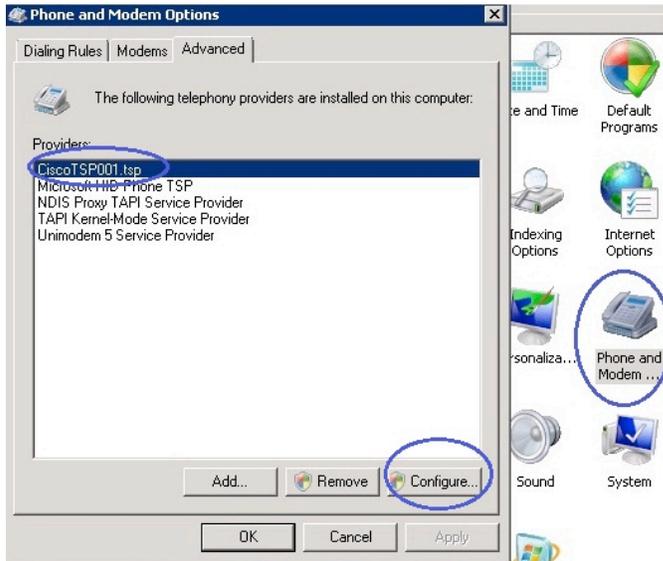


Figure 12

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

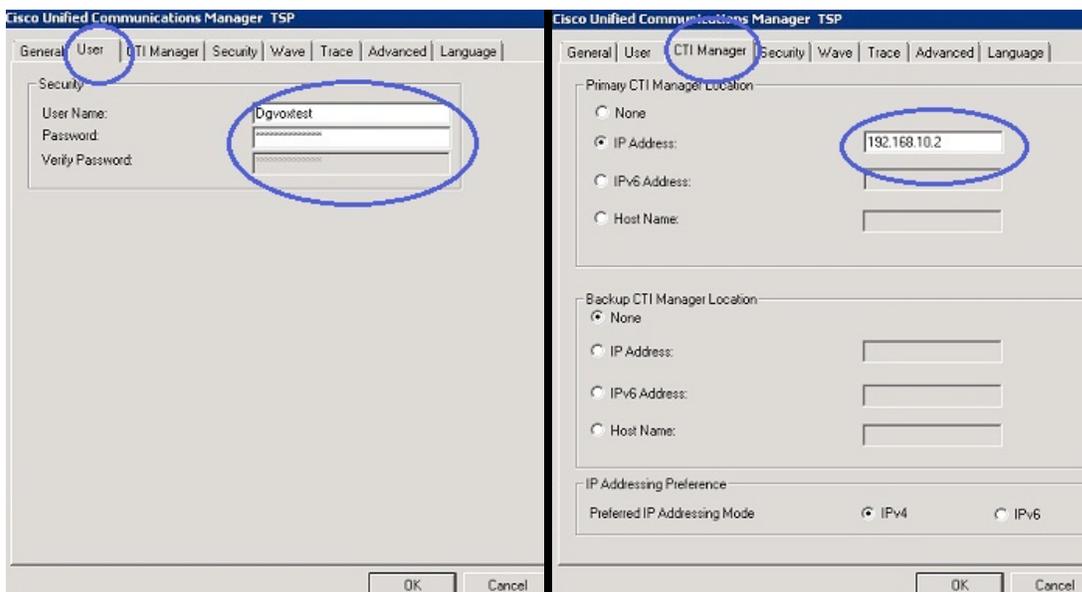


Figure 13

- 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.

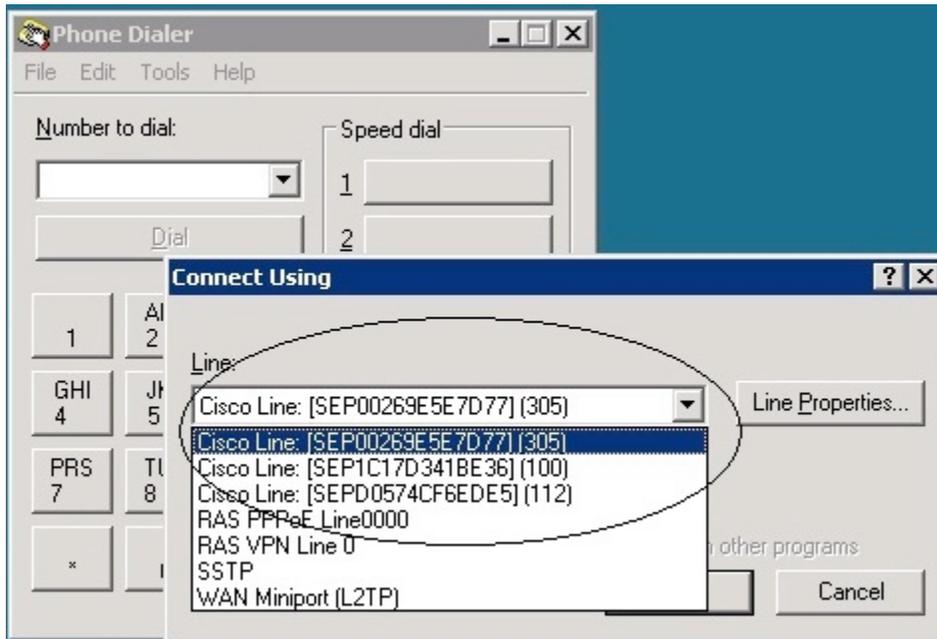


Figure 14

### **DGVox PORT LIST:**

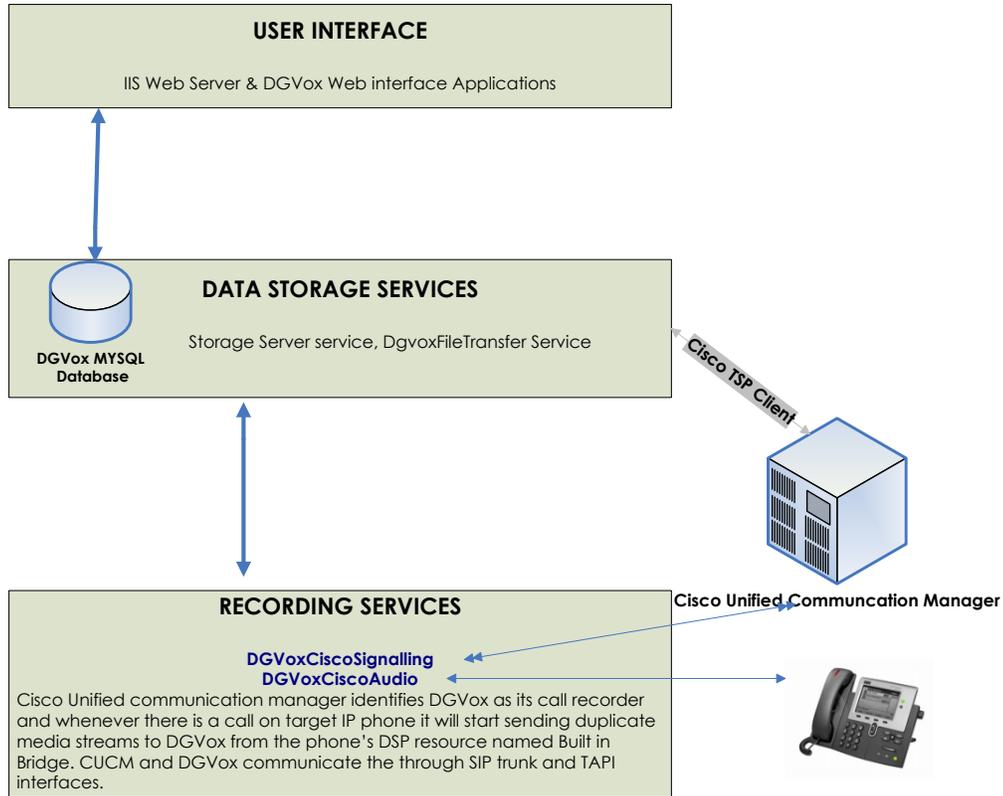
The following are the ports used by the DGVox application:

**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.**

