

LAB 2: PBX, Voicemail & Auto Attendant

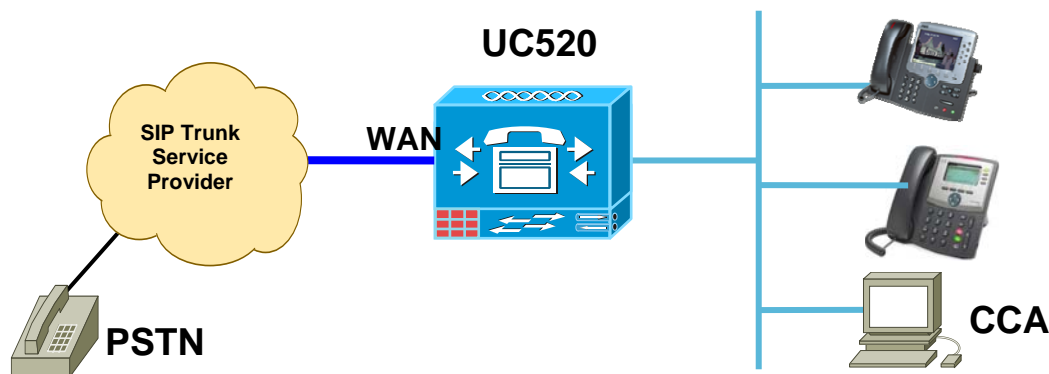
Introduction:

The UC520 supports two voice system configuration types – PBX type and Keysystem type. LAB2 focuses on the PBX system configuration. Keysystem configuration is included as an addendum in appendix B. 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.

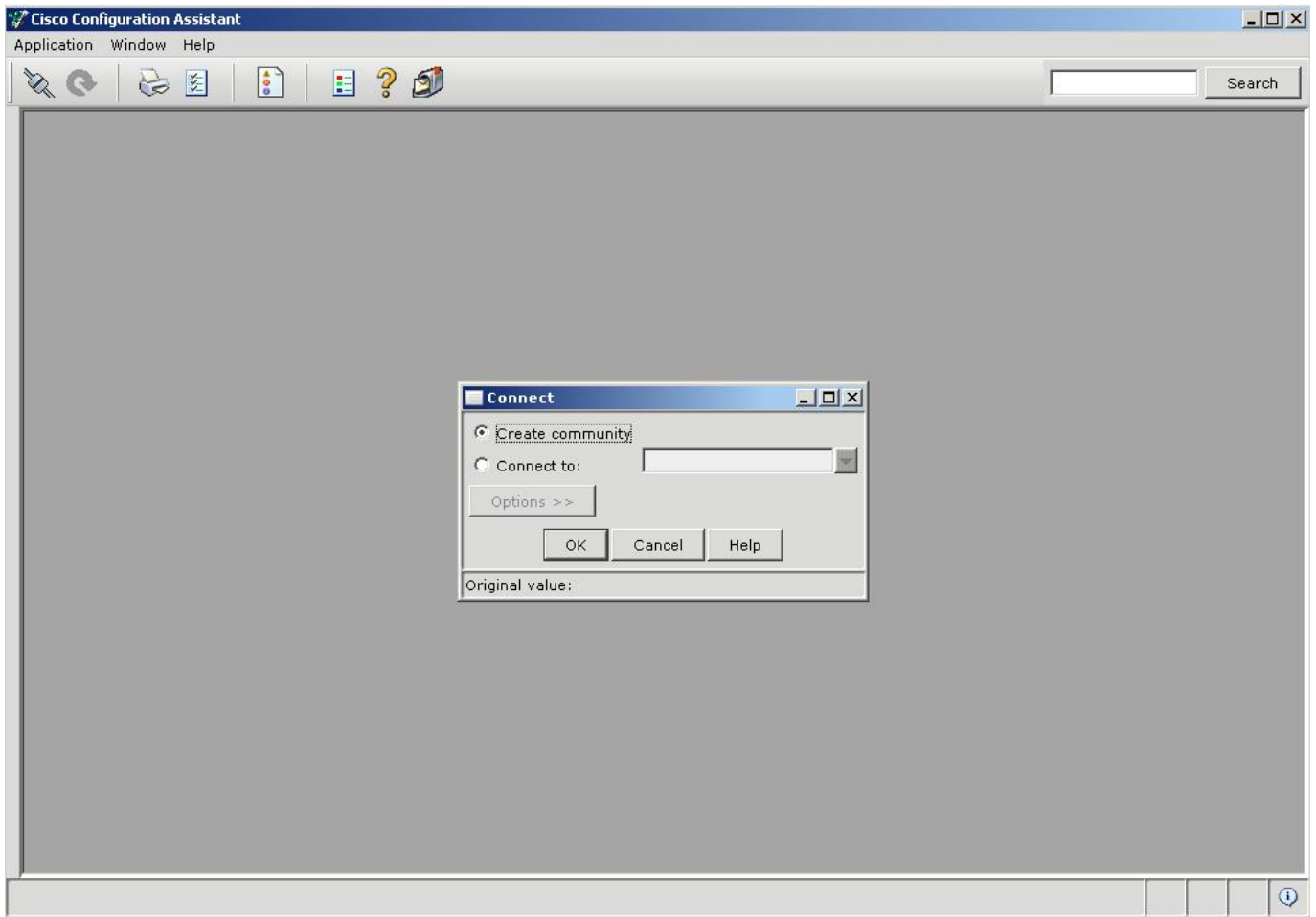
Topology:



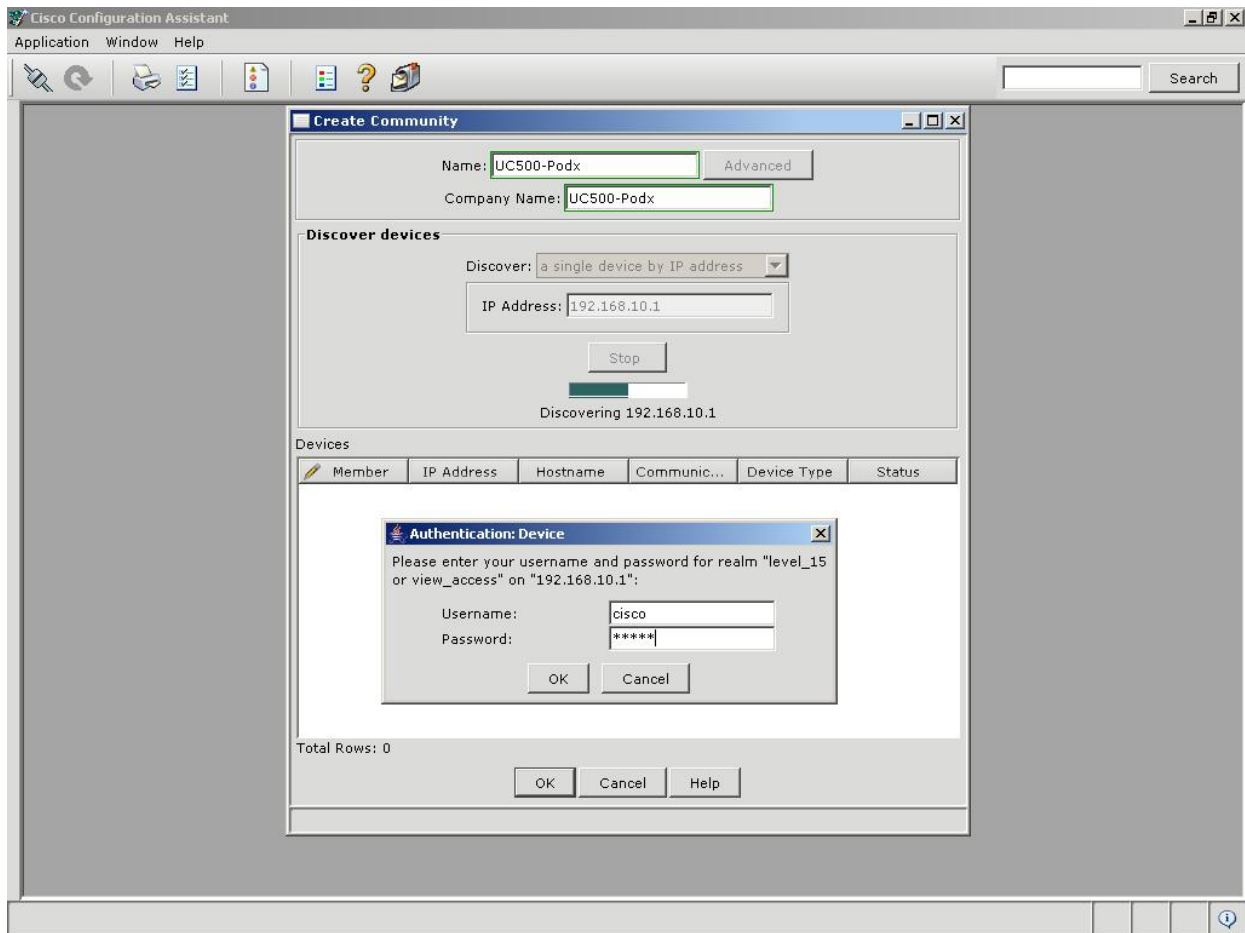
Setup steps:

1. Install Cisco Configuration Assistant (CCA) on your laptop.
2. Connect your laptop to one of the UC520 Ethernet ports or the Ethernet connection on the back of an IP phone (if not already connected) & launch CCA.

3. When you launch CCA, a Connect window will appear. Click on “Create community” and click on OK button.



1. Now you should see a dialog as below:
 - Create a community UC520-xx. For the Company name, use POD-xx (where 'xx' is your Pod #)
 - Choose discover option as “a single device by IP address” and use IP address 192.168.10.1
 - Click Start - now you should get a “Security Certificate Alert”, click ‘Always’.
 - Enter username/password as cisco/cisco.



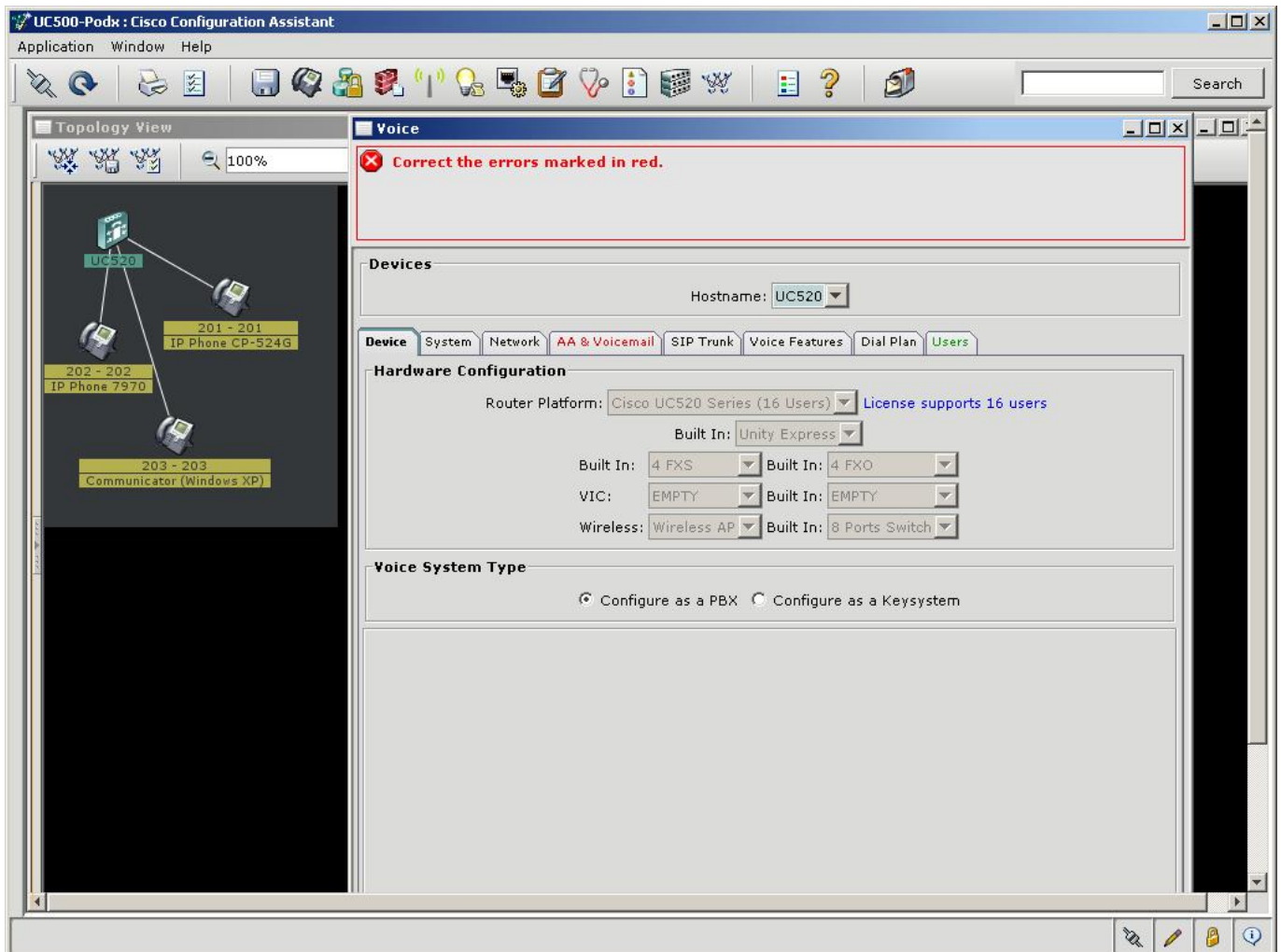
- Once discovery is complete (this may take a few minutes), click ‘OK’ to view the topology. The progress for discovery can be seen at the bottom of the screen.
- Check the topology to ensure all the phones show up. You may or may not see the CIPC device, depending on where your laptop is plugged in. If it is plugged into POE port of the UC520, the CIPC will show up. If the laptop is plugged into the back of an IP phone, the CIPC will not show up. The ‘switch’ device showing is the Lab Switch that we are using to simulate the Internet.
- Click on ‘Configure -> Telephony -> Voice’ on the side bar or click the ‘Phone’ icon on the top menu bar. In the next steps, we will configure the various tabs – “Device”, “System”, “AA & Voicemail”, “Voice Features”, “Dial Plan” and “Users”.

NOTE: Before making any changes in CCA, ensure that the ‘Apply’ and ‘Refresh’ buttons at the bottom of the screen are active. These are grayed out while CCA reads the configuration from the UC500.

NOTE: DO NOT CLICK “OK” or “Apply” until you complete steps 4 to 12.

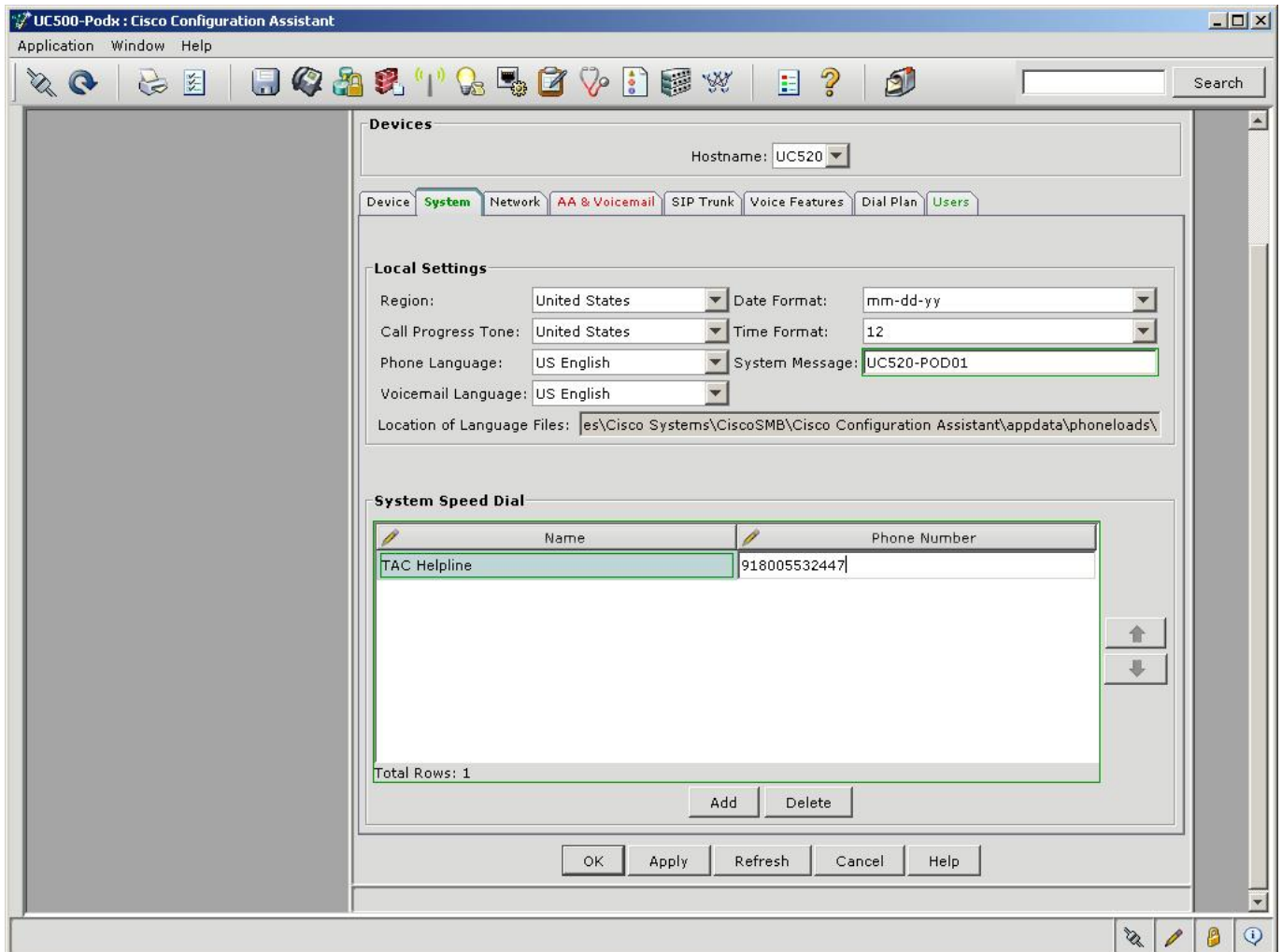
2. Check the topology to ensure all the connected phones are showing up. Unplug any lines that are connected to the FXO ports on the PODs.

3. Click on “Configure -> Telephony -> Voice” on the left navigation bar or click on the Phone icon on the top menu bar. Select “UC520” for hostname and click OK. 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:
 - o Change the System Message to “UC520-PODxx”
 - o Configure a system-wide speed dial. Click on the “Add” button and configure a name and a corresponding number (eg. “TAC Helpline”, Number: 918005532447). 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:
 - o Set the AA Access Extension as 400 & the AA PSTN number as 4085xx1200
 - o For the Auto Attendant script, select the aa_transfer.aef
 - o For the Menu Prompt, select “AAWelcome.wav”
 - o Set the following parameters for the Auto Attendant script:
 - Option 0: 203
 - Option 1: 201
 - Option 2: 202
 - Option 9: 401
 - o 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”

Device System Network **AA & Voicemail** SIP Trunk Voice Features Dial Plan Users

Auto Attendant

Auto Attendant Extension: 400

Auto Attendant PSTN Number: 4085011200

Auto Attendant Script: aa_transfer.aef

Script Parameters

Menu Prompt: AAWelcome.wav Browse

Option 0: 203 Option 1: 201

Option 2: 202 Option 3: None

Option 4: None Option 5: None

Option 6: None Option 7: None

Option 8: None Option 9: 401

Voicemail

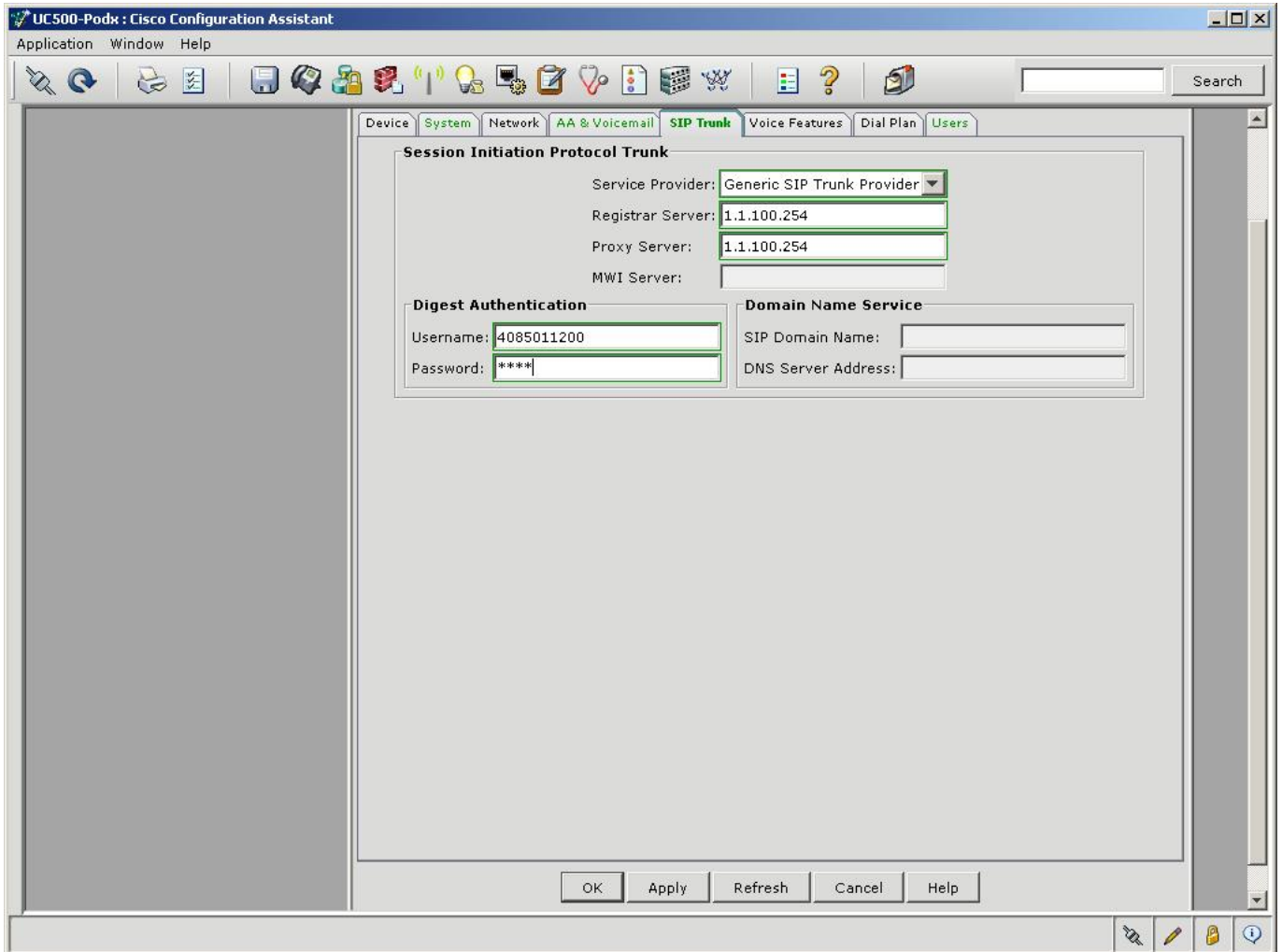
Voicemail Access Extension: 401

Voicemail Access PSTN Number: 4085011209

OK Apply Refresh Cancel Help

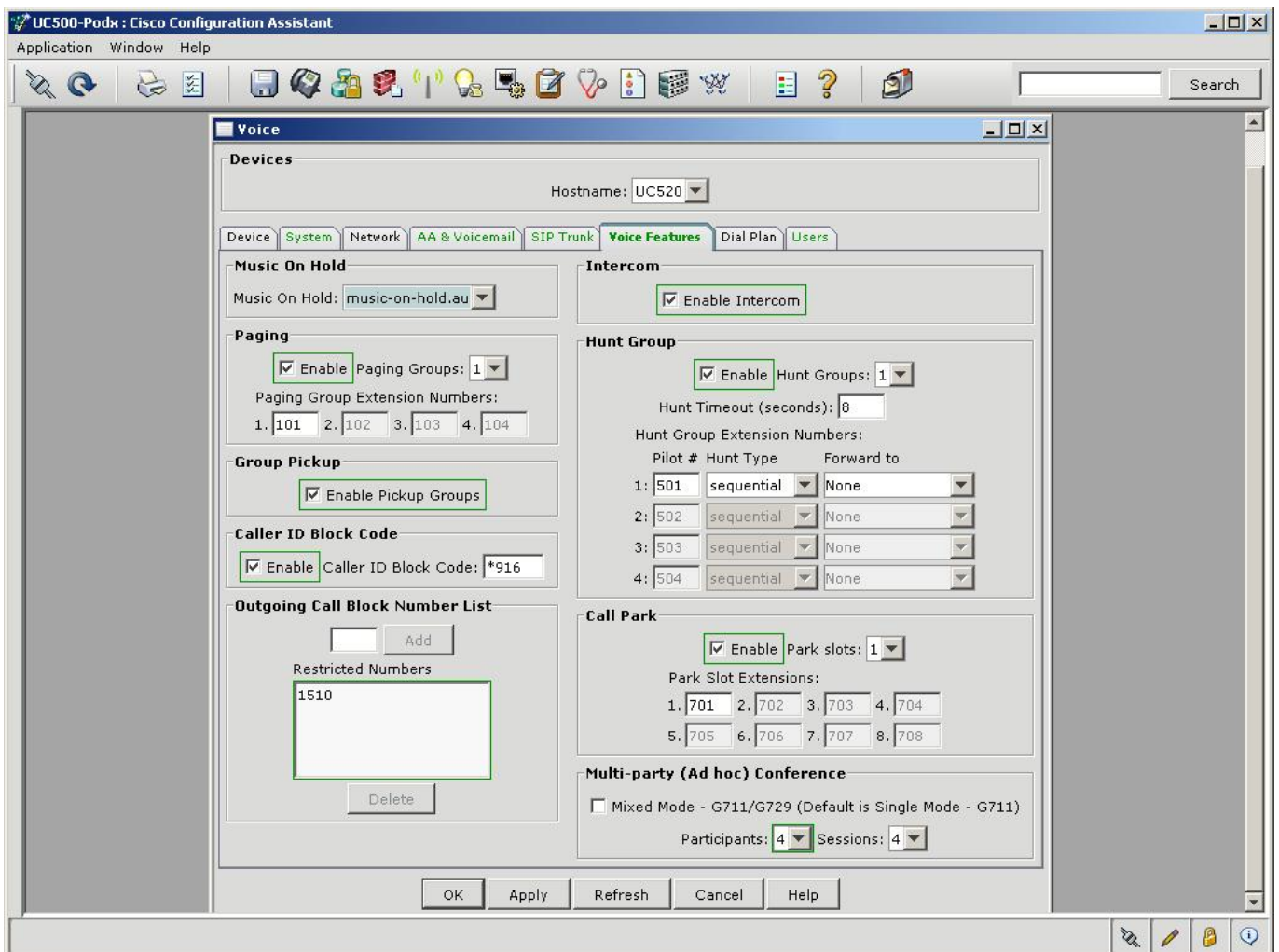
Note: Recording new menu prompts will be covered in Lab3.

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



8. Click on the “Voice Features” tab

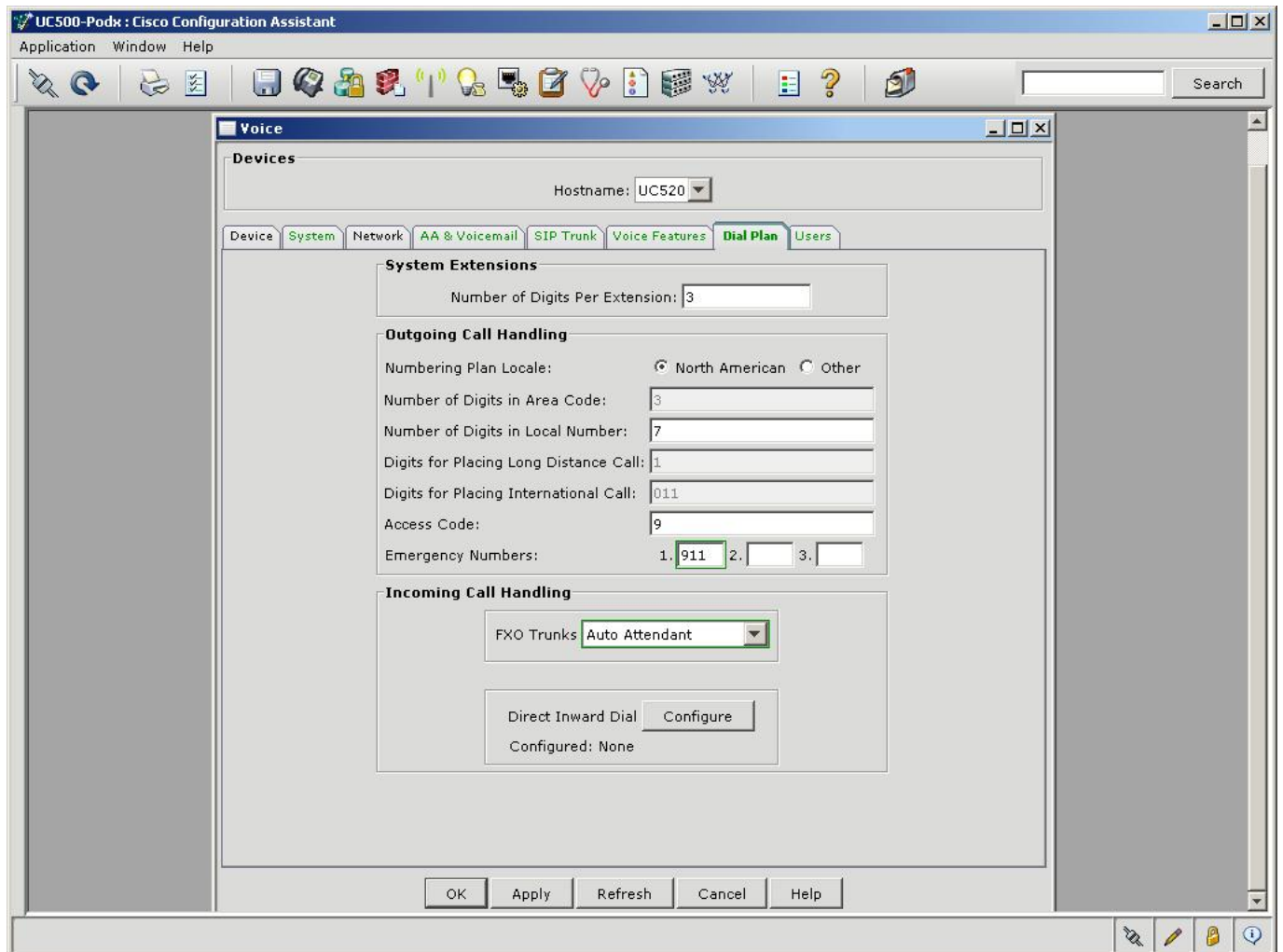
- Under the “Paging Parameters”, check the “Enable Number of Paging Groups”, select 1. Configure the first Paging Group Extension Number as 101.
- Under the “Group Pickup Parameters”, check the “Enable Number of Pickup Groups”, select 1.
- Under the “Caller ID Block Code”, check the “Enable Caller ID 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. Use the default Hunt type, sequential
- 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). Leave “Mixed Mode” check box blank.



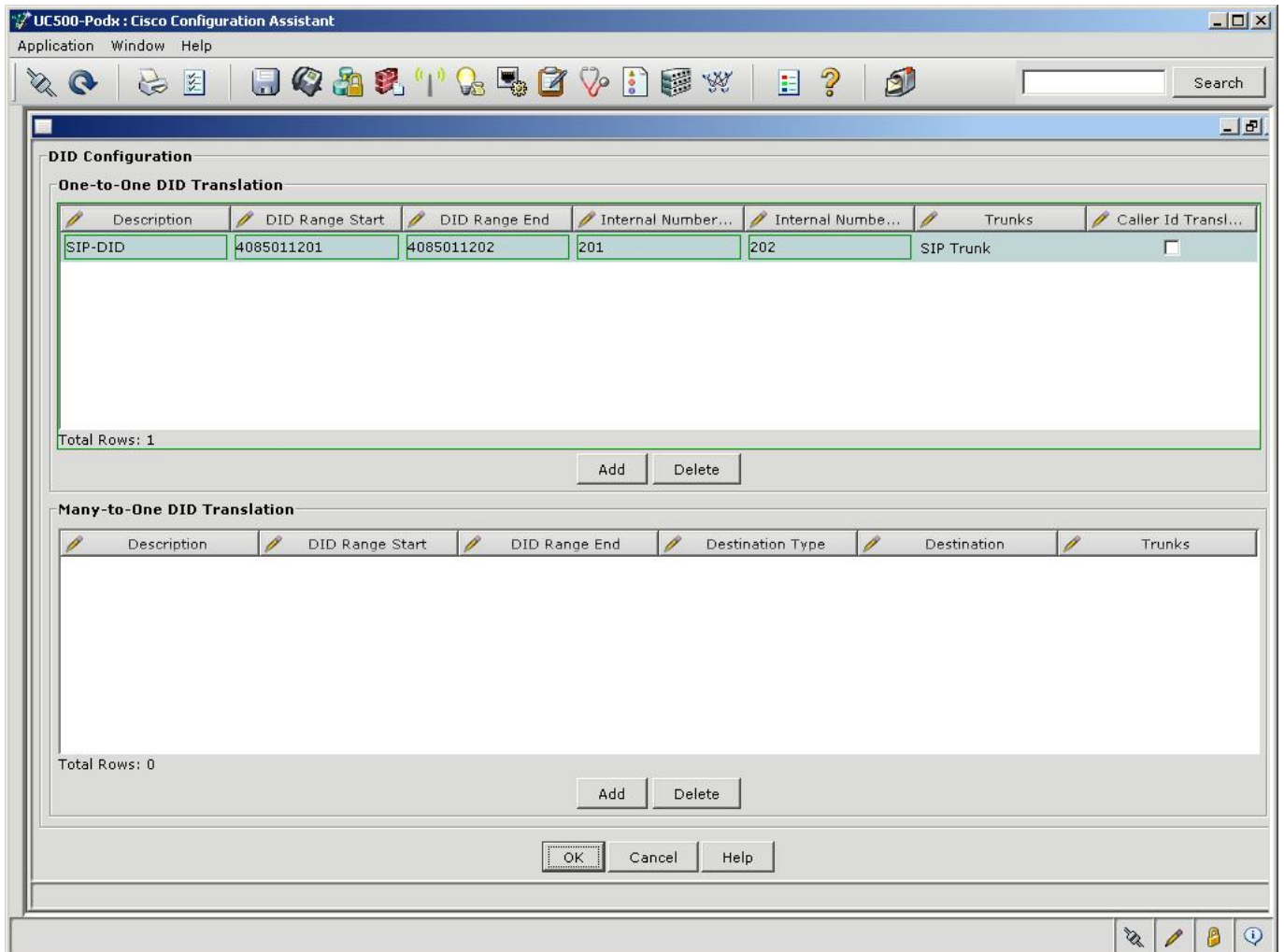
9. Click on the “Dial Plan” tab:

- 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.

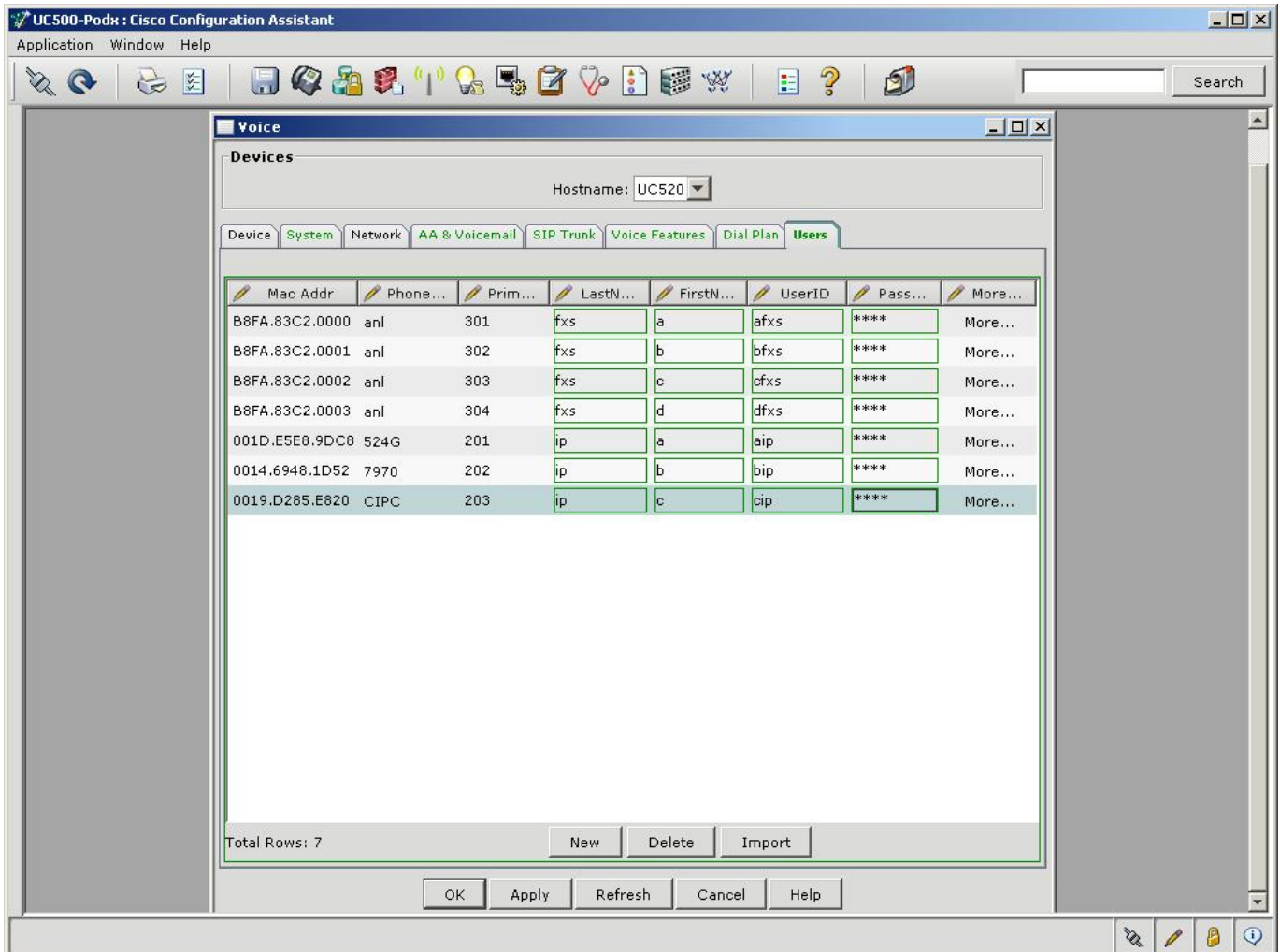


- Click on the “Configure” button for Direct Inward Dial. We will use this feature to add PSTN DID numbers to 2 of the IP phone users.
 - Click on Add range for “One-to-One DID Translation”
 - Enter a description such as SIPPSTN
 - Enter starting DID range as 4085xx1201 and ending as 4085xx1202
 - Enter 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



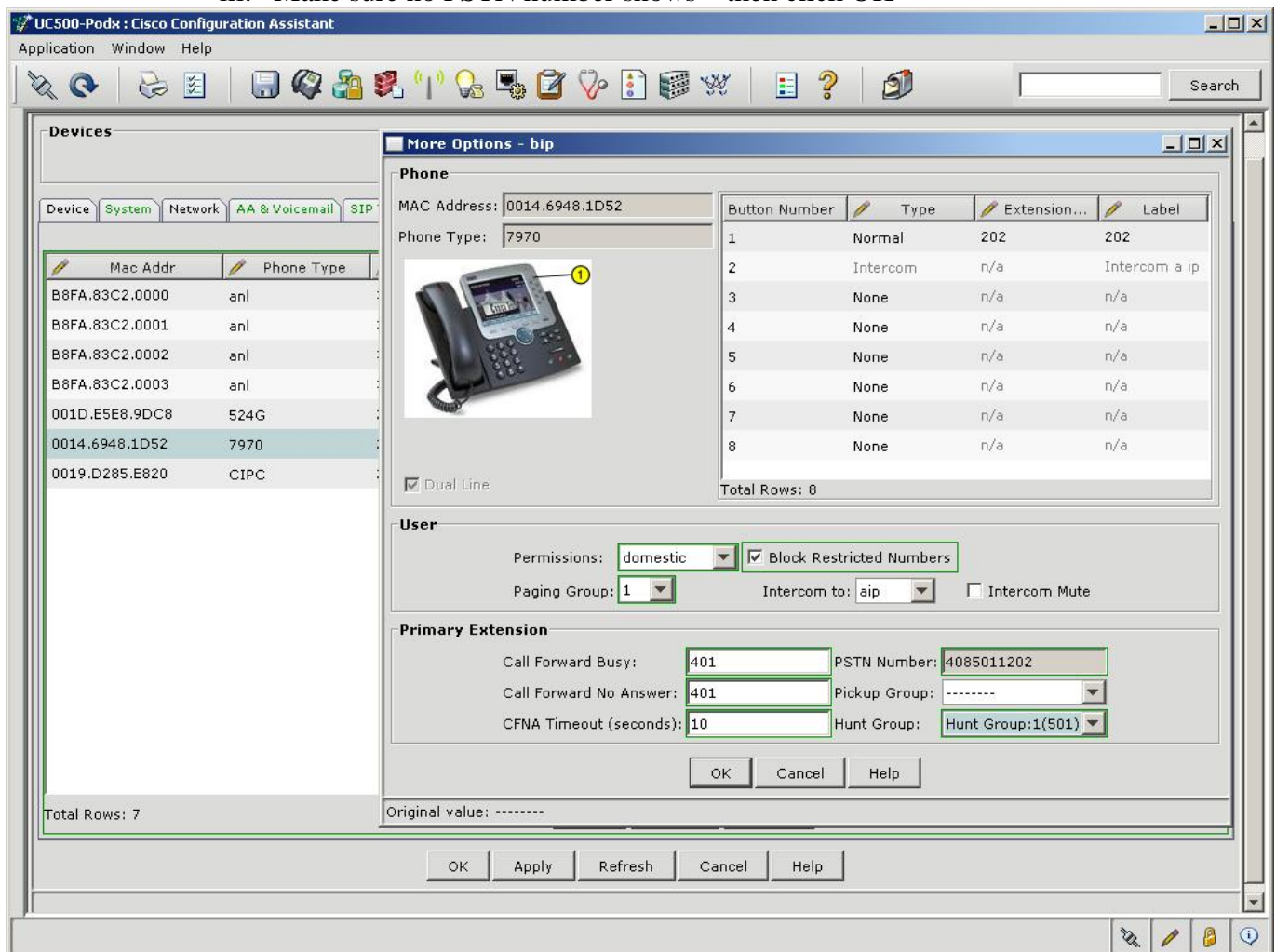
10. Click on the “Users” tab:

- 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. Verify that the Intercom to aip is configured
 - v. 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

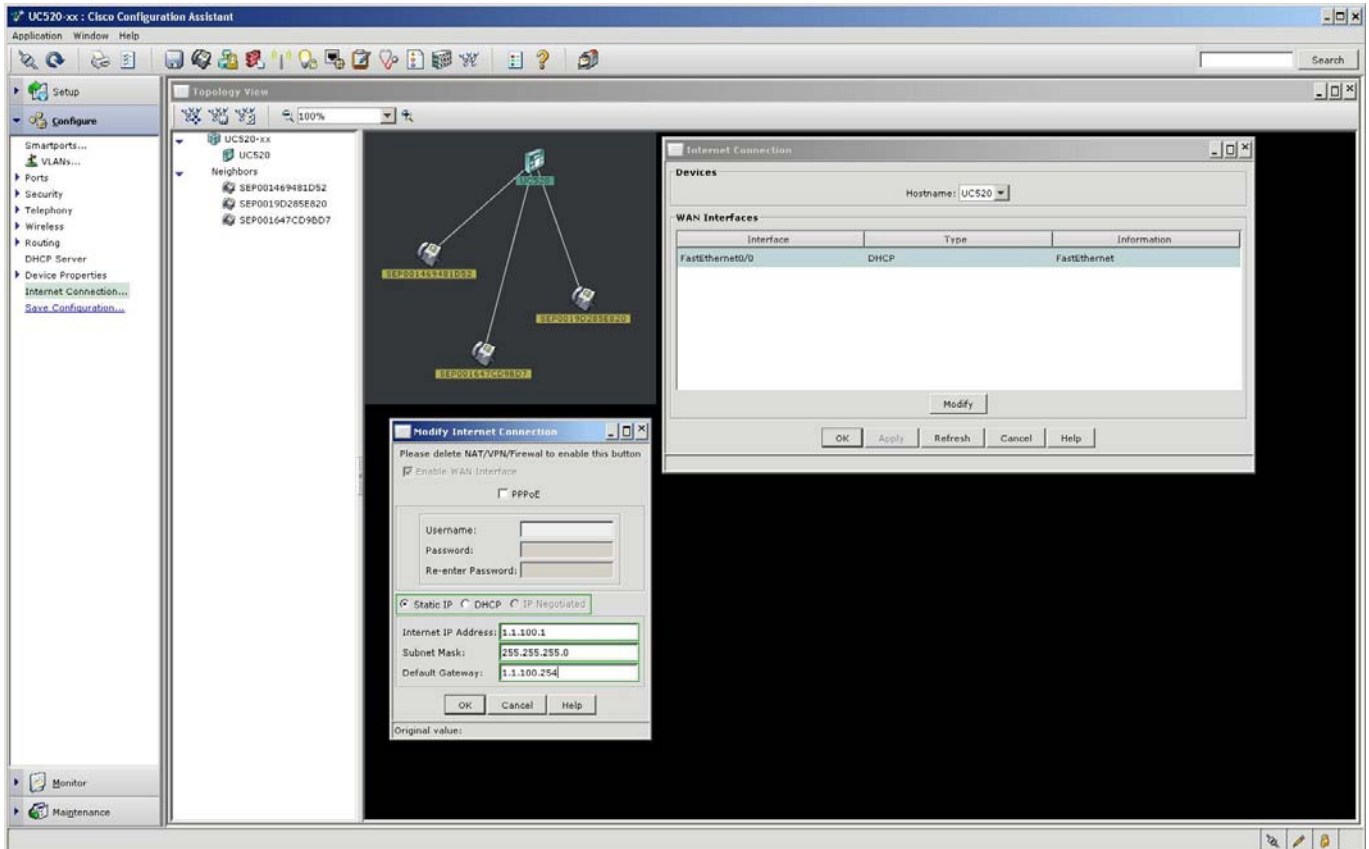


12. Click on “ok” button at the bottom of the screen and observe the progress bar. Click OK when the “Configuration successfully sent to UC520” message pops up. It may take several minutes for the configuration to be applied to the UC500.

13. Configure the WAN Connectivity. On the left pane go to Internet Connections

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

Note: This step is required for SIP trunk lab to function properly.



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.
6. Press "Services" key on the IP Phone and select the "CME Service URL" to launch Voice View Express (VVE). Use the password from step 3 for PIN. Follow the directions to optional labs as indicated at the bottom of this page.
7. From any phone, press "Directory" key and then select the "Local Speed Dial" option. Verify that the system-wide speed-dial configured in the system tab appears here.
8. Dial 101 from any phone to check the paging feature.
9. For 201, press the Intercom button to verify that intercom to 202 works fine.
10. For 201, dial *916202. On 202, the caller-id should show "From Private". This indicates that the callerID block code is working fine.
11. For 202, call 203. From 201, press *more* > *GPickup* to verify that call pickup works fine.
12. Simulate an incoming PSTN call over the SIP Trunk by going over to the PSTN phone (near proctor desk) and dialing any DID's for your POD [4085xx120y where xx is POD # and y goes from 0 to 39].
13. Try calling 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
```

14. 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.
15. 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)
16. From CIPC, dial 501 to test the hunt-group functionality. The call should hunt to 201 and 202.

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