VOIP: RINGTONE, VOICEMAIL, MUSIC ON HOLD, CALL OUT

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Purpose

The purpose of this lab is to continue practicing our ability to set up networks in a Virtual Machine, and understand different features and functions available in Cisco Unified Communications Manager. We also explored the role of Cisco Unity Connection Manager in setting up a voicemail system.

Background Information

It is normally not possible for an internal private phone line to be able to call an outside phone. It is like talking to a stranger. You are comfortable talking with your friends, because you are all in the same friend group and you all know each other by name. However, you would not normally talk to any stranger. But, if you and stranger had a mutual friend, it would be easier to communicate. The same is true with calling out. The VoIP phones are only basically configured to be able directory numbers within the network, but through the mutual friend, the gateway and a phone line, the phone is able to communicate with the outside world. Placing someone on hold is a very necessary feature in a business setting. When dealing with many customer requests, it is common courtesy to place customers on hold and play music to inform the customer that their call is important. A custom ringtone is a widely used feature important for identifying calls from various individuals. People tend to use their favorite tunes as ringtones for close family or friends.

Lab Summary

Since this lab was an extension of the basic VoIP configuration lab, I maintained the same setup with Cisco Unified Connection Manager and R1 as a DHCP server. First, I set up music on hold. To do so, I found a MP3 music file and converted it to a WAV file using Audacity. Then, I uploaded the file as a Media Resource under the MOH Audio File Management tab. Now that the file was a media resource, I could add it to the MOH Audio Source and add my file. To specify the MOH Audio file for a phone, I entered the Phones page and selected my newly added file as the User Hold MOH Audio Source.

Creating a custom ringtone was a similar process, but with more specificities in Audacity. First, I created a folder on my desktop to store all TFTP server files. Then, using tftpd32, I set my VM as the TFTP server and uploaded a remote file called 'Ringlist.xml'. Next, I found an appropriate MP3 file, and in Audacity, changed the project rate to 8000 Hz and exported the audio as an uncompressed file with a raw header and U-Law encoding. In the TFTP server folder, then I opened the Ringlist.xml file and added my raw file to the end of the list. Then, in CUCM, I uploaded the same ringtone raw file in TFTP file management. After restarting the TFTP server in Cisco Unified Serviceability, the new ringtone was added to the phone.

To set up voicemail, I first created a new VM and installed Cisco Unity Connection using the OVAs provided. CUC is necessary to create the service group and users in the voicemail system. The setup for CUC in the VM is nearly identical to the setup for CUCM, just with a different IP address within the same network. In CUCM, I created 2 message waiting numbers for each of my 2 phones. Then, I used the Cisco Voice Mail Port Wizard to set up the remaining steps as prompted by the wizard. In addition, I created a Line Group and Hunt List, and added the Line group to the Hunt List and Hunt

Pilot. Next, I created a Voice Mail Pilot with the same number as the Hunt Pilot. Then, I created a Voice Mail Profile and set the Pilot to be the Voice Mail Pilot that I created. After the CUC installation was complete, I created a new phone system, port group, and added 2 new ports. Then, I created 2 users that corresponded with the respective directory numbers on my 2 VoIP phones. Then, voicemail was setup and I was able to leave a message for the other user on my phone.

To set up call out, I first set my router as an H323 gateway in CUCM. Then I created a route pattern beginning with 9*! and set the gateway to the address of the router. On the router, I configured the gateway with the H323 protocol and enabled VoIP calls on the voice port.

Lab Commands

Both call out and call in:

`R1(config)#voice service voip`

This command allows you to enter the voice-service configuration mode

`R1(conf-voi-serv)#h323`

This command sets the gateway to use H.323 protocol, which provides interoperability between H.323 endpoints. H.323 uses TCP on port 1720.

`R1(conf-voi-serv)#allow-connection h323 to h323`

This command allows two H323 endpoints to communicate with each other through the gateway device

Call in Commands:

`R1(config)#voice call send-alert`

This command sets the gateway to send alert messages after it receives a call setup message, instead of sending a progress message. The default is to send a progress message.

`R1(config)#voice rtp send-recv`

In order for phones from an outside network to be able to call in, a two-way voice path must be established. This command establishes that path when an RTP channel is opened.

`R1(config)#voice-port 0/3/0`

This command allows you to enter the interface configuration mode for the voice port on the router. It should be a FXO (Foreign Exchange Office) voice port.

`R1(config-voiceport)#caller-id enable`

This command is used to allow the sending or receiving of caller ID information, and is set in voice port configuration mode.

```
R1(config-voiceport)#connection plar opx [directory number]
```

This command is used to specify a PLAR connection. PLAR stands for Private Line Automatic Ringdown, and is a mechanism to create switch VoIP calls without digit dialing.

```
R1(config)#dial-peer voice [number] voip
```

This command enters dial peer configuration mode and defines a VoIP dial peer, identified by the number. The voip keyword shows that the dial peer is using voice encapsulation over an IP network.

```
R1(config-dial-peer)#destination-pattern [directory number]
```

This command matches dialed digits to an internal phone directory number.

```
R1(config-dial-peer)#dtmf-relay h245-alphanumeric h245-signal
```

In terms of DTMF (Dual Tone –Multi Frequency) relay, the two types are in-band and out of band. By enabling this command, the DTMF tones are prevented from being distorted, because the DTMF tones are transported out of band.

```
R1(config-dial-peer)#session target ipv4:[ip address of CUCM Server]
```

This command sets the session target for voice multicast peers. In the case of this lab, the IP address was of the CUCM server.

Call out Commands:

```
R1(config)#dial-peer voice [number] pots
```

This command enters dial peer configuration mode and creates a local dial peer to connect to a POTS (Plain Old Telephone Service) interface. The number is used to identify the unique dial peer.

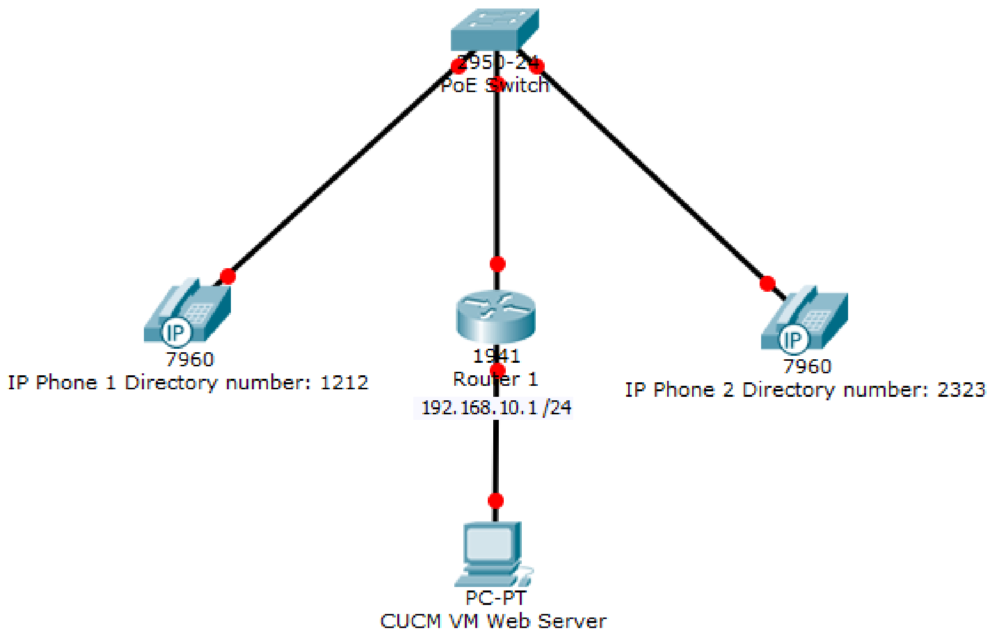
```
R1(config-dial-peer)#port 0/3/0
```

This command maps the dial peer to a specific interface. In the case of our lab, the port was the voice port with call in configured.

```
R1(config-dial-peer)#destination-pattern [string]T
```

This command matches dialed digits to a telephony device, and instantiates the Timer. The router will collect dialed digits until the timer of 10 seconds expires.

Network Diagram



Network: 192.168.10.0/24

Configurations

R1 show run:

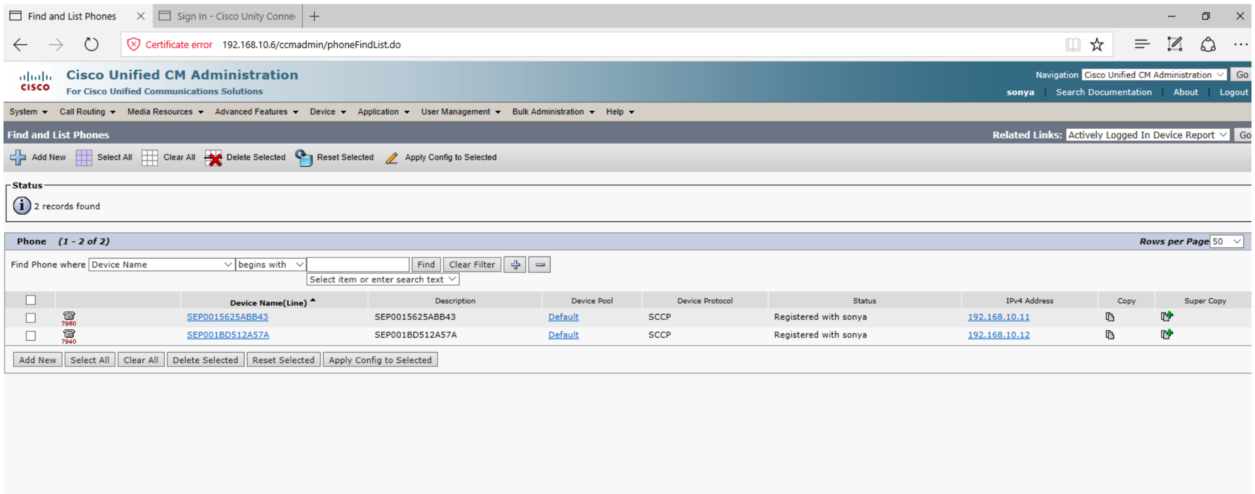
```
hostname R1
boot-start-marker
boot-end-marker
logging message-counter syslog
no aaa new-model
memory-size iomem 10
no network-clock-participate slot 1
dot11 syslog
ip source-route
ip cef
ntp master
ip dhcp excluded-address
192.168.10.1 192.168.10.10
ip dhcp pool VOIP
network 192.168.10.0 255.255.255.0
default-router 192.168.10.1
dns-server 192.168.10.1
option 150 ip 192.168.10.6
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
voice call send-alert
voice rtp send-recv
voice service voip
h323
voice-card 0
no dspfarm
voice-card 1
no dspfarm
vtp domain cisco
vtp mode transparent
archive
log config
hidekeys
interface FastEthernet0/0
ip address 192.168.10.1
255.255.255.0
duplex auto
speed auto
ip forward-protocol nd
no ip http server
no ip http secure-server
control-plane
voice-port 0/3/0
caller-id enable
dial-peer voice 1 voip
session target ipv4:192.168.10.6
dtmf-relay h245-alphanumeric h245-
signal
dial-peer voice 1 pots
destination-pattern 9T
prefix 9
port 0/3/0
line con 0
line aux 0
line vty 0 4
login
transport input all
scheduler allocate 20000 1000
ntp master
end
```

R1#show voice port 0/3/0

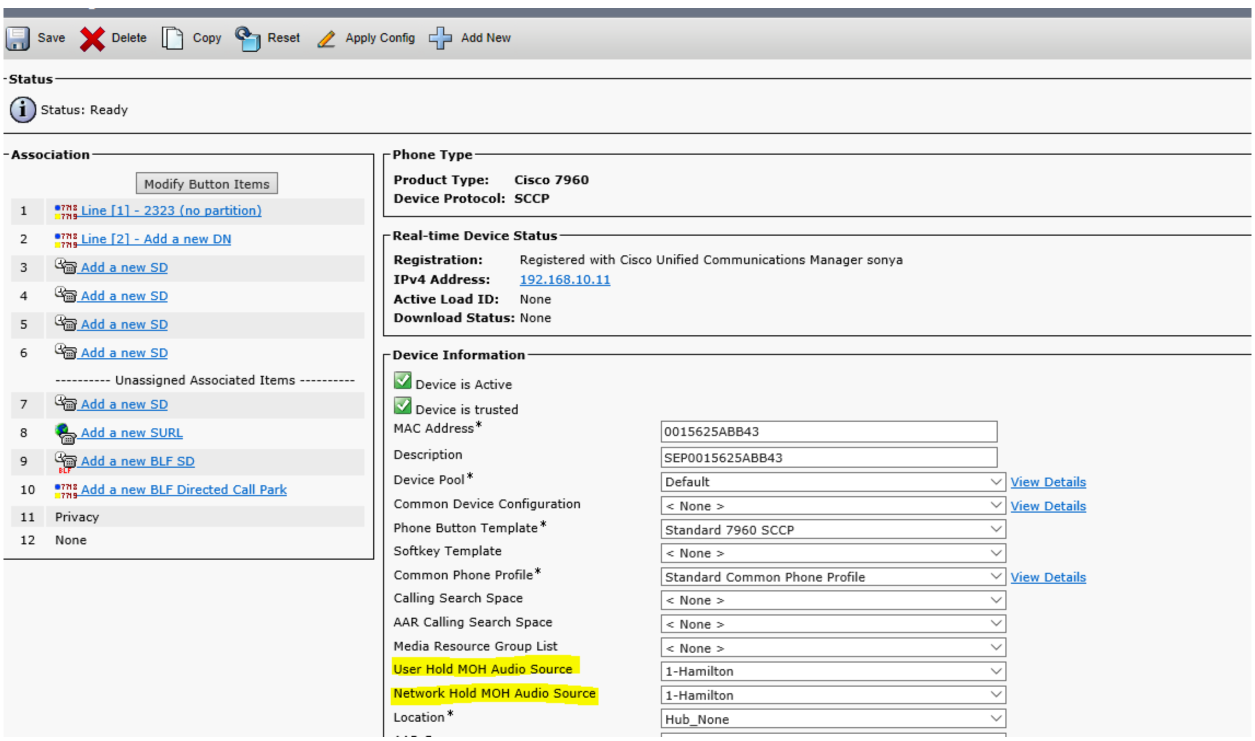
```
Foreign Exchange Office
 Type of VoicePort is FXO
 Operation State is DORMANT
 Administrative State is UP
 The Last Interface Down Failure
Cause is Administrative Shutdown
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to
-38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8
ms
 Connection Mode is plar
 Connection Number is 2000
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call-Disconnect Time Out is set to
60 s
 Ringing Time Out is set to 180 s
 Region Tone is set for US
 Analog Info Follows:
 Currently processing Voice
 Maintenance Mode Set to None (not
in mtc mode)
 Number of signaling protocol
errors are 0
 Impedance is set to 600r Ohm

 Voice card specific Info Follows:
 Signal Type is loopStart
 Number Of Rings is set to 1
 Supervisory Disconnect active
 Hook Status is On Hook
 Ring Detect Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Dial Type is dtmf
 Digit Duration Timing is set to
100 ms
 InterDigit Duration Timing is set
to 100 ms
 Pulse Rate Timing is set to 10
pulses/second
 InterDigit Pulse Duration Timing
is set to 750 ms
```

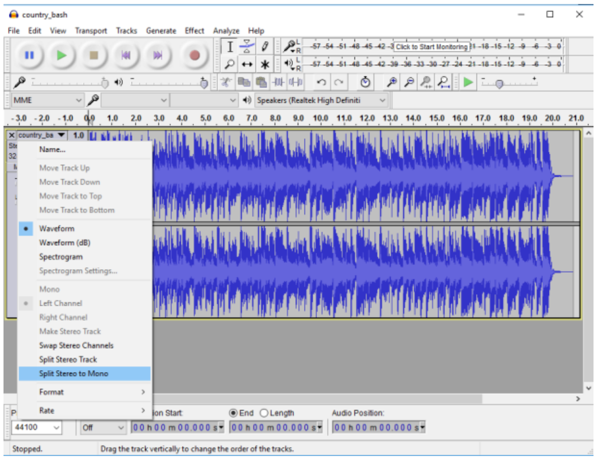
These are fully registered and basic configured phones, as a continuation from the VoIP part 1 lab.

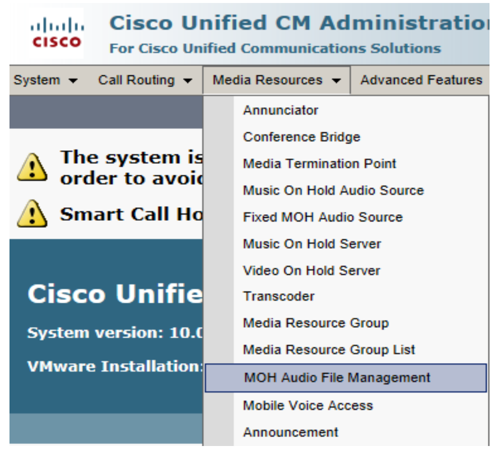


Music on Hold configuration – to verify, check in the phone profile:

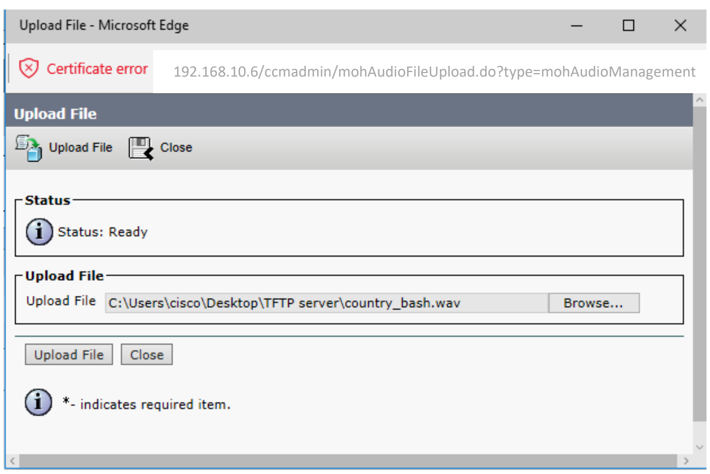


First, find a song and use Audacity to split the audio from stereo and mono. Then convert your file to a .wav

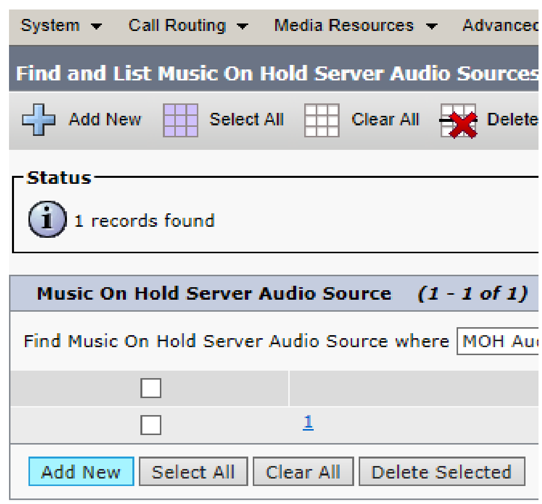




Then, in CUCM, hover over the Media Resources tab and select "MOH Audio File Management"



Next, upload your file to the page.

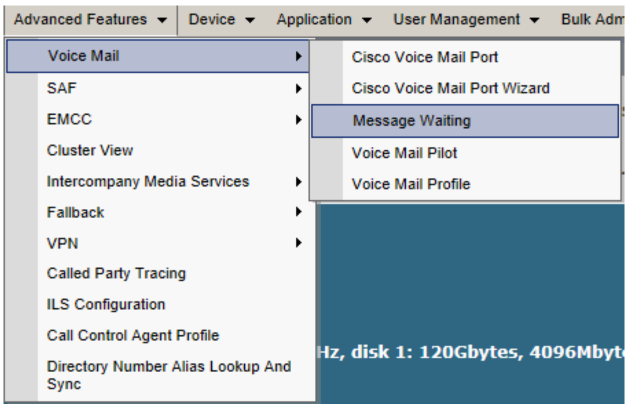


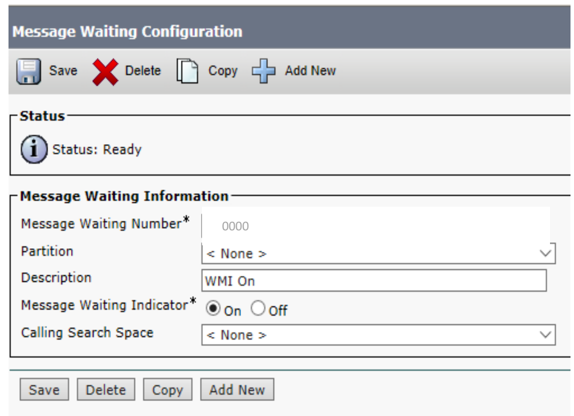
Hover over Media Resources > Music on Hold Audio Source. Then add your file again.

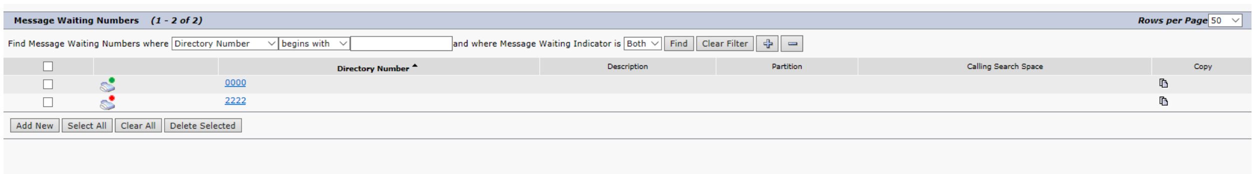
Then, navigate to the phones page and select your audio source for the User MOH Audio Source and Network Hold MOH Audio Source (see first image).

Voicemail Configuration:

First, I installed Cisco Unity Connection Manager, which was the same process as installing CUCM. Then, under Advance Features > Voice Mail > Message Waiting, add 2 message waiting numbers for each of the phones. I used the numbers 0000 and 2222. Turn one of the Message Waiting Indicators on.

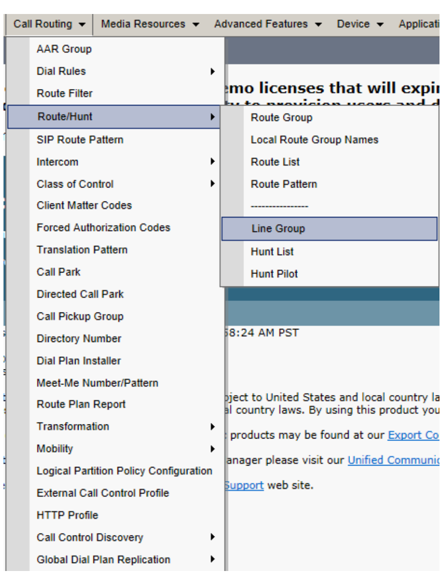


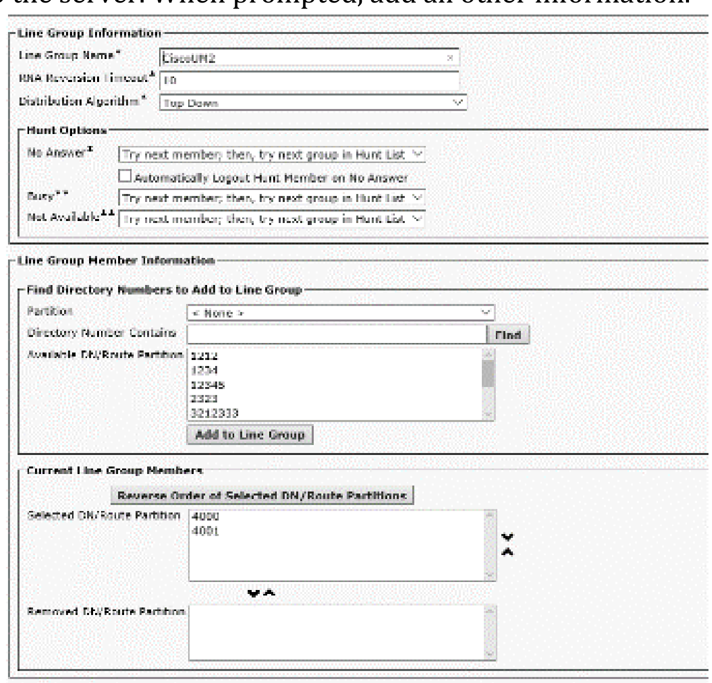




Next, go to Advanced Features > Voice Mail > Cisco Voice Mail Port Wizard

Name your Cisco Voice Mail Server and add 2 ports to the server. When prompted, add all other information.

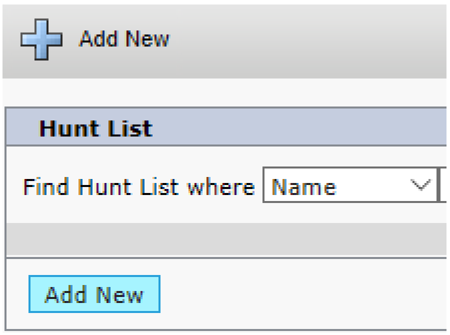


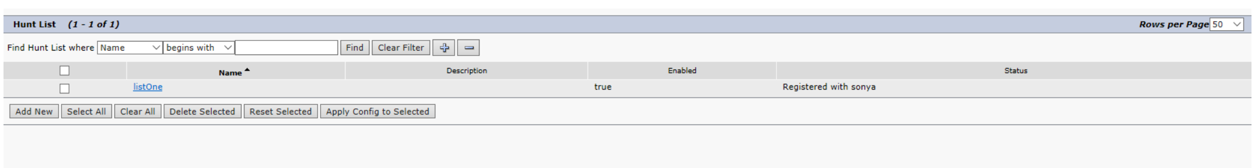


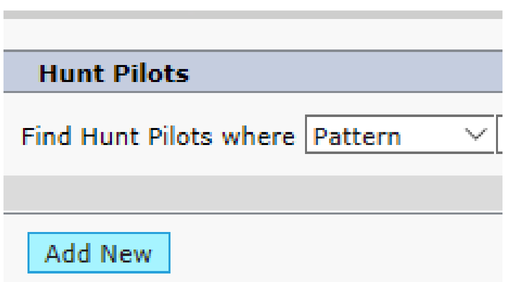
Then, go to Call Routing > Route/Hunt > Line Group

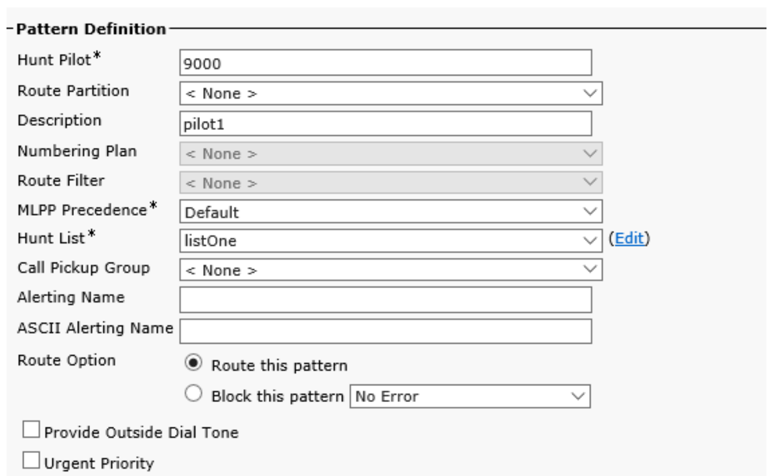
Add a new Line Group. For the Directory Numbers, these should be the 2 numbers that you created in the Cisco Voice mail Port Wizard.

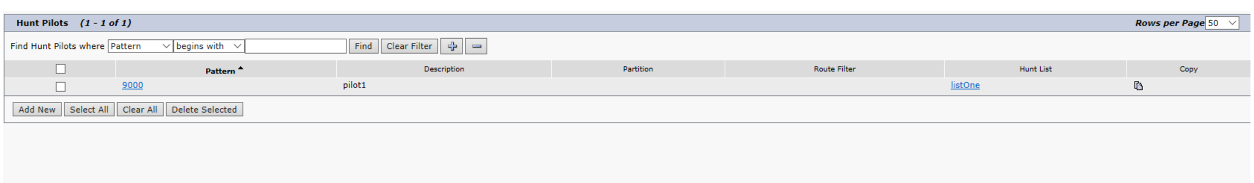
Then, go to Call Routing > Route/Hunt > Hunt List and add a Hunt List. Then, go to Call Routing > Route/Hunt > Hunt Pilot and add a Hunt Pilot. Then add your newly created hunt list to your hunt pilot. The number for the hunt pilot should be the same as your Voice mail Pilot.



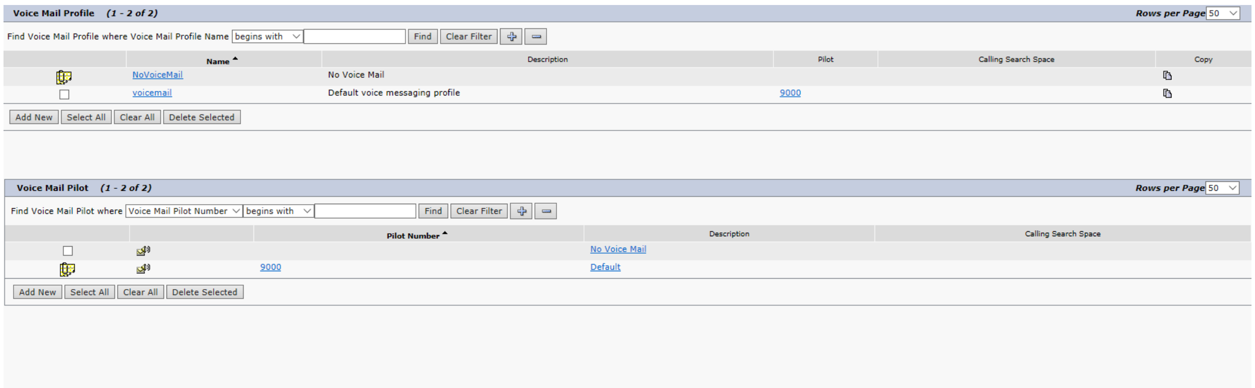




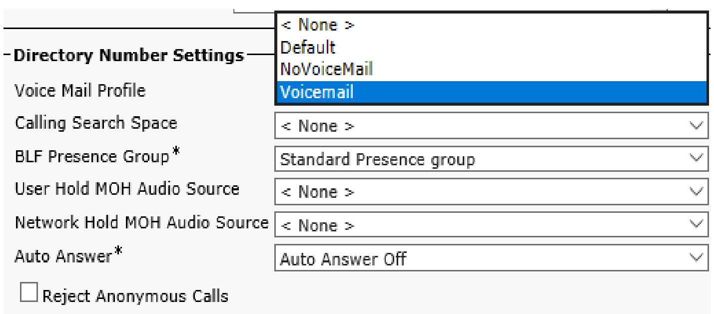


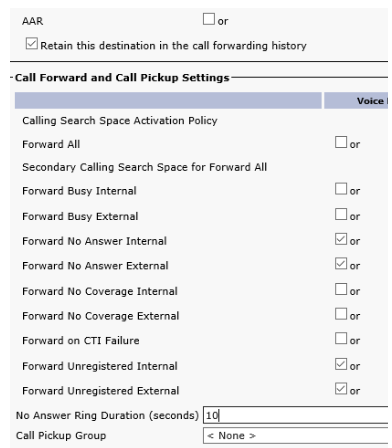


Next, create a Voice Mail Pilot and Voice Mail Profile. To do so, navigate to Advanced Features > Voice Mail > Voice Mail Pilot or Voice Mail Profile.

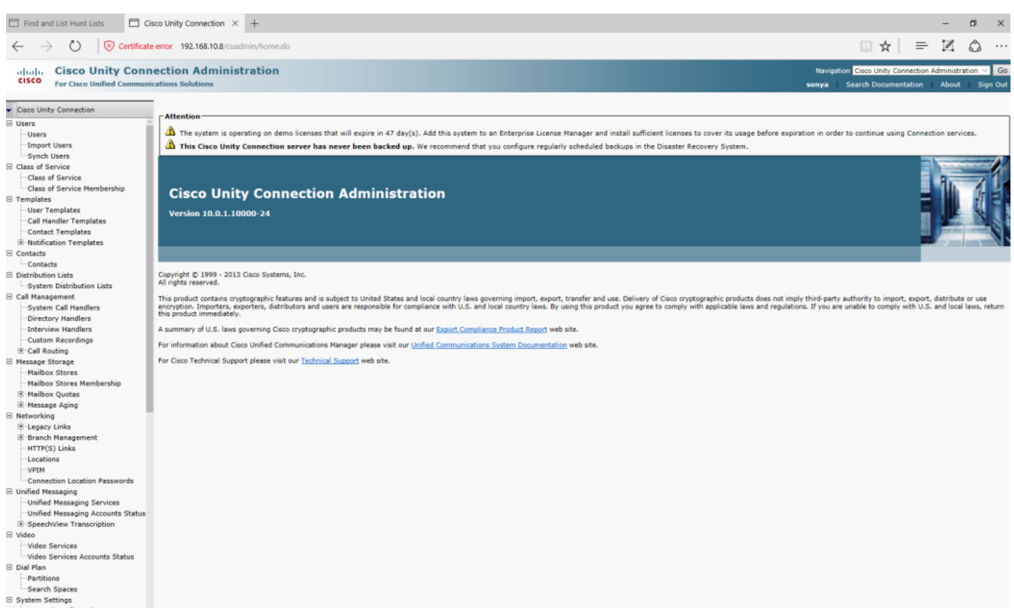


Then go to Device > Phone, select the phone to configure, and set the Voice Mail Profile to the one you created (located in the Line). Also, set the No Answer Ring Duration to 10 seconds and check the boxes listed below.

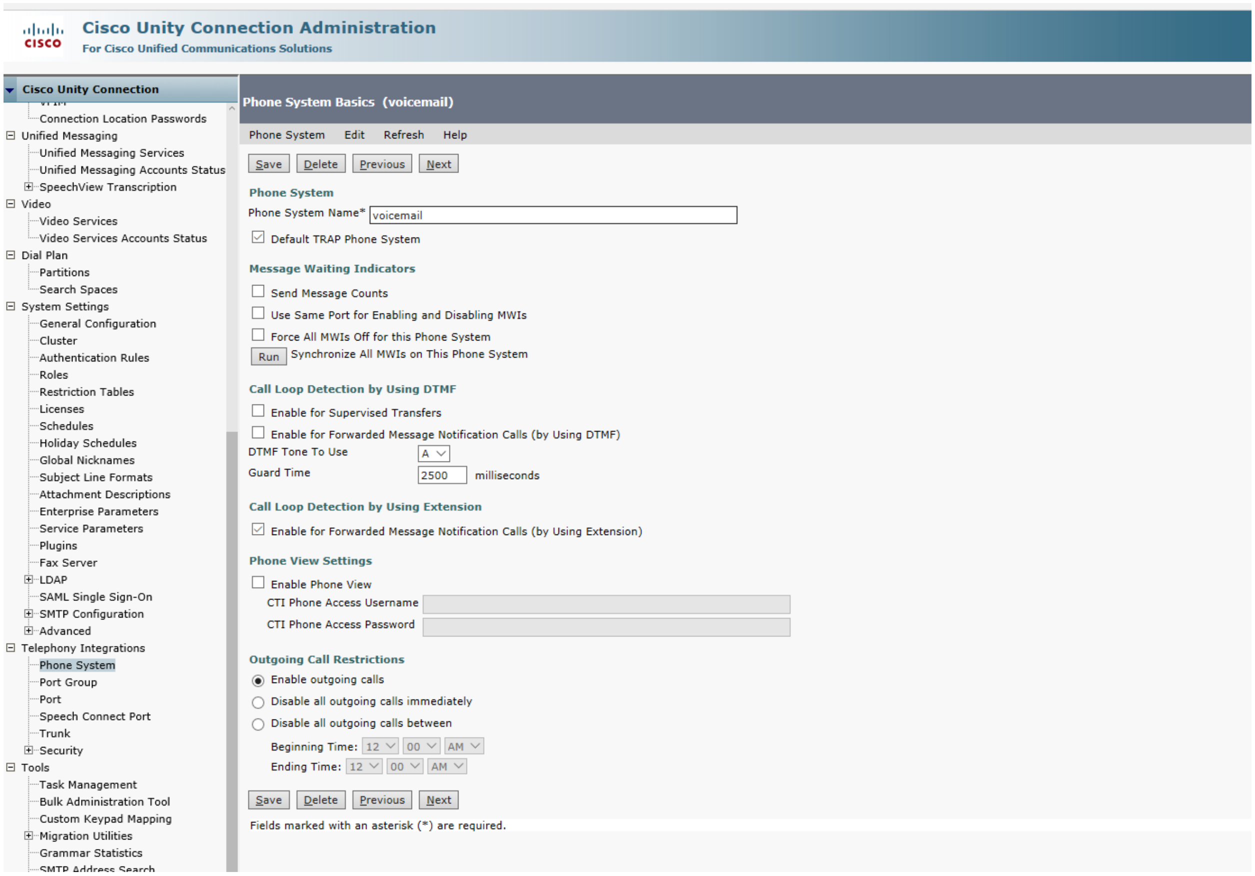




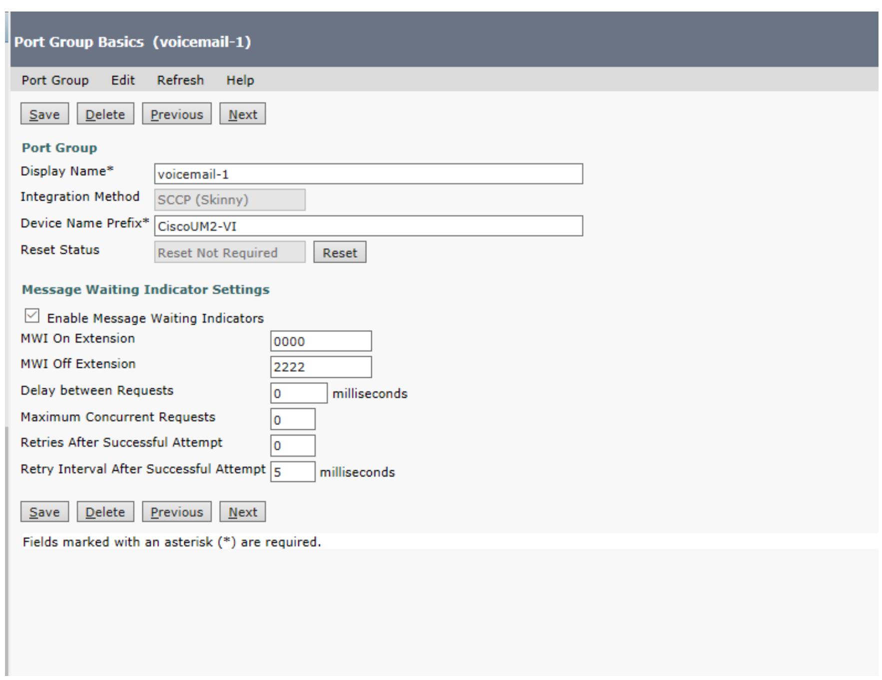
Then, enter Cisco Unity Connection Administration and select Phone System.



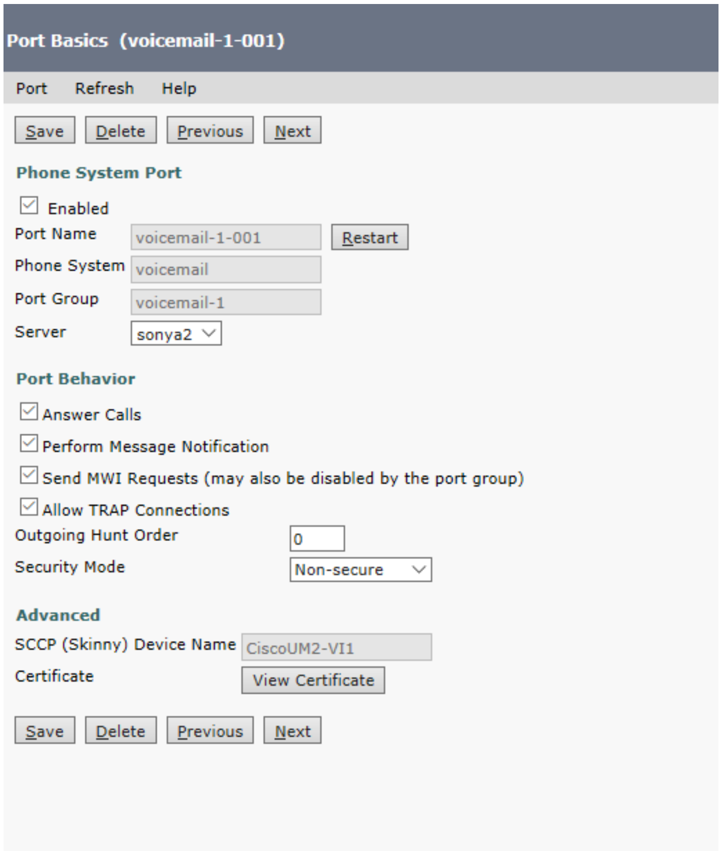
Create a new Phone System, and be sure to select “Enable outgoing calls”.

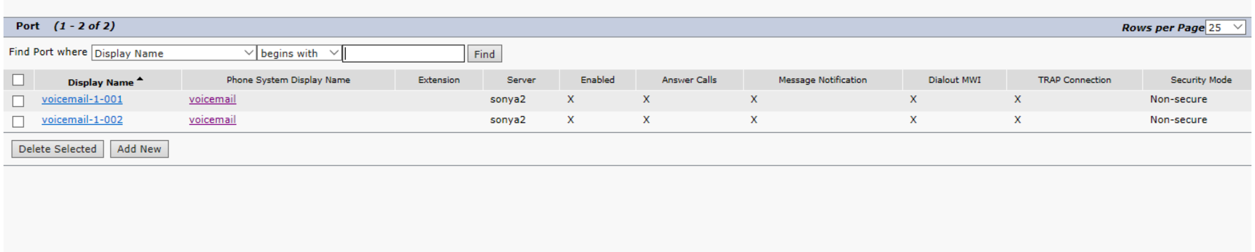


Then, add a port group using the Message Waiting Indicator numbers that you set earlier.

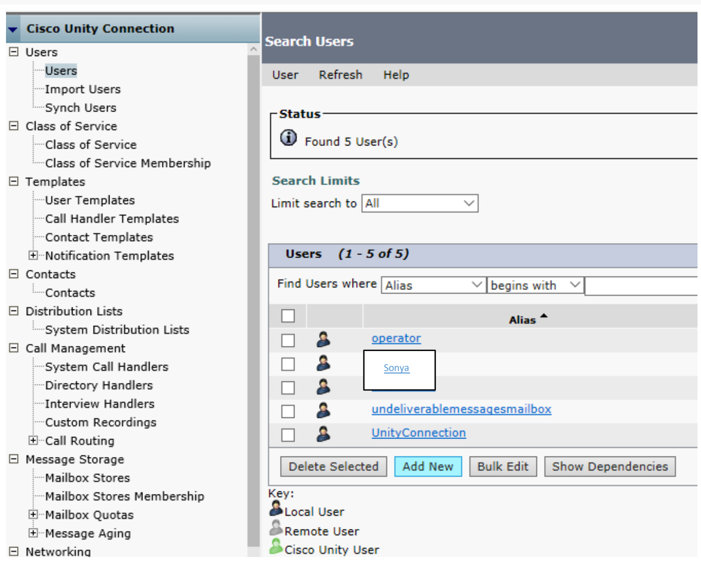


Now that the port group has been configured, you can now add ports. There should be 2 ports and the configuration should look like it does below.



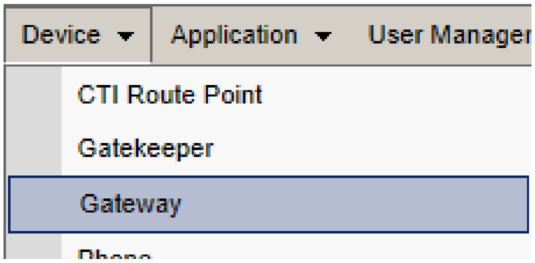


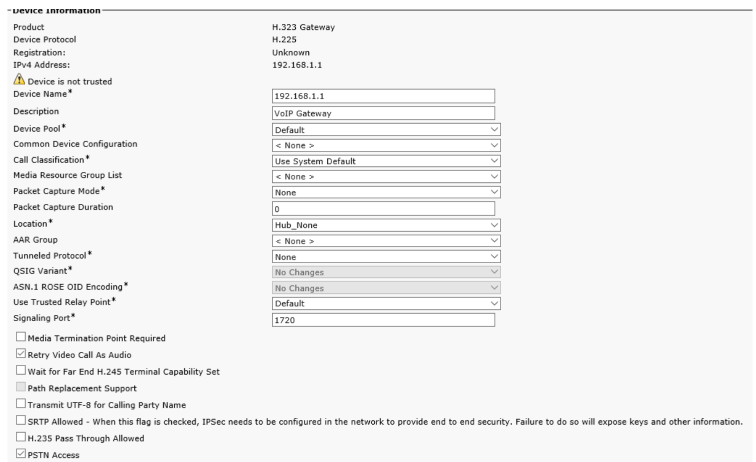
Finally, add a 2 new users and assign each user one directory number. Once this is completed, voicemail will be configured.



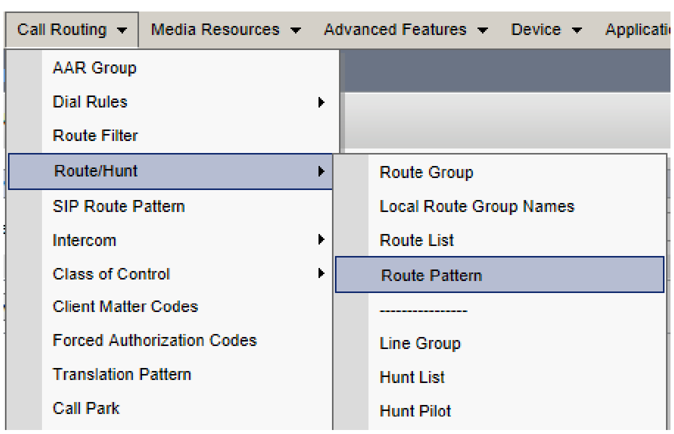
Call out Configuration:

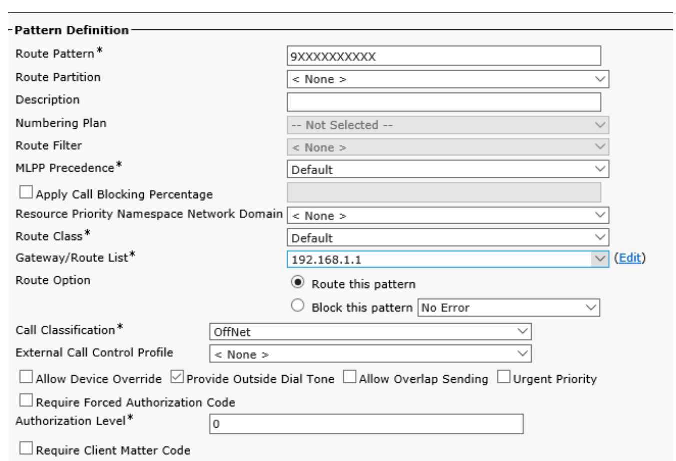
First, navigate to Device > Gateway and add a H.323 Gateway using the IP address of your router.





Then, go to Call Routing > Route/Hunt > Route Pattern, and add a Route pattern beginning with 9. Set the gateway to be the address of your router.

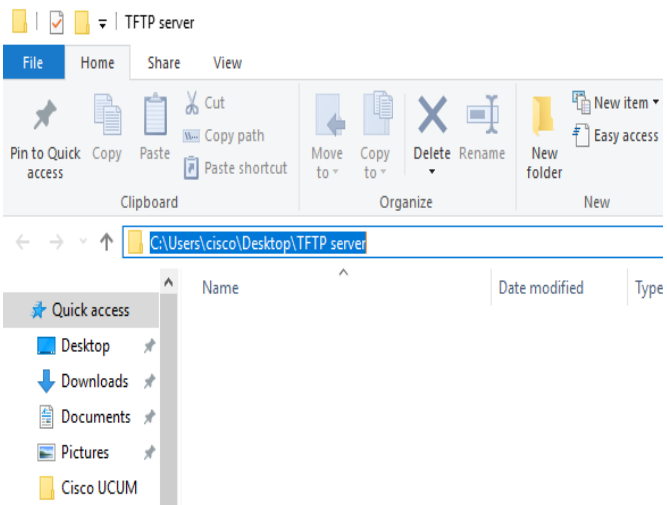


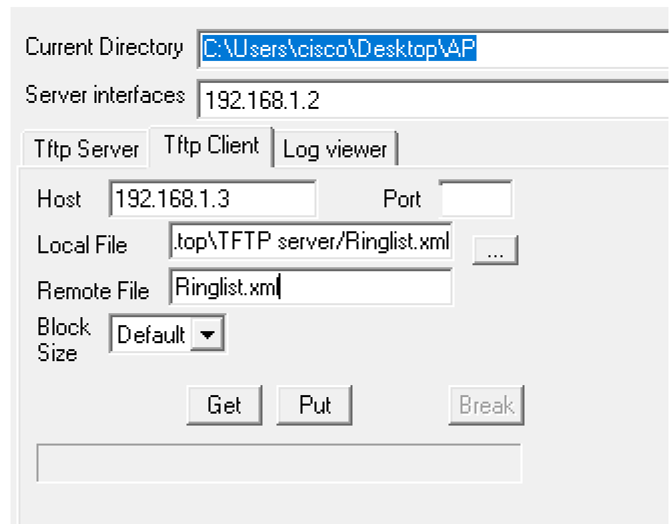


Then, enter the correct commands on your router to configure the voice port with VoIP calling features and connect the router to an external phone cable as the connection to the external network.

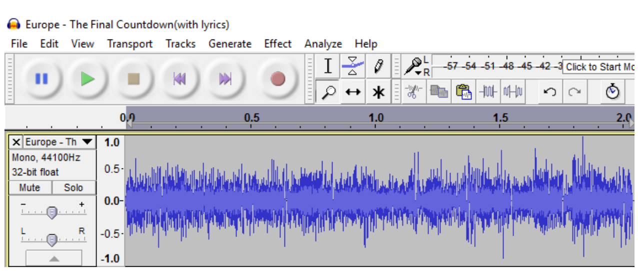
Ringtone Configuration:

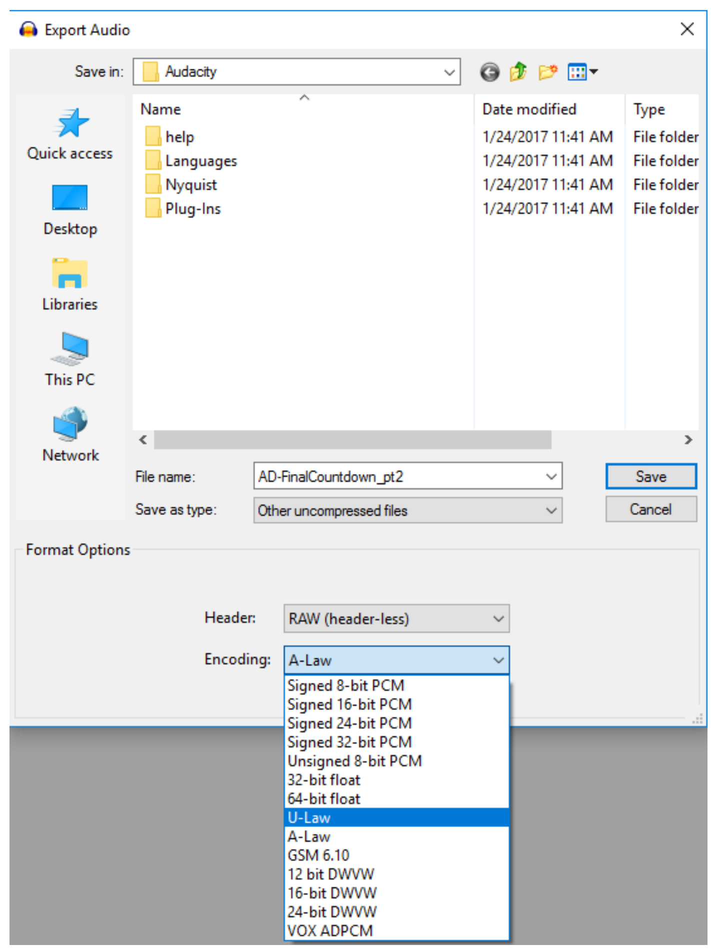
First, create a folder for all TFTP server files. Then, using tftpd32, create a TFTP server on your PC using the IP address of the router. Add the remote xml file Ringlist to the server.



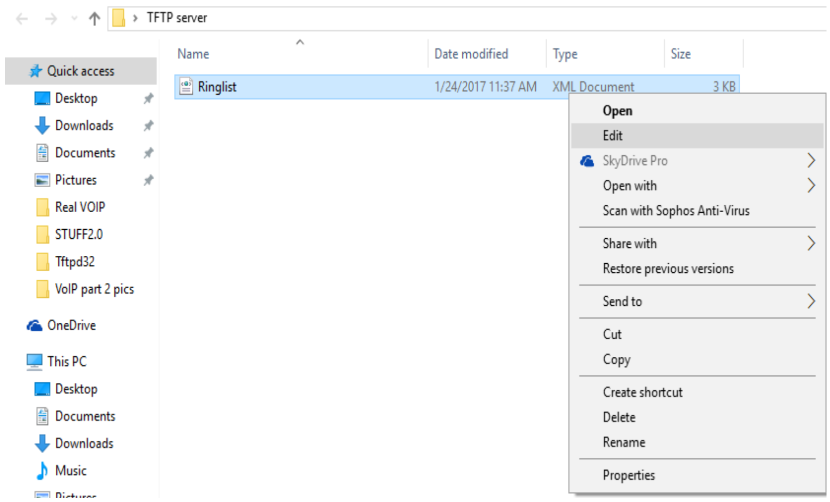


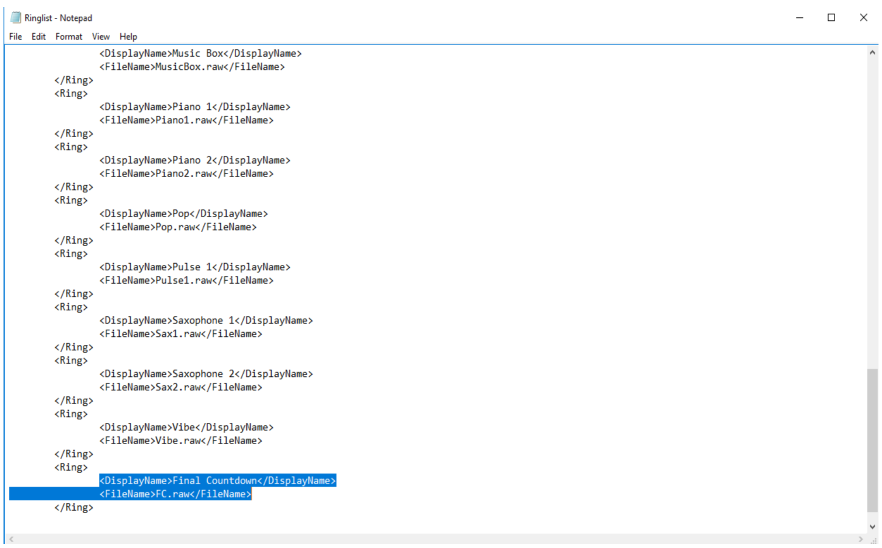
Next, in Audacity, compress your MP3 file to be less than 2 seconds with a Project Rate of 8000Hz. Save your file as an Other uncompressed file with a RAW header and U-Law encoding.



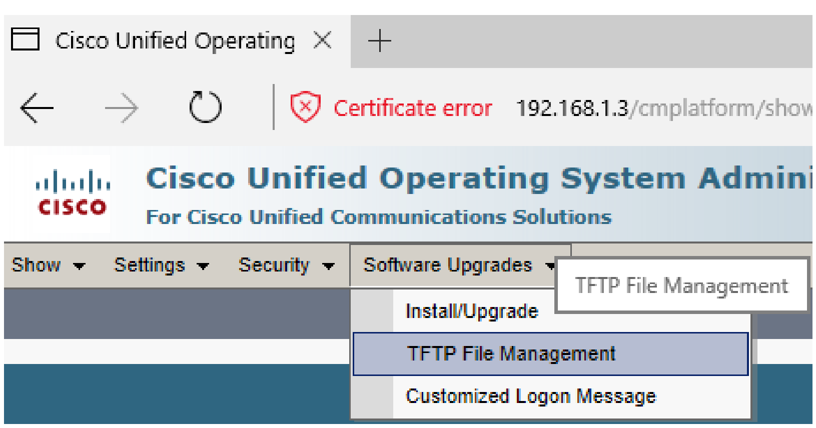


Open the TFTP server folder and edit the Ringlist xml file. Add your newly compressed raw file to the end of the list, following the format of the rest of the files.

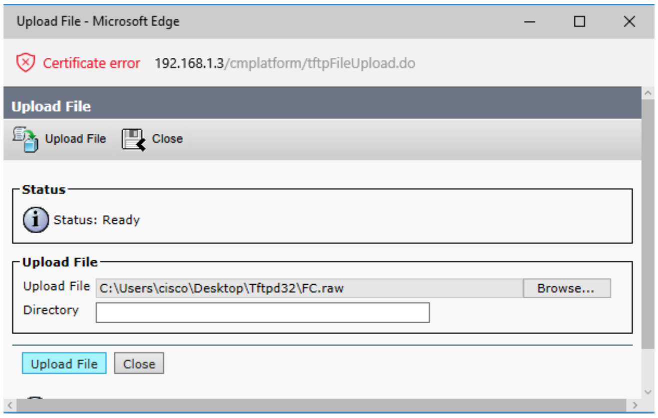




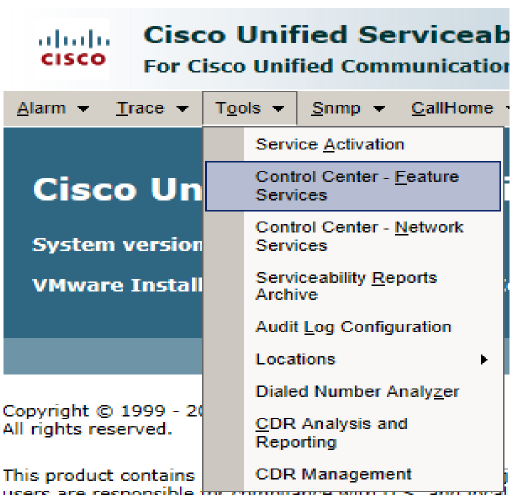
Then, enter CUCM and navigate to Software Upgrades > TFTP File Management



Upload your raw file to the TFTP server



Then, navigate to the Cisco Unified Serviceability, Tools > Control Center – Feature Services



Restart the TFTP server and once it has restarted, the new ringtone will be added to the phones.

Problems

In setting up my initial topology, I had a lot of trouble with seemingly faulty phones. Whenever I plugged the phones into the PoE switch, the phone was unable to find the correct file configuration from the TFTP server, and returned the message, “TFTP file not found.” To fix this issue, I configured portfast on the connected ports of the switch so that the switch configurations did not interfere with the phone’s file loading process. Also, I learned not to plug in the phone into the switch until I had DHCP running on the router and a profile set up for the phone in CUCM.

When I was configuring voicemail, in the beginning, the only setup I completed was through the Voicemail Pilot Wizard. I created all the necessary parts for voicemail – the voicemail pilot, the message waiting indicator, the hunt pilot and the hunt list, and I expected it to work. However, after searching online and completing more research, I realized that we also needed to install Cisco Unity Connection Manager. In installing Cisco Unity Connection, at first, I used the incorrect CUC ova file in my VM, but after watching a tutorial online, I was able to install CUC in VMware as a new server.

For call out, I was able to successfully call from my VoIP phone to my cell phone in the morning. However, when I tried to replicate the same results in the afternoon during tutorial, I was unable to I tried retracing my steps and adding every possible variation of options in the CUCM on the phone. It wasn’t until I added the command `allow-connection h323 to h323` that the call was successful. The command enabled for the h323 router to communicate with the h323 phone.

Conclusion

Learning to expand my Cisco VoIP knowledge is extremely important in the real world. It was an interesting opportunity to create my own ringtone, hold music, and voicemail system, and Call in/out. I am now aware of the ever-present Cisco VoIP in every classroom and in movies. I gained a better understanding of protocols involved in voice configurations. I did not realize how large of an impact our knowledge is on people worldwide.