

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

SONYA LAO
CCNP PER. 1/2

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.

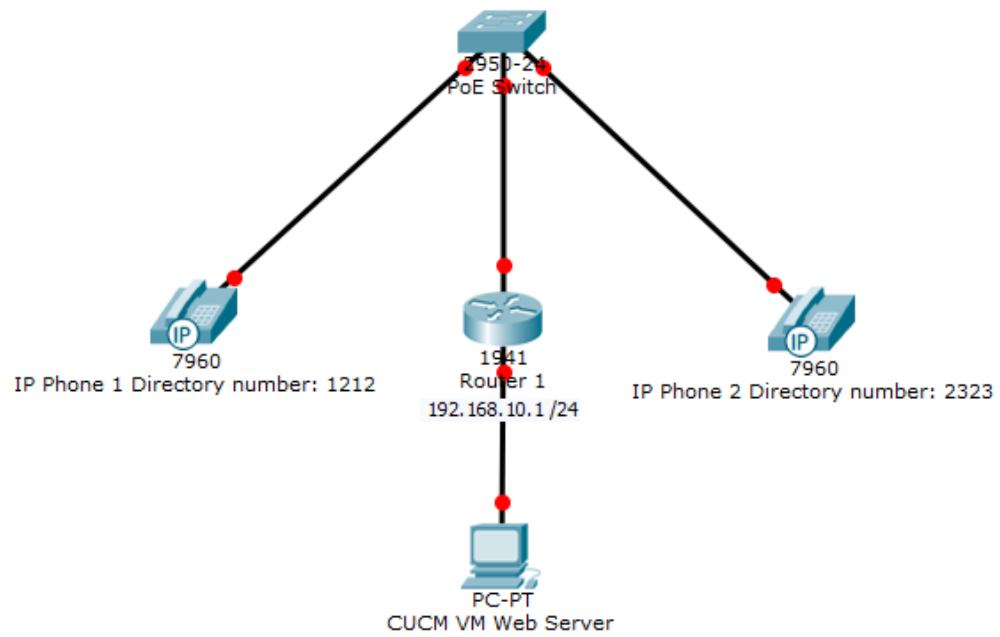
```
R1(config-dial-peer)#port 0/3/0
```

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 FX0
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.

Find and List Phones

Related Links: [Actively Logged In Device Report](#)

Status
2 records found

Phone (1 - 2 of 2)

Find Phone where Device Name begins with Find Clear Filter Select item or enter search text

| | Device Name(Line) | Description | Device Pool | Device Protocol | Status | IPv4 Address | Copy | Super Copy |
|--------------------------|----------------------|-----------------|-------------|-----------------|-----------------------|---------------|------|------------|
| <input type="checkbox"/> | 7960 SEP0015625ABB43 | SEP0015625ABB43 | Default | SCCP | Registered with sonya | 192.168.10.11 | | |
| <input type="checkbox"/> | 7960 SEP001BD512A57A | SEP001BD512A57A | Default | SCCP | Registered with sonya | 192.168.10.12 | | |

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

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

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Association
Modify Button Items

- 7960 Line [1] - 2323 (no partition)
- 7960 Line [2] - Add a new DN
- Add a new SD
- Add a new SD
- Add a new SD
- Add a new SD
- Unassigned Associated Items -----
- Add a new SD
- Add a new SURL
- Add a new BLF SD
- Add a new BLF Directed Call Park
- Privacy
- None

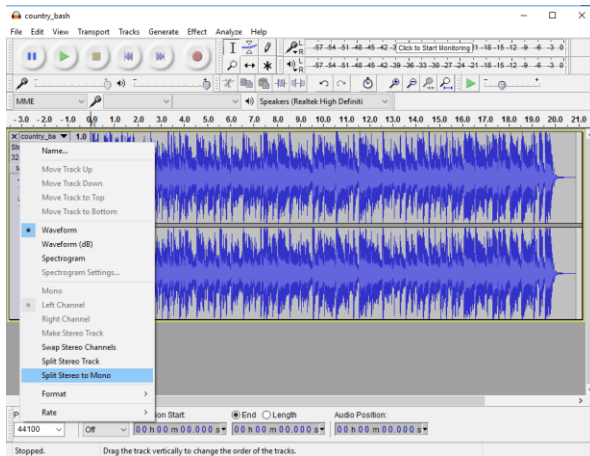
Phone Type
Product Type: Cisco 7960
Device Protocol: SCCP

Real-time Device Status
Registration: Registered with Cisco Unified Communications Manager sonya
IPv4 Address: 192.168.10.11
Active Load ID: None
Download Status: None

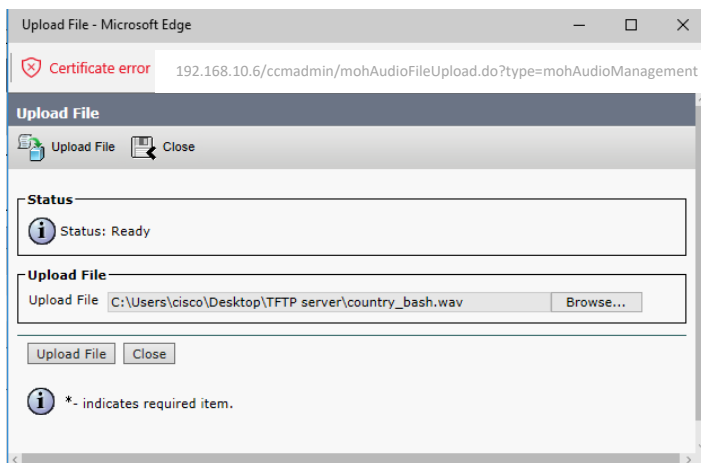
Device Information

| | |
|---|--|
| <input checked="" type="checkbox"/> Device is Active | |
| <input checked="" type="checkbox"/> Device is trusted | |
| MAC Address* | 0015625ABB43 |
| Description | SEP0015625ABB43 |
| Device Pool* | Default View Details |
| Common Device Configuration | < None > View Details |
| Phone Button Template* | Standard 7960 SCCP |
| Softkey Template | < None > |
| Common Phone Profile* | Standard Common Phone Profile View Details |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Media Resource Group List | < None > |
| User Hold MOH Audio Source | 1-Hamilton |
| Network Hold MOH Audio Source | 1-Hamilton |
| Location* | Hub_None |

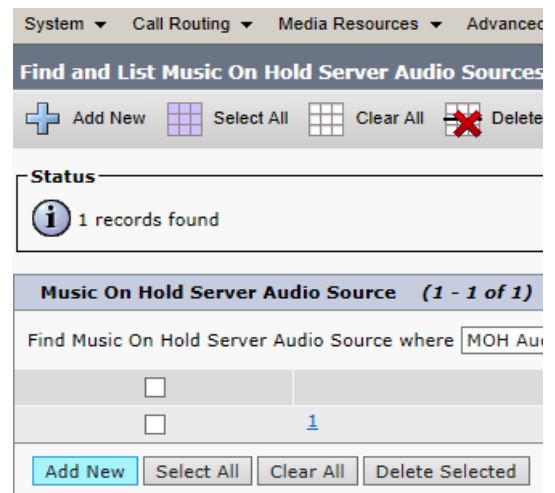
First, find a song and use Audacity to split the audio from stereo and mono. Then convert your file to a .wav



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.

The screenshot shows the 'Message Waiting Configuration' page in Cisco Unity Connection. On the left, a navigation menu is open to 'Voice Mail' > 'Message Waiting'. The main area shows the configuration for a specific message waiting number. The 'Status' is 'Ready'. Under 'Message Waiting Information', the 'Message Waiting Number' is set to '0000', 'Partition' is '< None >', 'Description' is 'WMI On', 'Message Waiting Indicator' is set to 'On' (radio button selected), and 'Calling Search Space' is '< None >'. At the bottom, there are buttons for 'Save', 'Delete', 'Copy', and 'Add New'.

The screenshot shows the 'Message Waiting Numbers' list in Cisco Unity Connection. The table has columns for 'Find Message Waiting Numbers where', 'Directory Number', 'Description', 'Partition', 'Calling Search Space', and 'Copy'. There are two entries: one with '0000' and another with '2222'. Below the table are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

| Find Message Waiting Numbers where | Directory Number | Description | Partition | Calling Search Space | Copy |
|------------------------------------|------------------|-------------|-----------|----------------------|------|
| <input type="checkbox"/> | 0000 | | | | |
| <input type="checkbox"/> | 2222 | | | | |

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.

The screenshot shows the 'Line Group Configuration' page in Cisco Unity Connection. On the left, a navigation menu is open to 'Call Routing' > 'Route/Hunt' > 'Line Group'. The main area is divided into two sections: 'Line Group Information' and 'Line Group Member Information'. In 'Line Group Information', 'Line Group Name' is 'CiscoUM2', 'RNA Revision Timeout' is '10', and 'Distribution Algorithm' is 'Top Down'. In 'Line Group Member Information', there is a section 'Find Directory Numbers to Add to Line Group' with a 'Find' button. Below this, 'Current Line Group Members' shows a list of 'Selected DN/Route Partition' with values '4000' and '4001'. There is also a 'Removed DN/Route Partition' section.

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.

Add New

Hunt List

Find Hunt List where

Add New

Hunt List

(1 - 1 of 1)

Rows per Page50

Find Hunt List where (Name) begins with

FindClear Filter

| | Name ^ | Description | Enabled | Status |
|--------------------------|-------------------------|-------------|---------|-----------------------|
| <input type="checkbox"/> | | | | |
| <input type="checkbox"/> | listOne | true | | Registered with sonya |

Add New

Select All

Clear All

Delete Selected

Reset Selected

Apply Config to Selected

Hunt Pilots

Find Hunt Pilots where

Add New

- Pattern Definition

| | |
|---------------------|---|
| Hunt Pilot* | <input type="text" value="9000"/> |
| Route Partition | <input type="text" value=" < None >"/> |
| Description | <input type="text" value="pilot1"/> |
| Numbering Plan | <input type="text" value=" < None >"/> |
| Route Filter | <input type="text" value=" < None >"/> |
| MLPP Precedence* | <input type="text" value="Default"/> |
| Hunt List* | <input type="text" value="listOne"/> (Edit) |
| Call Pickup Group | <input type="text" value=" < None >"/> |
| Alerting Name | <input type="text"/> |
| ASCII Alerting Name | <input type="text"/> |

Route Option

☒ Route this pattern
☐ Block this pattern

☐ Provide Outside Dial Tone
☐ Urgent Priority

Hunt Pilots

(1 - 1 of 1)

Rows per Page 50

Find Hunt Pilots where

Pattern

begins with

Find

Clear Filter

| <input type="checkbox"/> | Pattern * | Description | Partition | Route Filter | Hunt List | Copy |
|--------------------------|-----------|-------------|-----------|--------------|-------------------------|------|
| <input type="checkbox"/> | 9000 | pilot1 | | | listOne | |

Add New

Select All




Clear All

Delete Selected

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.

Voice Mail Profile (1 - 2 of 2) Rows per Page 50


Find Voice Mail Profile where Voice Mail Profile Name begins with Find Clear Filter

| | Name ^ | Description | Pilot | Calling Search Space | Copy |
|---|-------------|---------------------------------|-------|----------------------|---|
|  | NoVoiceMail | No Voice Mail | | |  |
| <input type="checkbox"/> | voicemail | Default voice messaging profile | 9000 | |  |

Add New Select All Clear All Delete Selected

Voice Mail Pilot (1 - 2 of 2) Rows per Page 50

Find Voice Mail Pilot where Voice Mail Pilot Number begins with Find Clear Filter

| | Pilot Number ^ | Description | Calling Search Space |
|---|----------------|---------------|----------------------|
| <input type="checkbox"/> | | No Voice Mail | |
|  | 9000 | Default | |

Add New Select All Clear All Delete Selected

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.

Directory Number Settings

Voice Mail Profile
Calling Search Space
BLF Presence Group*
User Hold MOH Audio Source
Network Hold MOH Audio Source
Auto Answer*
☐ Reject Anonymous Calls

< None >
Default
NoVoiceMail
Voicemail

< None >
Standard Presence group
< None >
< None >
Auto Answer Off

AAR ☐ or
☒ Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

| | Voice |
|--|--|
| Calling Search Space Activation Policy | |
| Forward All | <input type="checkbox"/> or |
| Secondary Calling Search Space for Forward All | |
| Forward Busy Internal | <input type="checkbox"/> or |
| Forward Busy External | <input type="checkbox"/> or |
| Forward No Answer Internal | <input checked="" type="checkbox"/> or |
| Forward No Answer External | <input checked="" type="checkbox"/> or |
| Forward No Coverage Internal | <input type="checkbox"/> or |
| Forward No Coverage External | <input type="checkbox"/> or |
| Forward on CTI Failure | <input type="checkbox"/> or |
| Forward Unregistered Internal | <input checked="" type="checkbox"/> or |
| Forward Unregistered External | <input checked="" type="checkbox"/> or |
| No Answer Ring Duration (seconds) | 10 |
| Call Pickup Group | < None > |

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

Find and Hunt Lists Cisco Unity Connection 192.168.10.8/roadmyn/home.do

Cisco Unity Connection Administration For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration Search Documentation About Sign Out

Attention

The system is operating on demo licenses that will expire in 47 day(s). Add this system to an Enterprise License Manager and install sufficient licenses to cover its usage before expiration in order to continue using Connection services.

This Cisco Unity Connection server has never been backed up. We recommend that you configure regularly scheduled backups in the Disaster Recovery System.

Cisco Unity Connection Administration
Version 10.0.1.10000-24

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.

- Cisco Unity Connection
 - Users
 - Import Users
 - Synch Users
 - Class of Service
 - Class of Service
 - Class of Service Membership
 - Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
 - Contacts
 - Contacts
 - Distribution Lists
 - System Distribution Lists
 - Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
 - Message Storage
 - Call Routing
 - Hallbox Stores
 - Hallbox Stores Membership
 - Hallbox Quotas
 - Message Aging
 - Networking
 - Legacy Links
 - Branch Management
 - HTTP(S) Links
 - Locations
 - VPSH
 - Connection Location Passwords
 - Unified Messaging
 - Unified Messaging Services
 - Unified Messaging Accounts Status
 - SpeechView Transcription
 - Video
 - Video Services
 - Video Services Accounts Status
 - Dial Plan
 - Partitions
 - Search Spaces
 - System Settings
 - General Configuration

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

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar contains a navigation tree with categories like 'Cisco Unity Connection', 'Unified Messaging', 'Video', 'Dial Plan', 'System Settings', 'Telephony Integrations', and 'Tools'. The 'Phone System' link under 'Telephony Integrations' is selected. The main content area is titled 'Phone System Basics (voicemail)' and includes tabs for 'Phone System', 'Edit', 'Refresh', and 'Help'. Below the tabs are 'Save', 'Delete', 'Previous', and 'Next' buttons. The configuration fields include: 'Phone System Name' (voicemail), 'Default TRAP Phone System' (checked), 'Message Waiting Indicators' (Send Message Counts, Use Same Port for Enabling and Disabling MWIs, Force All MWIs Off for this Phone System, and a 'Run' button to synchronize all MWIs), 'Call Loop Detection by Using DTMF' (Enable for Supervised Transfers, Enable for Forwarded Message Notification Calls (by Using DTMF), DTMF Tone To Use (A), and Guard Time (2500 milliseconds)), 'Call Loop Detection by Using Extension' (checked), 'Phone View Settings' (Enable Phone View, CTI Phone Access Username, and CTI Phone Access Password), and 'Outgoing Call Restrictions' (radio buttons for 'Enable outgoing calls', 'Disable all outgoing calls immediately', and 'Disable all outgoing calls between' with time selection fields for Beginning and Ending Time). At the bottom, there are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (*) are required.'

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

The screenshot shows the 'Port Group Basics (voicemail-1)' configuration page. It has a similar layout to the previous page with tabs for 'Port Group', 'Edit', 'Refresh', and 'Help', and 'Save', 'Delete', 'Previous', and 'Next' buttons. The configuration fields include: 'Display Name' (voicemail-1), 'Integration Method' (SCCP (Skinny)), 'Device Name Prefix' (CiscoUM2-VI), and a 'Reset Status' section with 'Reset Not Required' and a 'Reset' button. The 'Message Waiting Indicator Settings' section includes: 'Enable Message Waiting Indicators' (checked), 'MWI On Extension' (0000), 'MWI Off Extension' (2222), 'Delay between Requests' (0 milliseconds), 'Maximum Concurrent Requests' (0), 'Retries After Successful Attempt' (0), and 'Retry Interval After Successful Attempt' (5 milliseconds). At the bottom, there are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (*) are required.'

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.

Port Basics (voicemail-1-001)

Port Refresh Help

Save Delete Previous Next

Phone System Port

☒ Enabled

Port Name voicemail-1-001 Restart

Phone System voicemail

Port Group voicemail-1

Server sonya2

Port Behavior

☒ Answer Calls

☒ Perform Message Notification

☒ Send MWI Requests (may also be disabled by the port group)

☒ Allow TRAP Connections

Outgoing Hunt Order 0

Security Mode Non-secure

Advanced

SCCP (Skinny) Device Name CiscoUM2-VI1

Certificate View Certificate

Save Delete Previous Next

Port (1 - 2 of 2)

Rows per Page 25

Find Port where Display Name begins with Find

| | Display Name | Phone System Display Name | Extension | Server | Enabled | Answer Calls | Message Notification | Dialout MWI | TRAP Connection | Security Mode |
|--------------------------|-----------------|---------------------------|-----------|--------|---------|--------------|----------------------|-------------|-----------------|---------------|
| <input type="checkbox"/> | voicemail-1-001 | voicemail | | sonya2 | X | X | X | X | X | Non-secure |
| <input type="checkbox"/> | voicemail-1-002 | voicemail | | sonya2 | X | X | X | X | X | Non-secure |

Delete Selected Add New

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

Cisco Unity Connection

Users

- Users
- Import Users
- Synch Users

Class of Service

- Class of Service
- Class of Service Membership

Templates

- User Templates
- Call Handler Templates
- Contact Templates
- Notification Templates

Contacts

- Contacts

Distribution Lists

- System Distribution Lists

Call Management

- System Call Handlers
- Directory Handlers
- Interview Handlers
- Custom Recordings
- Call Routing

Message Storage

- Mailbox Stores
- Mailbox Stores Membership
- Mailbox Quotas
- Message Aging

Networking

Search Users

User Refresh Help

Status

Found 5 User(s)

Search Limits

Limit search to All

Users (1 - 5 of 5)

Find Users where Alias begins with

| | Alias |
|--------------------------|------------------------------|
| <input type="checkbox"/> | operator |
| <input type="checkbox"/> | Sonya |
| <input type="checkbox"/> | undeliverablemessagesmailbox |
| <input type="checkbox"/> | UnityConnection |

Delete Selected Add New Bulk Edit Show Dependencies

Key:

- Local User
- Remote User
- Cisco Unity User

Call out Configuration:

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

The screenshot shows the CUCM configuration interface. On the left, the 'Device' tab is selected, showing a list of devices: CTI Route Point, Gatekeeper, and Gateway. The 'Gateway' option is highlighted. On the right, the 'Add a new Gateway' wizard is displayed. The 'Gateway Type' is set to 'H.323 Gateway'. The 'Next' button is visible at the bottom of the wizard.

Device Information

| Field | Value |
|-----------------------------|--------------------|
| Product | H.323 Gateway |
| Device Protocol | H.225 |
| Registration | Unknown |
| IPv4 Address | 192.168.1.1 |
| Device Name* | 192.168.1.1 |
| Description | VoIP Gateway |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | < None > |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| QSIG Variant* | No Changes |
| ASN.1 ROSE OID Encoding* | No Changes |
| Use Trusted Relay Point* | Default |
| Signaling Port* | 1720 |

☐ Media Termination Point Required
☒ Retry Video Call As Audio
☐ Wait for Far End H.245 Terminal Capability Set
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
☐ H.235 Pass Through Allowed
☒ PSTN Access

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.

The screenshot shows the CUCM configuration interface. On the left, the 'Call Routing' tab is selected, showing a list of options: AAR Group, Dial Rules, Route Filter, Route/Hunt, SIP Route Pattern, Intercom, Class of Control, Client Matter Codes, Forced Authorization Codes, Translation Pattern, and Call Park. The 'Route/Hunt' option is highlighted, and the 'Route Pattern' option is selected. On the right, the 'Pattern Definition' wizard is displayed. The 'Route Pattern' is set to '9XXXXXXXXX'. The 'Gateway/Route List' is set to '192.168.1.1'. The 'Route Option' is set to 'Route this pattern'. The 'Next' button is visible at the bottom of the wizard.

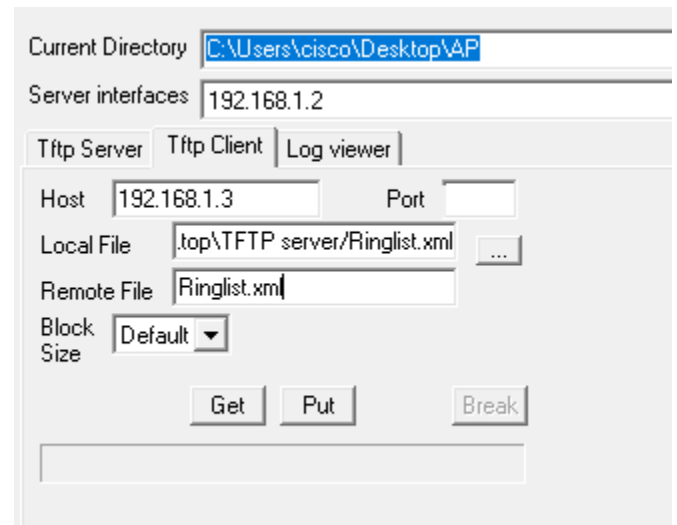
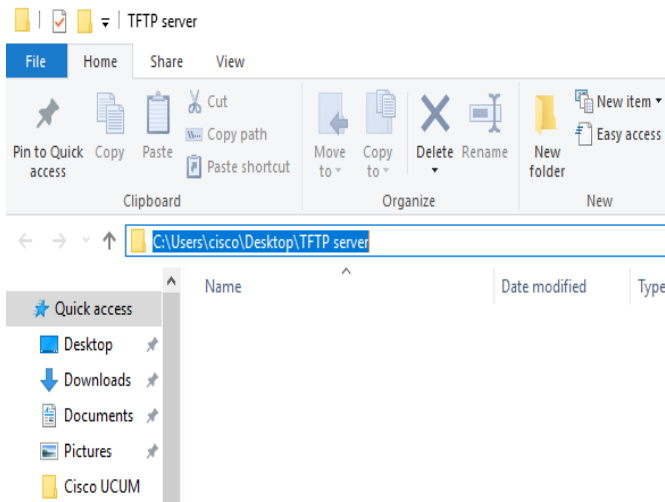
Pattern Definition

| Field | Value |
|--|--|
| Route Pattern* | 9XXXXXXXXX |
| Route Partition | < None > |
| Description | |
| Numbering Plan | -- Not Selected -- |
| Route Filter | < None > |
| MLPP Precedence* | Default |
| <input type="checkbox"/> Apply Call Blocking Percentage | |
| Resource Priority Namespace Network Domain | < None > |
| Route Class* | Default |
| Gateway/Route List* | 192.168.1.1 (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error |
| Call Classification* | OffNet |
| External Call Control Profile | < None > |
| <input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority | |
| <input type="checkbox"/> Require Forced Authorization Code | |
| Authorization Level* | 0 |
| <input type="checkbox"/> Require Client Matter Code | |

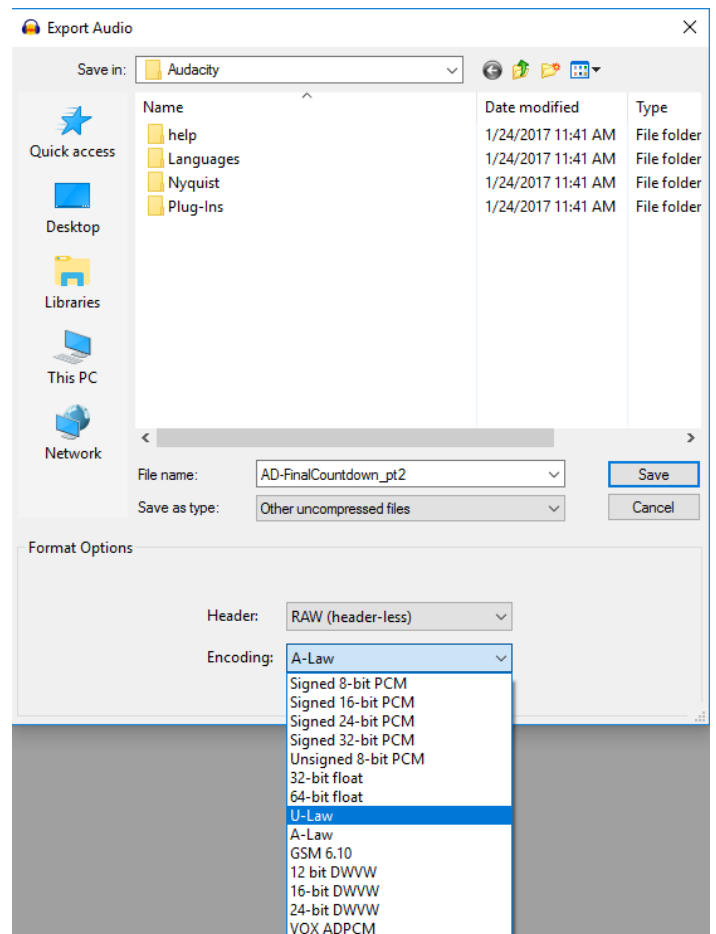
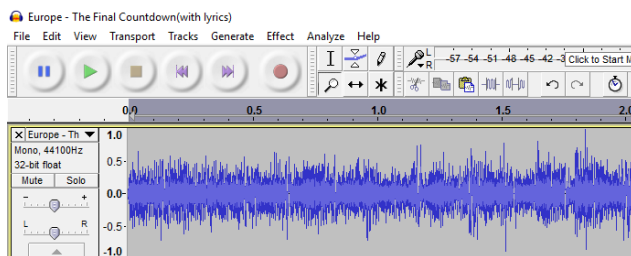
Then, enter the correct commands on your router to configure the voice port with VoIP calling features and connect the router to an external phone cable as the connection to the external network.

Ringtone Configuration:

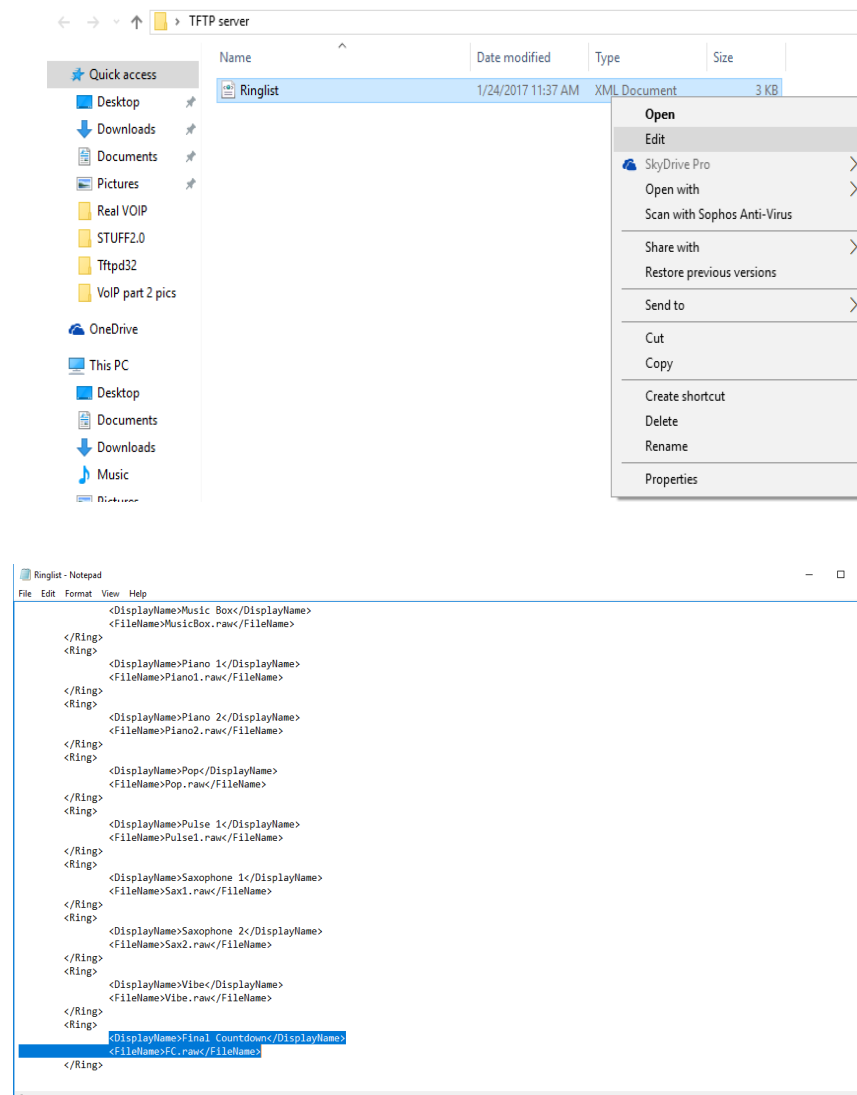
First, create a folder for all TFTP server files. Then, using tftpd32, create a TFTP server on your PC using the IP address of the router. Add the remote xml file Ringlist to the server.



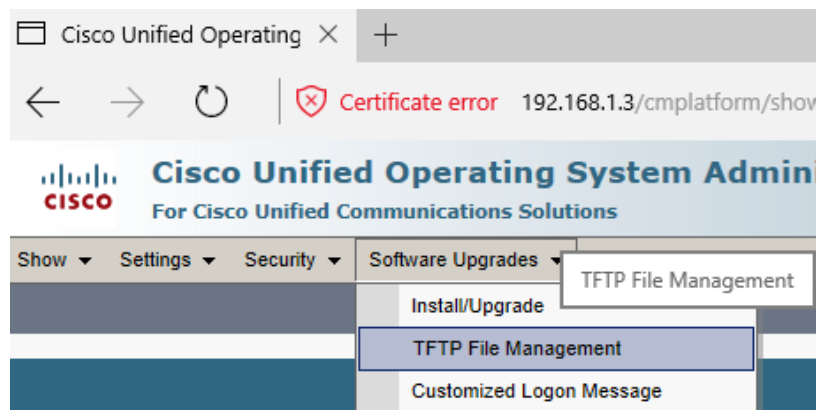
Next, in Audacity, compress your MP3 file to be less than 2 seconds with a Project Rate of 8000Hz. Save your file as an Other uncompressed file with a RAW header and U-Law encoding.



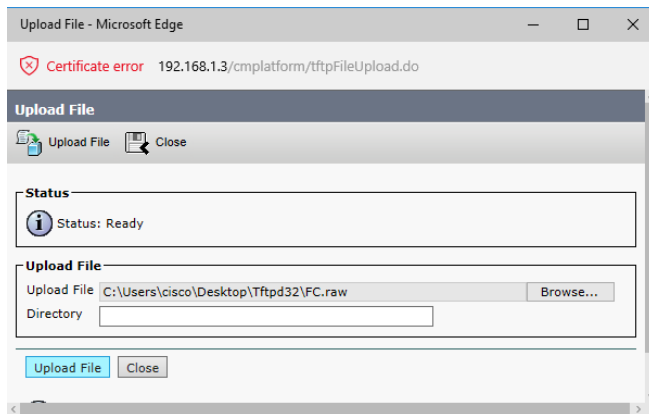
Open the TFTP server folder and edit the Ringlist xml file. Add your newly compressed raw file to the end of the list, following the format of the rest of the files.



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.