VOIP Part 2

CCNP Lab 8

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Cisco CCNP - Hoffman and Mason - Periods 6 and 7

VOIP Part 2 Lab 8

Purpose

The purpose of this lab was to implement VOIP (Voice Over IP) with PSTN (Public Switched Telephone Network). We also added new features to our preexisting VOIP phones, like voicemail, music on hold and custom ringtones.

Background Information

Voice Over IP, also known as VOIP, is a method of providing voice and multimedia communication over internet protocol (IP) services such as the internet. Through this, one can place free phone calls, by bypassing phone companies and their fees entirely. IP Phones, are special phones that have a RJ-45 ethernet port, rather than RJ11 port, which allows them to be directly connected to a host. The protocol used by the phones is called SCCP (Skinny Client Control Protocol). This is a proprietary protocol, originally made by Selsius Systems, but acquired later by Cisco. It allows for real time audio stream on clients, which use this protocol, giving them the capability to send audio back and forth, in real time. In this case the “clients” are the IP Phones. This protocol works in conjunction with the CUCM to achieve this real time stream of audio. The Cisco Unified Communications Manager (CUCM) is an Internet Protocol (IP) based system that allows for communication using voice, video and other data and media outlets. This software can be installed on a server to make it accessible from anywhere, allowing for secure and cost effective communication of any type, available in any network. To add new features such as music on hold (MOH), custom ringtones, and external dial out and in, the Cisco Unity Communications (CUC) platform must be used in conjunction with the preexisting CUCM. To add on, you can use other protocols to interface with VOIP. For example, you can use SMTP (Simple Mail Transfer Protocol) to send your voicemail files to your email.

Lab Summary

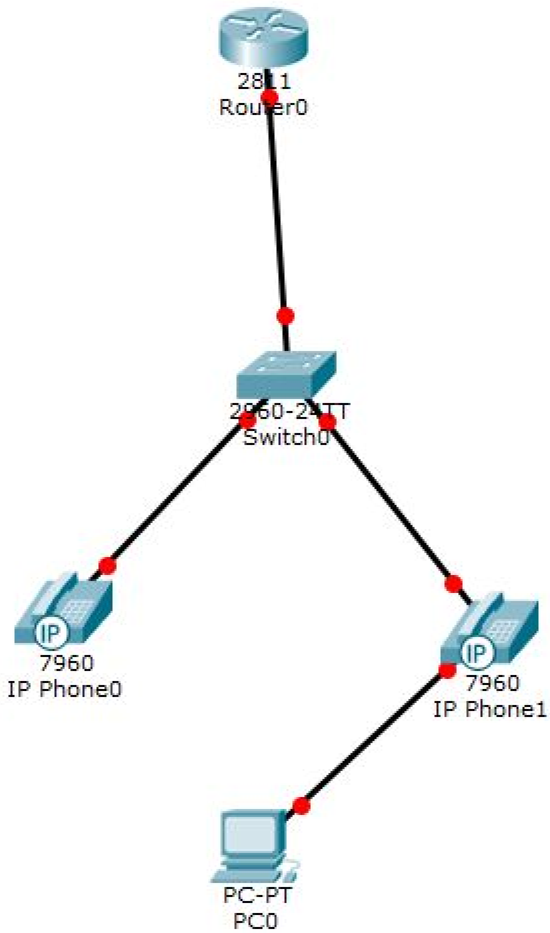
For this lab, we setup two ip phones and connected them to a 2811 router with FX0 ports that support voice-ports, VOIP specific routers must be used for this. We also setup two VMs with different IP addresses so that the phones could be connected to the CUCM and CUC. A switch with power over ethernet was also used, but only to power and route to the servers and routers.

Lab Commands

voice-port [port]	Interface that is to be configured as a voice port - the one that is connected to the phone.
signal [groundstart] timing hookflash-out 50 timing guard-out 1000	Type of signal wanting to be used
connection plar opx [phone number]	Sets the phone number to be reached
caller-id enable	Shows caller-ID - the phone number
ccm-manager config server [server address] ccm-manager config	Points router to CUCM server
mgcp call-agent [server address] service-type mgcp version [version number] mgcp profile default	Points the router to CUCM server
dial-peer voice 10 voip	Dial in setup start command

destination-pattern [phone number]	Phone number to be called (dial in)
session target ipv4:[server address]	Directs the call to the dial in phone number to go to the server address (dial in)
dmtf-relay h245-signal h245-alphanumeric	Allows for signal to be released (dial in)
dial-peer voice 1 pots	Dial out starting setup command
destination-pattern 9T direct-inward-dial	Sets the routing pattern, in this case 9 is the extension and T is timeout (dial out)
port [outgoing port interface]	Sets the outgoing interface (dial out)
forward-digits [number]	Sets the number of digits dialed (dial out)

Network Diagram



Configurations

Router 1 Configurations:

```
hostname R1
no aaa new-model
resource policy
memory-size iomem 10
no network-clock-participate slot 1
ip subnet-zero
ip cef
no ip dhcp use vrf connected
ip dhcp pool Voice
```

```
   network 172.16.0.0 255.255.255.0
   default-router 172.16.0.1
   option 150 ip 172.16.0.2
ip dhcp pool Data
   network 172.16.0.16 255.255.255.240
   default-router 172.16.0.17
   option 150 ip 172.16.0.2
voice call send-alert
voice rtp send-recv
voice service voip
 h323
interface FastEthernet0/1.1
 encapsulation dot1Q 10
 ip address 172.16.0.1 255.255.255.240
 no snmp trap link-status
interface FastEthernet0/1.2
 encapsulation dot1Q 20
 ip address 172.16.0.17 255.255.255.240
 no snmp trap link-status
voice-port 0/3/1
 signal groundStart
 timing hookflash-out 50
 timing guard-out 1000
 connection plar opx 1111
 caller-id enable
ccm-manager config server 172.16.0.20
ccm-manager config
mgcp call-agent 172.16.0.20 service-type mgcp version 0.1
mgcp profile default
dial-peer voice 10 voip
 destination-pattern 1111
 session target ipv4:172.16.0.20
 dtmf-relay h245-signal h245-alphanumeric
dial-peer voice 1 pots
 destination-pattern 9T
 direct-inward-dial
 port 0/3/1
 forward-digits 11
telephony-service
```

```
 max-ephones 2
 max-dn 2
 ip source-address 172.16.0.20 port 2001
 create cnf-files version-stamp 7960 Jan 22 2015 23:53:36
 max-conferences 8 gain -6
 transfer-system full-consult
scheduler allocate 20000 1000
ntp master
```

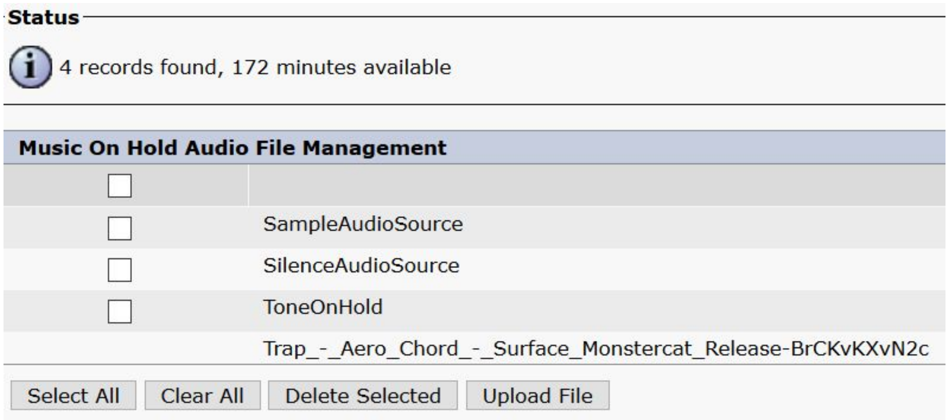
Switch 1 Configuration:

```
hostname S1
vlan 10
name Voice
vlan 20
name Data
int range fa0/3 - 4
switchport mode access
switchport voice vlan 10
switchport mode access
switchport access vlan 20
int fa0/20
switchport trunk encapsulation dot1q
switchport mode trunk
```

Configuration Screenshots

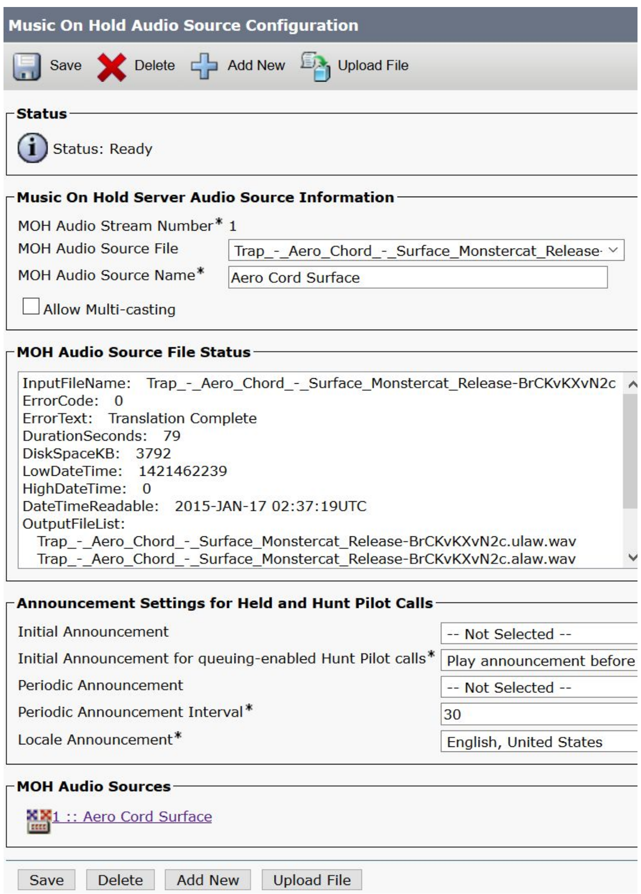
Music on Hold:

Step 1:



Upload the music file using the “Upload File” button and select the music file.

<u>Step 2:</u>



For “Music On Hold Audio Source Configuration”:

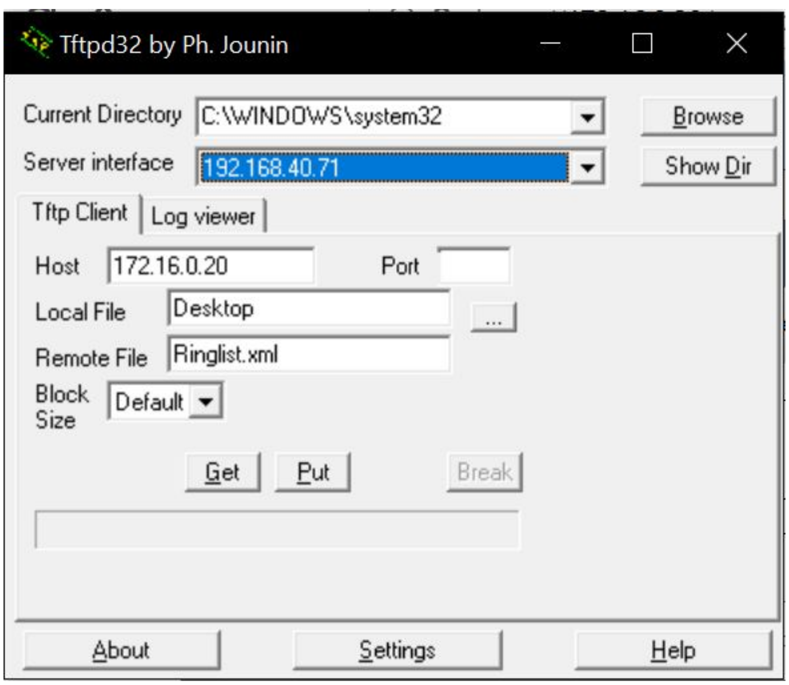
Set a name for the music file in the “MOH Audio Source Name” field.

In the “Initial Announcement” dropdown, leave it at “--Not Selected--”.

In the “Initial Announcement for queuing-enabled Hunt Pilot calls” dropdown, choose “Play announcement before”.
In the “Periodic Announcement” dropdown, leave it at “--Not Selected--”.
In the “Periodic Announcement Interval” dropdown, choose the number of seconds wanted to pause between each repeat of the announcement (default is 30 seconds).
In the “Locale Announcement” dropdown, choose “English, United States”.
Any fields with asterisks (*) should be filled in or selected.

Custom Ringtone Configuration:

Step 1:

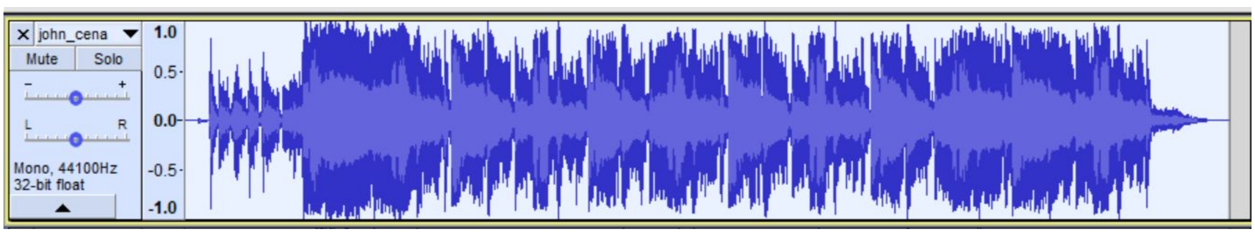


Use Tftpd32 (or any tftp software you are comfortable with) to retrieve the Ringlist.xml file from the VM server to edit.

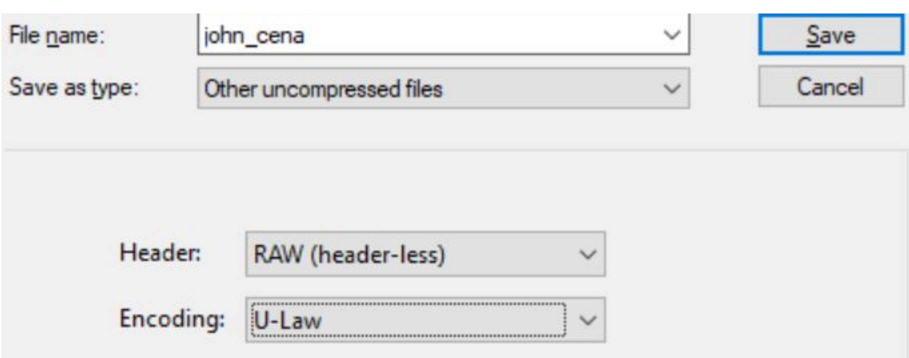
```
        <DisplayName>Saxophone 1</DisplayName>
        <FileName>Sax1.raw</FileName>
</Ring>
<Ring>
        <DisplayName>Saxophone 2</DisplayName>
        <FileName>Sax2.raw</FileName>
</Ring>
<Ring>
        <DisplayName>Cena</DisplayName>
        <FileName>John.raw</FileName>
</Ring>
```

Set the filename between the “<FileName>” and “</FileName>” tags to the exact file name saved on the computer, if it is not exactly the same, it will not be read correctly. The display name between the “<DisplayName>” and “</DisplayName>” tags can be set to anything you prefer.

<u>Step 2:</u>



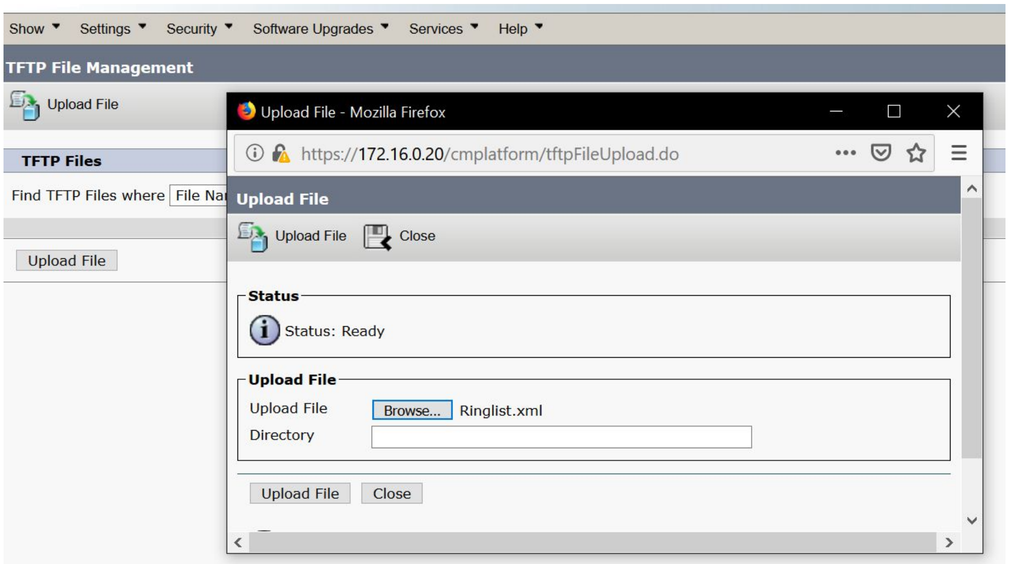
Using Audacity, import the ringtone file being used. Right click in Audacity and select “Mono”. It should split into 2 parts, delete one of the parts. Set the frequency to 8000 Hz at the bottom left, and shorten the clip to less than 10 seconds.

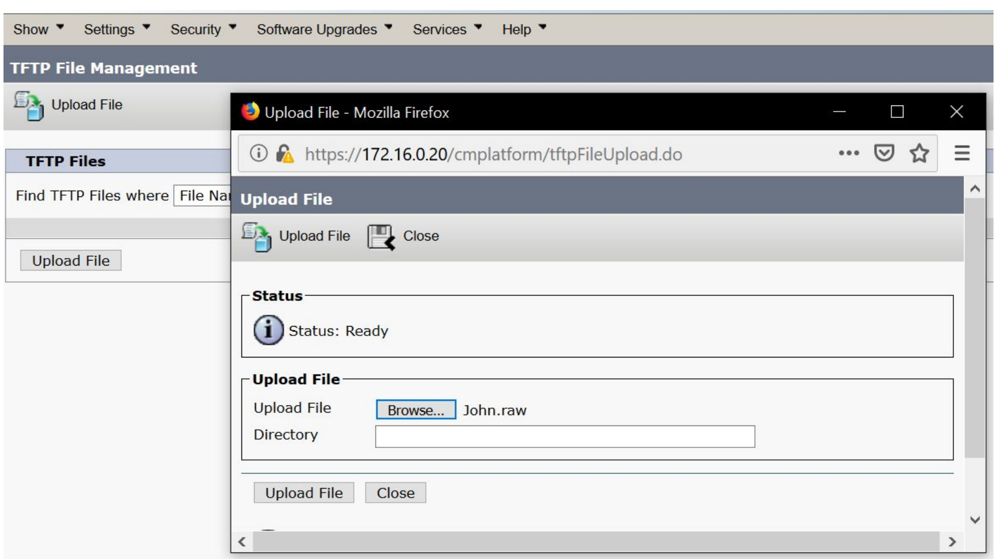


Once editing the file is done, click “File” and select “Export Media”. In the “Save as type” dropdown, choose “Other uncompressed files”. For the

"Header" dropdown, choose "Raw (header-less)" and for the "Encoding" dropdown choose "U-Law". If these settings are not copied exactly, CUCM will not read the file in the server.

Step 3:

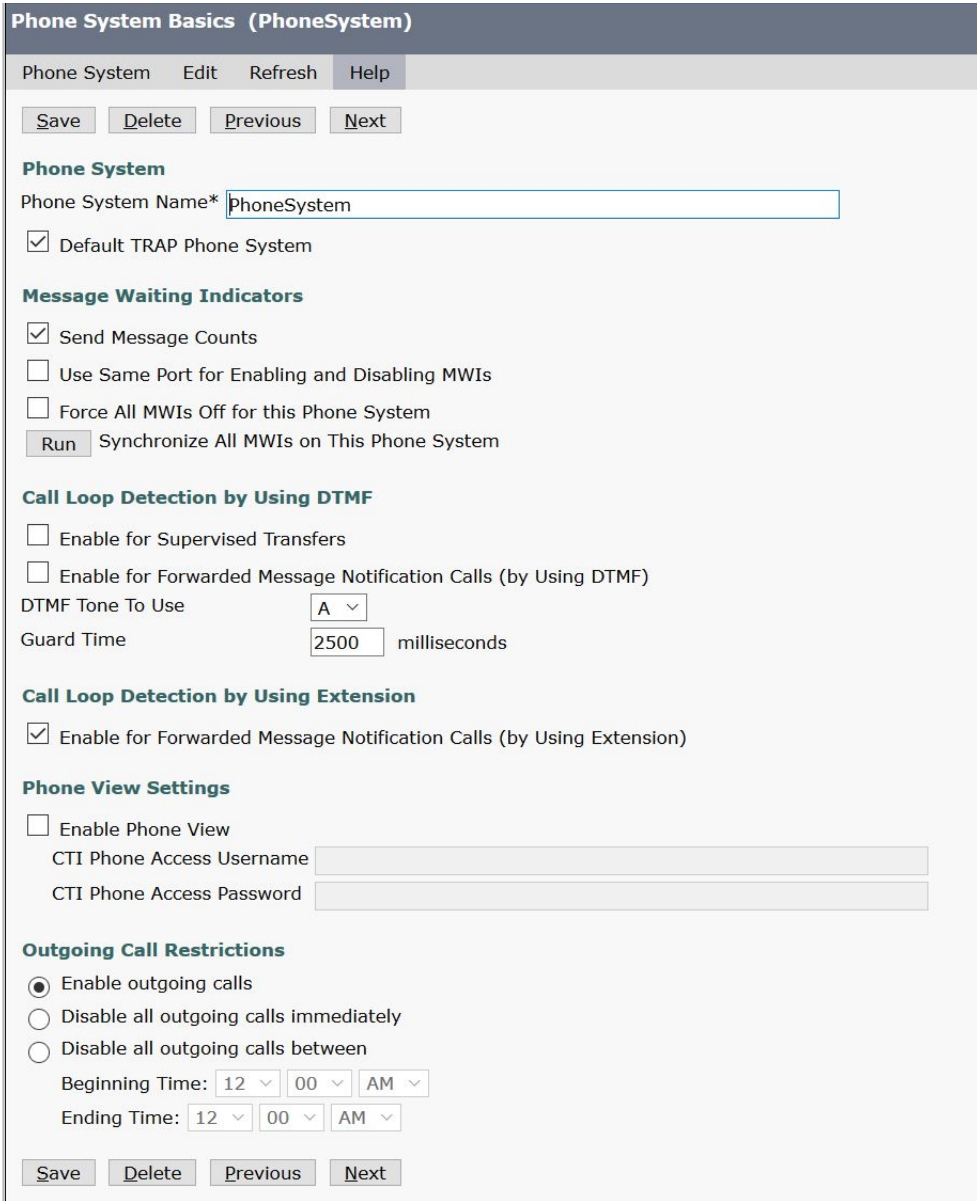




In the navigation tab of CUCM serviceability, choose "TFTP file management" and upload both Ringlist.xml and the ringtone (RAW) file.

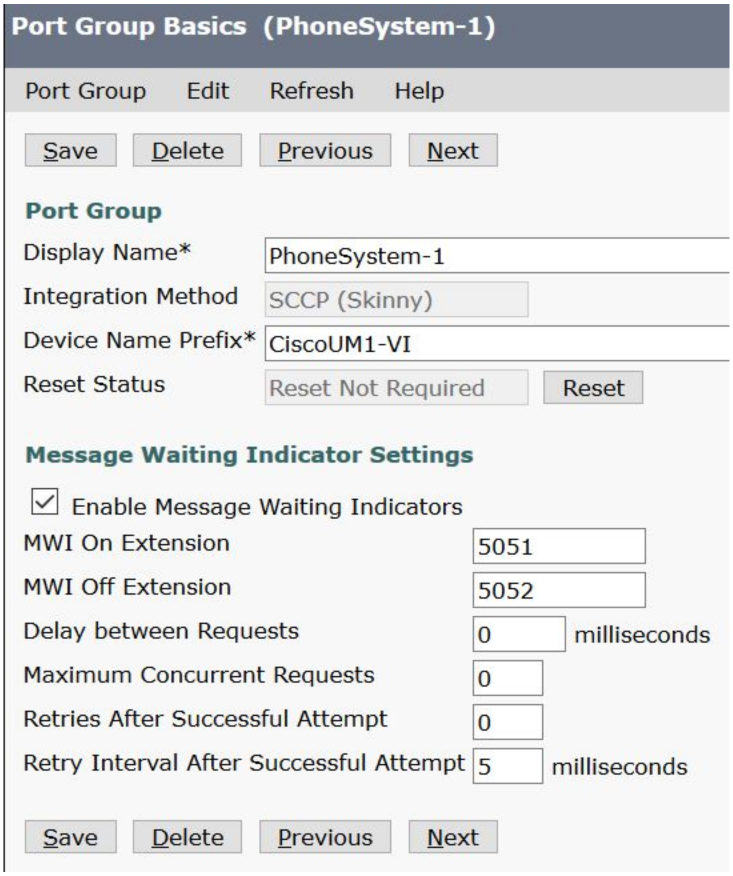
Voicemail Configuration:

Step 1:



Using the CUC, start by creating a new phone system. The wizard should look like the one shown above. Fill out all the necessary information.

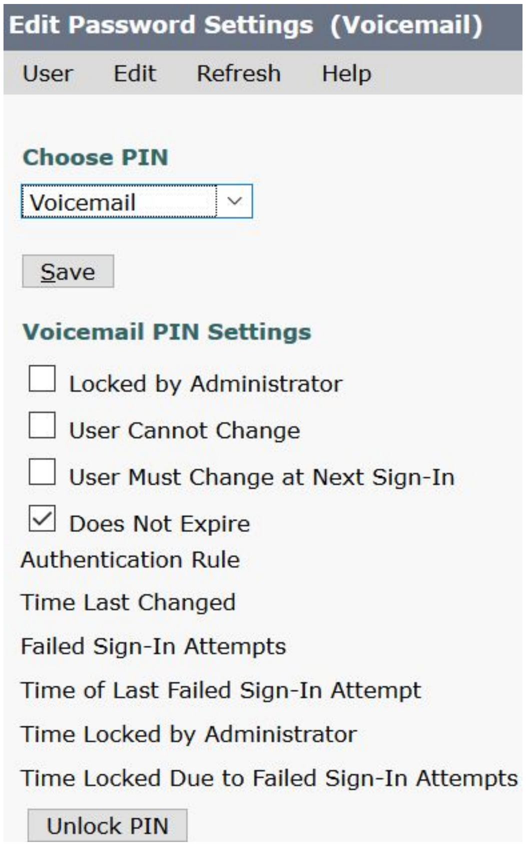
Step 2:



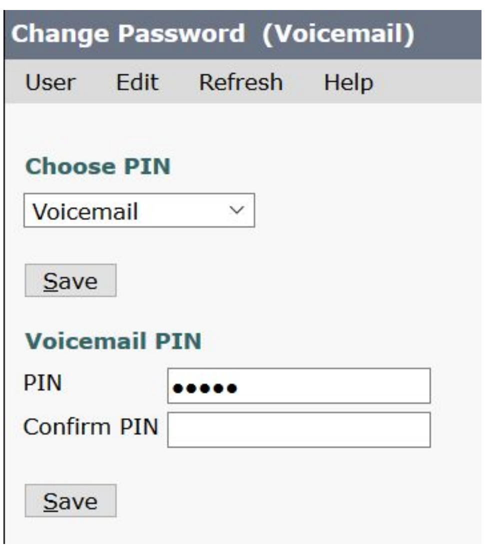
In the Port Group, set the “Display Name” and the “Device Name Prefix” for the phone. MWI stands for “Message Waiting Indication” which tells you if you have voicemail when you call the number. You can choose to turn this feature on or off.

Set up the ports for your two phones.

<u>Step 3:</u>

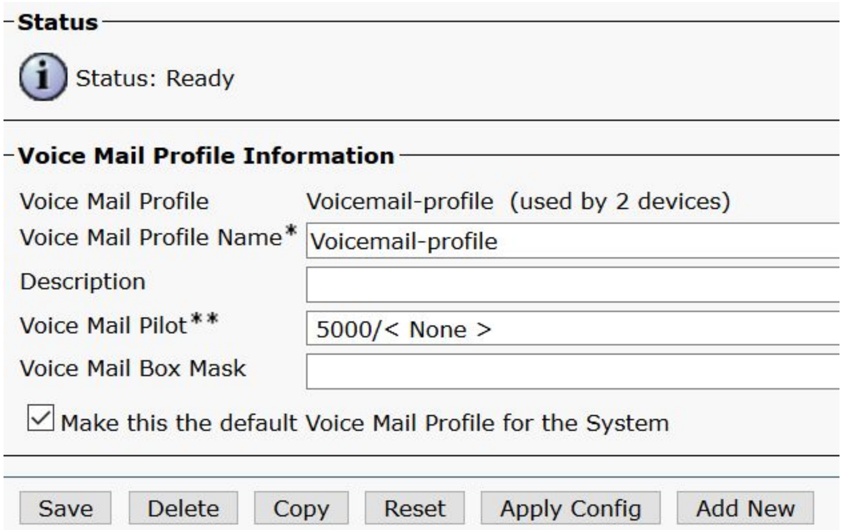


Set up a user profile for both phones.

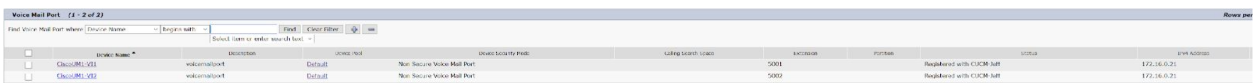


Set a PIN to access voicemail on both phones.

<u>Step 4:</u>

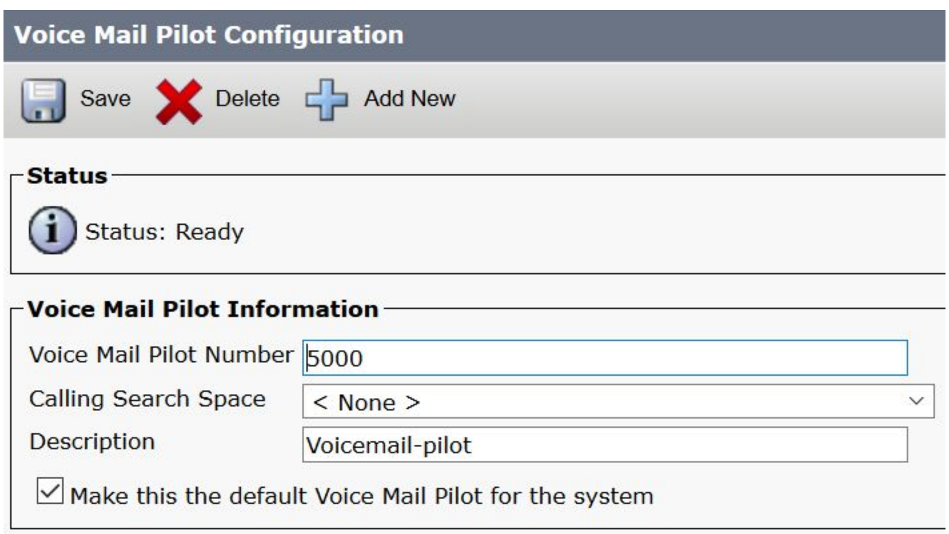


Move to the CUCM for the next steps. Start by setting up the voicemail profile as shown. Make sure you see this wizard.



You should see voicemail ports after making the voicemail profile.

<u>Step 5:</u>



Voicemail pilot is the directory number you dial to access your voicemails.

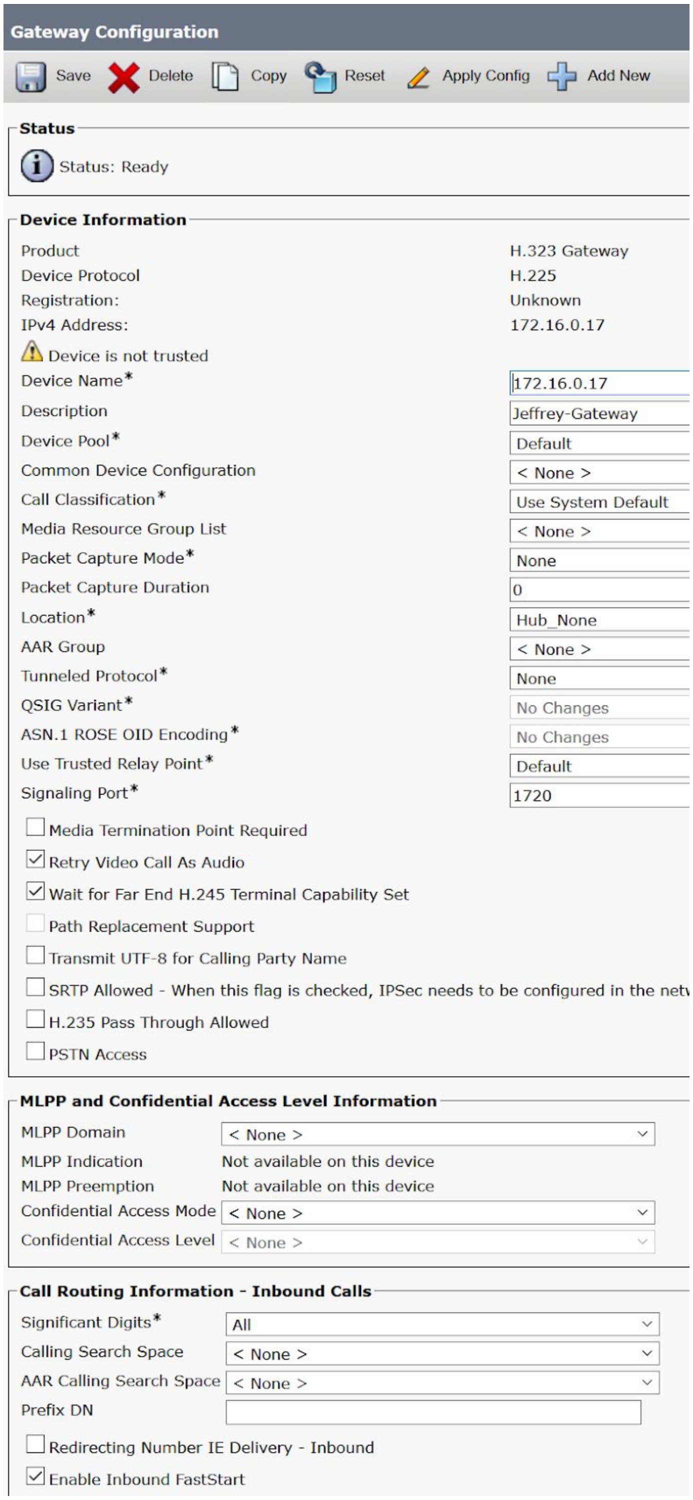
Set up as shown in the screenshot.

When trying to access your voicemail:

1. Press the * button.

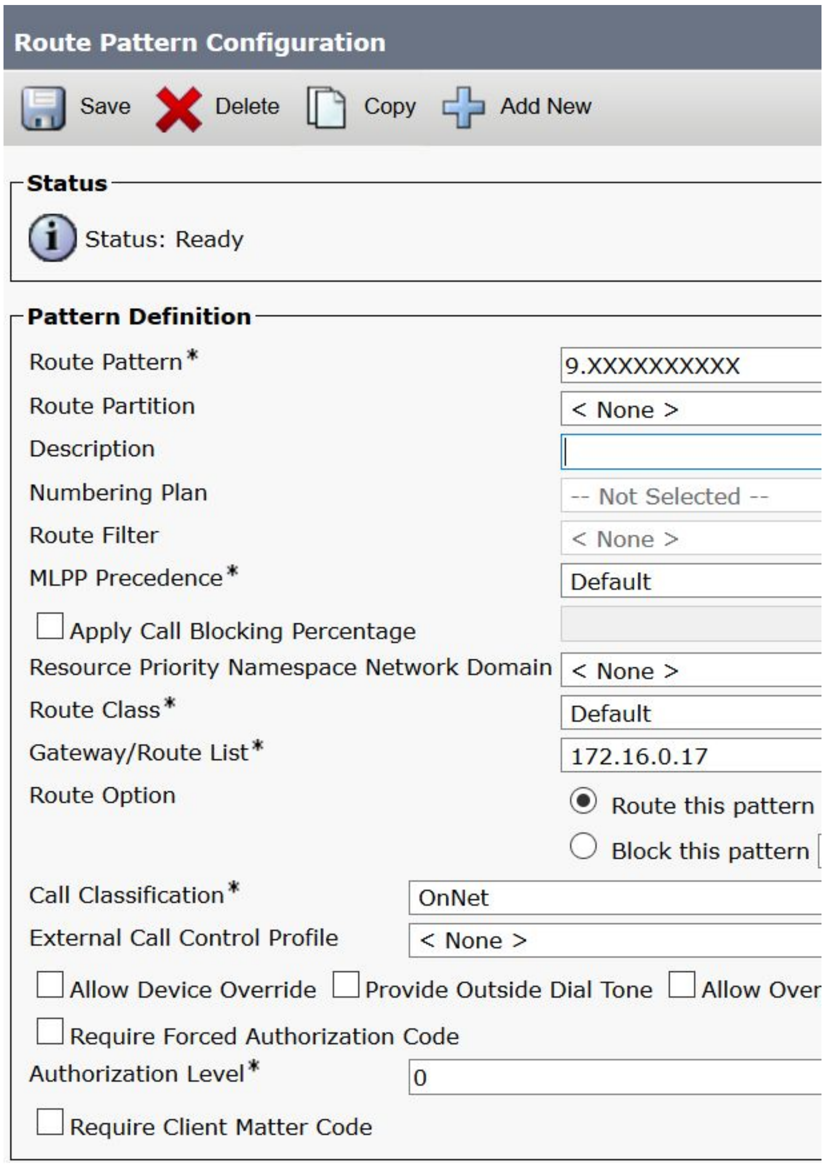
2. Dial your phone’s number.
3. Enter the pin you set earlier to finally access your messages.

<u>Step 6:</u>



Time for dial in and out. First step is to create a h.323 gateway. In the device tab of the CUCM, choose that option. Set the device name to your router's ip address. Make sure to check the "Enable Inbound FastStart" checkbox. Fill out all the other asterisk marked fields as shown, or however you prefer.

<u>Step 7:</u>



In the "Route and Hunt" tab, choose to create a route pattern. This wizard should pop up. In the "Route Pattern" field enter the text shown above. 9 is the extension, the period represents that everything after the 9 is removed, and the 10 X's are the phone's dial number.

Problems

The first main problem was figuring out how to correctly set up the CUC. It took a couple tries, but we found the right hardware settings to do so. We also had to find the right router to use, as not all routers support voice ports, so that took a lot of time, to find a new router and duplicate our setup on that new one. At first, it was also hard to look for setting up a pin number for voicemail, but after doing some research, we finally figured it out. A couple times our CUC VM needed to be restarted due to incorrect configurations.

Conclusion

In conclusion, we set up two IP Phones according to a topology to use VOIP to communicate. We also added new features such as custom ringtones, music on hold, and voicemail. To set up voicemail, we learned how to use the CUC along with the CUCM.