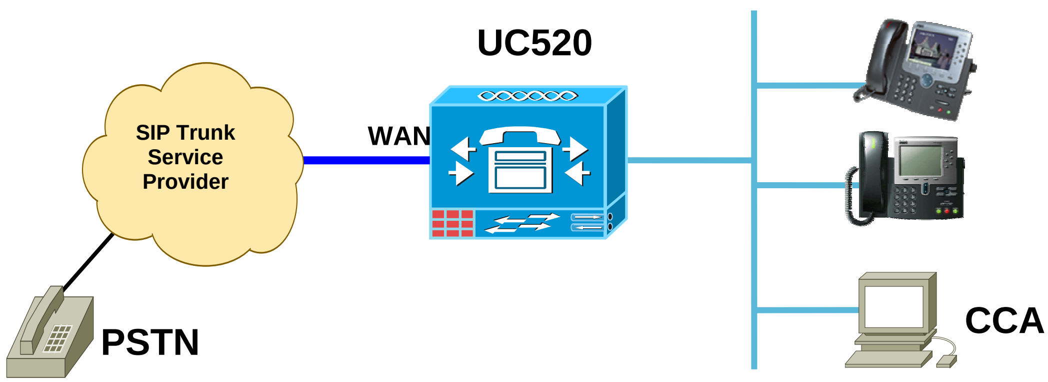
LAB3: PBX, VOICEMAIL & AUTO ATTENDANT

Introduction:
The UC520 supports two voice system configuration types – PBX type and Keysystem type. LAB3 focuses on the PBX system configuration. A typical PBX system involves an Auto Attendant that handles incoming calls from a PSTN line (eg. Analog FXO trunks) and transfers the caller to one of the internal extensions. Outbound local/LD/international calls are routed out through the PSTN line. In addition to this, local extensions have voicemail boxes where callers can leave messages.

Objective:
The main objective of this lab is to configure the basic PBX system, voicemail and Auto Attendant features. These features will be configured using Cisco Configuration Assistant (CCA). After completion of this lab – you will be able to setup the SBCS system as a PBX, place calls between extensions, setup SIP Trunk on the UC520 to a Service Provider (Lab Simulation), place calls in & out to the PSTN via the SIP trunk.

Topology:



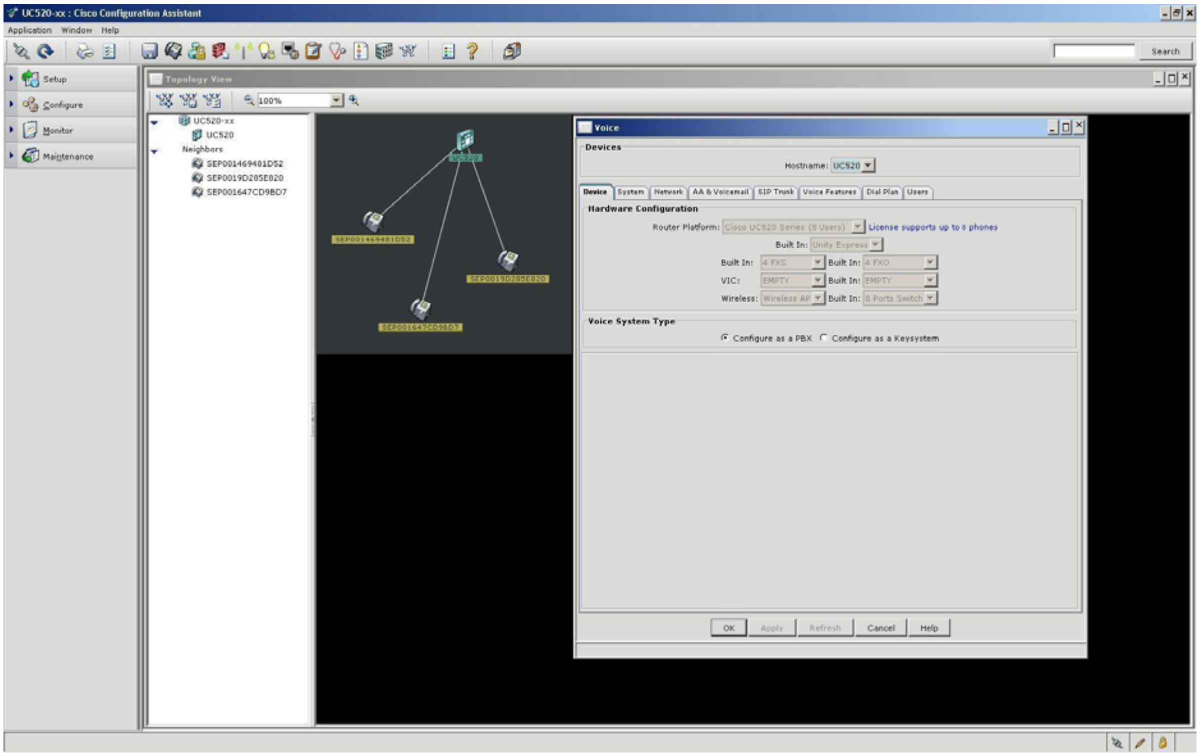
Setup steps:

1. Launch CCA, connect to the created community and enter the username and password (cisco/cisco) and let it discover the network and connected devices. Now launch CIPC and make sure it gets extension 203 (x203)

2. Check the topology to ensure all the connected phones are showing up. Unplug any lines that are connected to the FXO ports on the PODs.

3. Click on “Configure -> Telephony -> Voice” on the left navigation bar or click on the Phone icon on the top menu bar. Wait until the “Refresh” and “Apply” buttons are enabled before making any changes. In this lab,, we will configure the various tabs - “Device”, “System”, “Dialplan”, “AA & Voicemail”, “SIP Trunking” “Voice Features”, and “Users”.

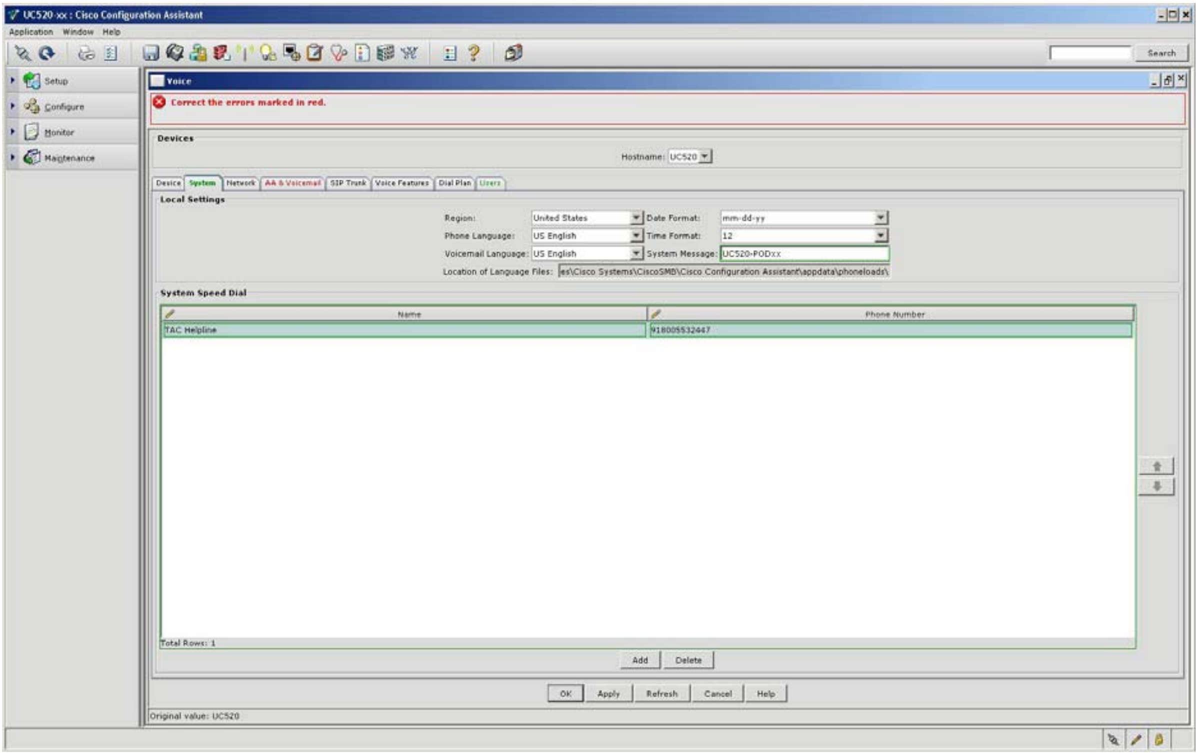
4. Click on the “Device” tab & select the System Type as a “PBX”.



5. Click on “System” tab:

- Change the System Message to "UC520-PODxx"
- Configure a system-wide speed dial. Click on the "Add" button and configure a name (eg. "TAC Helpline") and a corresponding number. Remember to include the PBX Access code and the long distance prefix for the speed-dial. These parameters are defaulted to 9 and 1 respectively for North American dialing and can be changed in the "Dial Plan" tab.

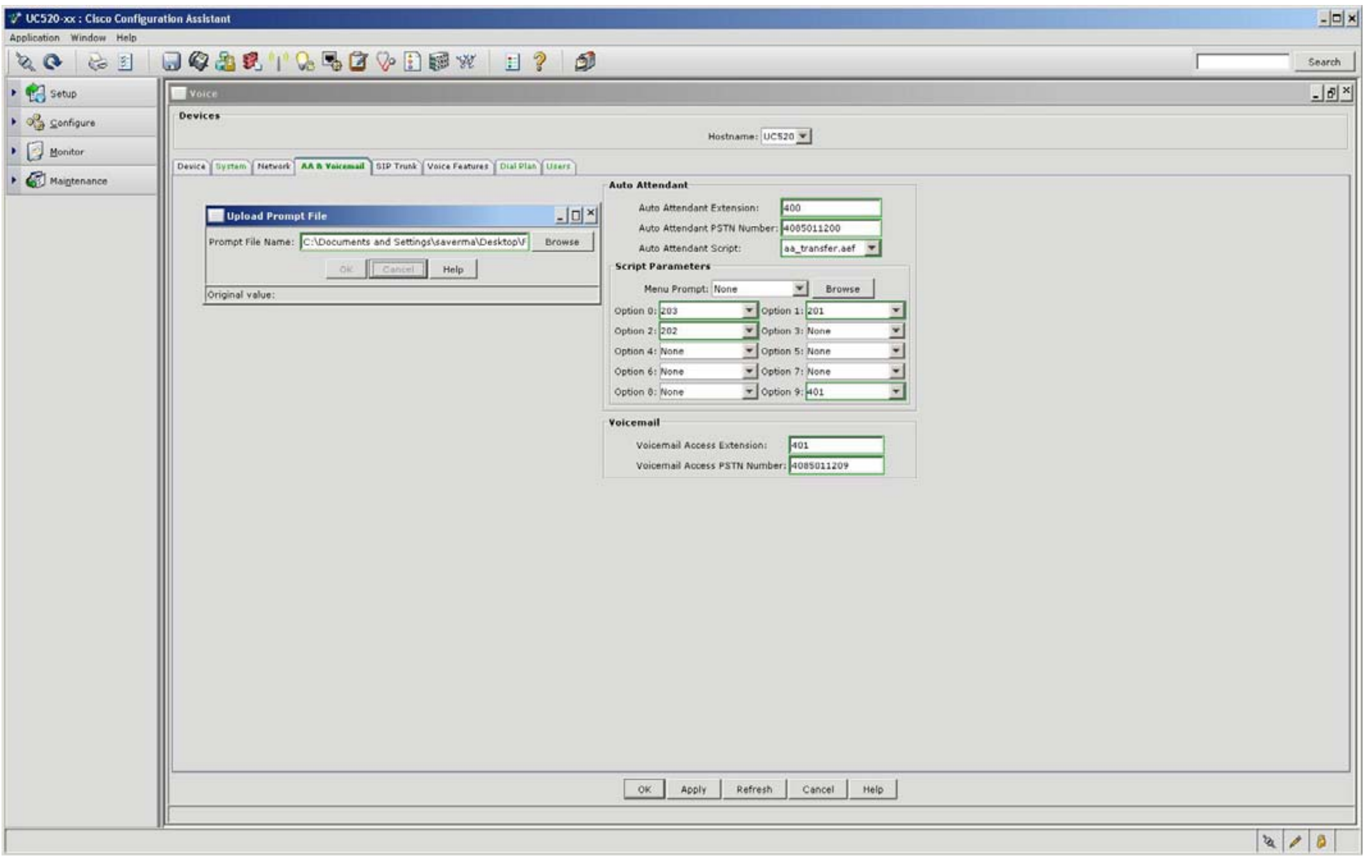
Note: Don't click on "OK" or "Apply" until you have configured all the voice tabs



6. Click on the "AA & Voicemail" tab:
    - Set the AA Access Extension as 400 & the AA PSTN number as 4085xx1200

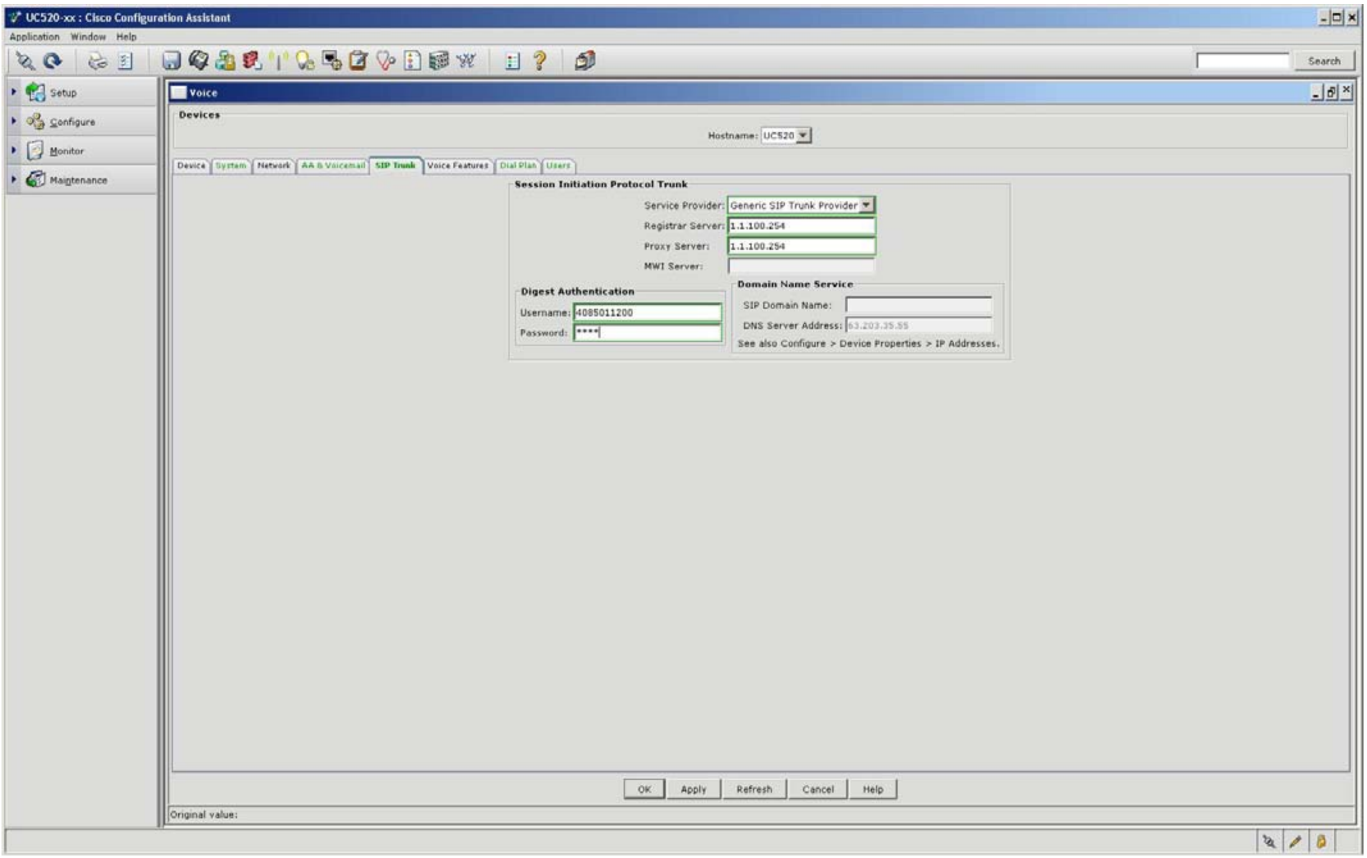
- For the Auto Attendant script, select the aa_transfer.aef
- Set the following parameters for the Auto Attendant script:
    - Option 0: 203
    - Option 1: 201
    - Option 2: 202
    - Option 9: 401
- Set the VoiceMail Access Extension to 401 & the VM PSTN number to 4085xx1209

Note: The pull-down menu assists by showing the existing extensions on the system. In addition, you can configure any number by overwriting "None"



Note: The Menu Prompt will be configured towards the end of this lab.

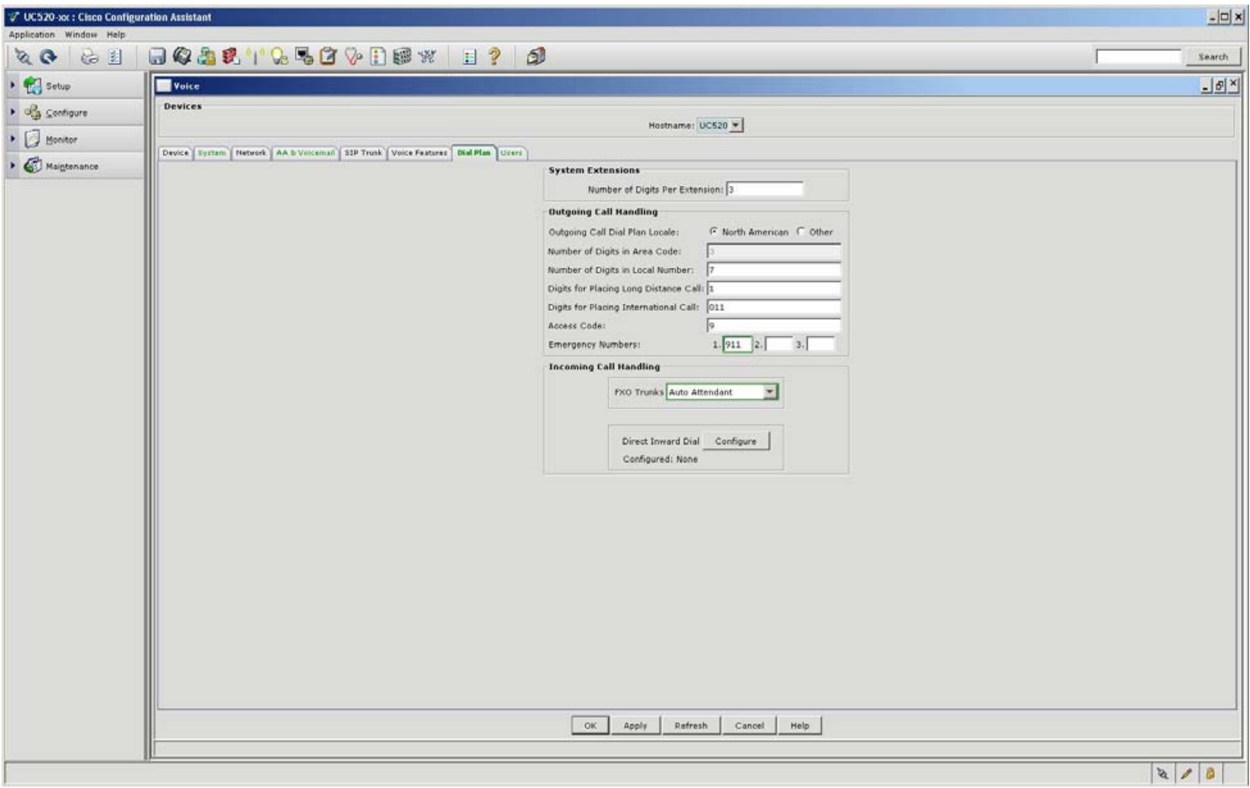
7. Click on the SIP Trunk tab and setup the UC520 for SIP trunking to a Service Provider for PSTN access
    - Select "Generic SIP Trunk Provider" from the Service Provider pull down menu
    - Set the Registrar Server and Proxy server fields with: 1.1.100.254 which is the SIP PSTN network setup in the lab
    - Under Digest Authentication, set username to 4085xx1200 & password to 1234



8. Click on the "Dial Plan" tab:
    - Ensure the following defaults are set for North American dialing:

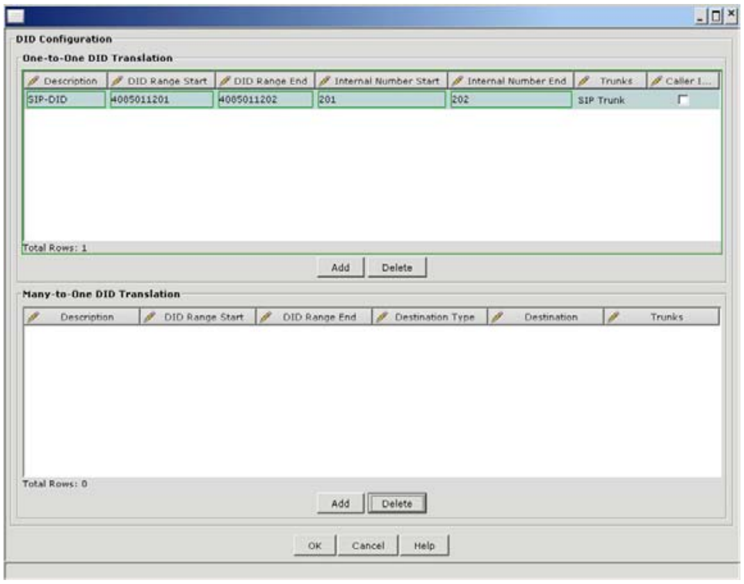
Number of digits per Extension	3
Number of digits in Area Code	3
Number of digits in Local Number	7
Digits for placing Long Distance calls	1
Digits for placing International calls	011
Access code	9

- Configure the "Emergency Number" to be "911"
- For "Incoming Call Handling", note that the "FXO Trunks" can be changed from "Custom Configuration" to "Auto Attendant" or "Operator". When you select "Operator", you get an option to configure the Operator's extension. For this lab, choose "Auto Attendant" from the pull-down menu.



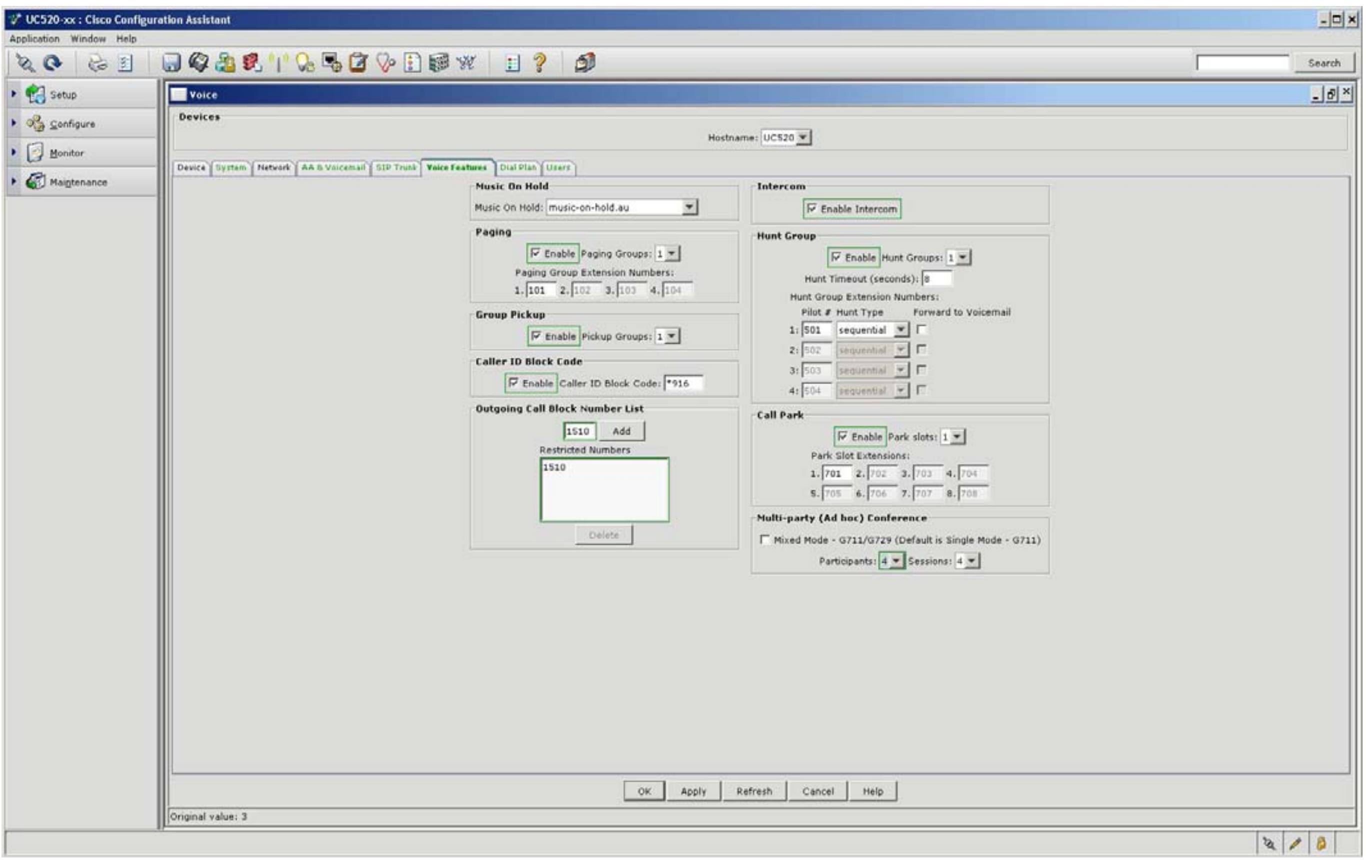
- Click on the "Configure" button for Direct Inward Dial. We will use this feature to add PSTN DID numbers to 2 of the IP phone users.

- Click on Add range for “One-to-One DID Translation”
- Enter a description such as SIPPSTN
- Enter starting DID range as 4085xx1201 and ending as 4085xx1202
- Enter starting Internal Extension range as 201 and ending as 202
- Choose SIP Trunk from the trunk pulldown
- Do not Check Caller ID and then click OK

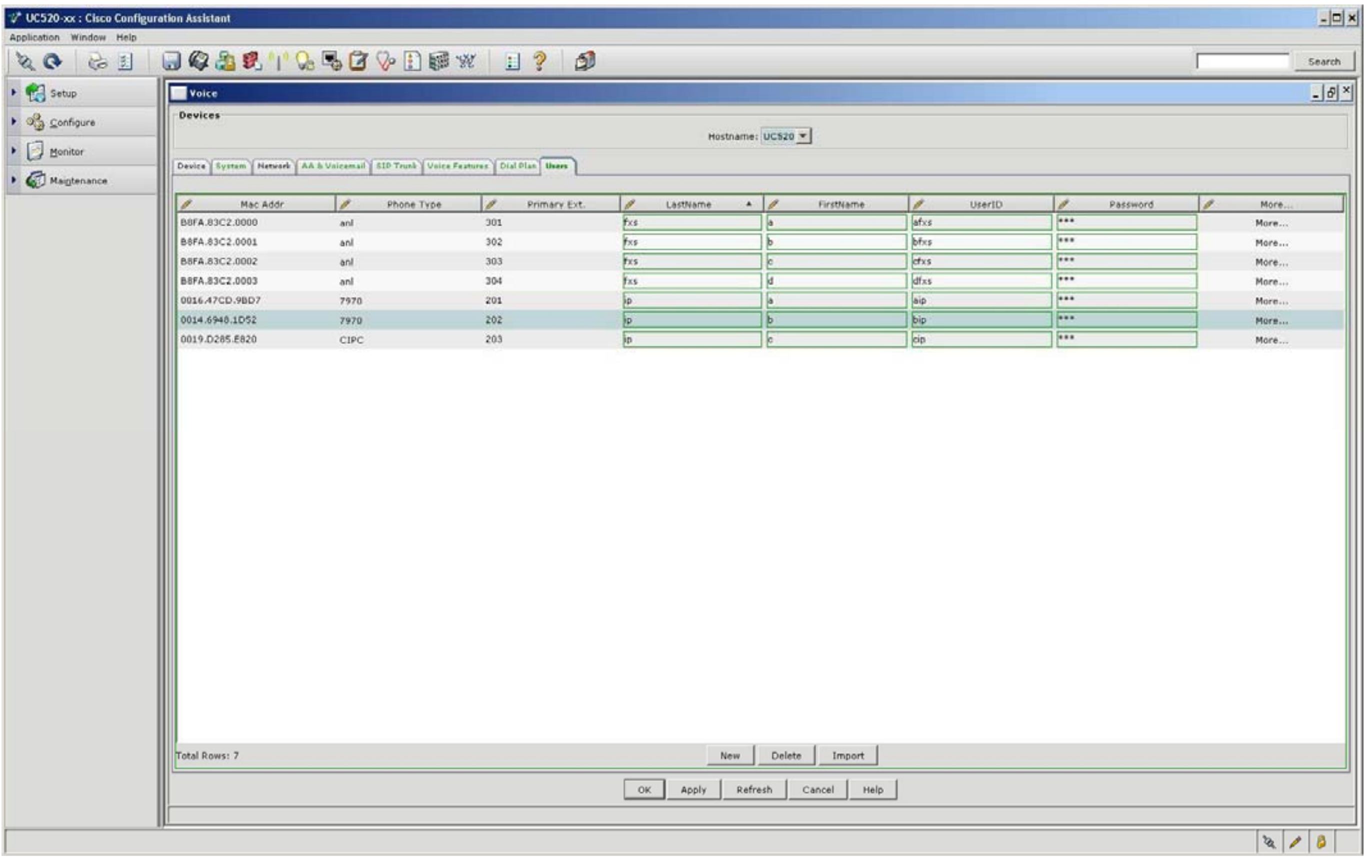


9. Click on the “Voice Features” tab
    - Under the “Paging Parameters”, check the “Enable Number of Paging Groups”, select 1. Configure the first Paging Group Extension Number as 101.

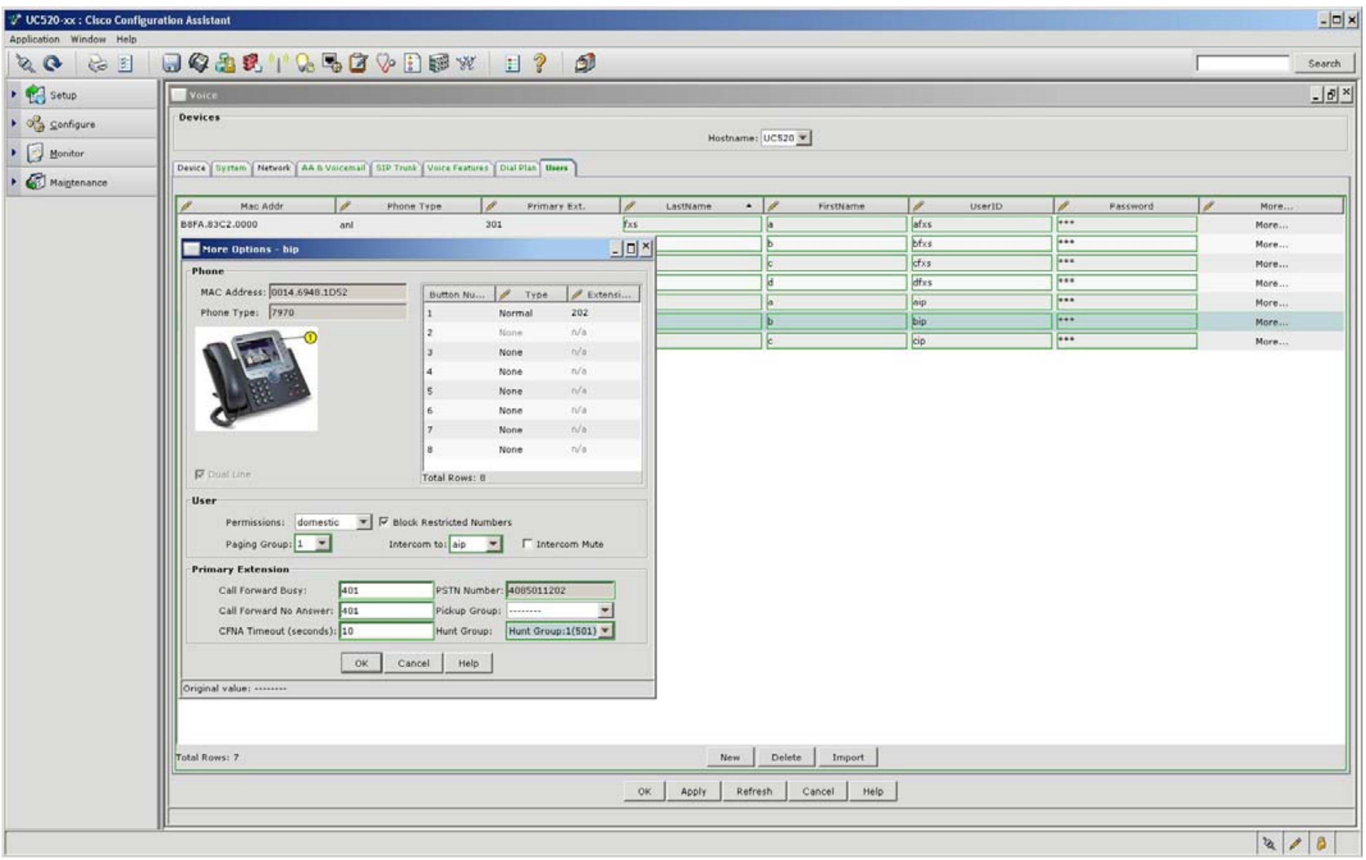
- Under the “Group Pickup Parameters”, check the “Enable Number of Pickup Groups”, select 1.
- Under the “Caller ID Block Code”, check the “Enable Caller ID Block Code”. Use the default value.
- For “Outgoing call block number list”, enter “1510” and click on the “Add” button. This will block all calls to 510 areacode. Note that you need to specify the long-distance prefix in front of the areacode.
- Check the “Enable Intercom” box.
- Under the “Hunt Group Parameters”, check the “Enable Number of Hunt Groups”, select 1. Configure the first Pilot # as 501
- Under the “Call Park Parameters”, check the “Enable Number of Park Slots”, select 1. Configure the first Park Slot Extension as 701.
- For the “Multi-Party (Ad-hoc) Conferencing”, change from the default of 3x8 (8 sessions of up to 3-party conference) to 4x4 (4 sessions of up to 4-party conference)



10. Click on the “Users” tab:
   - For all the Analog Phones (none connected but configured by default on the UC520):
      i. Configure the LastName for all Analog phones as fxs
      ii. Configure the FirstName as a, b, c, d.
      iii. Configure the userid as afxs, bfxs, cfxs, dfxs.
      iv. Set the passwords to 1234.
   - For all the IP Phones:
      i. Configure the LastName for all IP phones as ip
      ii. Configure the FirstName as a, b, c.
      iii. Configure the userid as aip, bip, cip.
      iv. Set the passwords to 1234.



11. Select the “More” button for the IP Phones one at a time as below
   - For IP phone with x201,
      i. Select the Paging Group from the pull down
      ii. Select the Hunt Group from the pull down
      iii. Set Permissions to International. Make sure “Block restricted number” is checked
      iv. Set the intercom between users aip and bip
      v. Make sure the DID number configured previously shows up – then click OK
   - For IP Phone with x202
      i. Select the Paging Group from the pull down
      ii. Select the Hunt Group from the pull down
      iii. Set Permissions to Domestic. Make sure “Block restricted number” is checked
      iv. Make sure the DID number configured previously shows up – then click OK
   - For the CIPC phone with x203
      i. Select the Pickup Group from the pull down
      ii. Set Permissions to Domestic. Make sure “Block restricted number” is checked
      iii. Make sure no PSTN number shows – then click OK

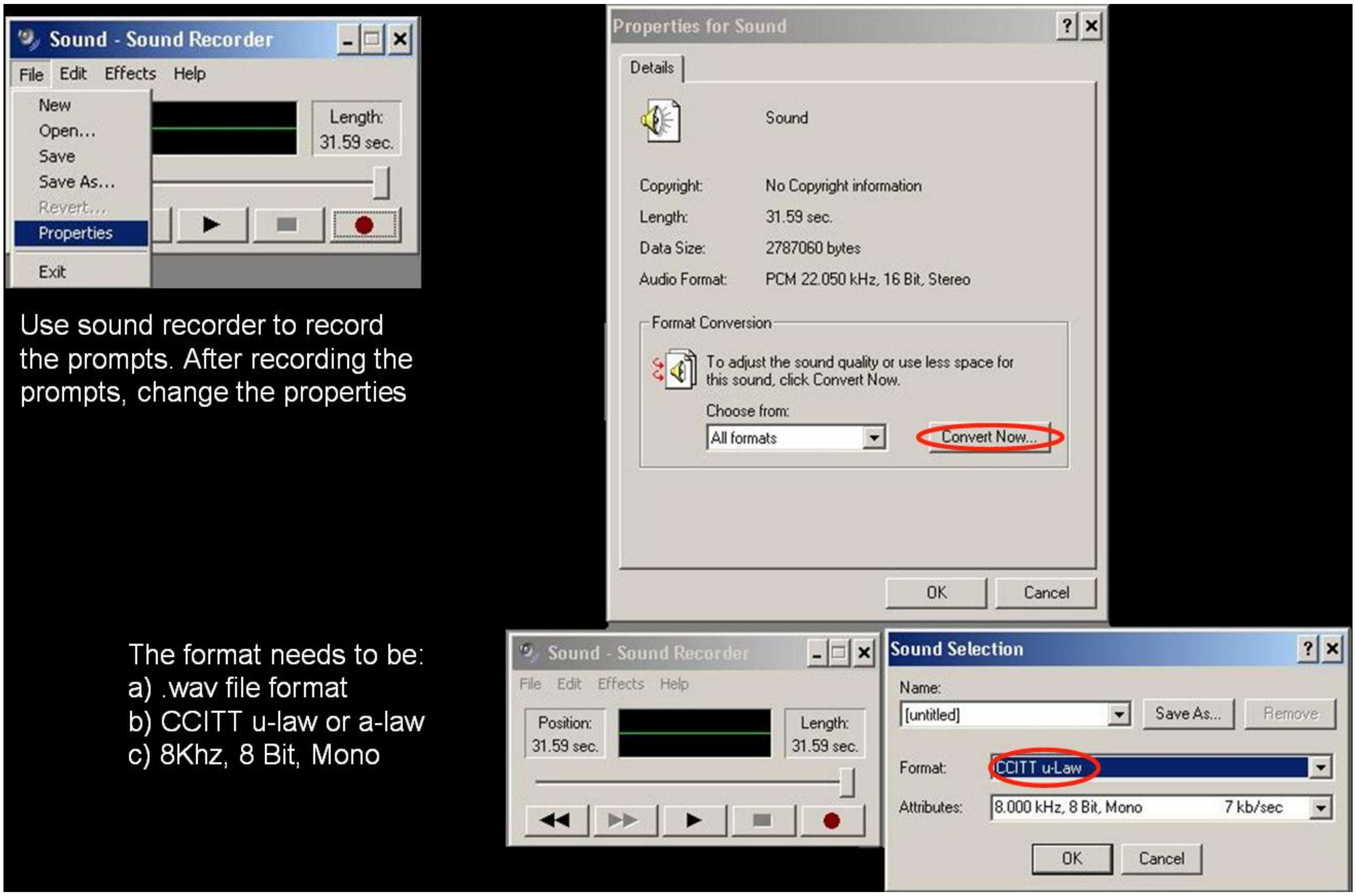


12. Click on “ok” button at the bottom of the screen and observe the progress bar. Click OK when the “Configuration successfully sent to UC520” message pops up.

13. Follow the steps below to setup prompts for Auto Attendant. There are 2 ways to setup prompts, you can either use **"Sound Recorder"** utility on windows and or you could setup a **"prompt management"** system.

Use the **"Sound Recorder"** tool to record the prompts.

- Click the "Red" Record button to start recording
- After the prompts are recorded, click on File > Properties and change the Audio format
- In the Format Conversion section, click on "Convert Now"
- In the "Sound Selection window" scroll up and select CCITT u-law", then click on OK
- Make sure the "Audio Format" in the Properties page reflects CCITT u-law, 8Khz, 8 Bit, Mono

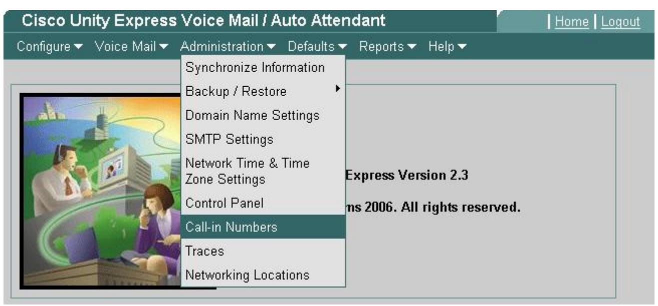


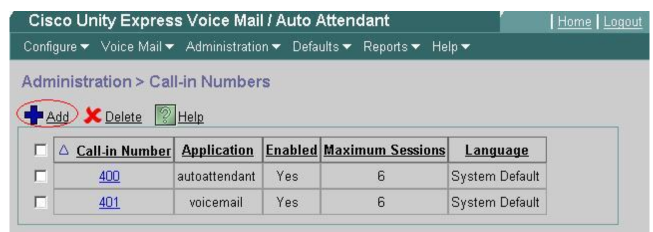
The other way to record prompt for Auto Attendant script is to use the **"Prompt Management"** script on CUE. For this we will use the CUE Web GUI. The configuration steps are divided into 3 main sections.

Section A configures a Call-in number for the prompt-management script
Section B assigns admin rights to a specific extension and
Section C guides you through CLI configuration on the UC500

SECTION A – Configure the Call-in number (trigger) for the prompt-management script
Step1: Launch a web browser and go to ***http://10.1.10.1***
Step2: Login using administrator username/password (by default this is cisco/cisco)

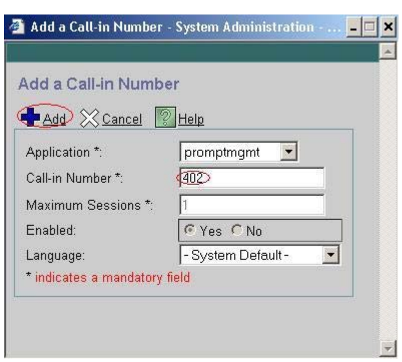
Step3: Click on Administrator > Call-in Numbers
Step4: Click on Add

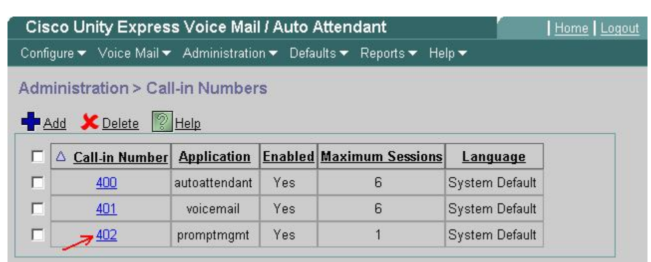




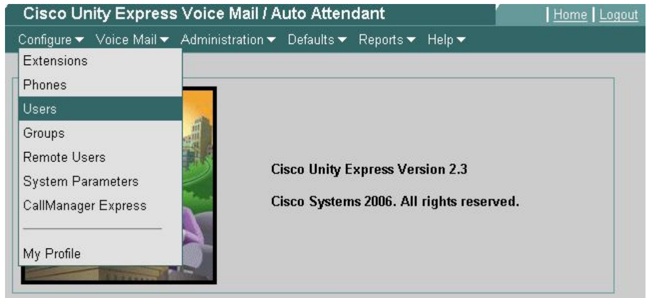
Step5: Configure the "Add a call-in Number" window:

- From the "Application" pull-down menu select "promptmgmt"
- Configure the "Call-in Number" as 402
- Once done, click on the Add icon.
- Make sure that the new call-in number appears under Administrator > Call-in Numbers

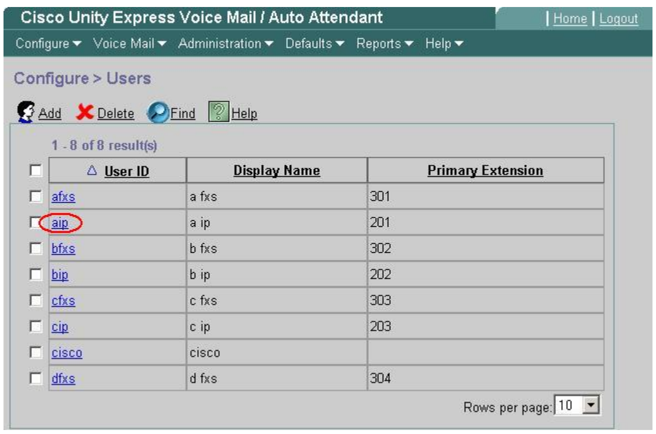


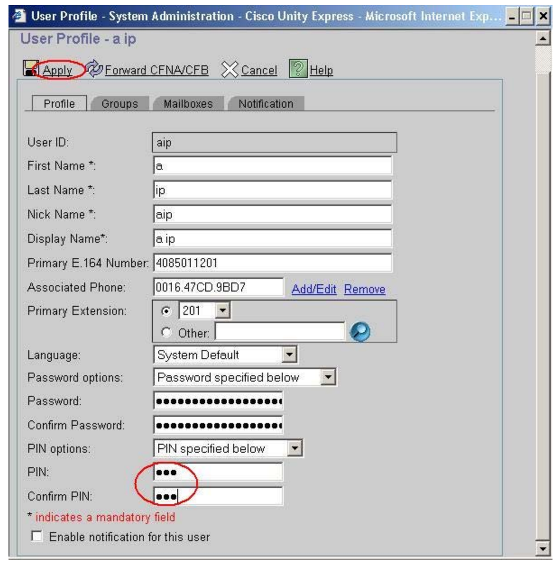


SECTION B – Assign administrator rights to extension 201
Step6: Click on Configure > Users

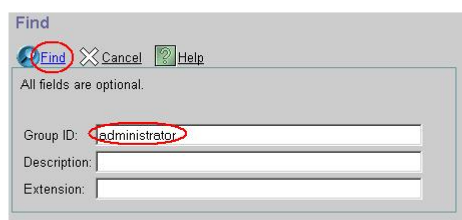


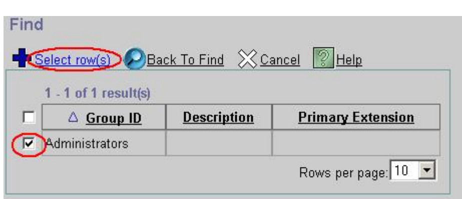
Step7: Click on the user aip. A “User Profile” window that displays all the parameters for a ip will pop up. On this window, set the PIN number to 789, and click on Apply icon.





Step8: Click on Groups tab, notice that the list contains IMAPgrp. Click on "Subscribe as member".
Step9: In the “Find” window, enter "Administrator" for Group ID and click on Find. Check the Group ID "Administrators" and click on "Select row(s)". Ensure aip shows as a member of administrator group and close the “User Profile” window.



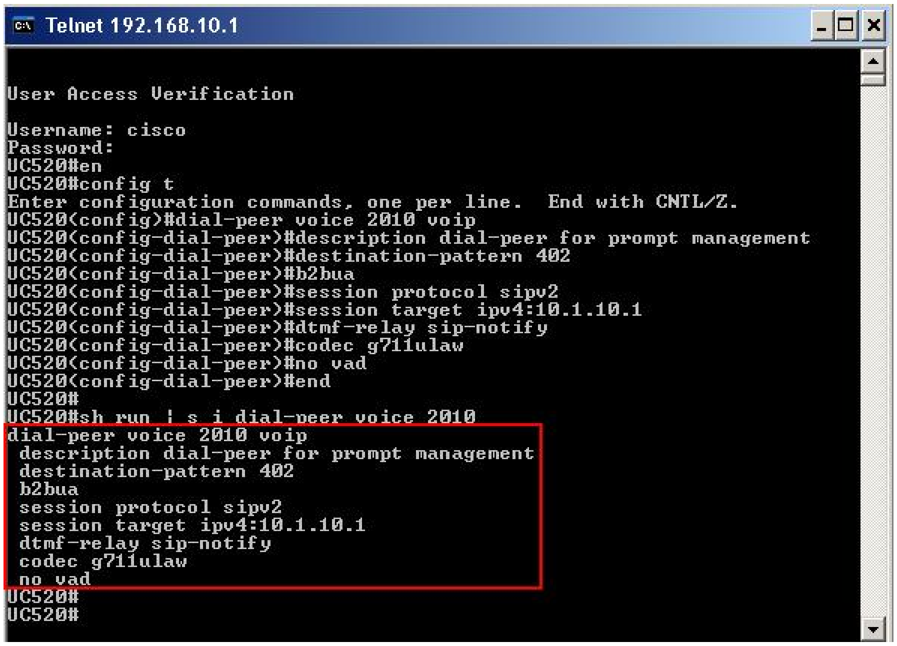


SECTION C

Step10: Telnet to UC500 using MS-DOS prompt. The default IP address for UC500 is 192.168.10.1 and the passwords are cisco/cisco. Get into the enable mode.

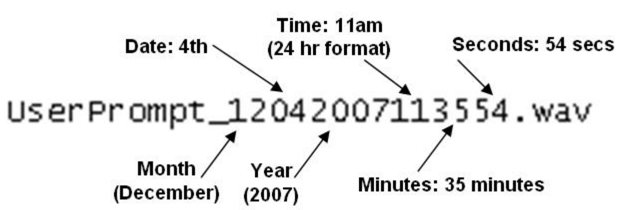
Step11: Use the following CLI to add a dial-peer for prompt-management:

```
config t
dial-peer voice 2010 voip
description dial-peer for prompt management
destination-pattern 402
b2bua
session protocol sipv2
session target ipv4:10.1.10.1
dtmf-relay sip-notify
codec g711ulaw
no vad
end
```



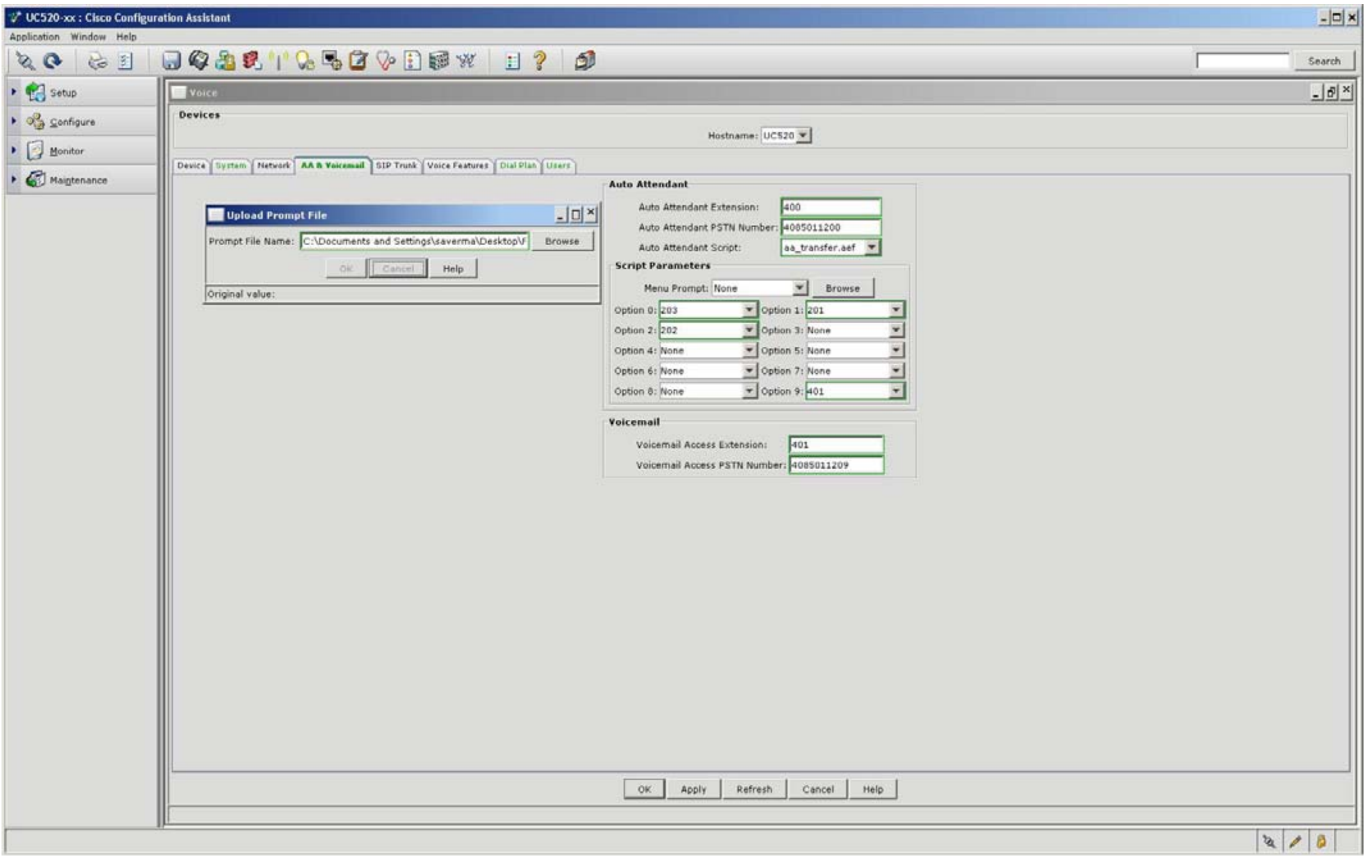
Step12: Record a prompt

- Go off-hook on any phone and dial 402 to trigger the prompt-management script
- The script will prompt for an extension/pin (followed by #). Use extension 201 and pin 789.
- Select option 2 to Administer Custom Prompt.
- Press 1 to record a new prompt.
- After recording the prompt, press 1 to save the prompt before terminating the call.
- The prompts recorded are in the format below

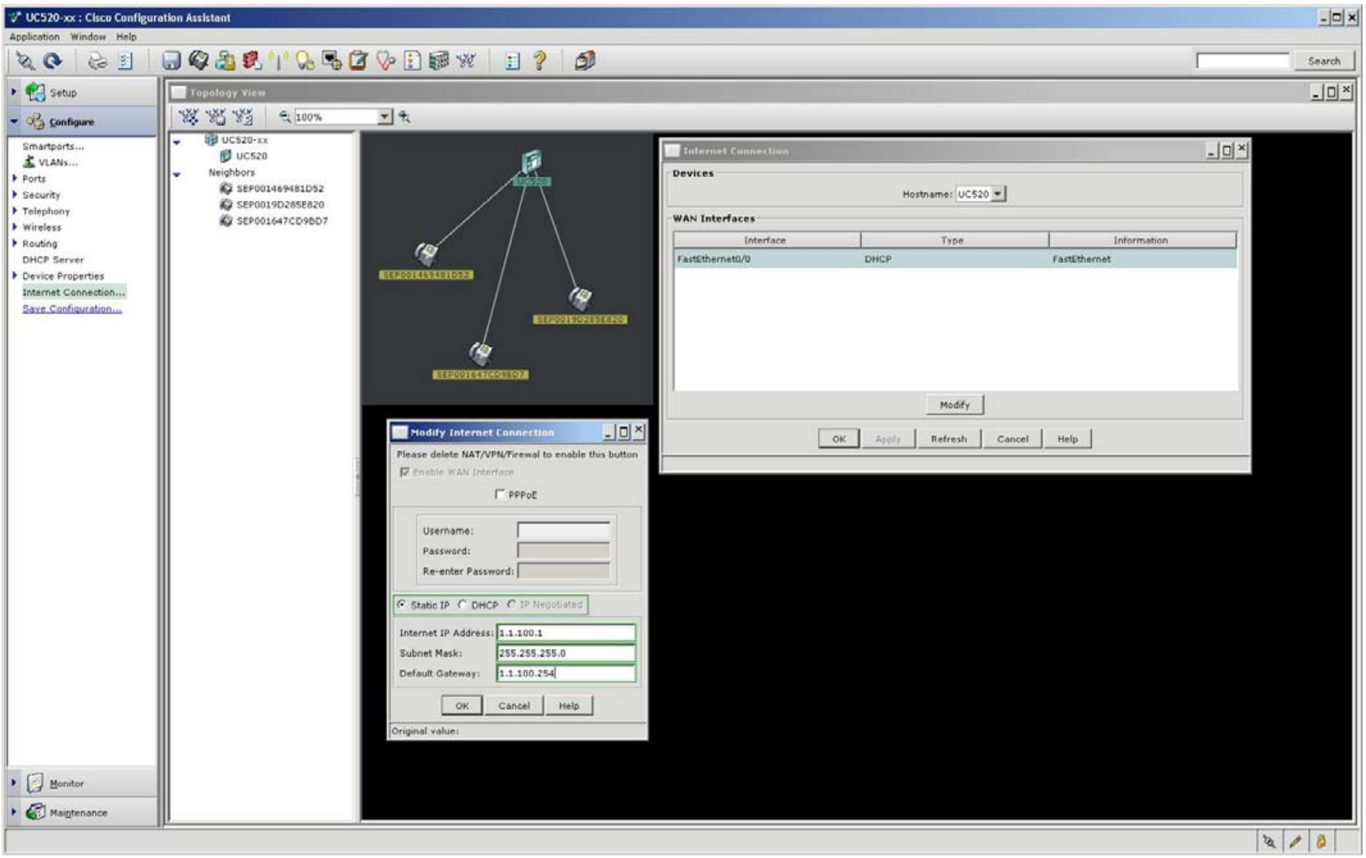


If you recorded the prompt using “Sound Recorder” tool, to upload the prompt, click on the Browse button next to the menu prompt.

If you recorded the prompt using “Prompt Management”, then you need to first refresh the voice window. You can refresh the voice window either by hitting the refresh icon on top left corner. Or you can close and re-launch the voice window. Once the window is refreshed, you can select the prompt from the "Menu Prompt" pull-down menu.



14. Configure the WAN Connectivity. On the left pane go to Internet Connections
- o Click on FastEthernet 0/0 and hit Modify
- o Click on Static IP radio button
- o Enter IP address as 1.1.100.xx (xx is POD # - drop the leading 0, only enter 1 for POD 01)
- o Enter Subnet mask as 255.255.255.0
- o Enter Default Gateway as 1.1.100.254
- o Click on OK to apply changes
- o Click on Apply on the Internet Connection window



Verify Steps:

1. On the IP Phones, make sure the FirstName and the LastName is visible.
2. Call the voicemail access number (401) & Auto Attendant (400) internally and test if calls work. Make sure the AA transfers the call as configured.
3. Press the 'Messages" key on each phone and enroll the users. Use "789" as a password.
4. Place calls between IP Phones and ensure that calls roll over to VM when not answered.
5. Check that the MWI light turns on when a message is left and check messages.
6. Check the paging, call park and pickup features work fine
7. Simulate an incoming PSTN call over the SIP Trunk by going over to the PSTN phone (near proctor desk) and dialing any DIDs for your POD [4085xx120y where xx is POD # and y goes from 0 to 9].
8. Try calling the your AA number from PSTN 4085xx1200 and make sure you hear the prompt recorded earlier.

Note: If you do not hear the prompt, you may need to remove access-list from the CUE interface. The CLI for removing the access-list is listed below:

```
config t
interface Integrated-Service-Engine0/0
no ip access-group 100 in
end
```

9. Call outbound from x202 to the PSTN by dialing the below patterns (there may be a slight delay in getting the calls setup)
    a. 911 (emergency call)
    b. 9911 (emergency call)
    c. 97771000 (local call)
    d. 916507772000 (long distance call) – this should fail in theory. However, there is a limitation with using the Generic SIP Trunk provider pulldown where COR does not apply.

10. Call outbound from x201 to the PSTN by dialing the below patterns

    a. 916507772000
    b. 915102221234 (calls to 510 areacode should fail based on blocked patterns)
    c. 9011441234512345 (international call)

11. Test the hunt-group functionality.

OPTIONAL VoiceView Express (VVE) and IMAP labs are outlined in Appendix B

Note: It is NOT necessary to reset the system at the end of this lab!