

Advanced Cisco Collaboration Guide V3.0

(On-Prem, Edge & Cloud Collaboration Solutions)
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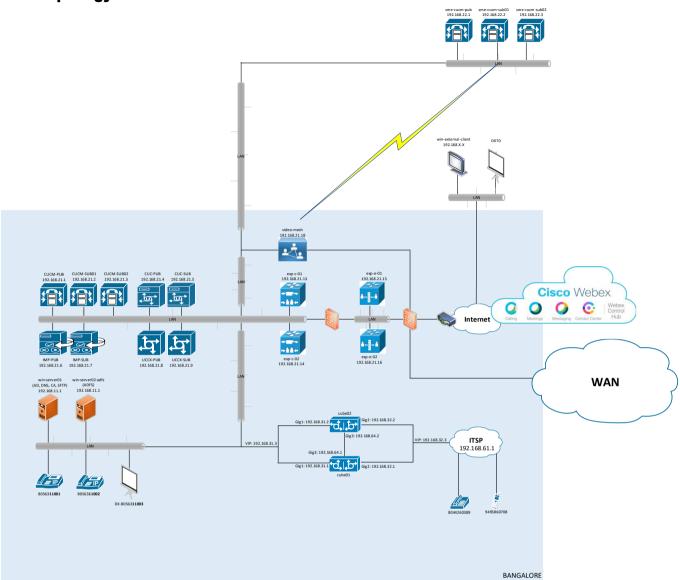
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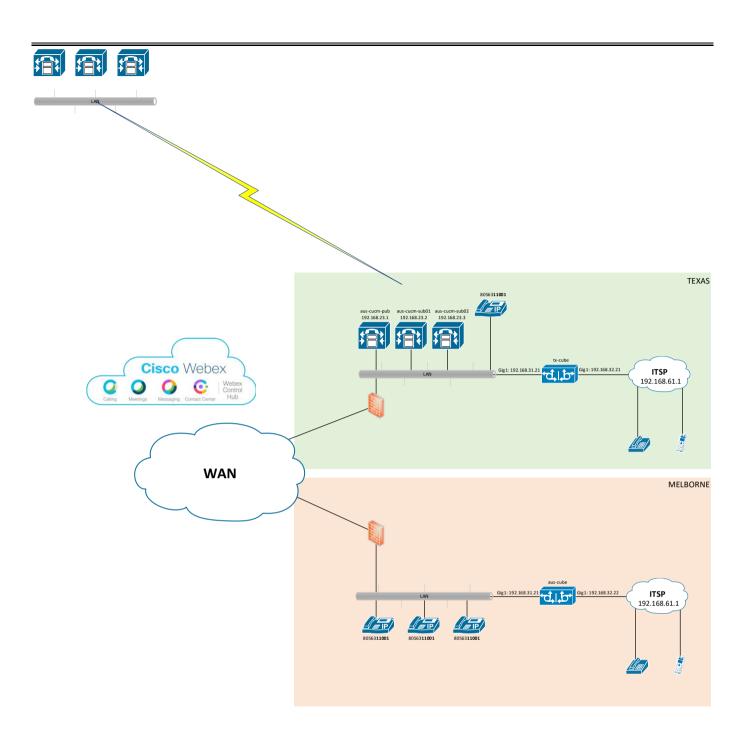
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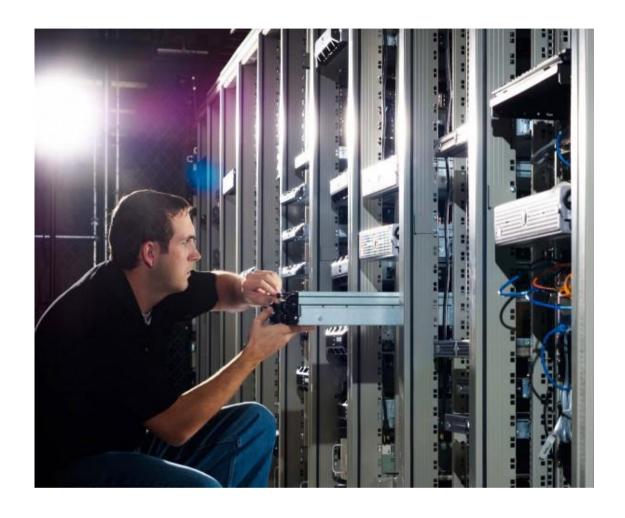
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Lab Topology





Chapter 1 Module 1 - Windows, vmware Windows, AD, DNS, vmware Basics



Introduction to vmware Virtualization



- VMware, Inc. is an American virtualization and cloud computing technology company headquartered in California. VMware was the first commercially successful company to virtualize the x86 architecture
- Vmware technology allows to run multiple Operating Systems on single hardware machine thus enabling better resource management
- vmware developed ESXi (Elastic Sky X Integrated) Hypervisor operating system that create virtualization layer on top of a physical machine.
- The hardware components (CPU, RAM, HDD, Network, etc.) of the physical machine are shared to the multiple operating systems running on ESXi
- Virtualization provides high availability, Cloning, Templating, Storage Optimization, Network Optimization, etc.

Components of vmware Virtualization



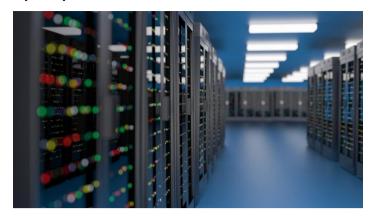
- Virtual Machine (VM): Software representation of a computer as a set of files. It is portable
- Hypervisor: Operating System or Software that enables us to run multiple VMs on it
- Type 1 Hypervisor: Operating system and runs directly on the hardware. Examples vmware ESXi,
 Microsoft Hyper-V, Oracle VM
- Type 2 Hypervisor: Runs as a software layer on an operating system. Example vmware
 Workstation, Microsoft Virtual PC, Oracle VirtualBox
- ESXi: Popular, widely adopted Type 1 Hypervisor developed by vmware
- vCenter: A tool (Operating System) that used to manage and orchestrate multiple ESXi Hosts
- vSphere: Suit of protocols and tools from vmware that makes virtualization and management of virtual environment. Example combination of ESXi and vCenter
- Open Virtualization Format (OVF): Open standard for packaging and distributing a virtual Machines as files
- Open Virtualization Appliance (OVA): OVF Package in single file archive file format. It may contain complete operating system (e.g. Expressway OVA) or just the virtual hardware (e.g. CUCM OVA)
- Datastore: Storage space visible to ESXi OS. This can be DAS (Direct Attached Storage) or NAS (Network Attached Storage)
- Virtual Switch: It is like a network switch used to connect VMs. Virtual Switch makes networking
 possible in ESXi. Multiple Virtual Switch can be crated. Virtual Switch interacts to the external
 network using uplinks.

Windows Active Directory (AD)



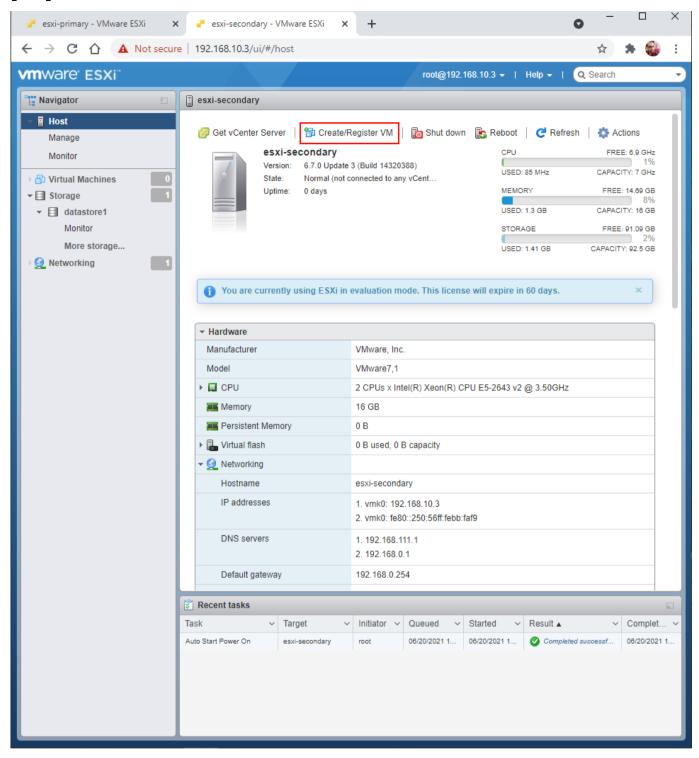
- Active Directory is a directory service developed by Microsoft; it stores the information about the user account
- A server running the Active Directory Domain Service (AD DS) role is called a domain controller
- It Authenticates and Authorizes all users and computers in a Windows domain network
- It can be used by other systems for Authentication via LDAP (Light Weight Directory Access Protocol)
- ADDS is the first server that most organizations install in their Data Center
- We should have a Windows Server to setup AD (I have used Windows Server 2019)
- Organizational Unit (OU): Represented by a folder in AD, provide hierarchy of a domain
- Objects: Users and Computers
- LDAP Filter: Query string to specify certain objects (or to filter certain objects)

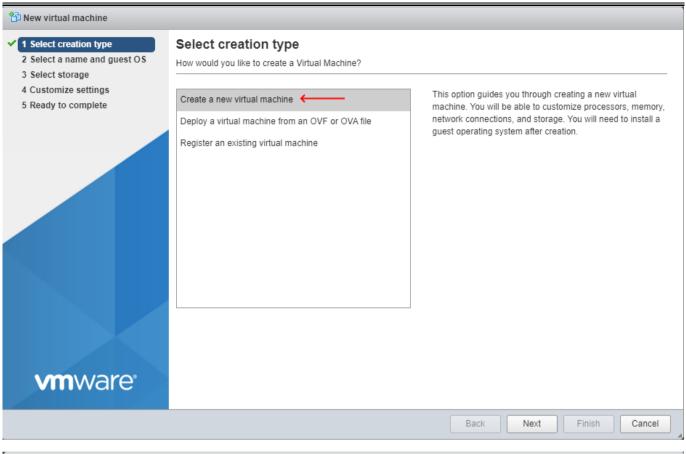
Domain Name System (DNS)

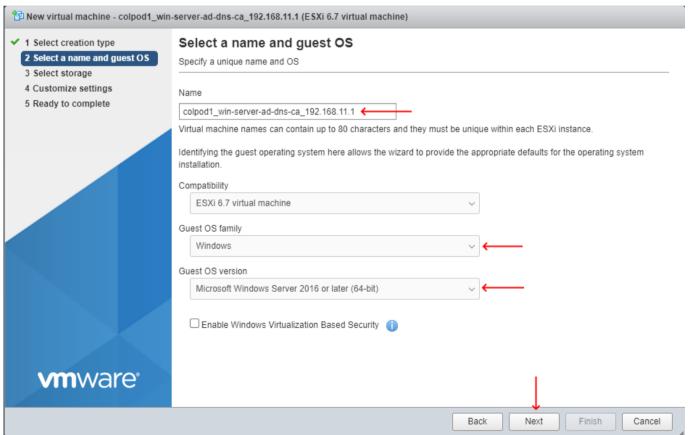


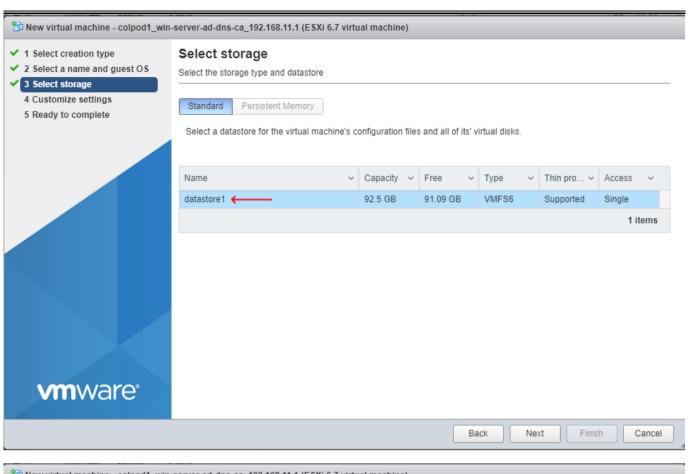
- DNS facilitates IP to Domain name and Domain name to IP mapping
- An internal DNS server hold all the server names and IP addresses for their domains and enable
 DNS lookup (IP to Domain Name and vice versa)
- A Public DNS Server is a central part of the internet, providing Public IP address to corresponding DNS name and reverse

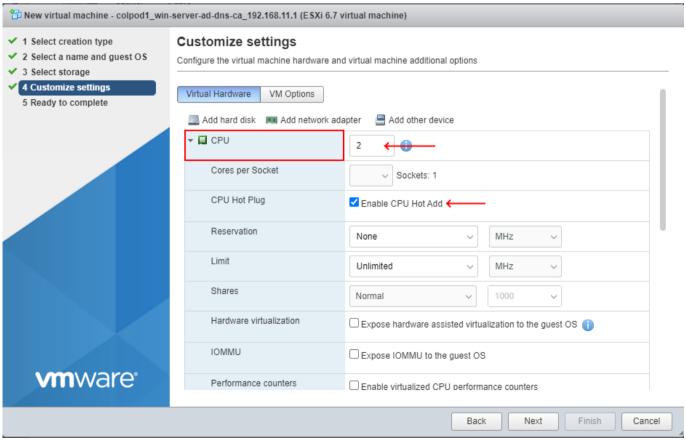
[Lab] Install Windows Server on vmware ESXi

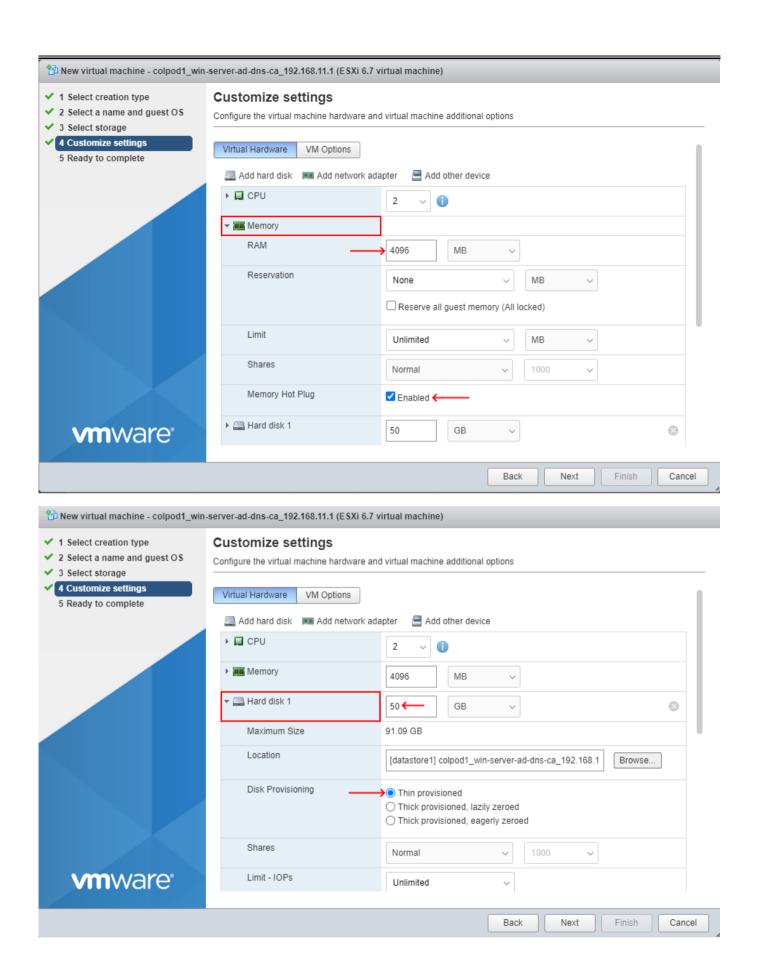


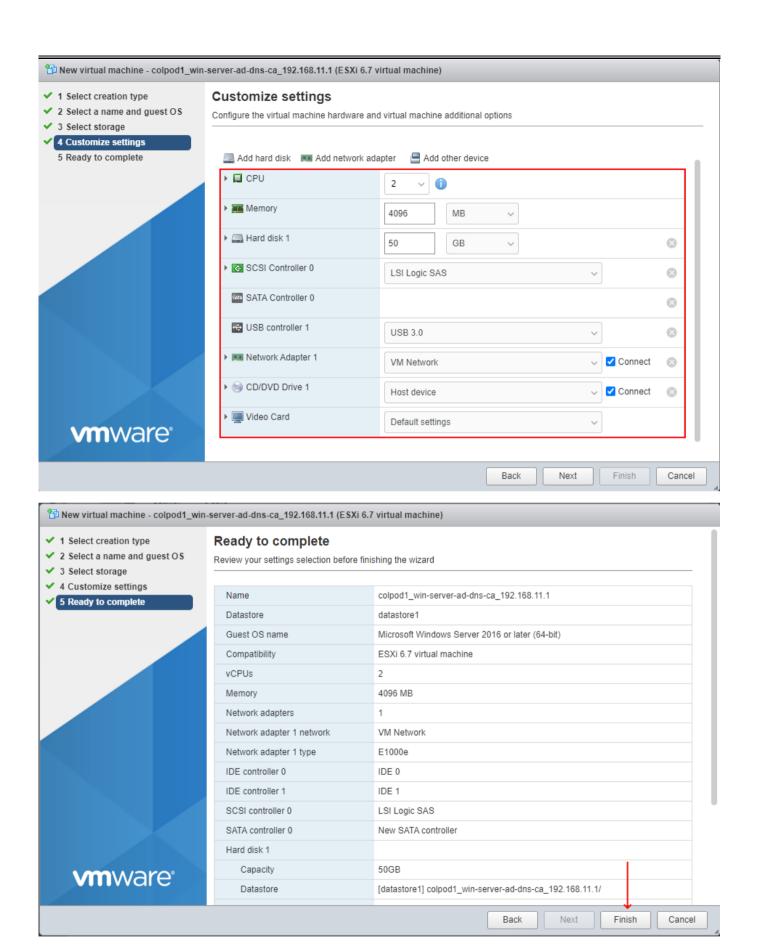




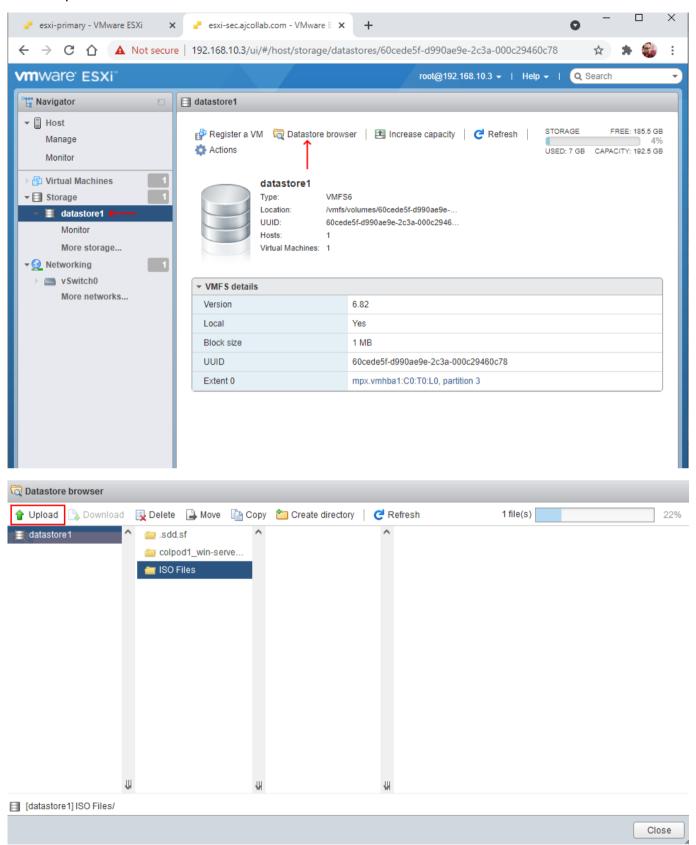




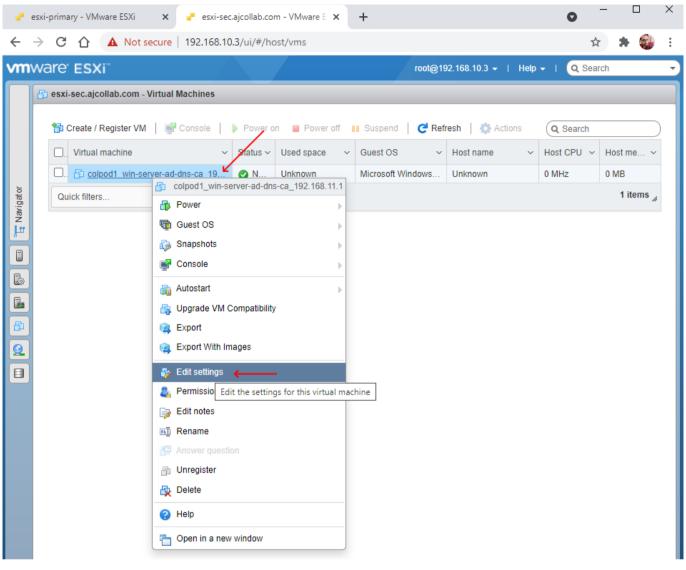


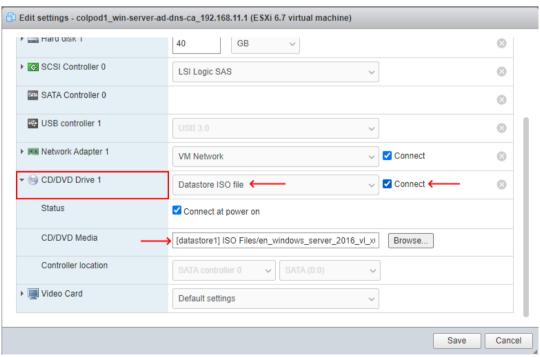


• Upload Windows Server ISO File to the Datastore

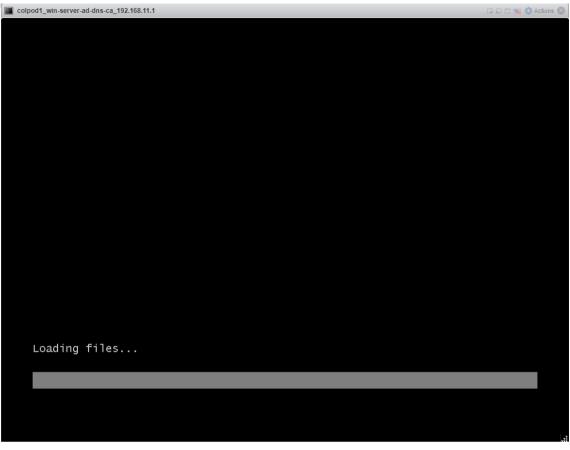


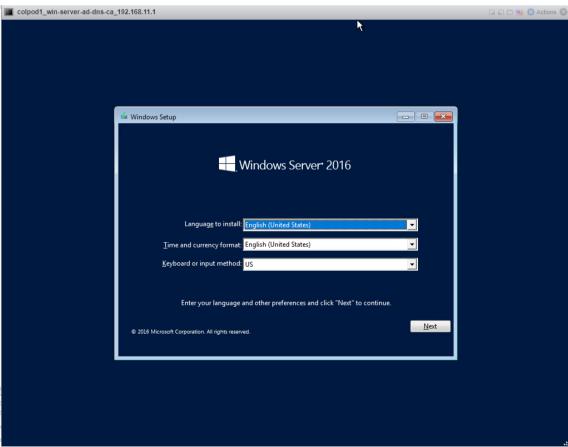
Mount the Iso File to the VM that we created

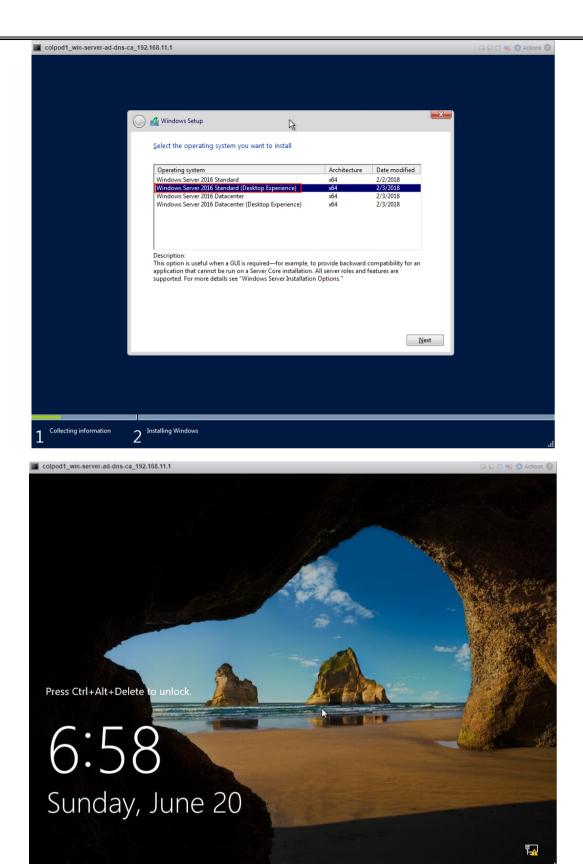




• Power on the virtual machine and continue installing Windows



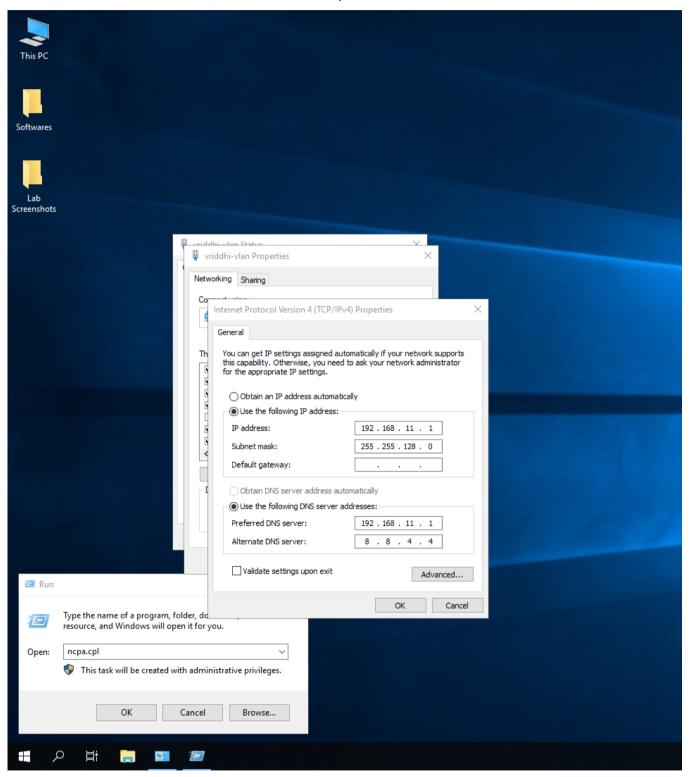




 Now you can configure IP Address and enable RDP (Remote Desktop Protocol) on the Windows from the vmware console and access it remotely

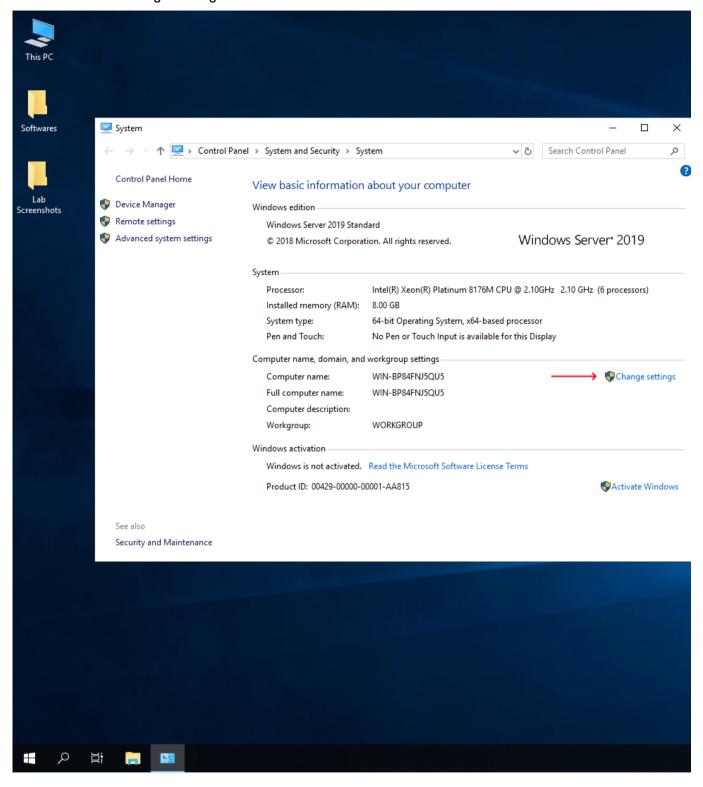
[Lab] Configure AD & DNS in Windows Server 2019

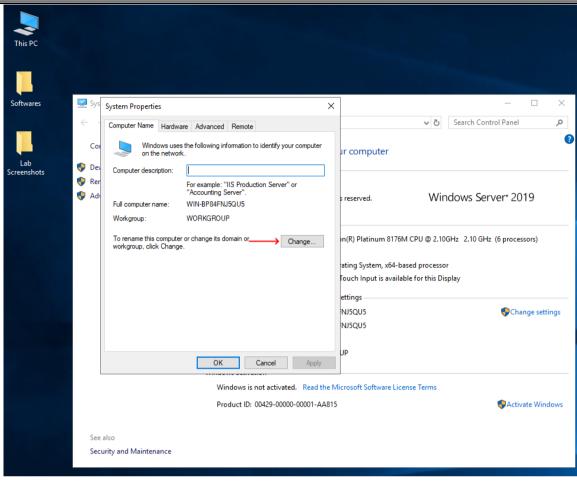
- We configure AD and DNS together on a Windows Server 2019. This will be our main server for many different purposes
- Installation of Windows Server is not the scope of this article

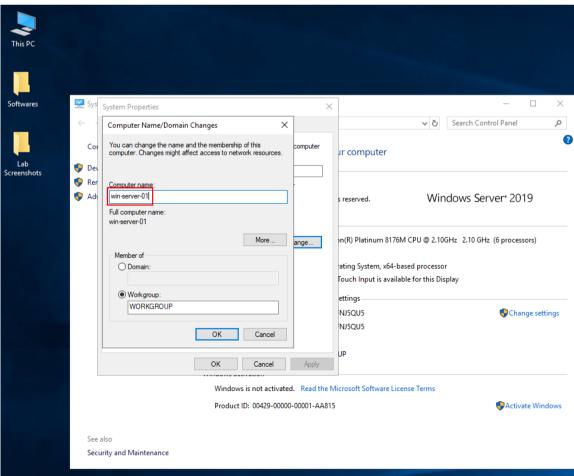


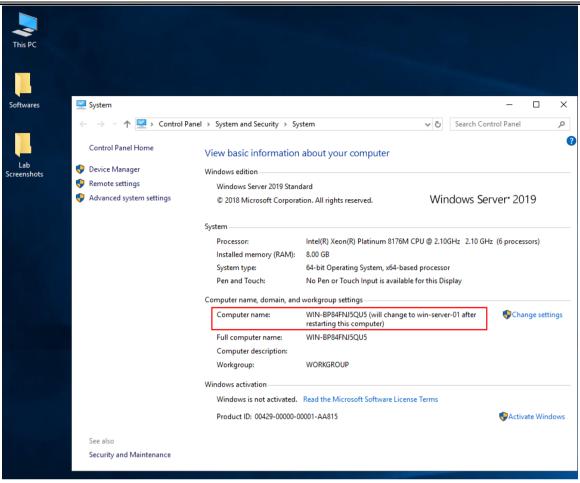
- After you done with Windows Server 2019 installation, make sure it has a static IP configured
- It is always recommended to have static IPs for servers

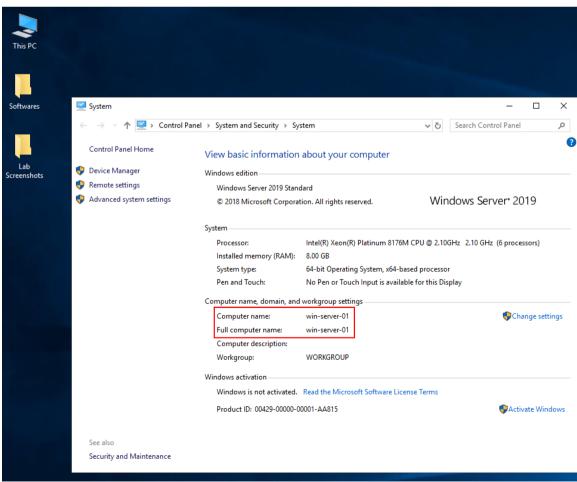
- Let's configure a hostname for the server
- Click on the Change Settings >>

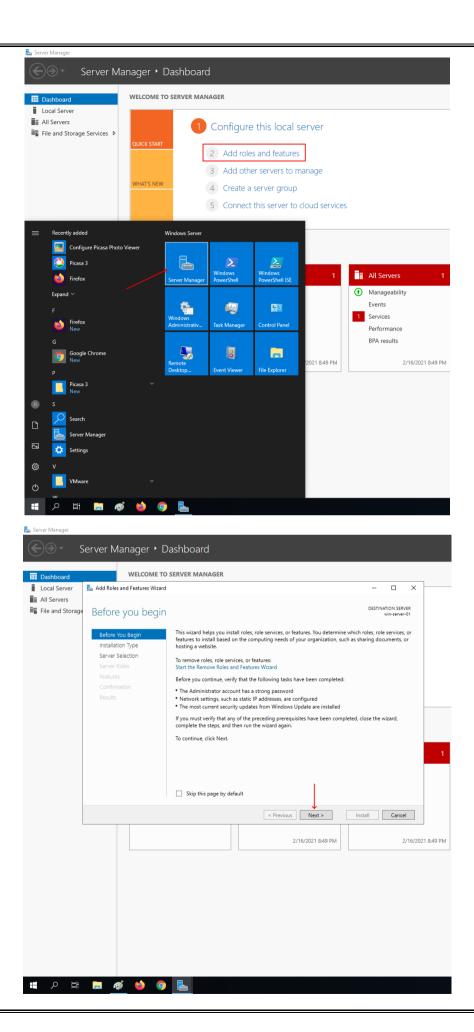


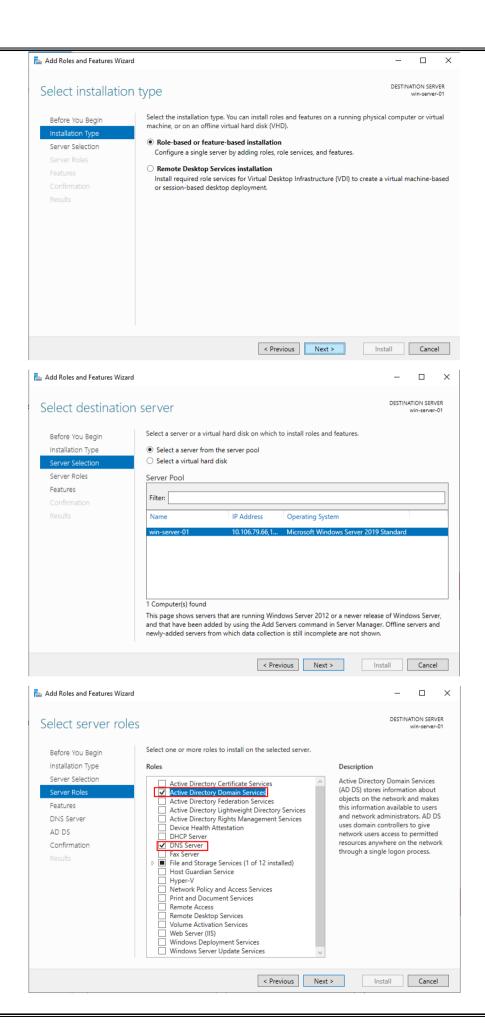


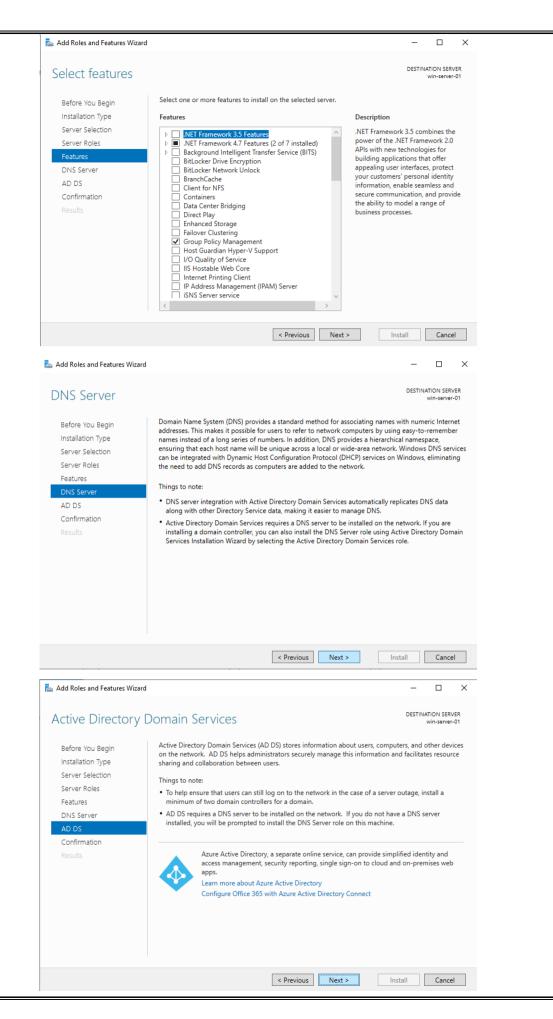


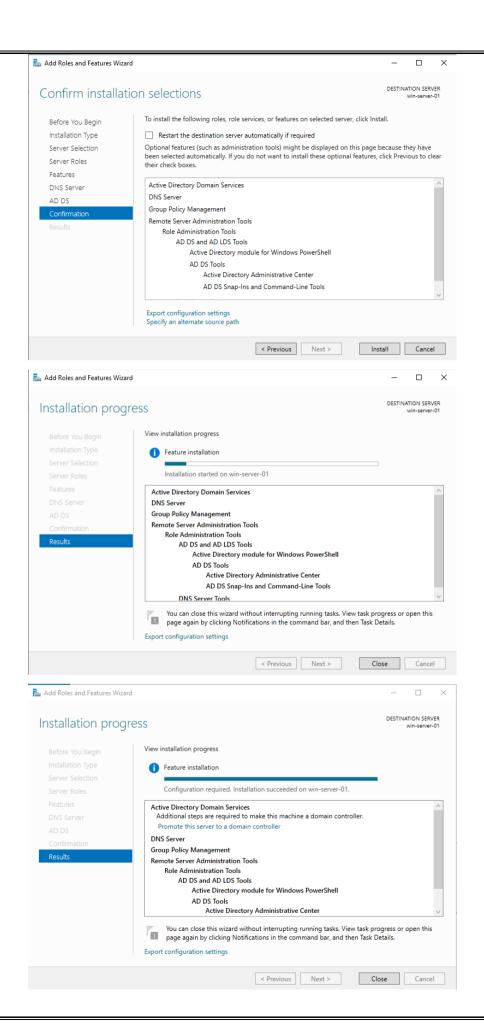


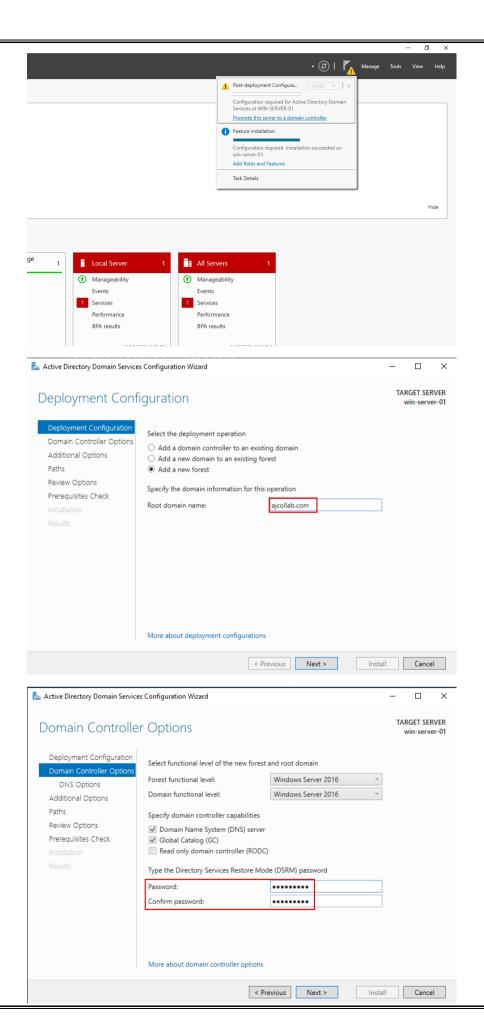


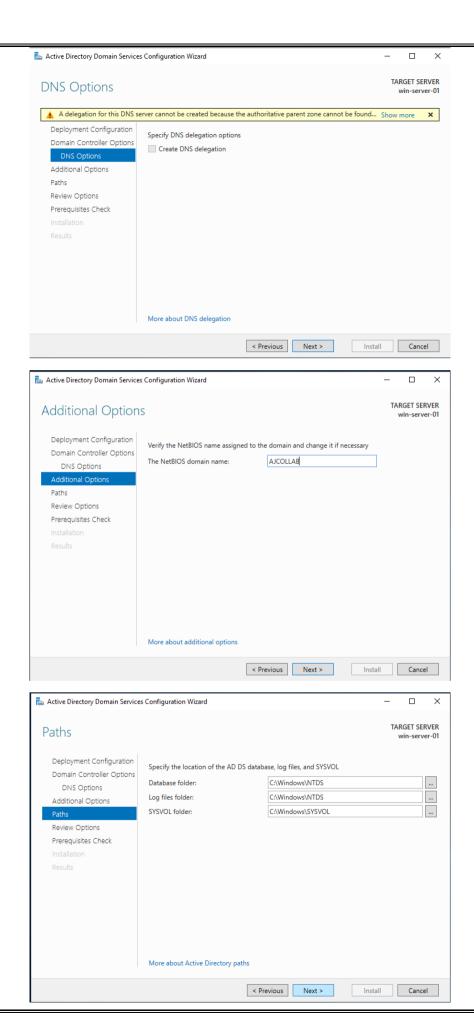


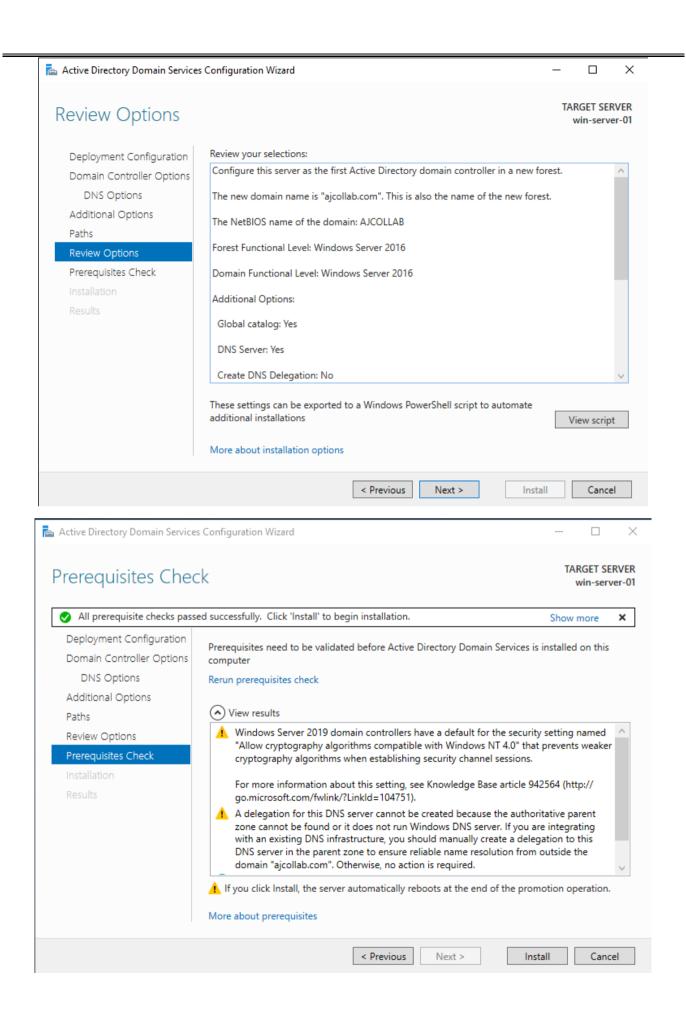


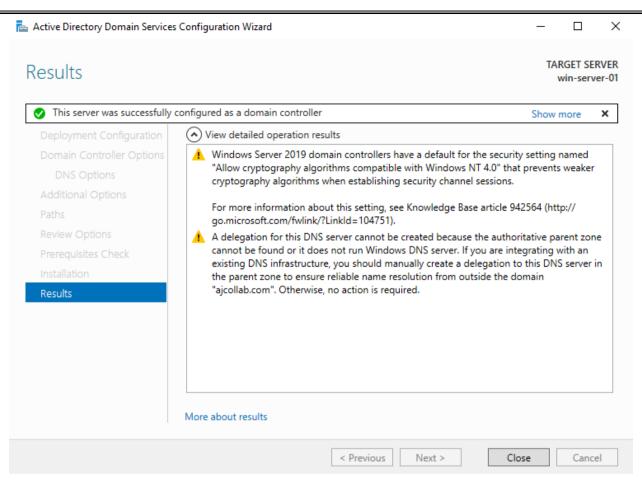


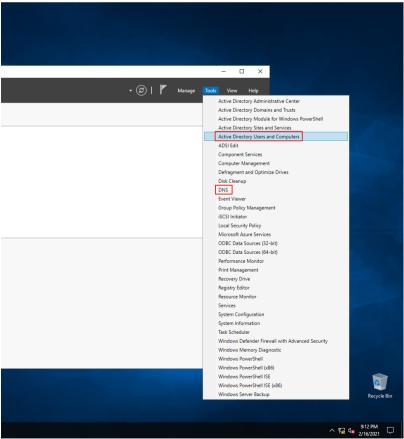


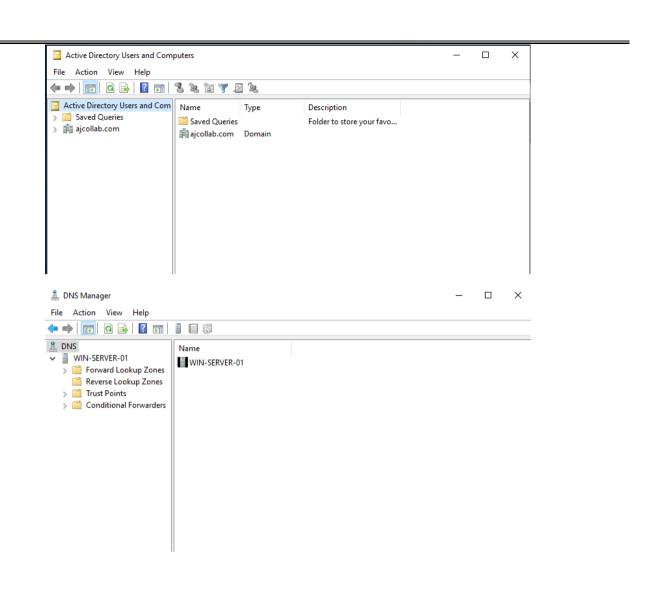






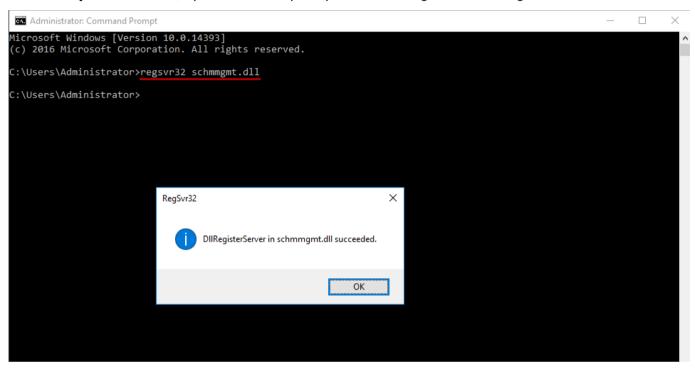






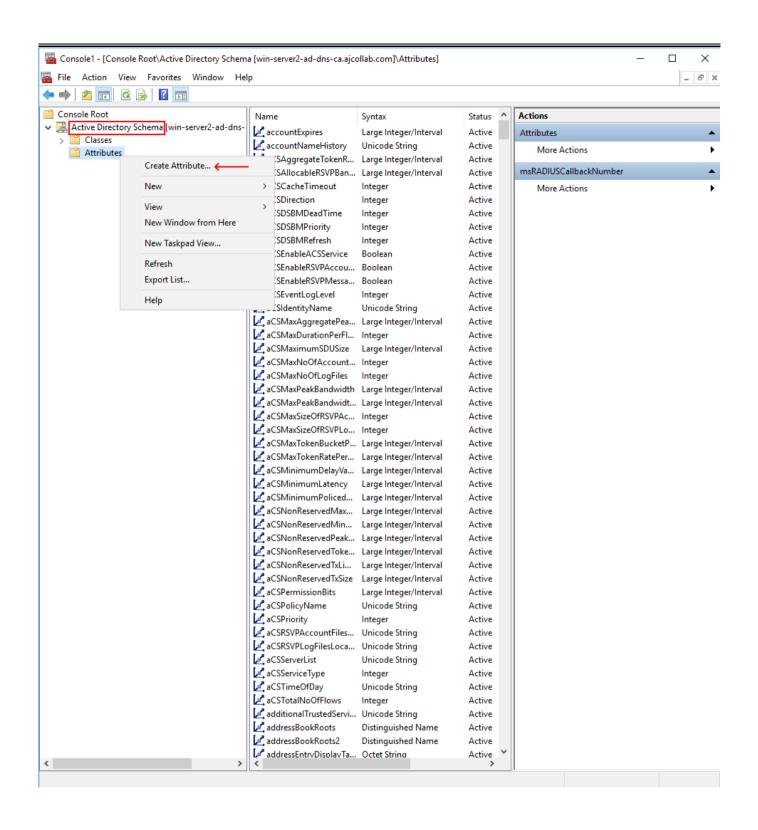
[Lab] Extending Active Directory Schema

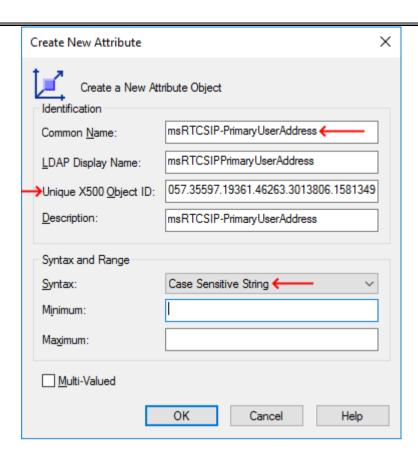
- Ideally an attribute in AD called 'msRTCSIP-PrimaryUserAddress' maps to CUCM as SIP URI. You
 can simply map email ID as well
- The attribute 'msRTCSIP-PrimaryUserAddress' is not available by default, for that you need to extend your AD Schema
- On your AD Server, open command prompt and enter regsvr32 schmmgmt.dll

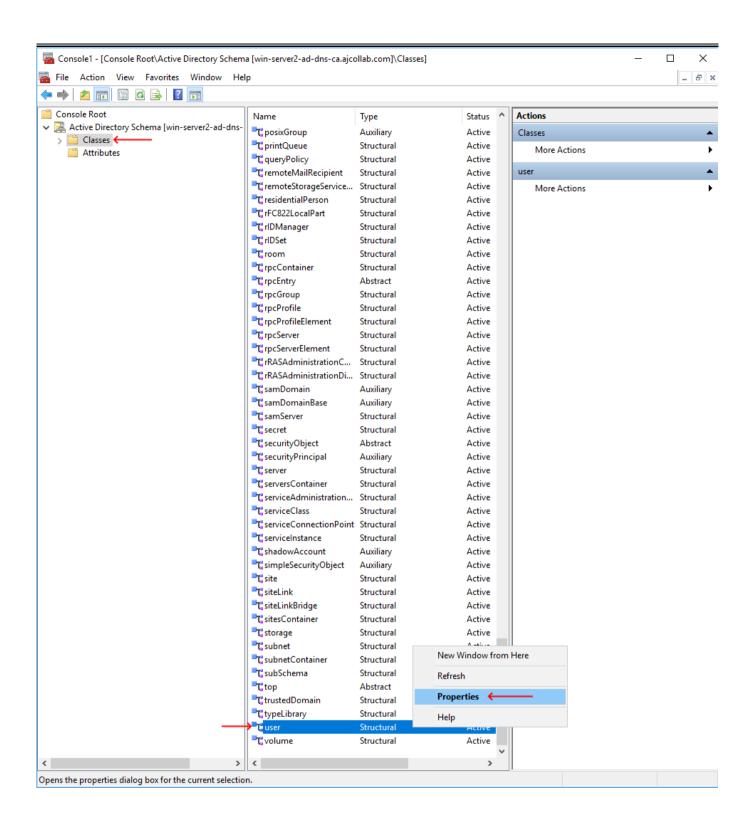


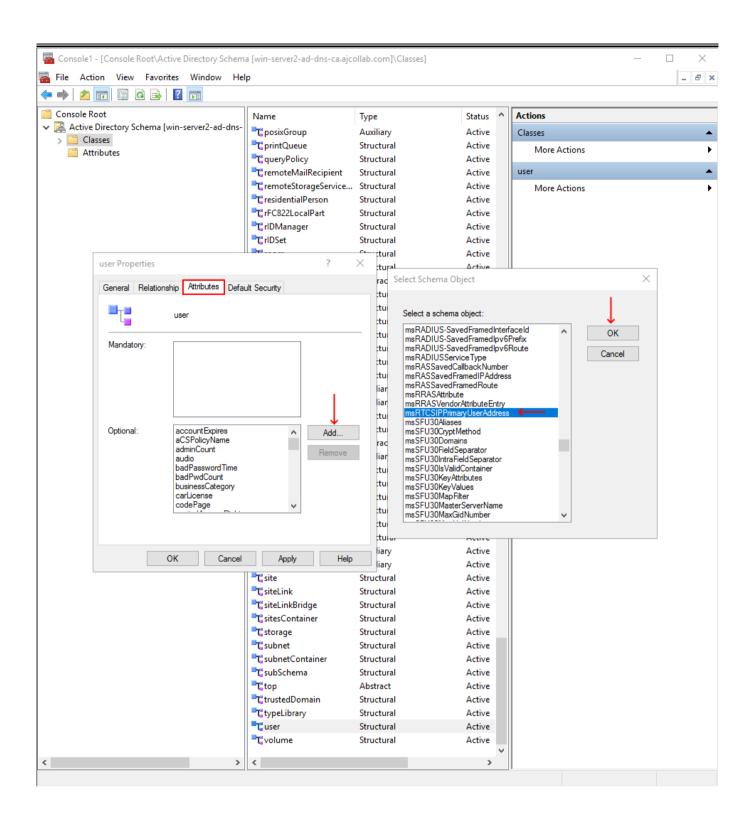
- Go to the link and copy the Script, paste it in to Notepad and save as OID.vbs
- Run the script and note the OID of your System

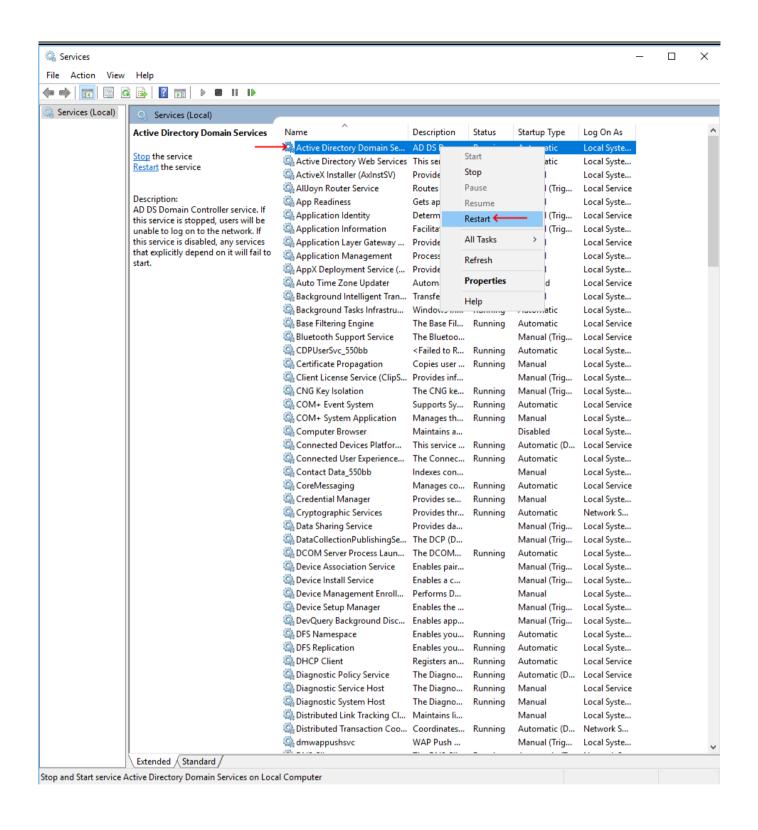
```
oidInfo - Notepad
                                                                                                              П
                                                                                                                    ×
File Edit Format View Help
Your root OID is:
1.2.840.113556.1.8000.2554.18227.64057.35597.19361.46263.3013806.1581349
This prefix should be used to name your schema attributes and classes. For example: if your prefix is "Microsoft", y
You can create subsequent OIDs for new schema classes and attributes by appending a .X to the OID where X may be any
If your assigned OID was: 1.2.840.113556.1.8000.2554.999999
then classes could be under: 1.2.840.113556.1.8000.2554.999999.1
which makes the first class OID: 1.2.840.113556.1.8000.2554.999999.1.1
the second class OID: 1.2.840.113556.1.8000.2554.999999.1.2
Using this example attributes could be under: 1.2.840.113556.1.8000.2554.999999.2
which makes the first attribute OID: 1.2.840.113556.1.8000.2554.999999.2.1
the second attribute OID: 1.2.840.113556.1.8000.2554.999999.2.2
Here are some other useful links regarding AD schema:
Understanding AD Schema
http://technet2.microsoft.com/WindowsServer/en/Library/b7b5b74f-e6df-42f6-a928-e52979a512011033.mspx
```

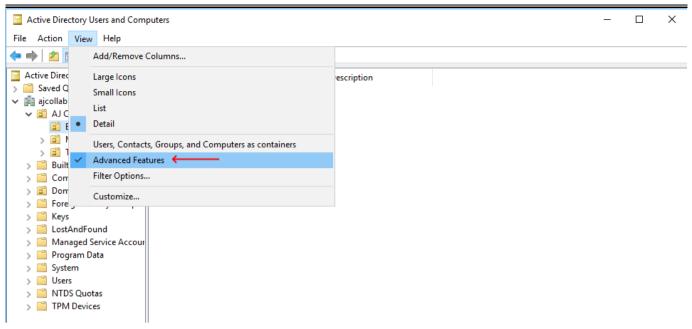


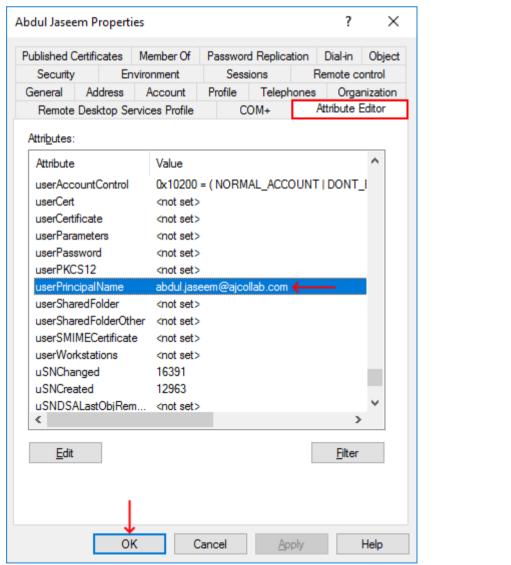






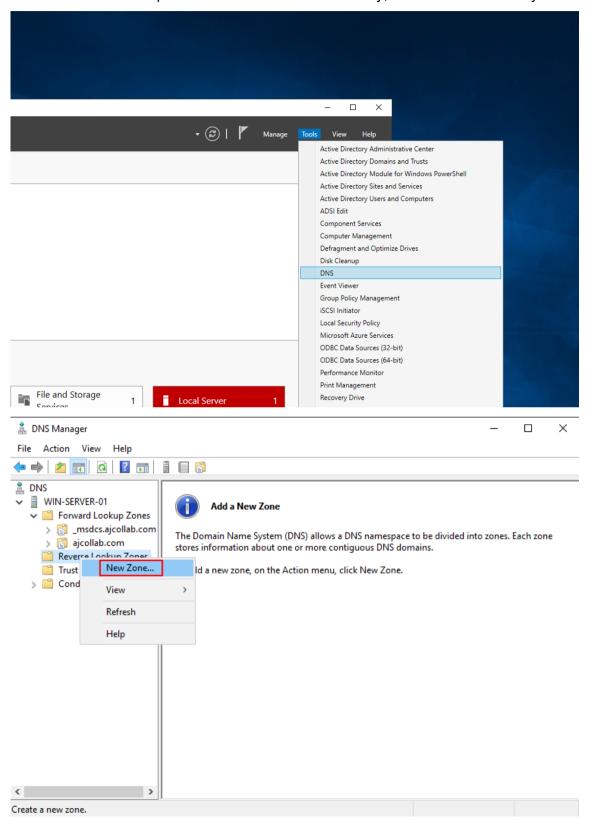


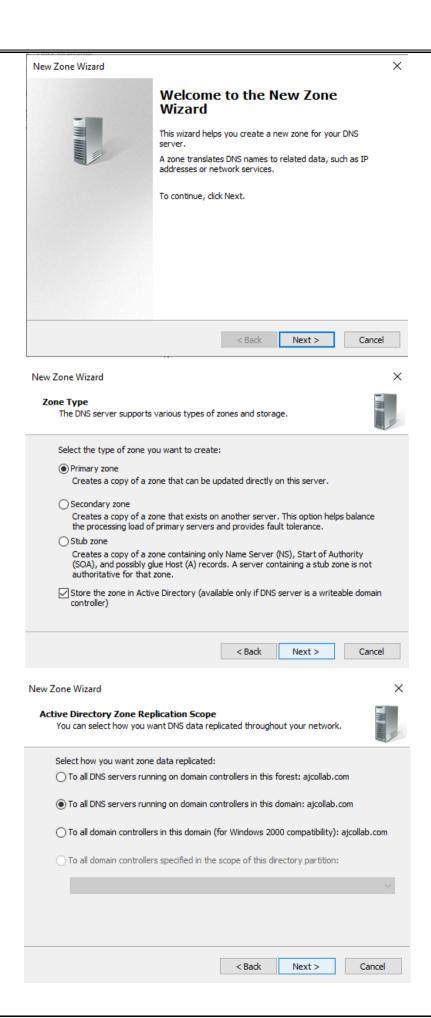


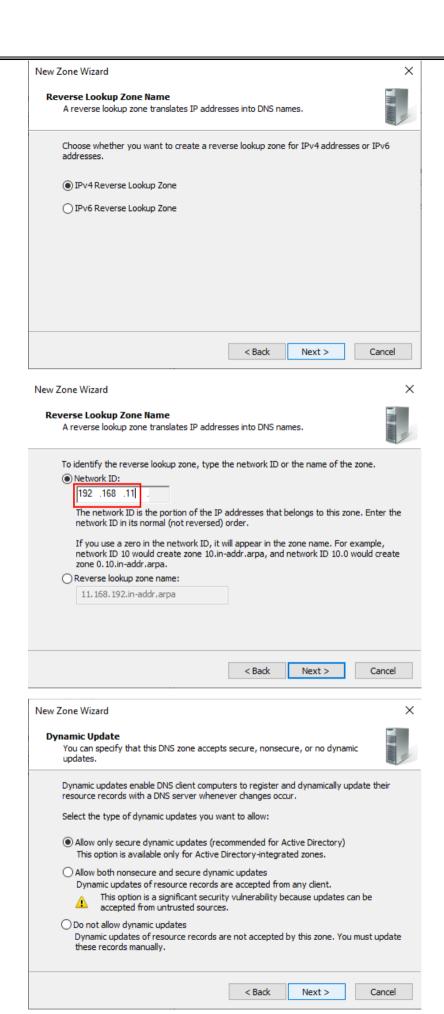


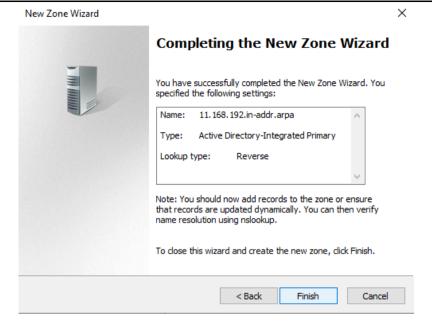
[Lab] Configure Reverse Lookup Zone in Windows DNS Server

- After adding DNS feature on the Windows Server, we need to manually add Reverse Lookup zone for the domain
- Forward Lookup Zone will be created automatically, we don't need to worry about it

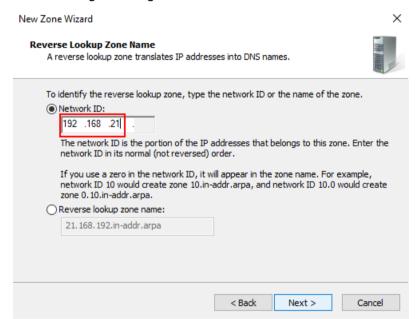






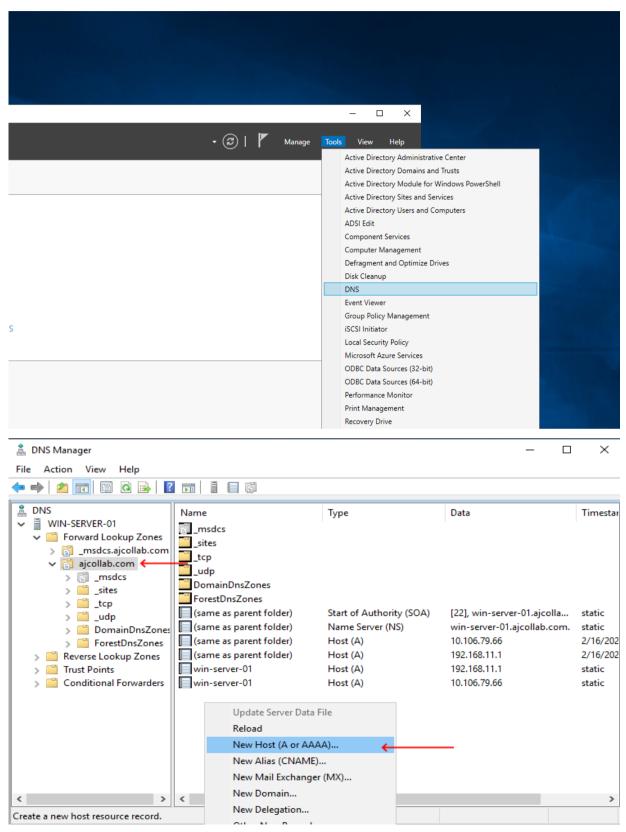


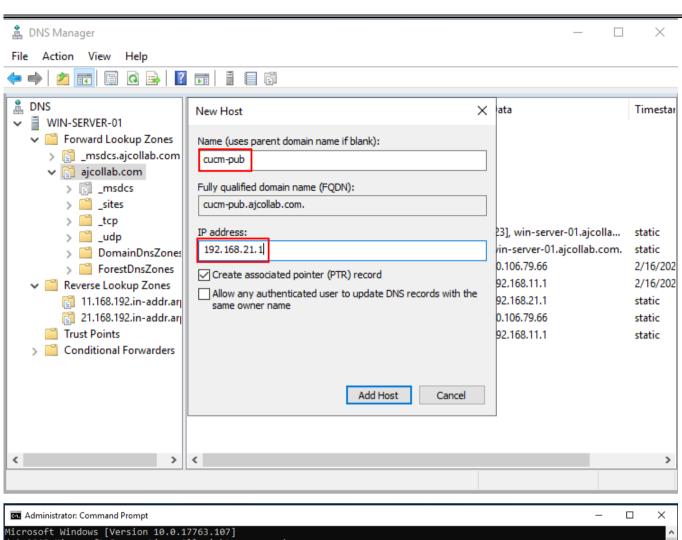
- You need to add other IP Ranges as well
- I have added 192.168.21.X range here again



[Lab] Add UC Servers DNS Entries

- Before installing CUCM or any other UC Server, we need to make sure that we have proper DNS records for the server that we are going to install
- In this Lab we will add CUCM-PUB server's IP Address as a DNS A Record

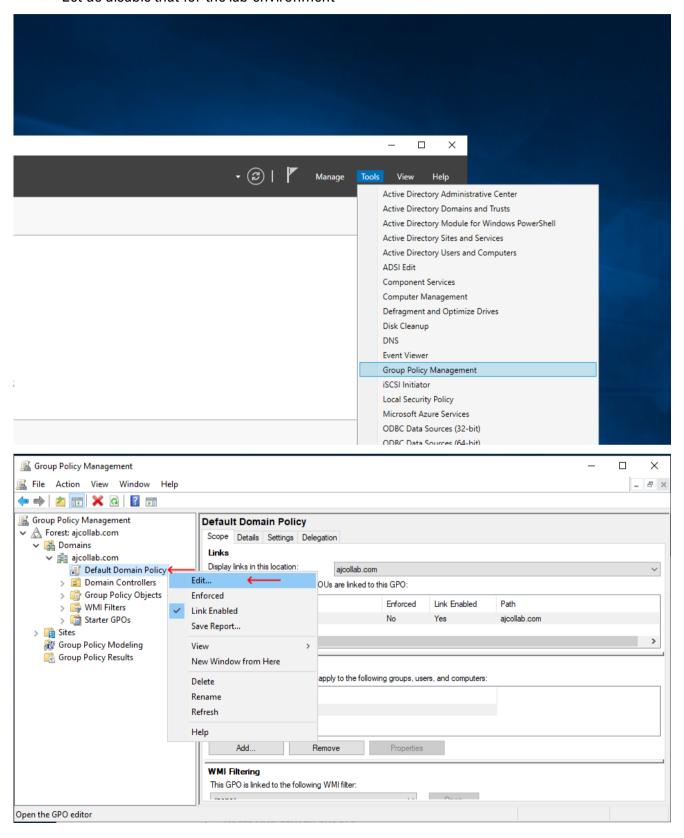


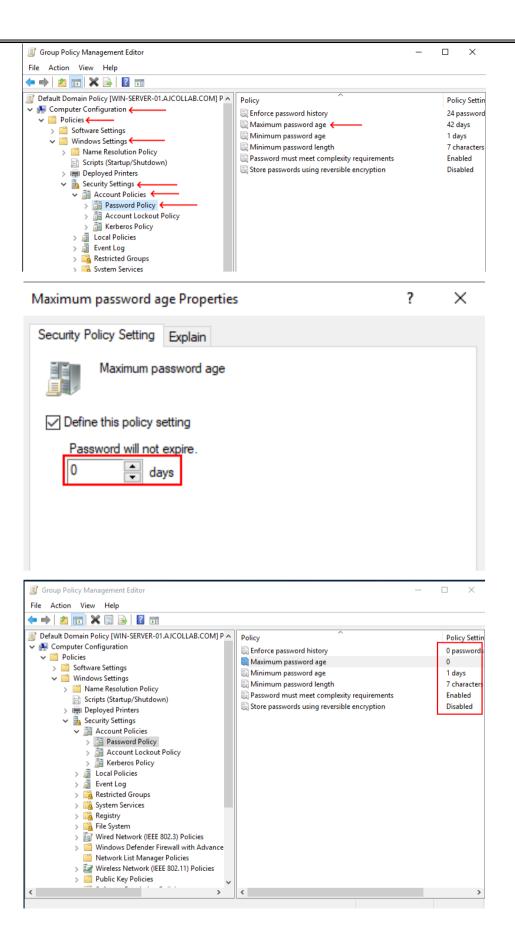


```
Microsoft Windows [Version 10.0.17763.107]
(c) 2018 Microsoft Corporation. All rights reserved.
serial = 0
       refresh = 28800 (8 hours)
retry = 7200 (2 hours)
expire = 604800 (7 days)
default TTL = 86400 (1 day)
Server:
        UnKnown
Address:
Name: cucm-pub.ajcollab.com
Address: 192.168.21.1
C:\Users\Administrator>
C:\Users\Administrator>nslookup cucm-pub.ajcollab.com
serial = 0
       refresh = 28800 (8 hours)
       retry = 7200 (2 hours)
expire = 604800 (7 days)
default TTL = 86400 (1 day)
Server:
        UnKnown
Address:
        cucm-pub.ajcollab.com
Name:
Address: 192.168.21.1
```

[Lab] Disable Windows Password Auto Expiry Group Policy

- By default, there is a password policy in-place for every Windows Server. Our password will be expired after some days
- Let us disable that for the lab environment

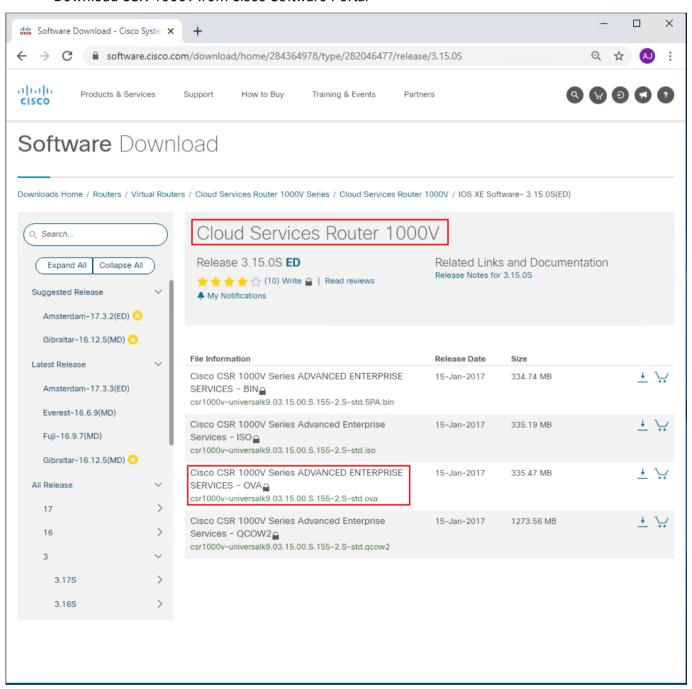


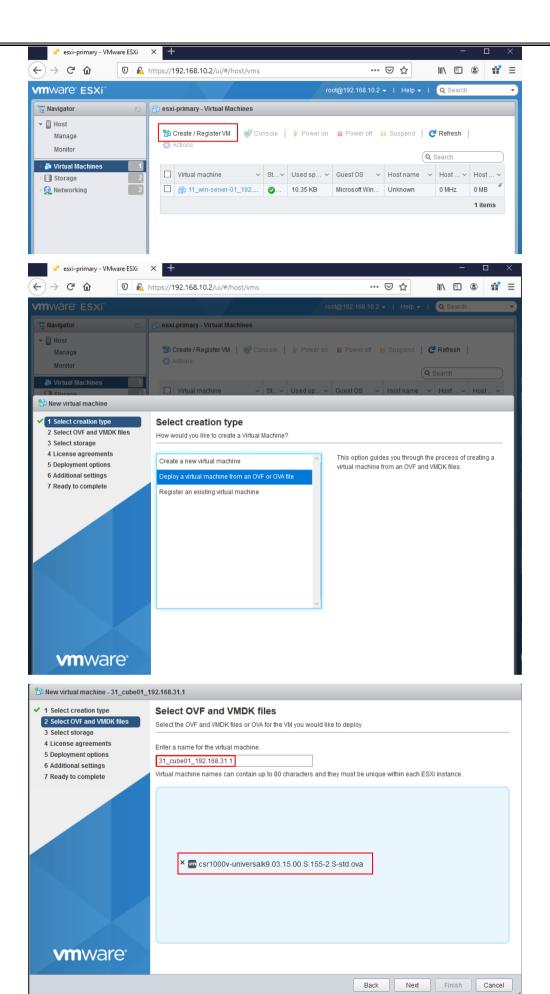


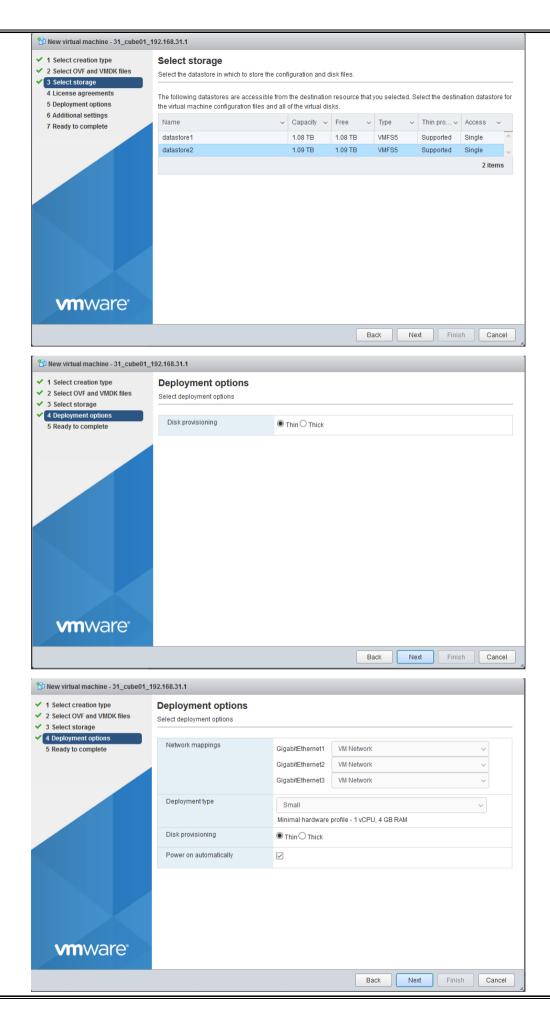
- You can customize other password policies here
- E.g. Password Complexity

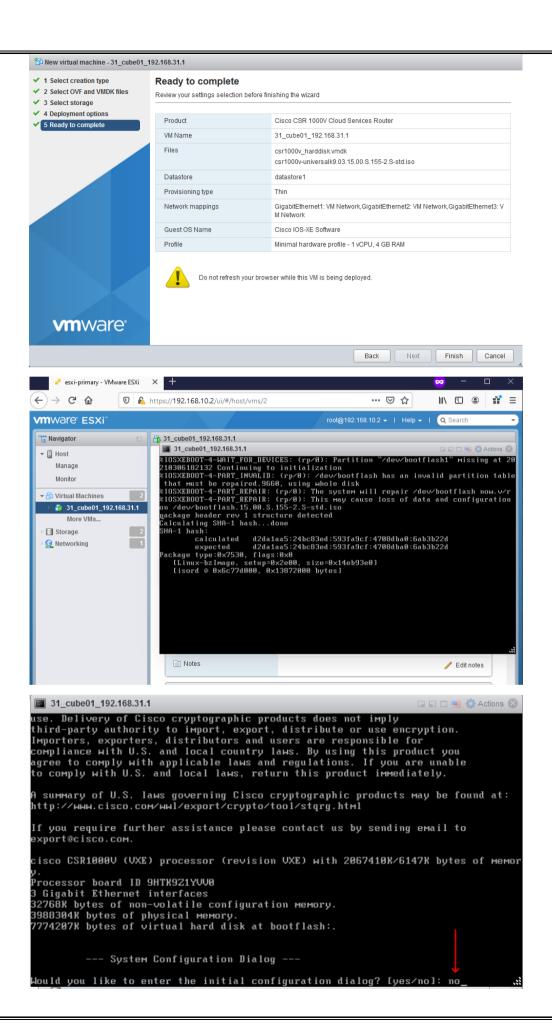
[Lab] Deploying CSR 1000v as NTP Server

- While setting up any UC lab, NTP is an important component, it maintains accurate time across all servers
- NTP can be configured on any Cisco Routers, Linux. Windows based NTPs are not supported for UC infrastructure
- I have used Cisco CSR 1000v as my NTP Server, it is an IOS XE Virtual Router. I will be using the same router as CUBE in future lectures
- Download CSR 1000v from Cisco Software Portal





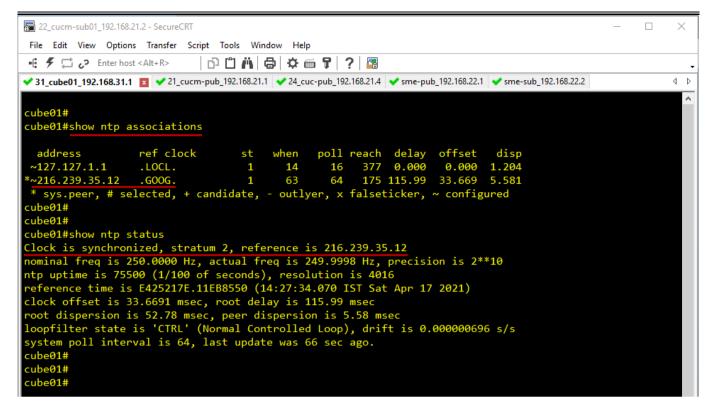




```
cube01(config)#line vt
cube01(config)#line vty 0 4
cube01(config-line)#login lo
cube01(config-line)#login lo
cube01(config-line)#trans
cube01(config-line)#trans
cube01(config-line)#transport in
cube01(config-line)#
cube01(config-line)#
cube01(config-line)#
cube01(config-line)#
cube01(config-line)#
cube01(config-line)#
cube01(config-line)#
cube01(config)#ntp source gi
cube01(config)#ntp source gigabitEthernet 1
cube01(config)#ntp server 192.168.31.1 sou
cube01(config)#ntp server 192.168.31.1 source gi
cube01(config)#
cube01(config)#
cube01config)#
cube01config)#
cube01configofiguration...
*Mar 6 19:08:49.505: %SYS-5-CONFIG_I: Configured from console by console[OK]
cube01#__
```

Below configurations will enable SSH and NTP on the virtual router

```
hostname cube01
enable secret Ajcollab@1
username admin password 7 Ajcollab@1
interface GigabitEthernet1
ip address 192.168.31.1 255.255.224.0
no shut
ip domain name ajcollab.com
ip name-server 192.168.11.1
ip name-server 192.168.0.1
crypto key generate rsa
ip ssh version 2
line vty 0 4
login local
transport input all
ntp server time.google.com source gigabitEthernet 1
ntp master 2
ntp source gigabitEthernet 1
clock timezone IST +5 30
```



- If you are going with ISO based CSR 1000v VM, below are the virtual hardware specification
 - Compatibility: ESXi 5.5 Virtual Machine
 - Guest OS Family: Linux
 - Guest OS Version: Other 2.6 Linux (64 bit)
 - o 1 vCPU
 - 4 GB RAM (2GB RAM will also work)
 - o 8GB HDD
 - o 3 NICs

How to Build Your Own Home UC Lab in vmware Workstation

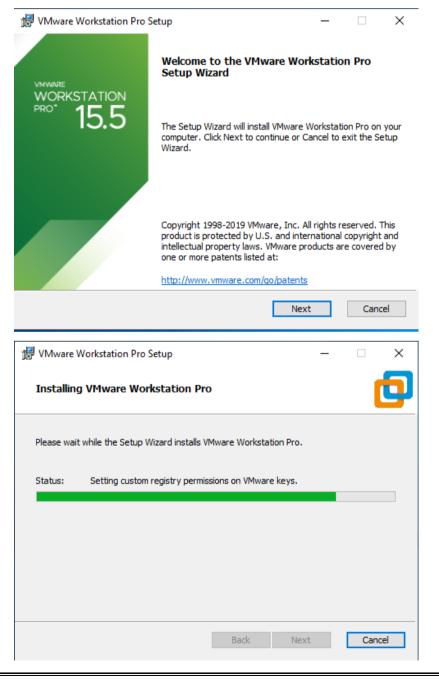
- The complete UC lab you can set up on your laptop / workstation if you have enough RAM
- Even without large amount of RAM, you can still setup a miniature lab for practice
- You need to have vmware Workstation to build the lab, download and install vmware workstation

Step 1: Enable virtualization Intel VT-X from BIOS

- This step differs for each laptop / desktop manufacture
- You should find a way to enable Intel VT-X from BIOS. Google to find the way to enable it based on you manufacture

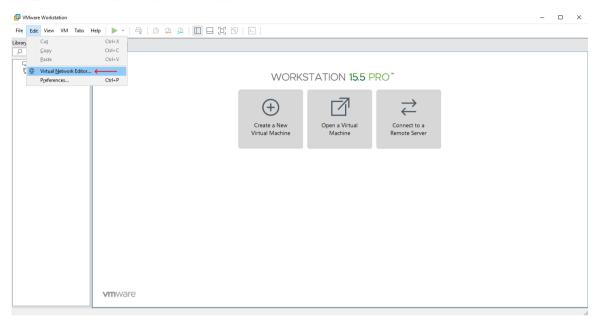
Step 2: Install vmware Workstation

Installation is pretty straight forward, just Next, Next, Finish

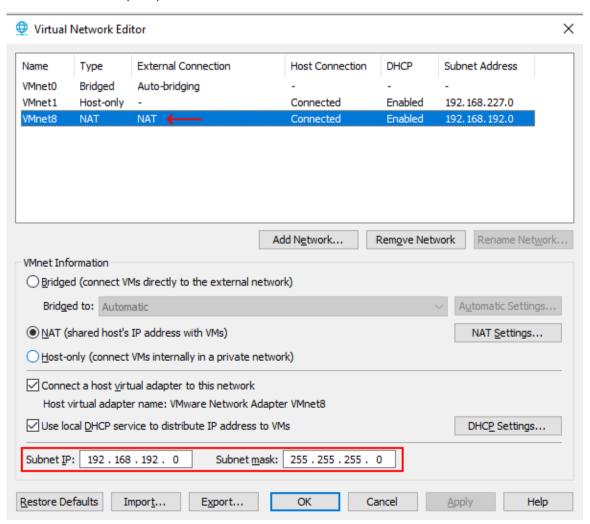


Step 3: Identify vmware NAT Network Settings

- Once you complete the installation, open vmware Workstation
- Go to Edit >> Virtual Network Editor



Select the VMnet8 (NAT) and check the IP Subnet



 Here my NAT Network is 192.168.192.0 and Subnet Mask is 255.255.255.0, hence below IP configurations can be used for UC Servers

 Network
 192.168.192.X/24

 Subnet Mask
 255.255.255.0

 Starting IP Address
 192.168.192.3

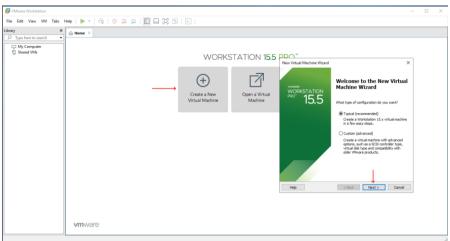
 Ending IP Address
 192.168.192.254

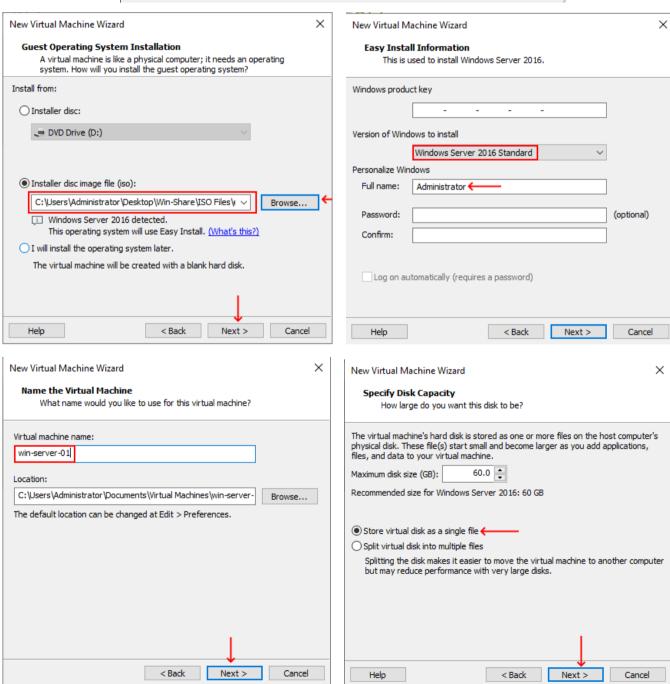
 Default Gateway
 192.168.192.2

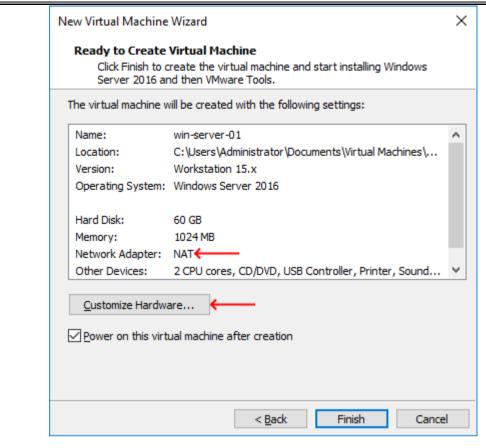
NTP 216.239.35.8 (Google NTP)

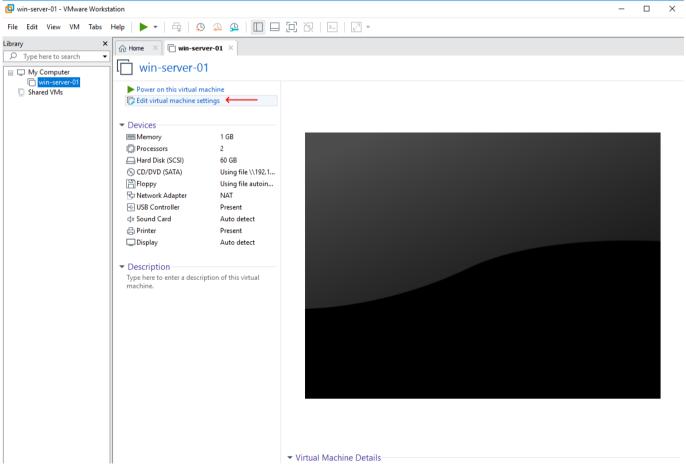
- While installing UC Servers, make sure you are connected to Internet since we use Public NTP that is available over internet
- Now you can deploy Windows Server for AD and DNS purpose and then UC Servers can be deployed
- Network Adapter of all VMs must be in NAT
- For CUCM installation, DNS server is optional, you can install without DNS Server as well but for UCCX installation, DNS is mandatory
- Here you don't need local any NTP Server, we have used public NTP

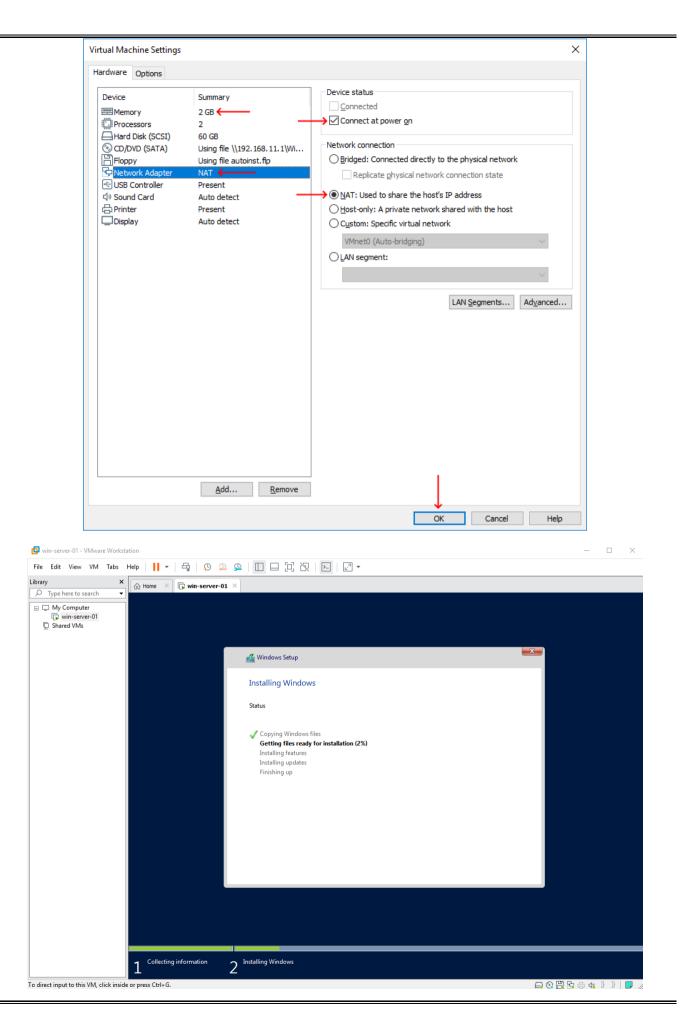
Step 4: Install Windows Server







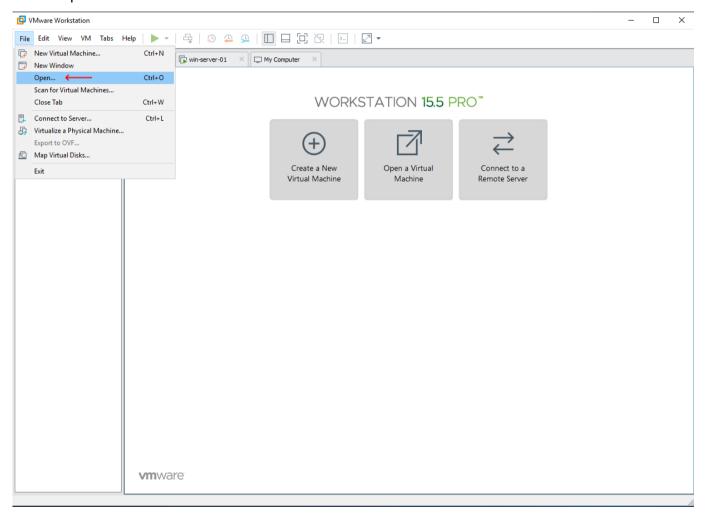


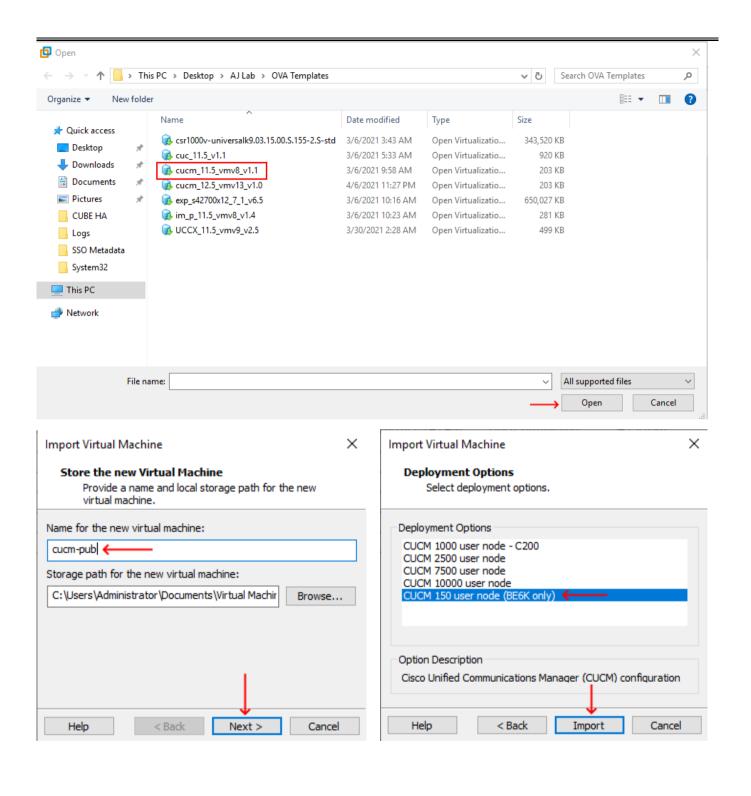


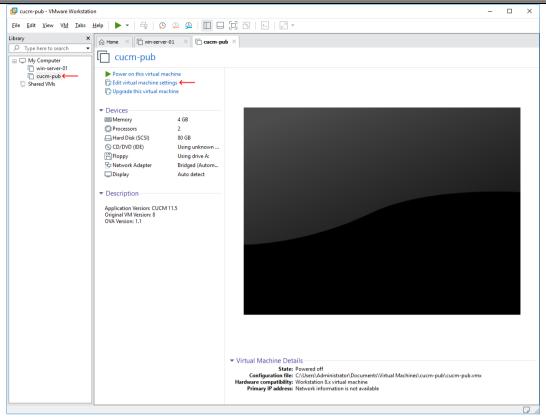
- Once windows installation completed, set a static IP address for it with default gateway 192.168.192.2
- Make sure you are able to access internet from the Windows Machine
- Setup AD and DNS on the windows server and add UC Server DNS A Records before installing CUCM

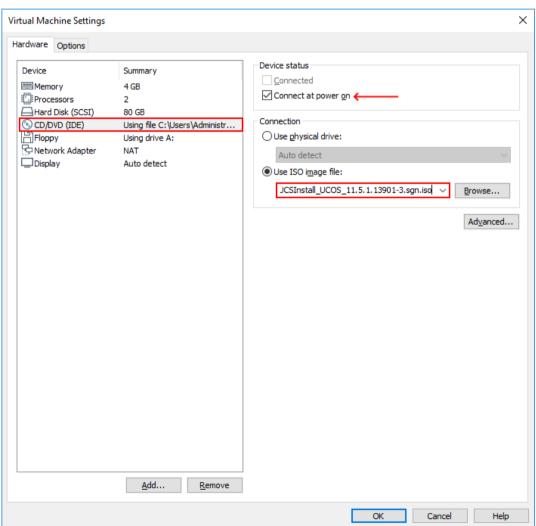
Step 5: Install CUCM

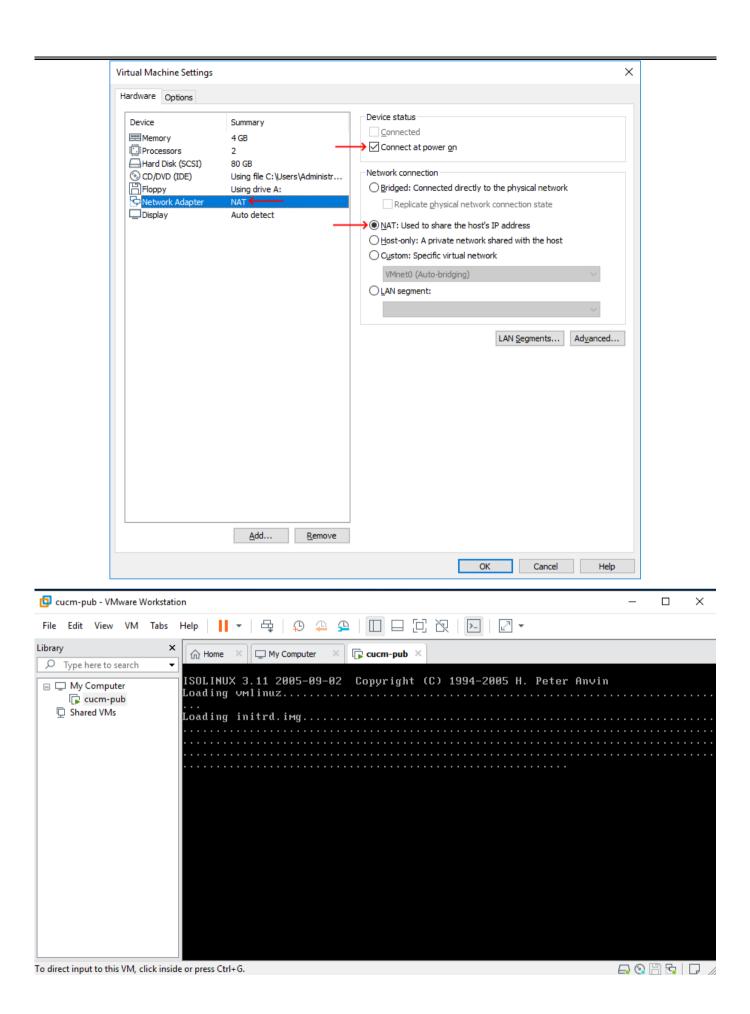
- Download the CUCM OVA
- Open the OVA from vmware Workstation

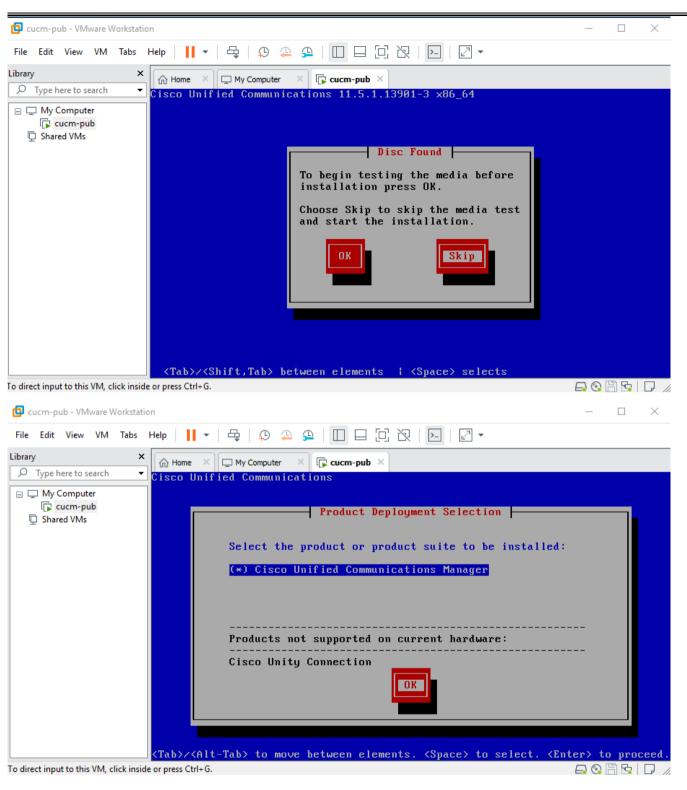












You can continue the installation, with below Network Settings

 IP Address
 192.168.192.X

 Subnet Mask
 255.255.255.0

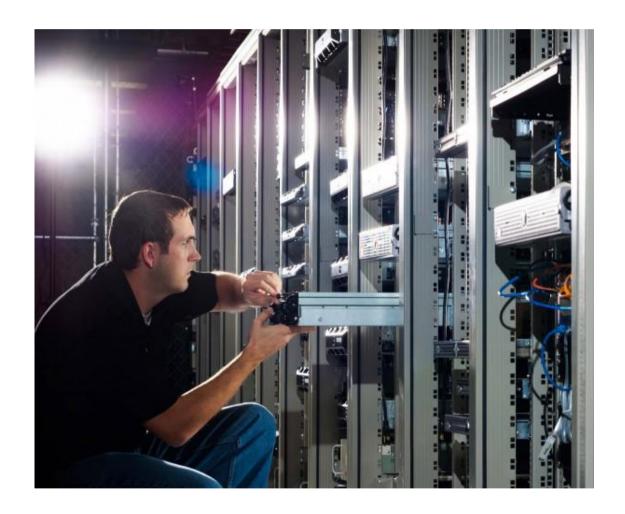
 Default Gateway
 192.168.192.2

 NTP
 216.239.35.8

 DNS Server
 Windows Server IP

It is important to select Network Adapter as NAT

Chapter 1 Module 2 - CUCM Cisco On-Premise Collaboration Solution Cisco Unified Communications Manager (CUCM)



62

Cisco Unified Communications Manager (CUCM)



CallManager

- Cisco Unified Communications Manager (CUCM or CallManager) is an IP-based communications system. It enables VoIP (Voice and Video Over IP Network)
- Acts as a soft switch that switches voice and video traffics
- CUCM is deployed as a cluster and the primary server in the cluster is called Publisher and other servers are known as Subscribers
- We can have multiple subscriber servers in the cluster based on the number of users and features
- CUCM Publisher maintains read-write copy of Database where CUCM Subscriber retains read only copy of Database
- IBM IDS (Informix Dynamic Server) is the database of CUCM
- IP Phones and Telepresence endpoints register with CUCM using SIP protocol (SCCP protocol was used in the older IP Phones)
- External PSTN integration is possible via PRIs on Voice Gateways (Voice Capable IOS Routers) and SIP Trunking on CUBE (Cisco Unified Border Element)
- Voice Gateway uses SIP, MGCP or H.323 to communicate with CUCM whereas CUBE uses only SIP

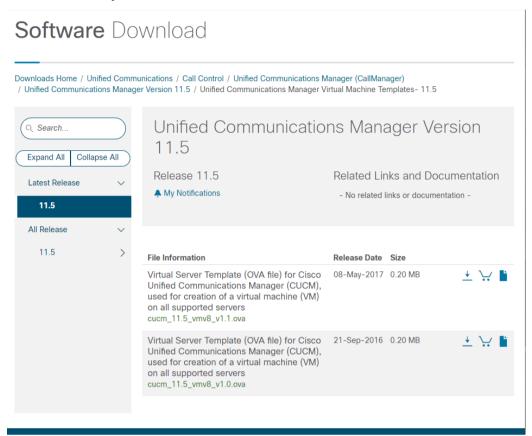
CUCM Installation - Publisher

- CUCM OS is a Cisco customized RedHat or CentOS Linux and installed as a Virtual Machine in vmware environment
- We need to download CUCM OVA (preconfigured virtual hardware file without OS installed) and then install the CUCM OS on top of it in a virtualized environment
- We must have a proper network connectivity and a working NTP server to complete CUCM installation
- Production grade CUCM cluster is installed on vmware ESXi. For learning purpose, you can install
 in on vmware workstation
- We will be dealing with CUCM 11.5 Version throughout this course

[Lab] Installing CUCM on vmware ESXi

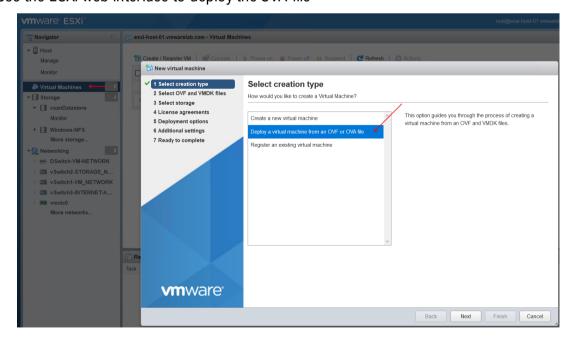
Step 1: Download the CUCM 11.5 OVA from Cisco Downloads

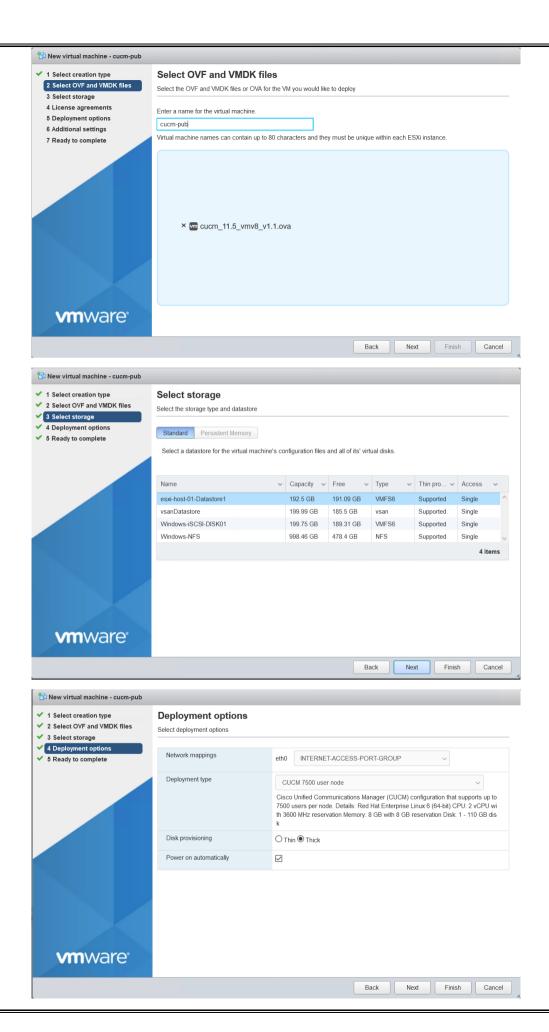
• For lab environment, you can download it from here.

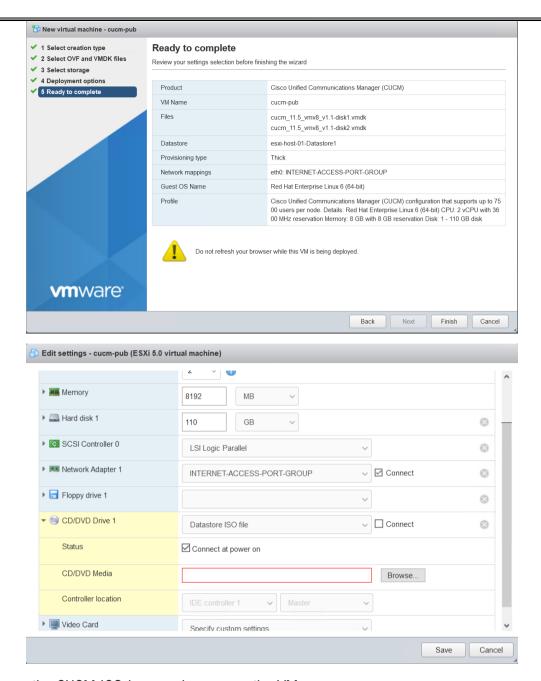


Step 2: Deploy the OVA File in vmware ESXi

Use the ESXi web interface to deploy the OVA file

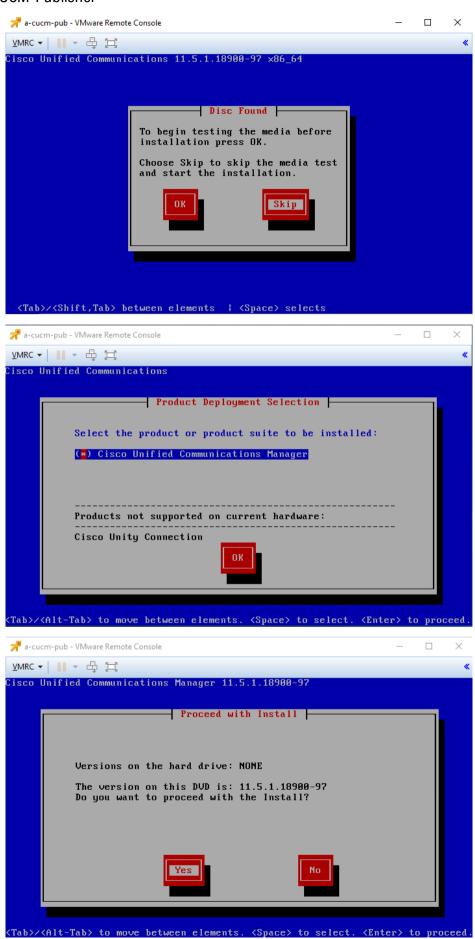






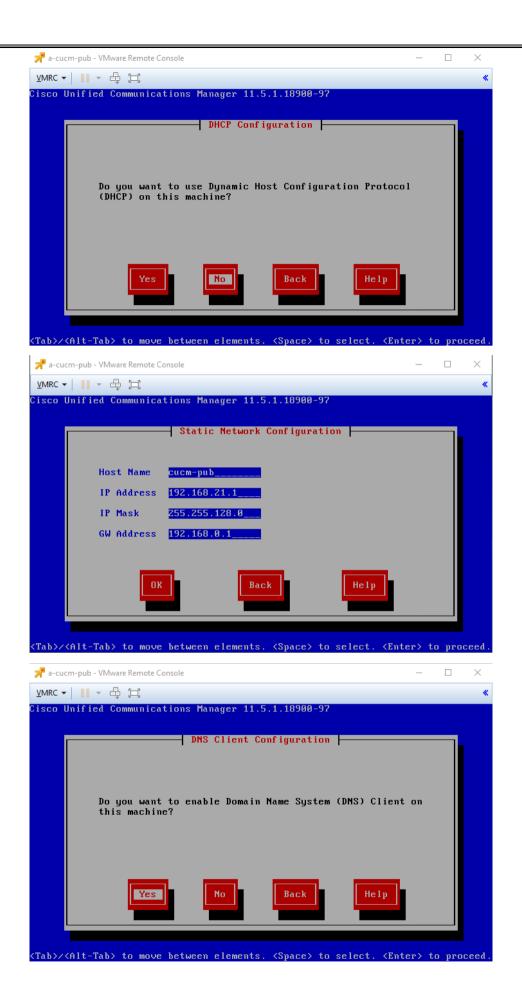
- Browse the CUCM ISO here and power on the VM
- CUCM ISO can be arranged from Cisco Partners or from TAC
- Bootable CUCM ISO is not available for direct download

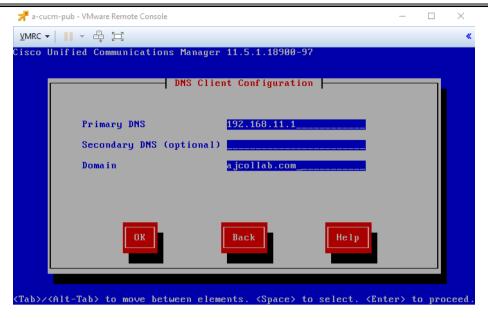
Step 3: Install CUCM Publisher



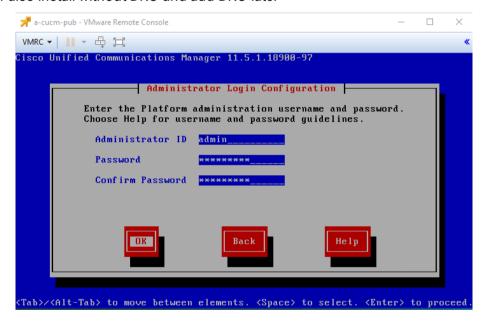




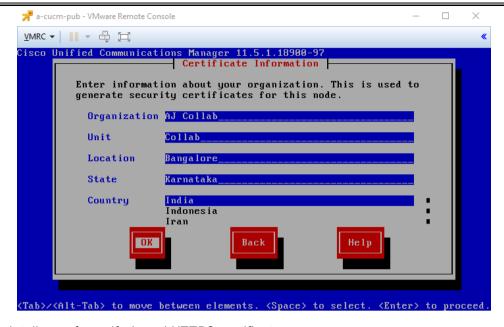




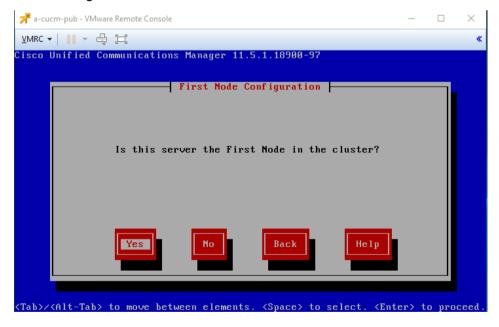
You can also install without DNS and add DNS later



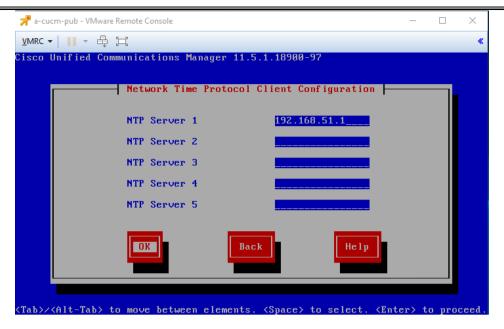
- This is the Linux OS Admin password.
- Used to login to CUCM Node CLI while troubleshooting
- This is node specific, means we need to enter this while installing CUCM Pub and CUCM Sub



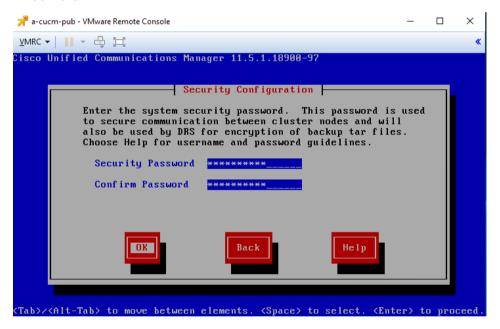
- These details are for self-signed HTTPS certificate.
- CUCM internally generates a self-signed certificate during the installation
- · We can install CA Signed certificate later



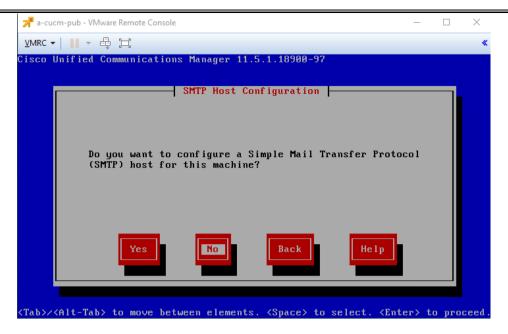
• This is the place we decide whether this is Pub or Sub.



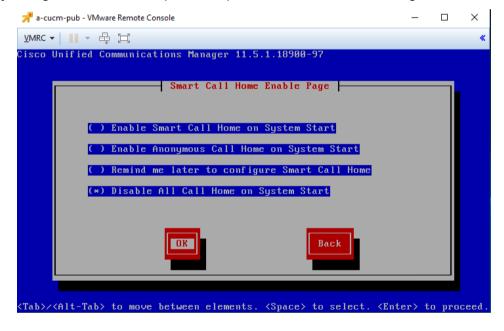
A working NTP server is mandatory. You could configure a Cisco IOS Router as NTP server or use
 Public NTP servers

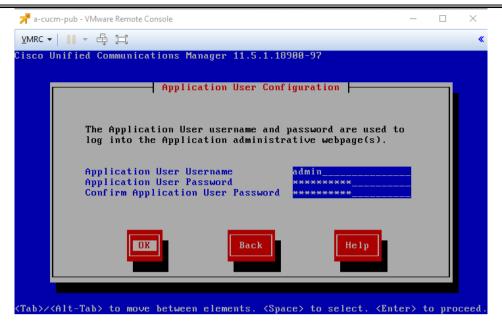


- This is the cluster security password.
- While adding other nodes (subscriber servers) to the cluster, this password is used to authenticate database each other
- During the installation of Subscriber, we should provide the exact same password

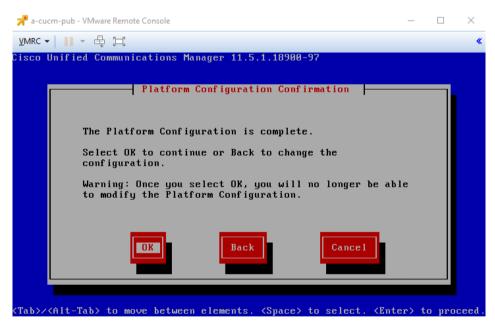


- This is used to send auto email from CUCM nodes
- Usually, we ignore this unless specific requirements are there to configure this

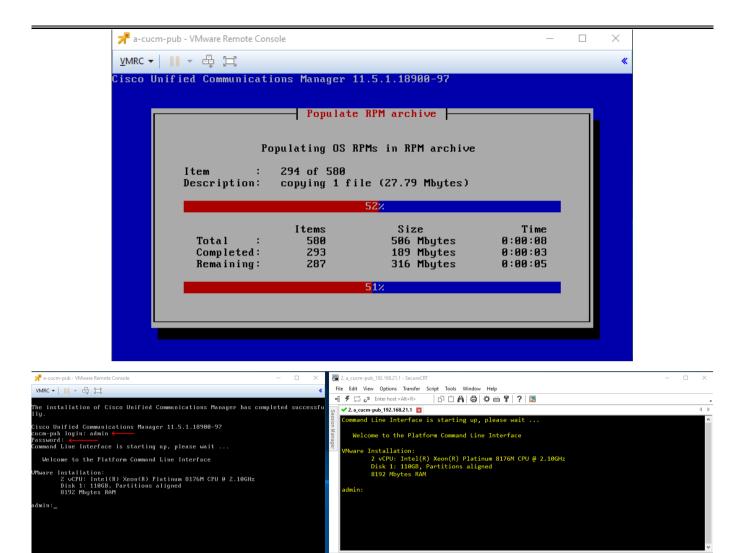




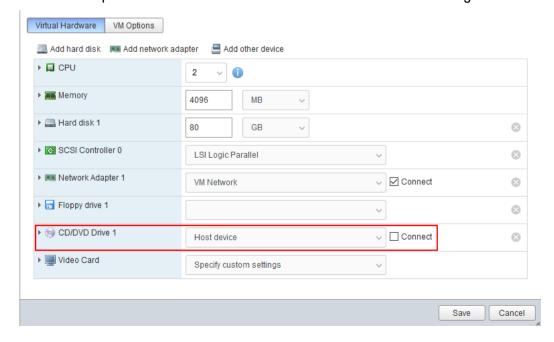
- This is the cluster wide application credentials
- After the installation we use this to login to CUCM Administration GUI
- This step is only in Publisher installation
- This credential is globally replicated hence we can use the same credentials to login to subscriber
 GUI

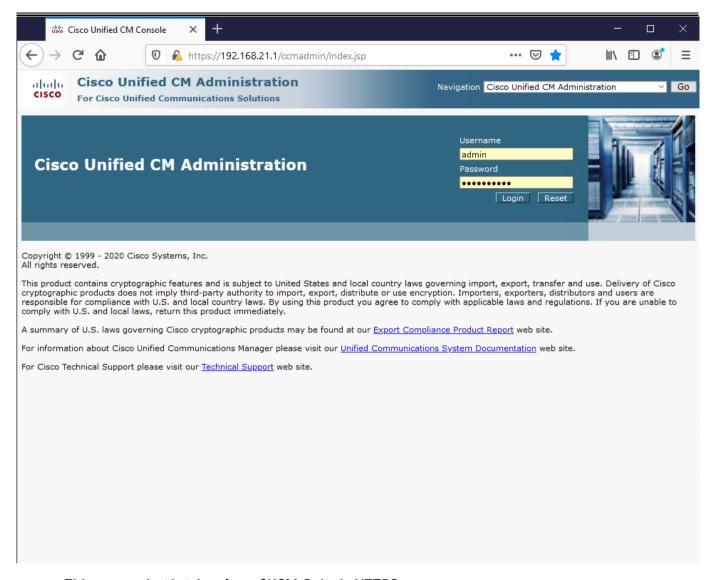


- This is the last stage and once you click 'OK', it will take some considerable amount of time to get the OS installed.
- Once it is completed, you can see CUCM Admin CLI in vmware console
- We use Platform Administration credential to login to the CLI either via vmware console or via SSH



- Right screenshot is taken from CUCM Pub via SSH. Similar interface is available on CUCM Sub as well
- This is the node specific Admin CLI where we use Platform credential to login

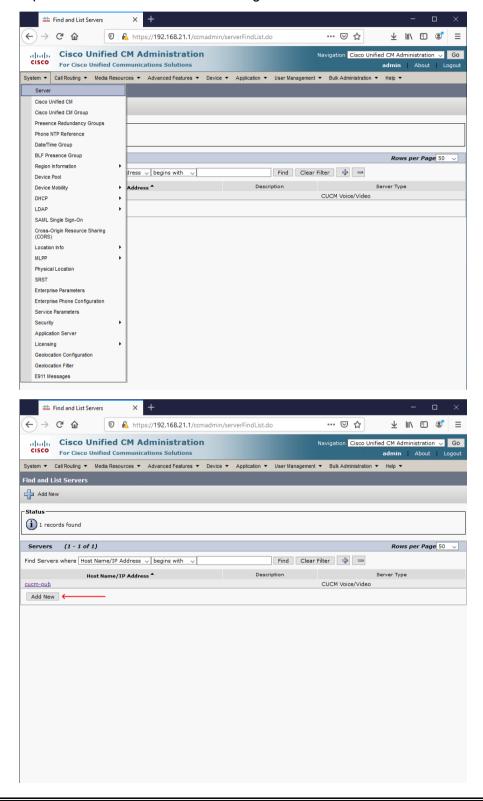


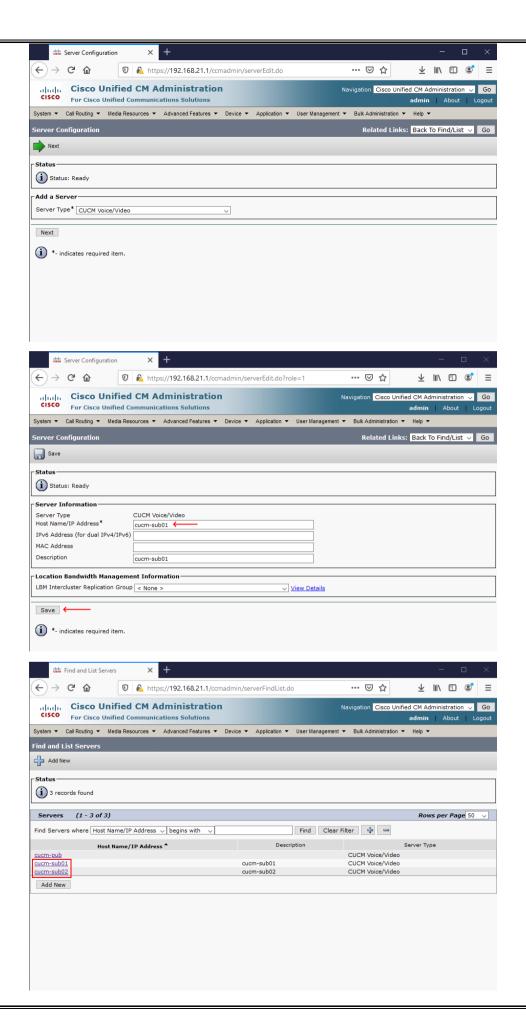


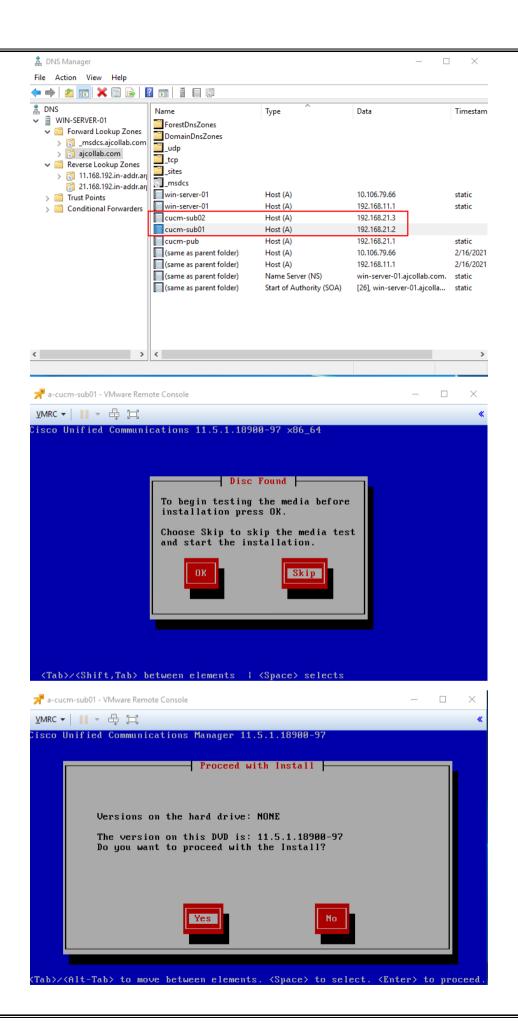
- This screenshot is taken from CUCM Pub via HTTPS
- This will be the main administration platform for CUCM
- CLI is used for some specific purpose and troubleshooting situations

Step 4: Install CUCM Subscriber

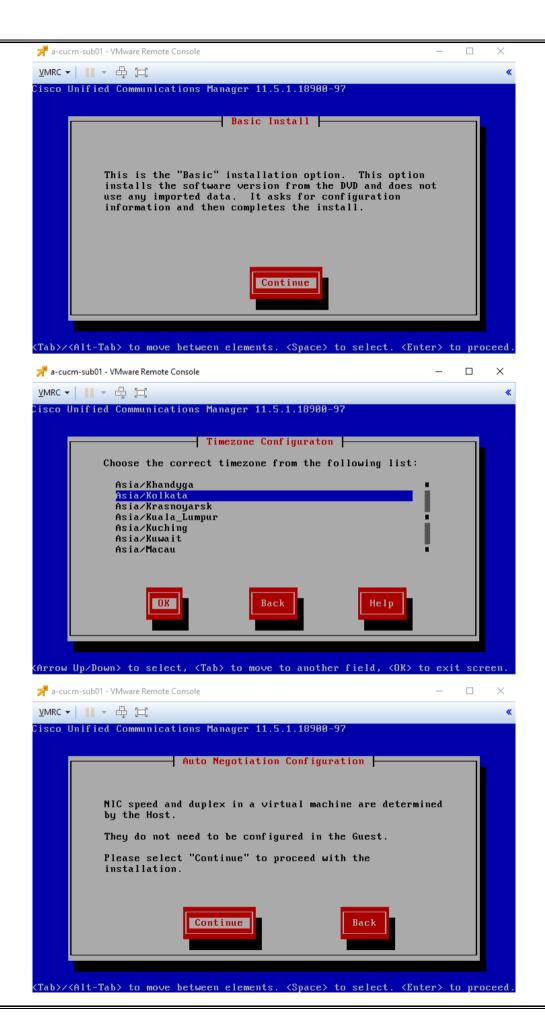
- We use the same ISO and OVA files for installing CUCM Subscriber server
- Deploy the OVA file again and map the CUCM ISO file, then power on the VM
- Most of the steps are like CUCM Publisher installation
- Install CUCM Pub first and then Subscribers, during the installation, Sub will check the connectivity to Pub. Multiple Subscribers can be installed together



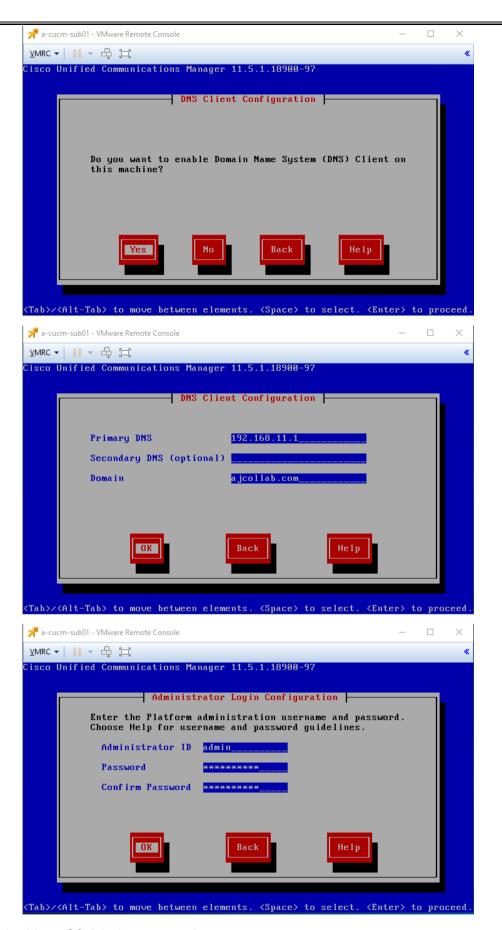




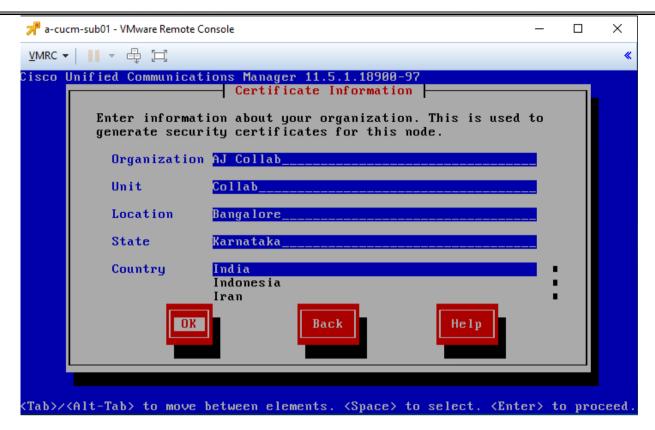




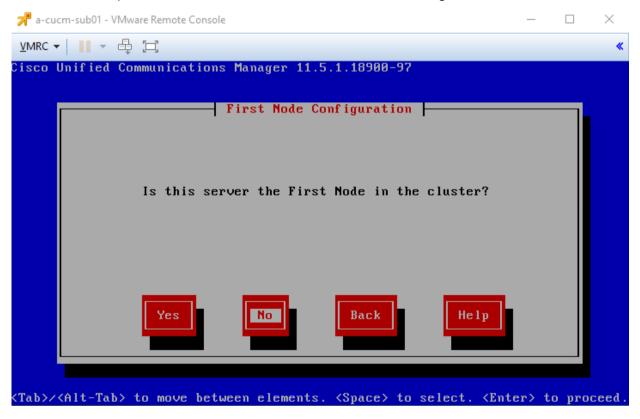




- This is the Linux OS Admin password
- Used to login to CUCM Node CLI while troubleshooting

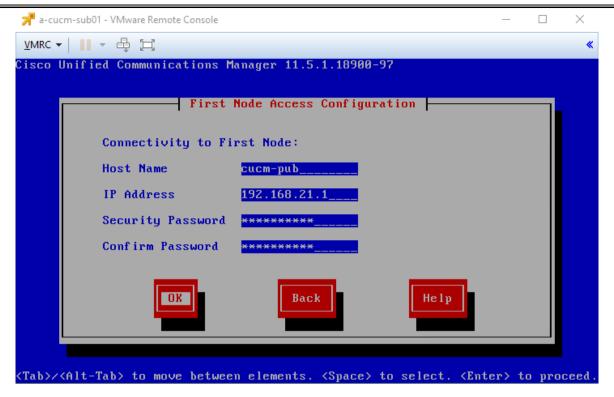


This is node specific, means we need to enter this while installing CUCM Pub and CUCM Sub

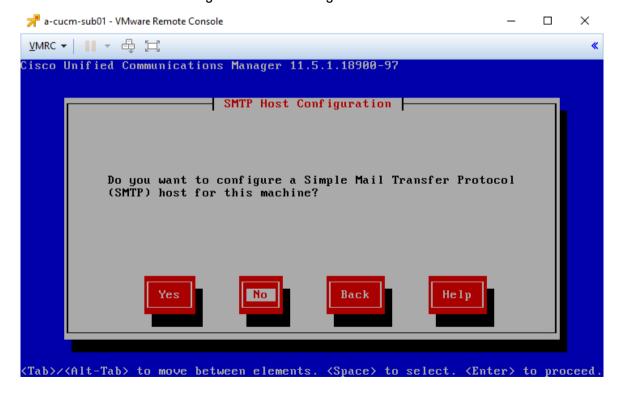


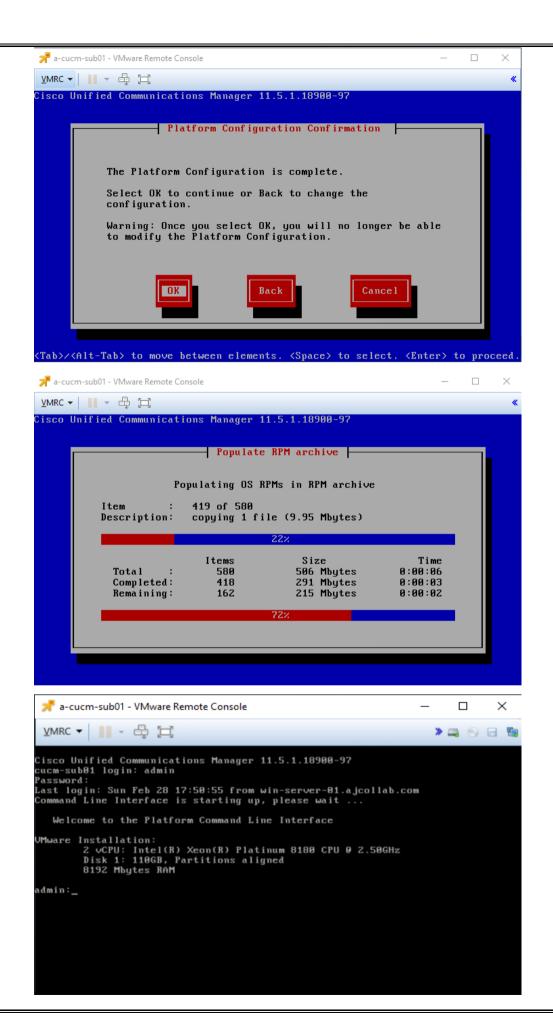
- Till here CUCM PUB and SUB installations are same
- This is the 1st difference in CUCM Sub installation
- We select this is NOT a First node in the cluster

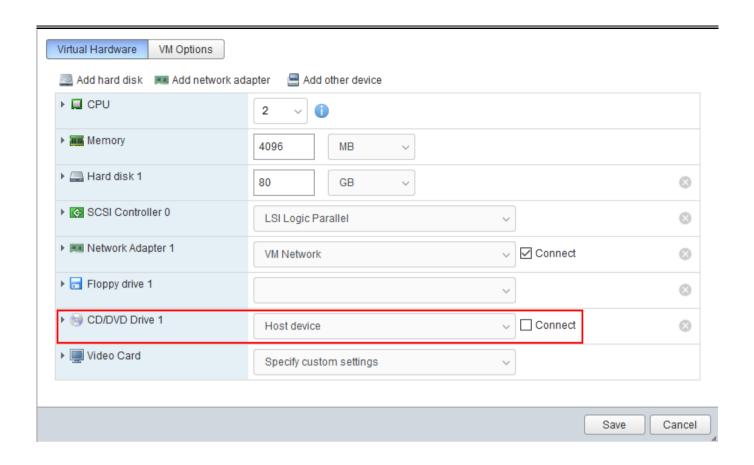




- We must give the CUCM Pub details here
- During the installation, Sub will do network connectivity validation
- CUCM Pub must be ON state
- Make sure you added CUCM-SUB01 in the System >> Server of CUCM-PUB
- We should enter the security password here
- This must match the one we gave while installing CUCM Pub







[Lab] Some Useful CLI Commands

CLI Commands Use

Status Commands

show statusGet the node statusshow myselfGet node detailsshow network eth0Get node IP Details

show network cluster To know other node details run sql select * from processnode To know other node details

utils ntp status Get NTP status, stratum should be <=5 show web-security To know tomcat certificate information

Config Commands

set network ip eth0 192.168.21.1 255.255.128.0 Change IP
set network gateway 192.168.0.1 Change Gateway
set network hostname cucm-pub Change Hostname
set network domain ajcollab.com Change domain name
set network dns primary 192.168.11.1 Change DNS primary server
set network dns secondary 192.168.11.2 Change DNS secondary server

Troubleshooting Commands

utils service listList all service statusutils diagnose testNode diagnostic informationshow process loadTo know the process load

show process using-most cpu
show process using-most memory

To know process consuming more CPU
To know process consuming more RAM

Operational Commands

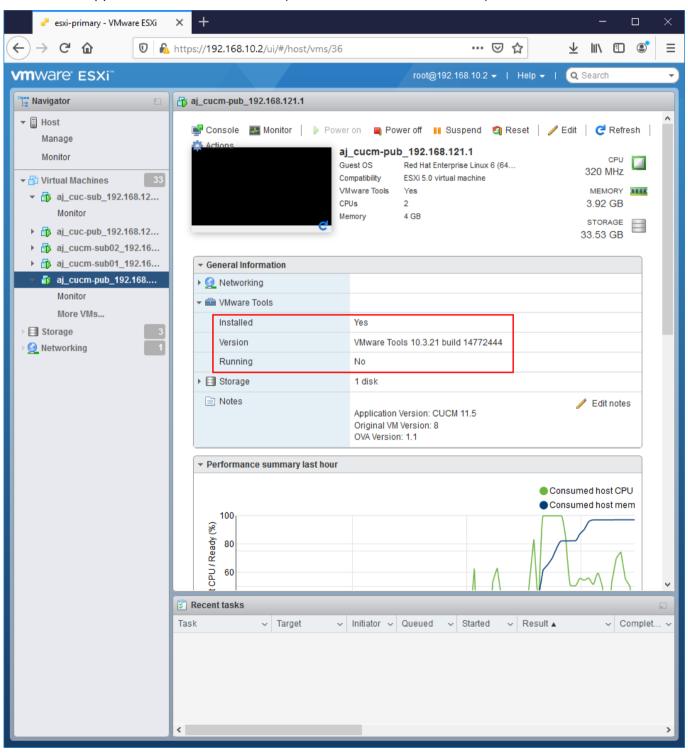
utils system shutdown

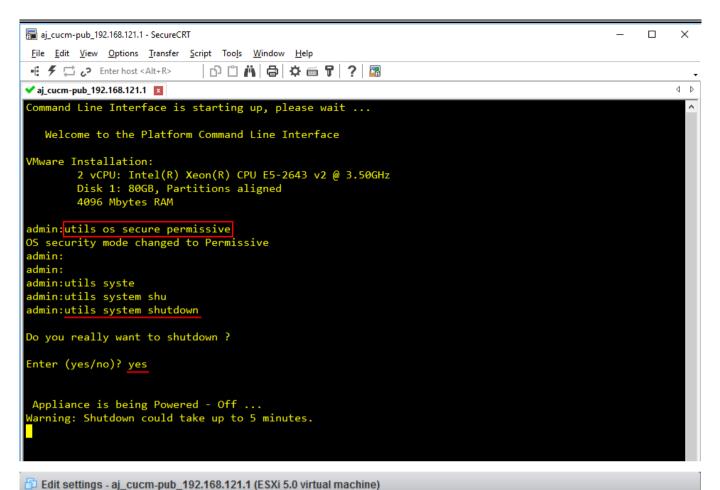
utilsservice restart Cisco TomcatRestart a Service (e.g. Cisco Tomcat)utilssystem restartNode restart

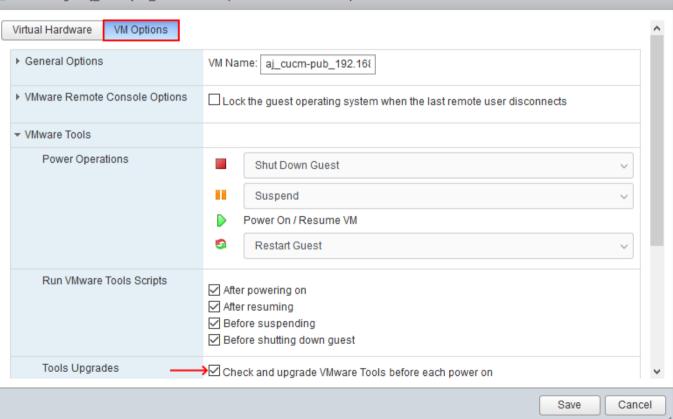
Node shutdown

Troubleshooting vmware Tools on UC Servers [Lab Workaround]

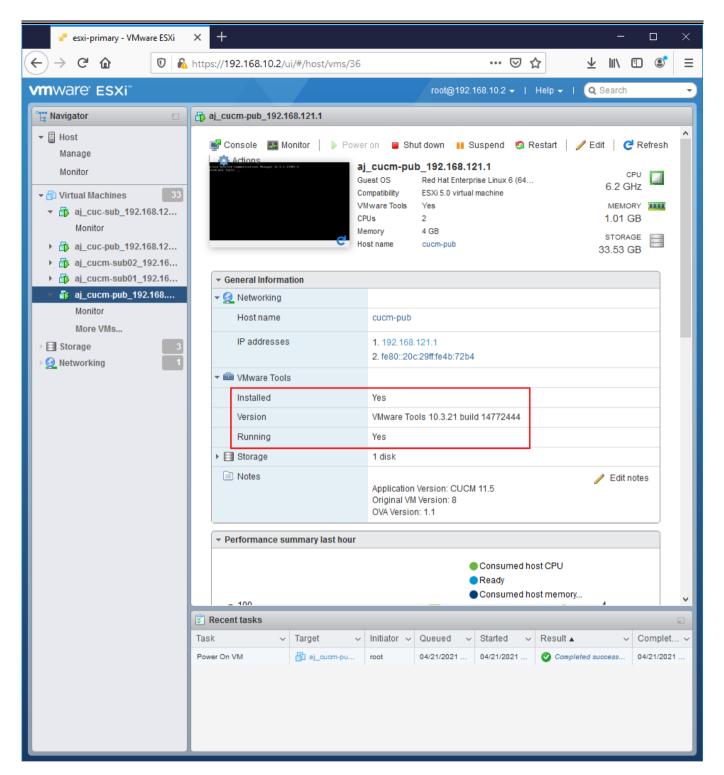
- VMware Tools is a suite of utilities that enhances the performance of the virtual machines guest operating system and improves management of the virtual machine
- vmware Tools provides better display resolution, fetches the host name, IP Address, MAC Address of the guest OS
- Full support for ESXi action buttons (Shutdown Guest OS from ESXi)



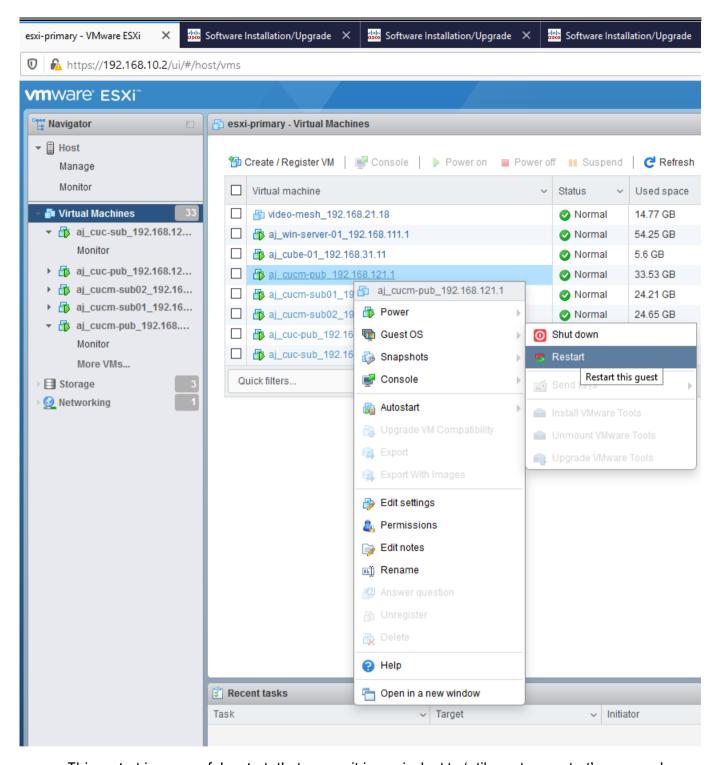




Now power on the virtual machine



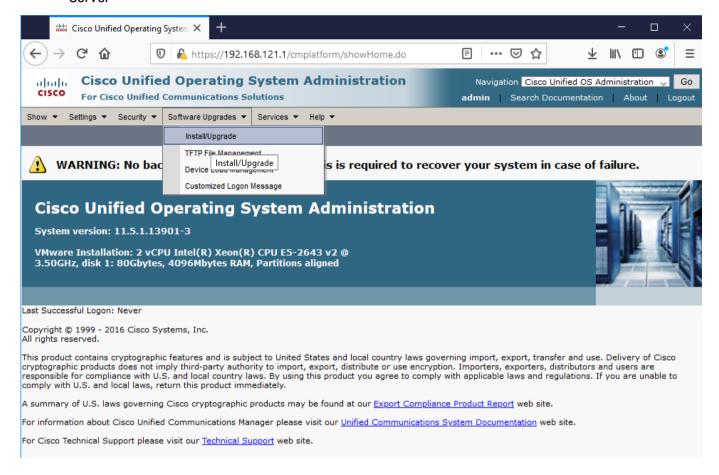
- It is not recommended to disable SE Linux in production cluster, hence this can be applied in Lab servers
- There are some known issues (CSCvq17528, CSCvh55176, CSCvm52977, and CSCvb21486) for CUCM 11.5 vmware Tools, hence for production servers, kindly install the COP File 'ciscocm.CSCvq17528_vmtools_initramfs_v1.3.cop'

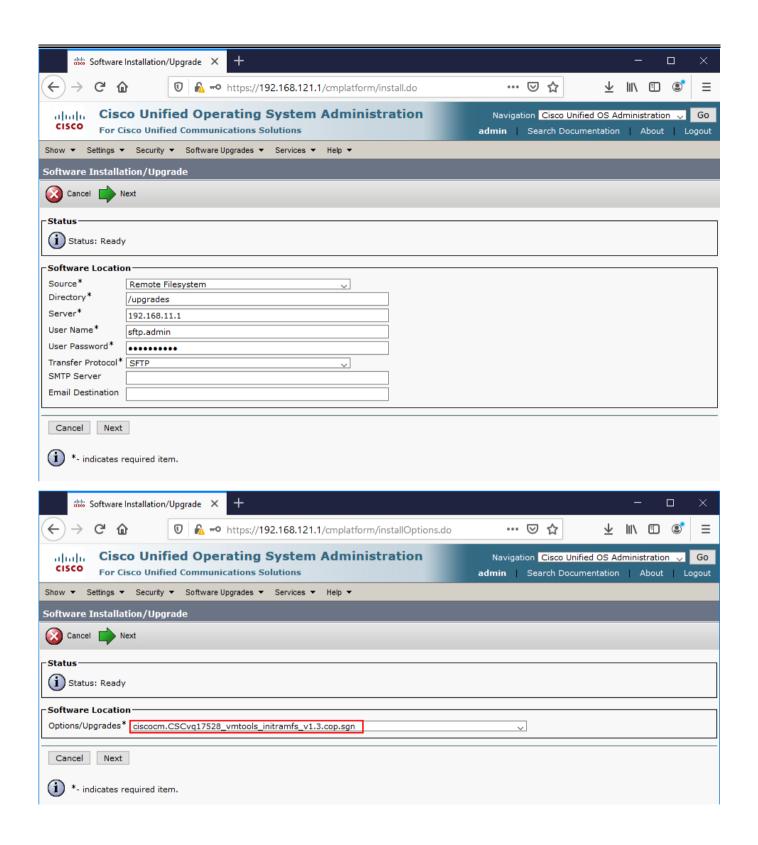


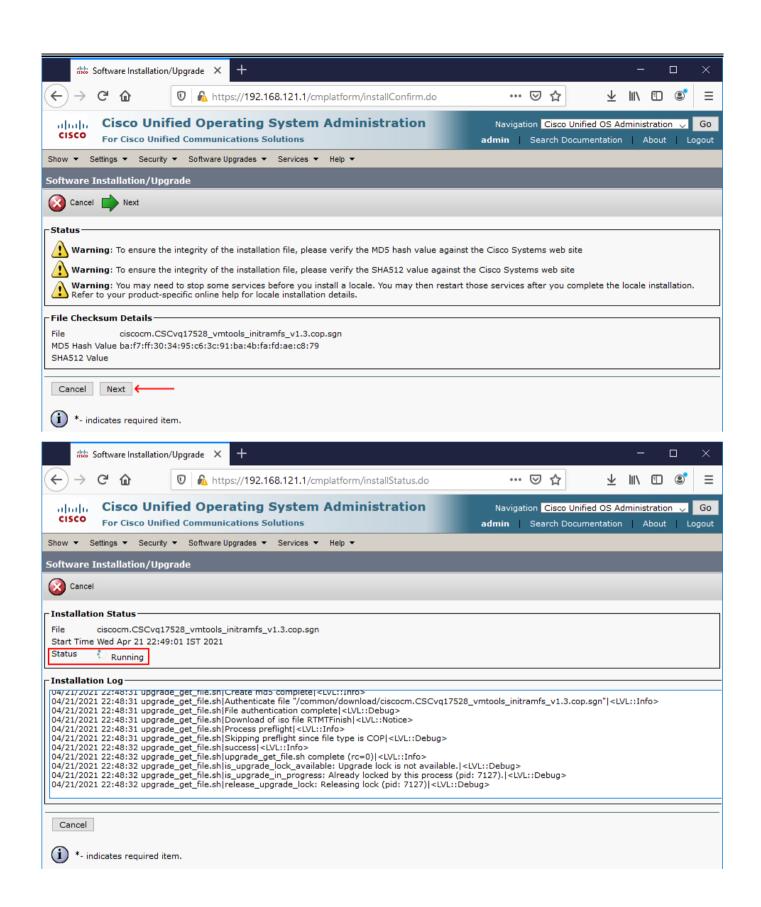
· This restart is a graceful restart, that means it is equivalent to 'utils system restart' command

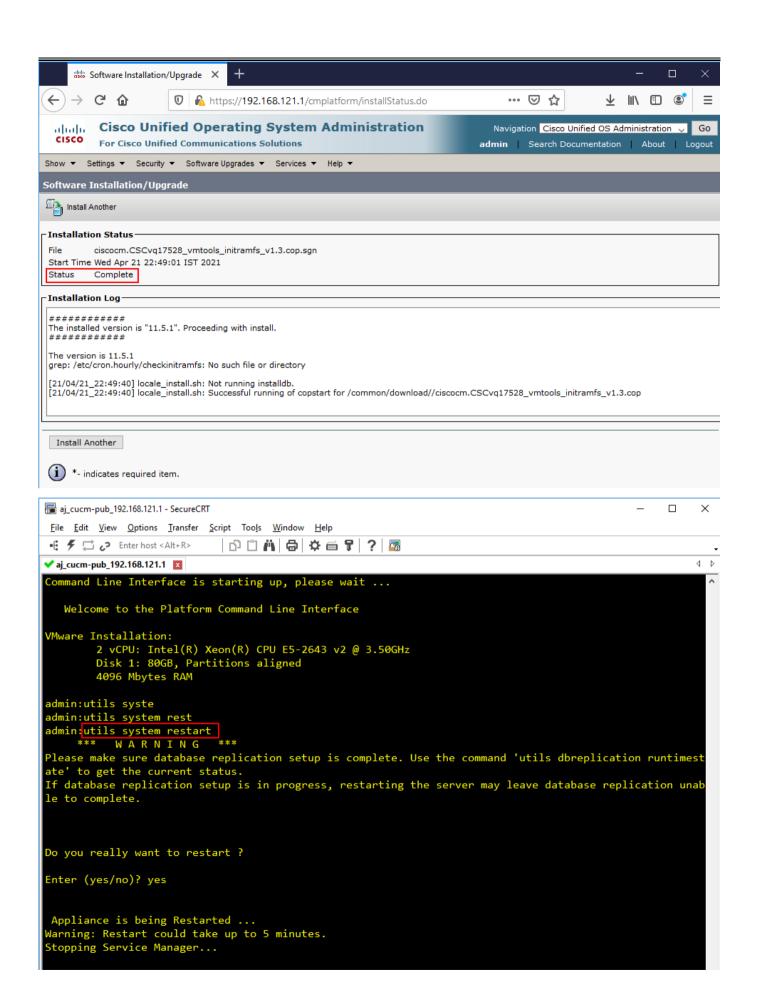
Troubleshooting vmware Tools on UC Servers [Production Workaround]

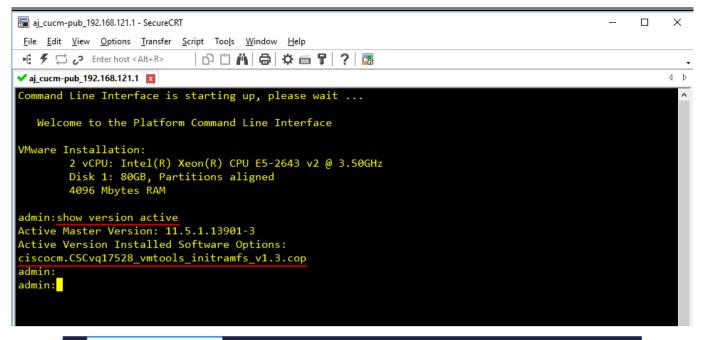
 Download the COP File ciscocm.CSCvq17528_vmtools_initramfs_v1.3.cop.sgn and place it in SFTP Server

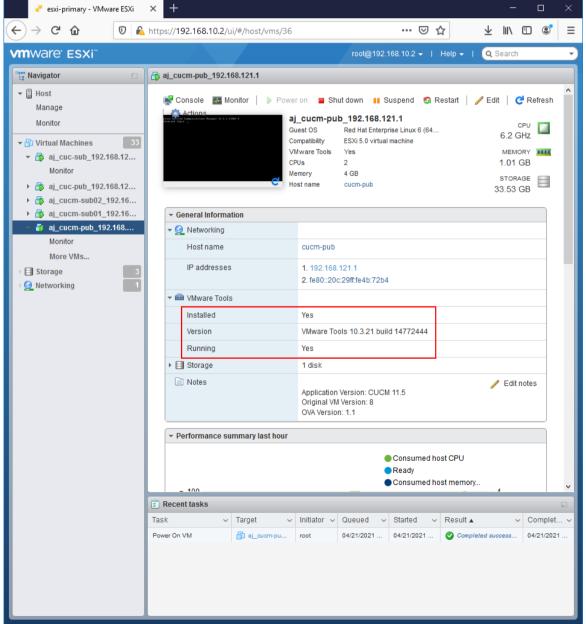


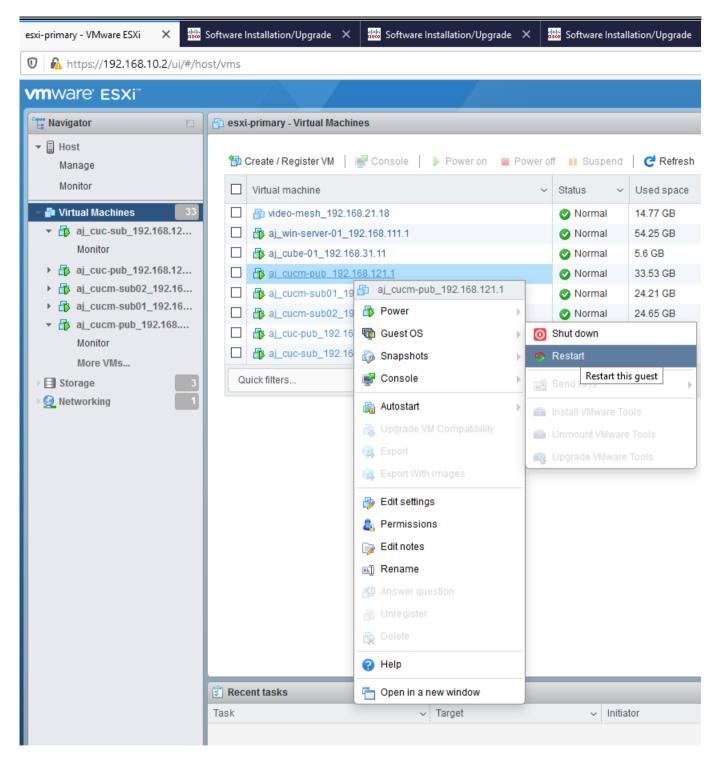








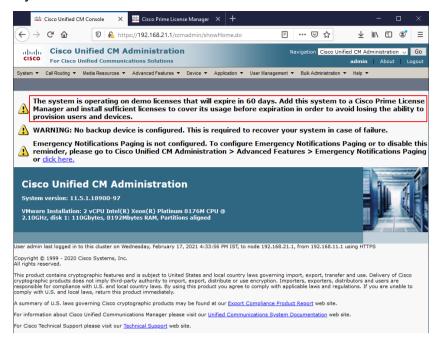




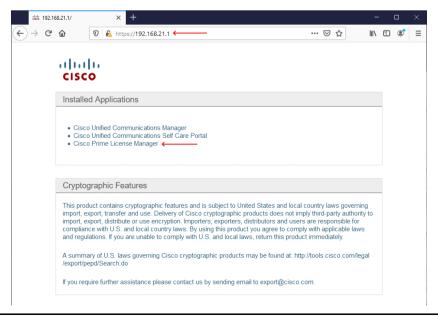
• This restart is a graceful restart, that means it is equivalent to 'utils system restart' command

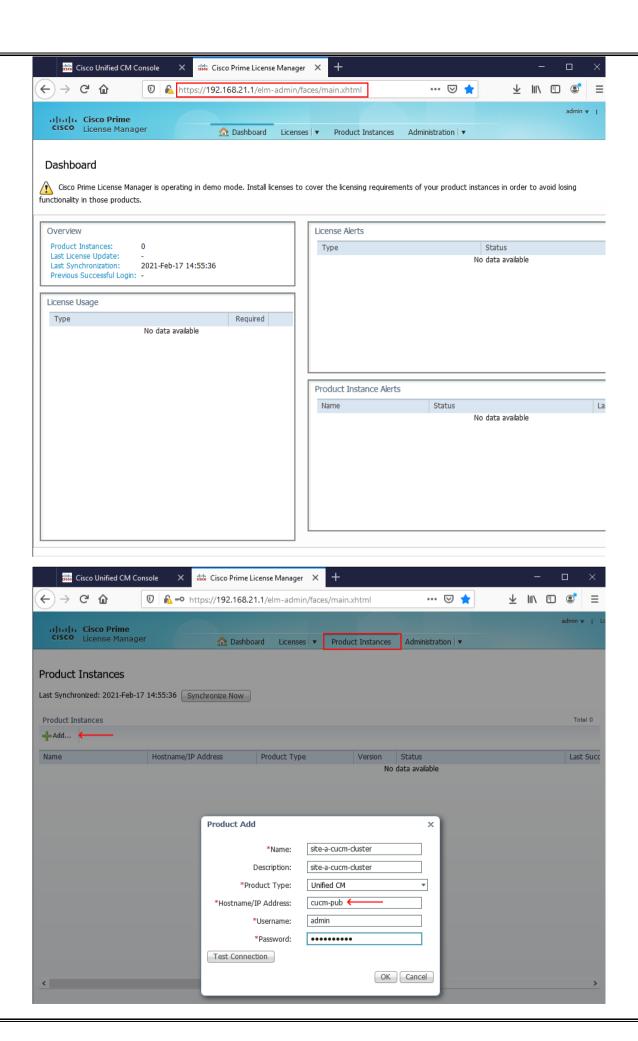
[Lab] CUCM Licensing

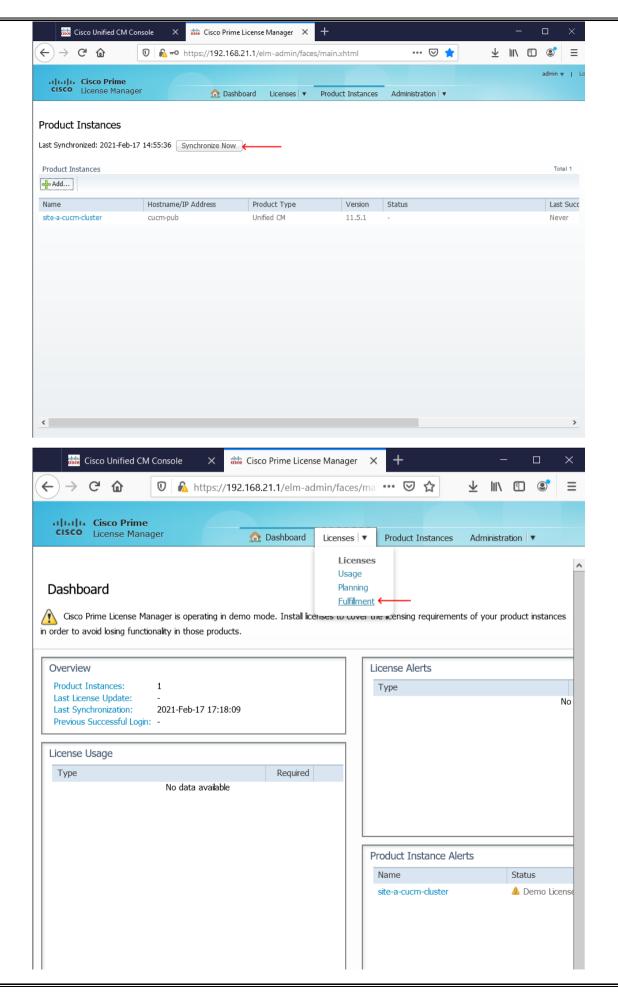
 Right after installing CUCM Cluster, we can see "The system is operating on demo license that will expire in 60 days"

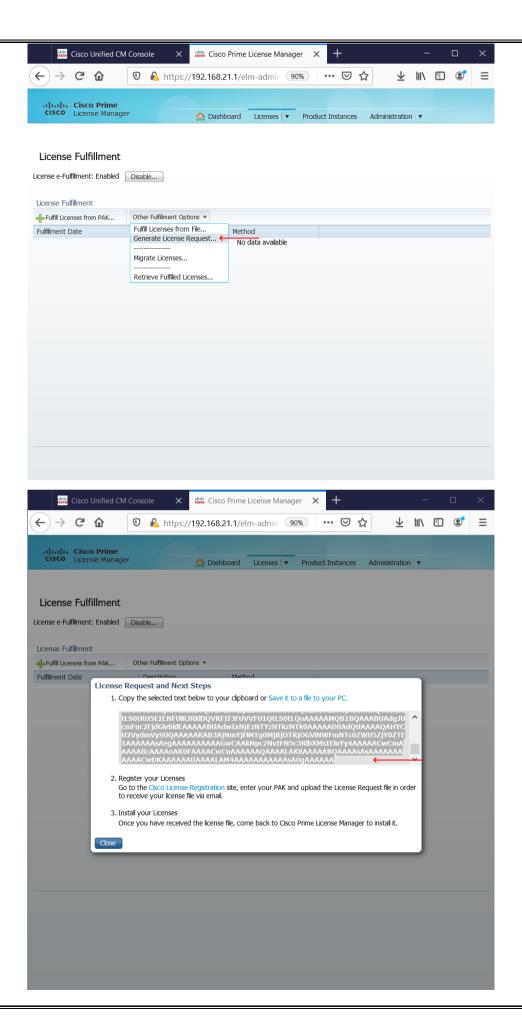


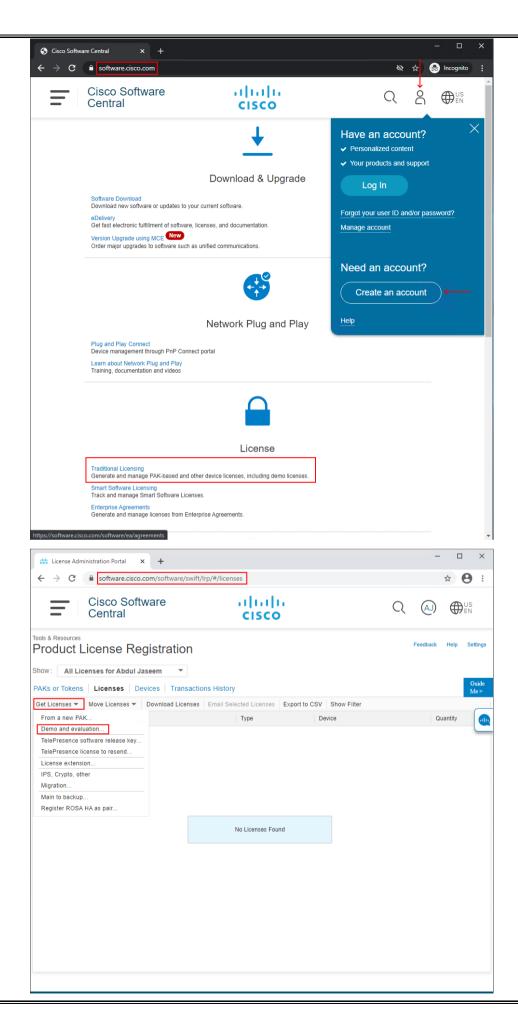
- We must configure the licensing for UC cluster to get rid of from this warning as well as to get licensed features
- The license is managed by Cisco Prime License Manager which is co-installed with CUCM
- You can also maintain separate node for Cisco Prime License Manager
- CUCM Cluster is synchronized with prime Cisco Prime License Manager
- In our scenario, I'm using co-installed Cisco Prime License Manager on CUCM PUB, that means I
 browse https://cucm-pub/elm to access Cisco Prime License Manager. Platform credentials are
 used to login to PLM

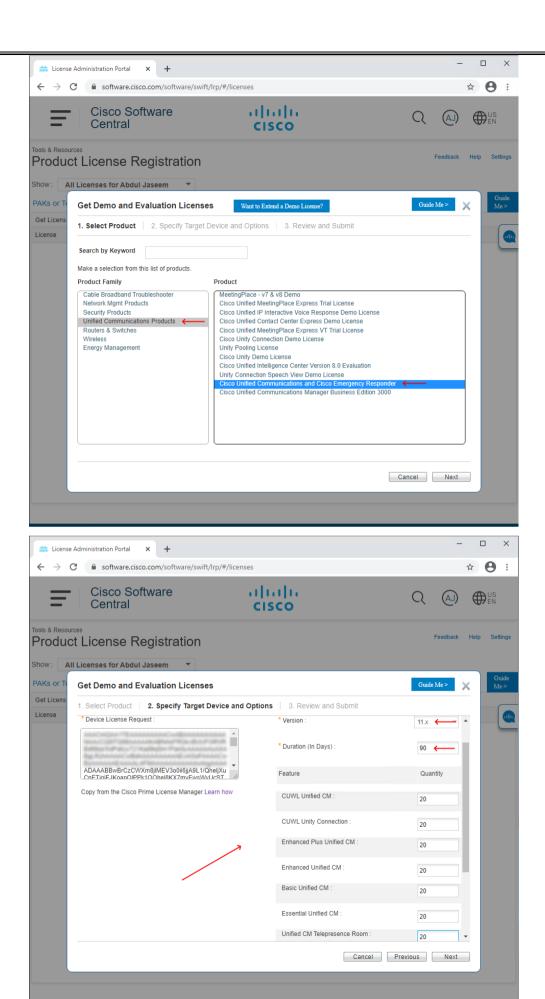


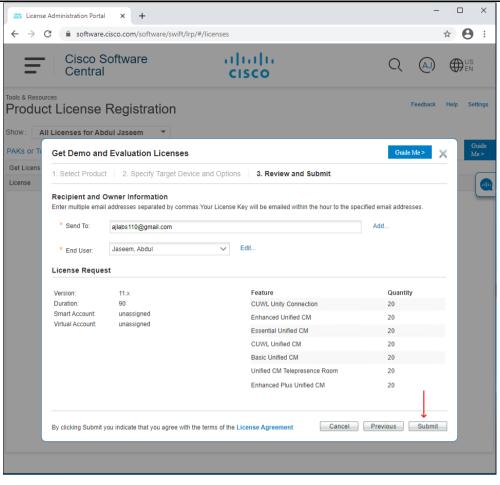


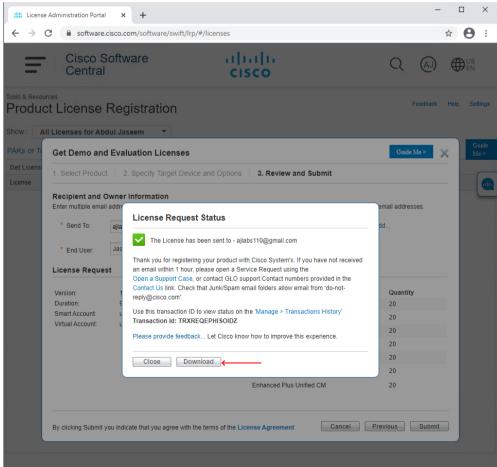


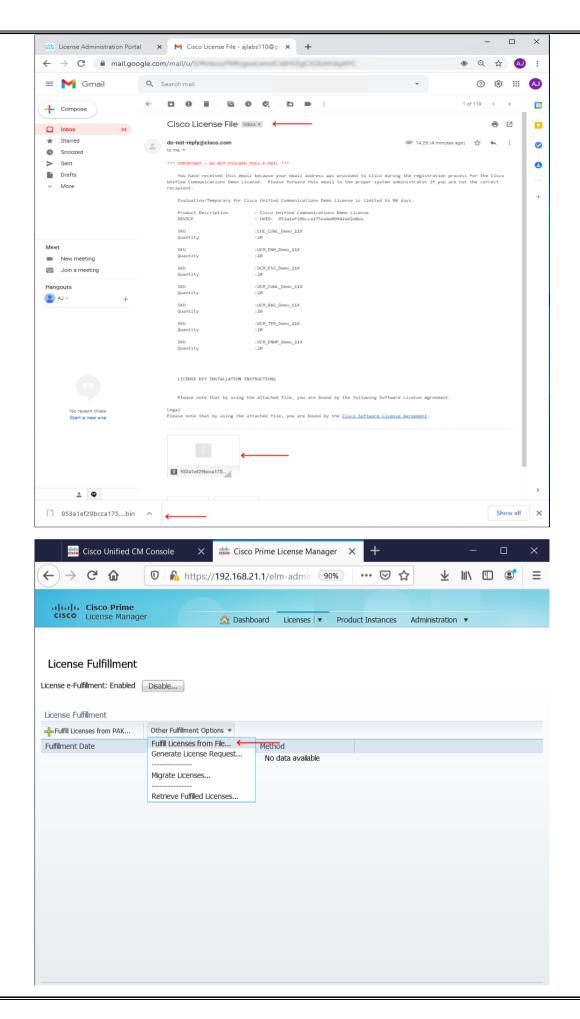


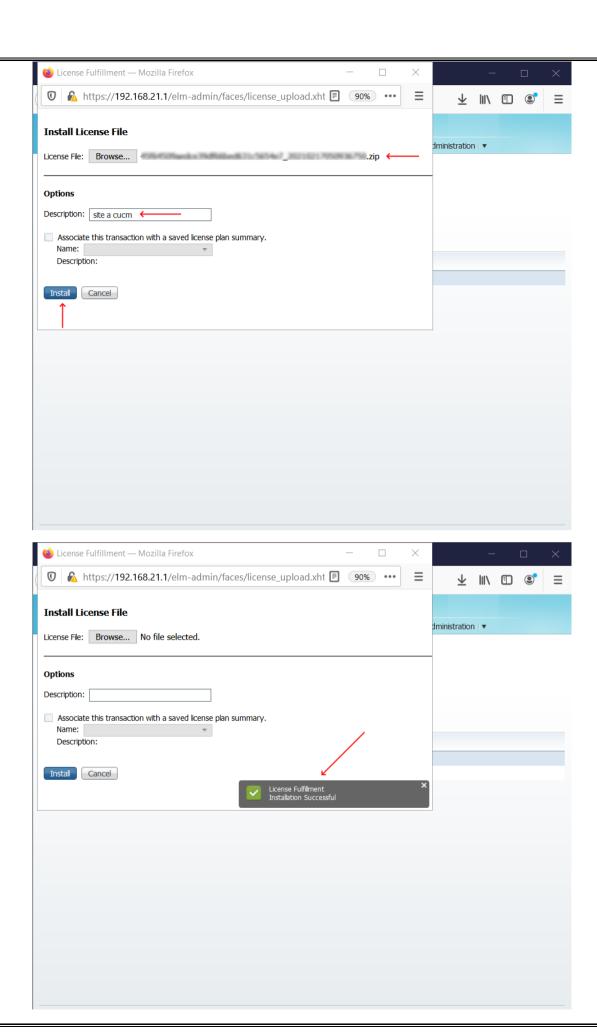


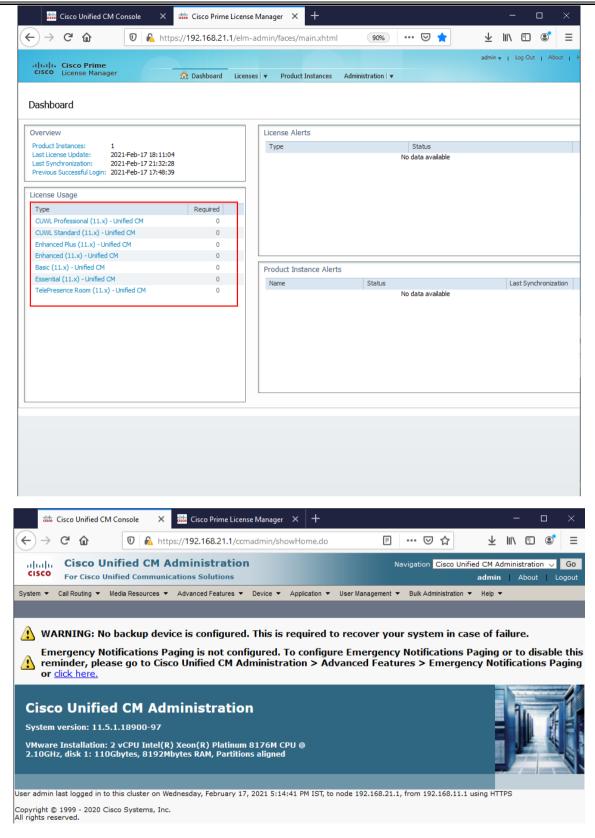








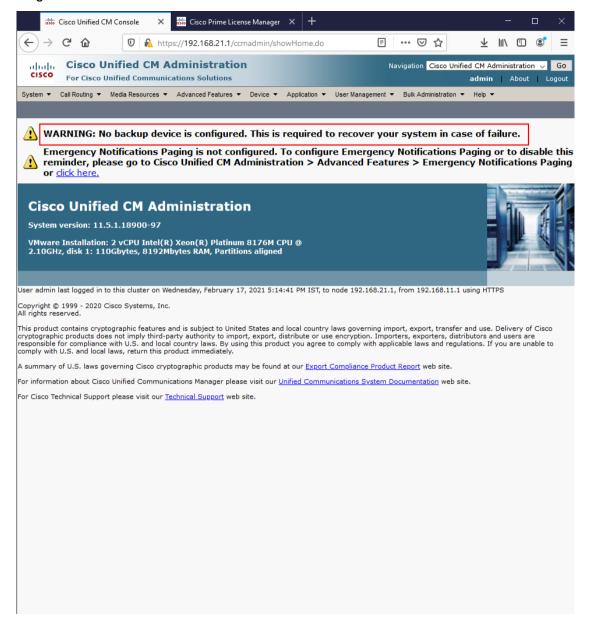




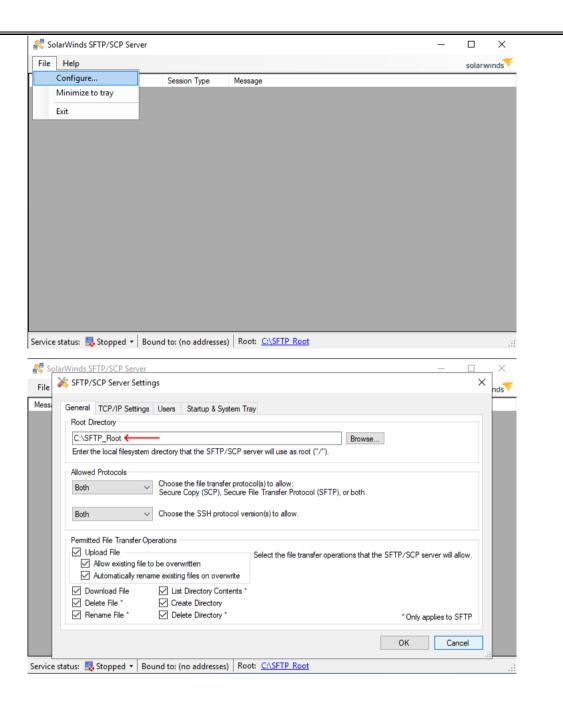
- Details about different license type and license consumption will be covered later in this document (License Consumption section)
- We need some endpoints to be registered to understand the license usage

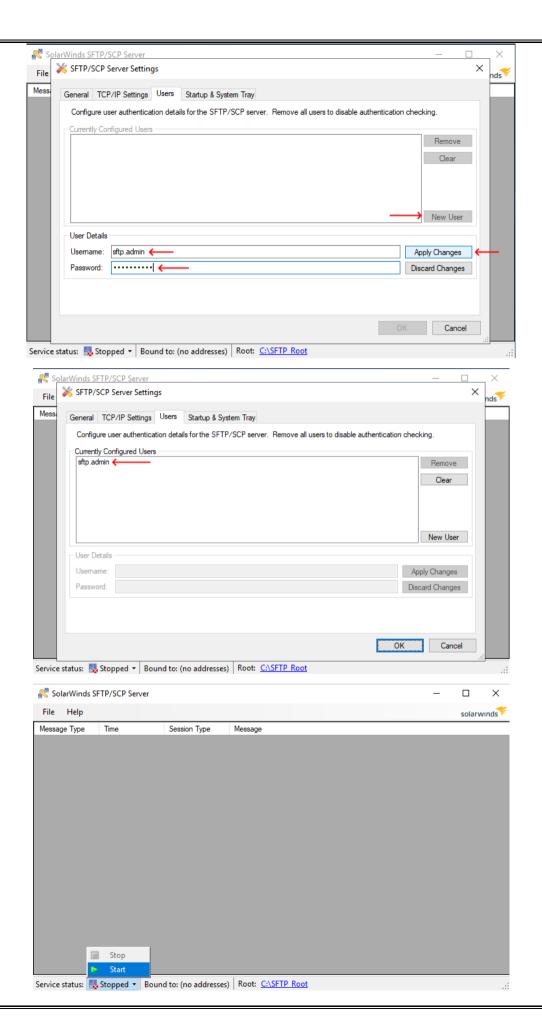
[Lab] Disaster Recovery System (DRS) Backup of CUCM Cluster

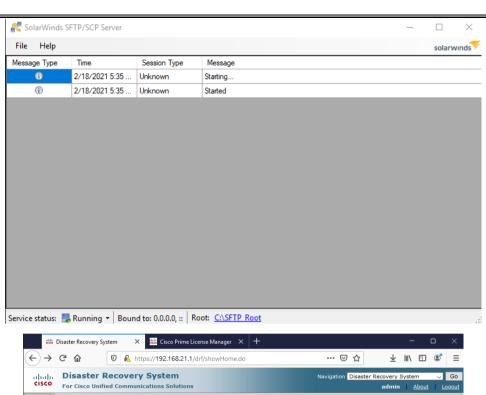
 After the installation of CUCM cluster, another warning you have seen is "No Backup device is configured"

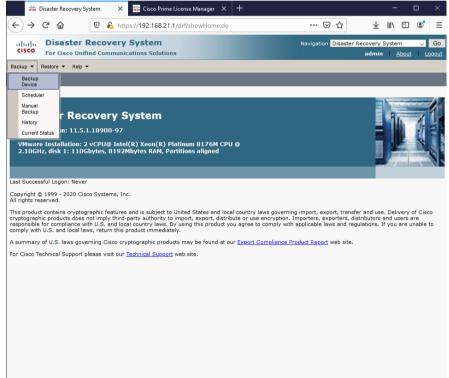


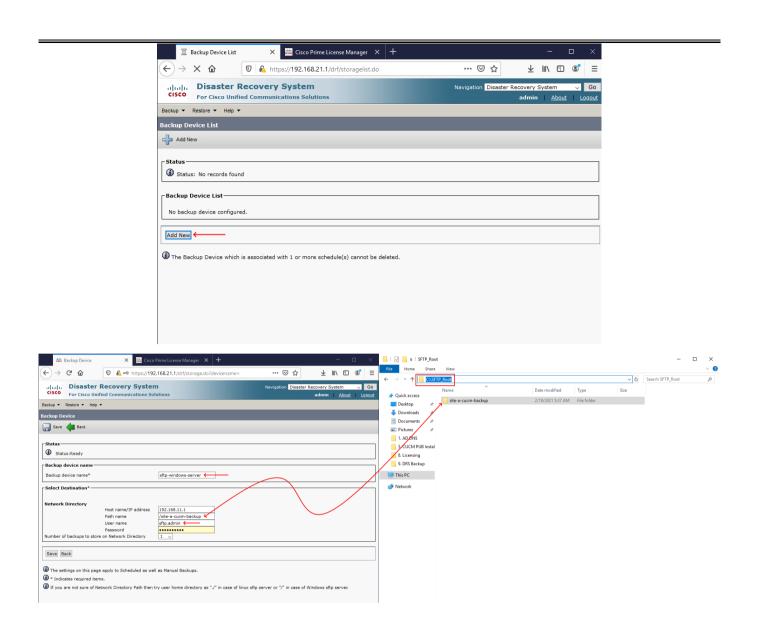
- DRS or Disaster Recovery System provides full data backup and restore capabilities for all servers in the cluster
- The DRS allows you to perform regularly scheduled automatic or user-invoked data backups
- The Disaster Recovery System contains two key functions, Master Agent (MA) and Local Agent (LA). The Master Agent coordinates backup and restore activity with Local Agents
- Disaster Recovery Framework (DRF) Master runs on CUCM-PUB and DRF Local runs on all the Subscribers
- Hence CUCM Publisher should be alive to perform backup

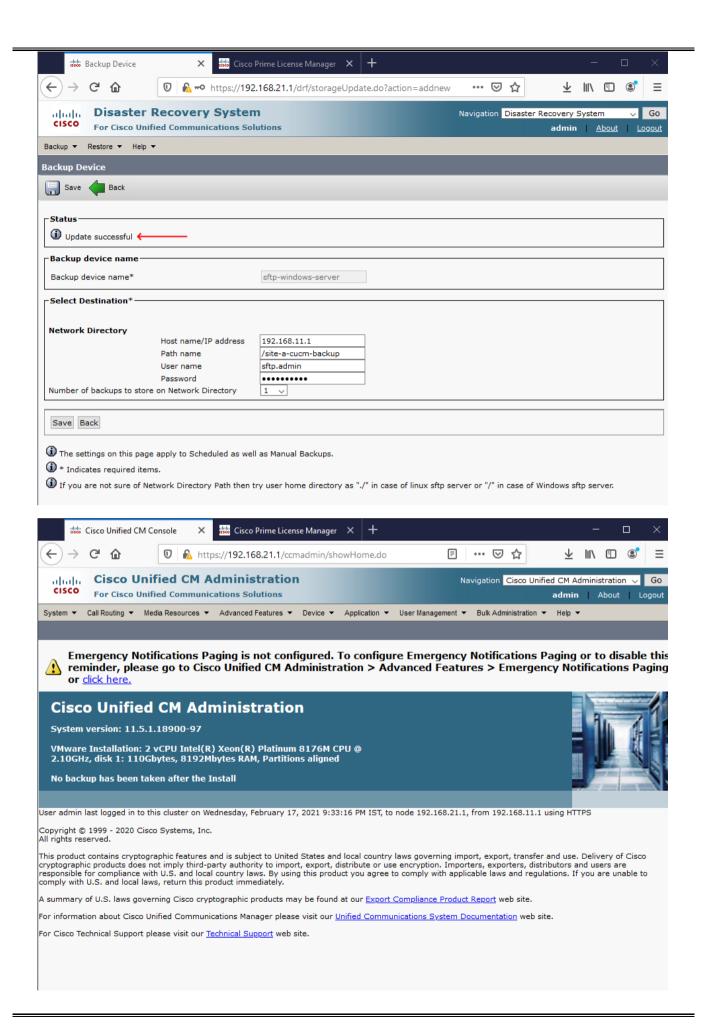






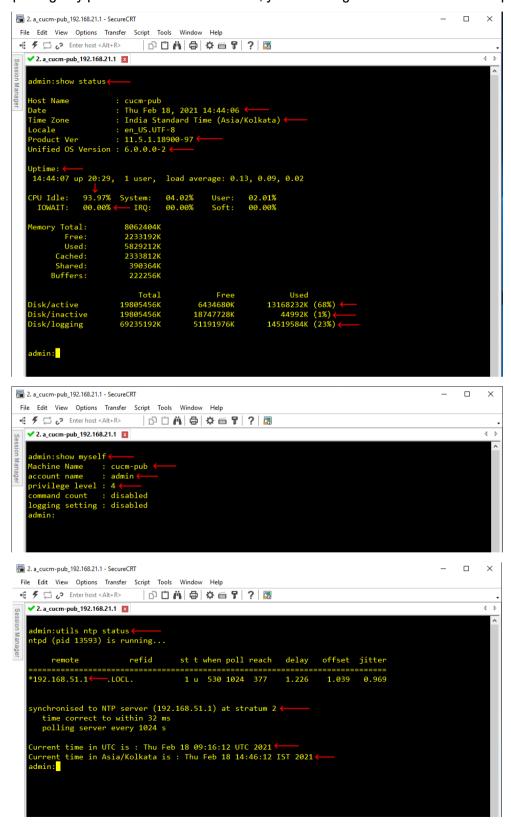


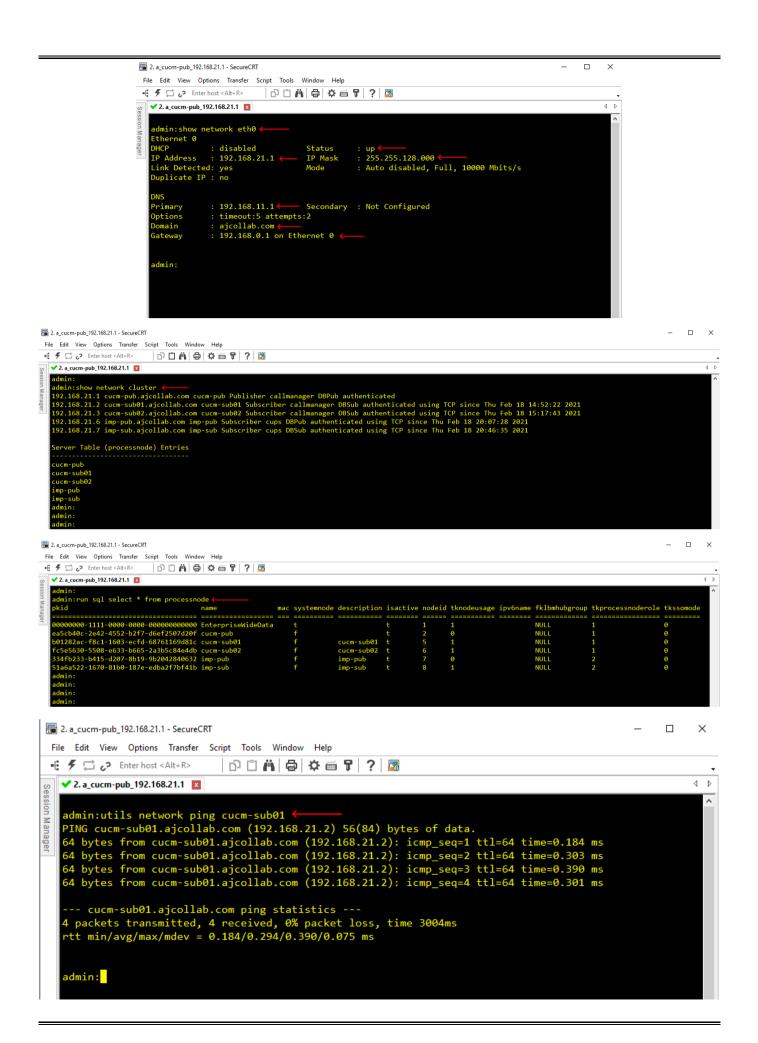


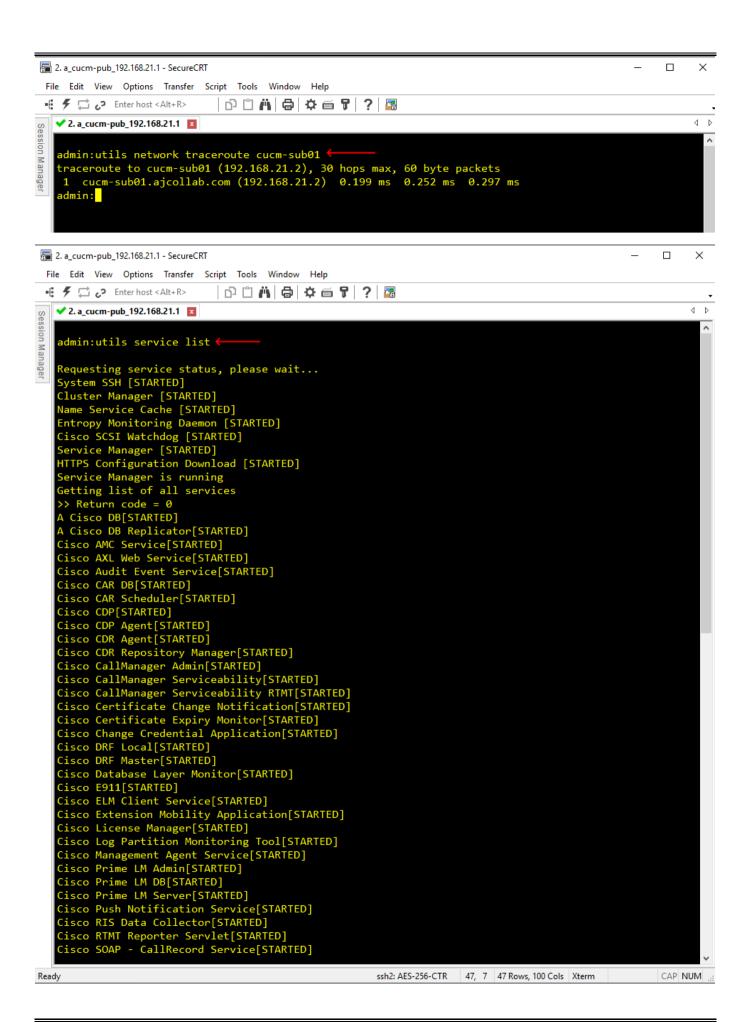


[Lab] Basic Health Check of UC Infrastructure & Understand the Cluster

- When you get started with a new job role, it is important to understand your UC infrastructure. I have given some CLI commands to help knowing the cluster
- As a regular health check, some engineers performs this every day when they start working
- While opening any platform related TAC cases, you should give these command outputs







```
File Edit View Options Transfer Script Tools Window Help
    •€ # □ • Enter host <Alt+R>
         dmin:show web-security←
            Version: V3
Serial Number: 511D2EC283C99AEAC43885A88884D4E8
SignatureAlgorithm: 5HA256withRSA (1.2.840.113549.1.1.11)
ISSUER Name: L=Bangalore, ST=Karnataka, CN=cucm-pub.ajcollab.com, OU=Collab, O=AJ Collab, C=IN
Validity From: Wed Feb 17 14:32:19 IST 2021
Subject Name: L=Bangalore, ST=Karnataka, CN=cucm-pub.ajcollab.com, OU=Collab, O=AJ Collab, C=IN
Key: RSA (1.2.840.113549.1.1.1)
Key value: 3082018a028201000be940f450c2bc0dae6e073f8644b9744de049bbea5e9a3d10f8eb9999f70fb5069
393ec336274402937bf26af00a347e4808831d565dae0cd225cb151ad7f553abba5325076eebcf5e097f88f001c39e65511
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1893468829b7717beefe55141bd6b7733782c6d06efb0203010001
                  Extension: KeyUsage (OID.2.5.29.15)
                  Critical: false
Usages: digitalSignature, keyEncipherment, dataEncipherment, keyCertSign,
                  Extension: ExtKeyUsageSyntax (OID.2.5.29.37)
                  Critical: false
Usage oids: 1.3.6.1.5.5.7.3.1, 1.3.6.1.5.5.7.3.2,
                 Extension: SubjectKeyIdentifier (0ID.2.5.29.14)
Critical: false
keyID: 2592e7a6a54770ff7cb00867d5c9a86c67b830cc
                 Extension: BasicConstraints (OID.2.5.29.19) Critical: true cA: true pathLenConstraint: \theta
               gnature:

900: 36 e5 49 42 01 ef 37 d3 9f 42 63 8a 95 b4 17 d1 [6.IB..7..Bc....

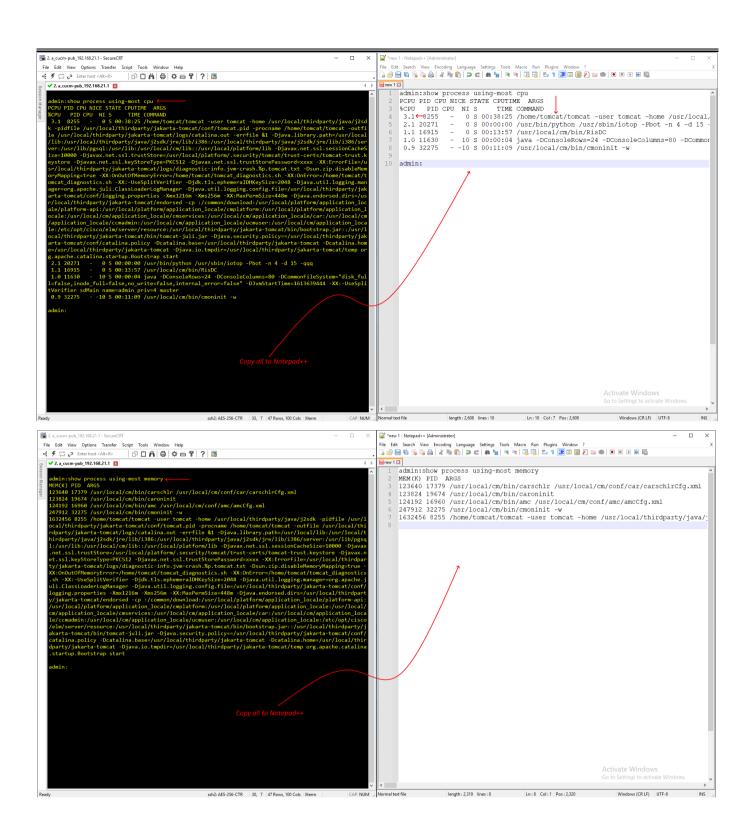
110: f5 d6 56 c1 6f 4d 75 1d 6f 6c 7d a3 ab 59 03 f3 [..v.oMu.ol]....

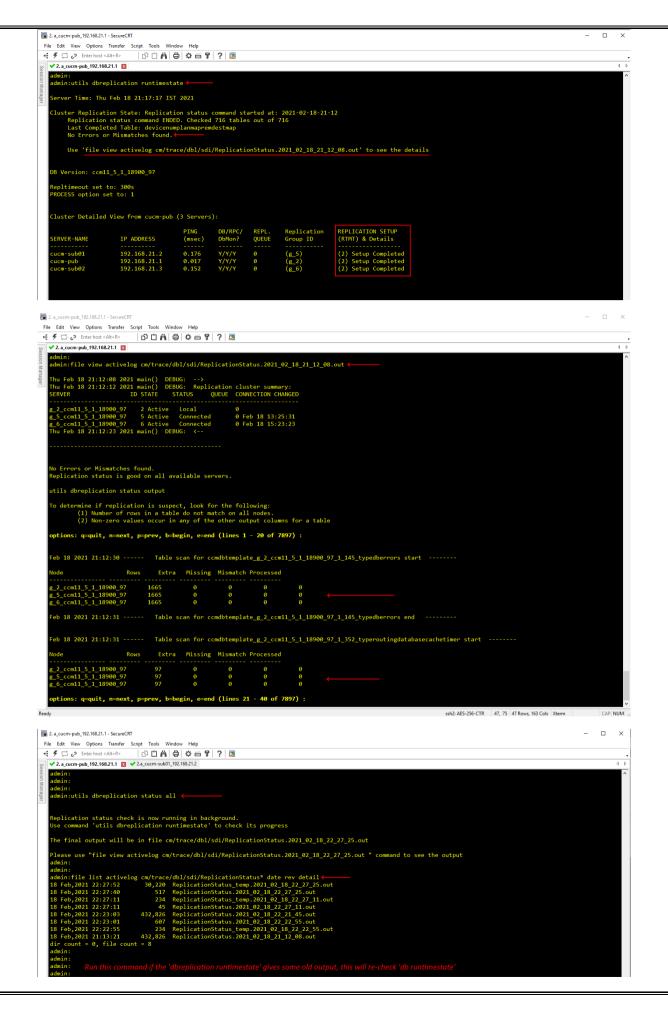
120: b1 b4 3d 2c 8e 8a 9a fb bf 42 8b 88 27 7d 0b 8f [...,...B...]

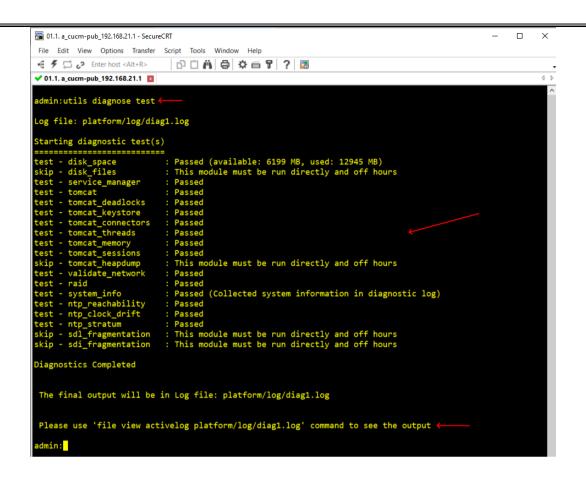
130: 9a ee 03 85 06 4c 9b 22 51 dd 06 ca 1d db 0c 9f [...,..."0....

140: 1b 9d 77 ab d2 f2 3b 85 6a d3 e1 d7 69 c8 f3 0a [...,..;j..i...]

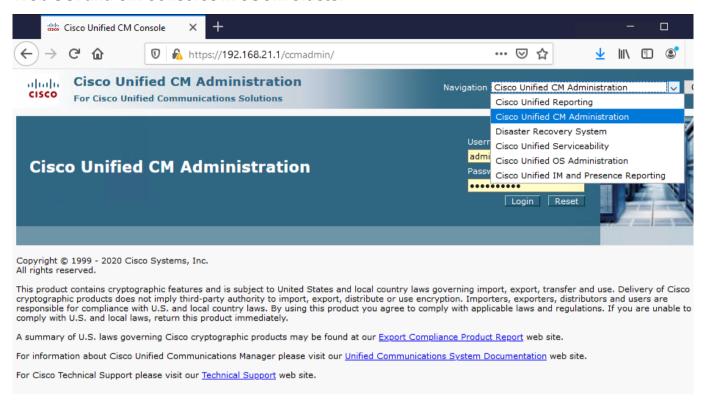
150: d8 45 ad 45 21 b2 8d be 06 c6 78 44 4f fe d6 f8 [.E.E!...x00...
                                                                                                                       a_cucm-pub_192.168.21.1 - SecureCRT
  File Edit View Options Transfer Script Tools Window Help
                                                          | D C A | D | O = 7 | ? | 3
         dmin:utils network connectivity cucm-sub01←
        This command can take up to 3 minutes to complete. Continue (y/n)?y \longleftarrowRunning test, please wait ...
                 ork connectivity test with cucm-sub01 completed successfully. \longleftarrow
            min:
min:utils network connectivity cucm-sub02 —
           his command can take up to 3 minutes to complete. Ontinue (y/n)?y ——
unning test, please wait ...
            twork connectivity test with cucm-sub02 completed successfully. \leftarrow
                                                                                                                                                                                                                  - □ ×
a 2. a cucm-pub 192.168.21.1 - SecureCRT
 | D B A | B | $ € ₹ | ? | 8
      ✓ 2. a cucm-pub 192.168.21.1 🛛
```



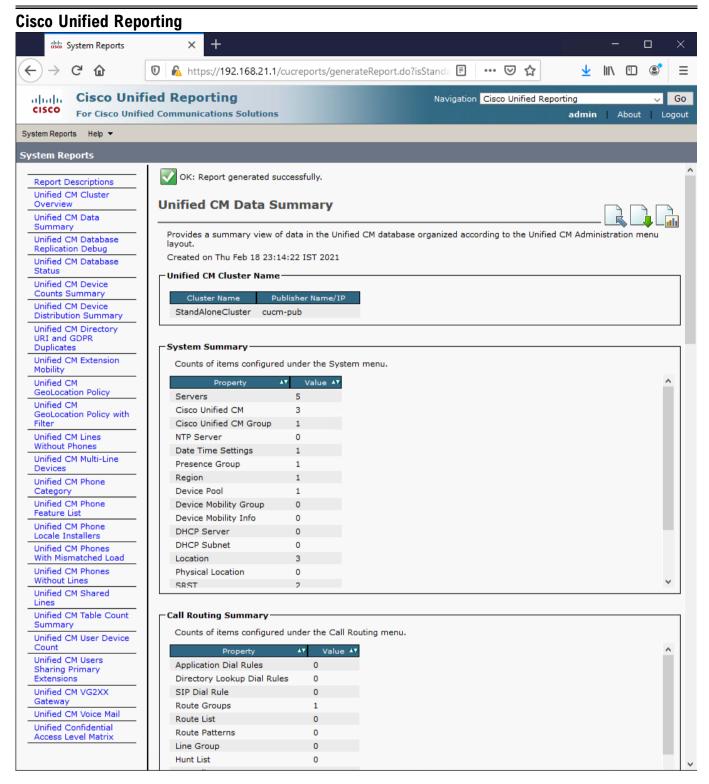




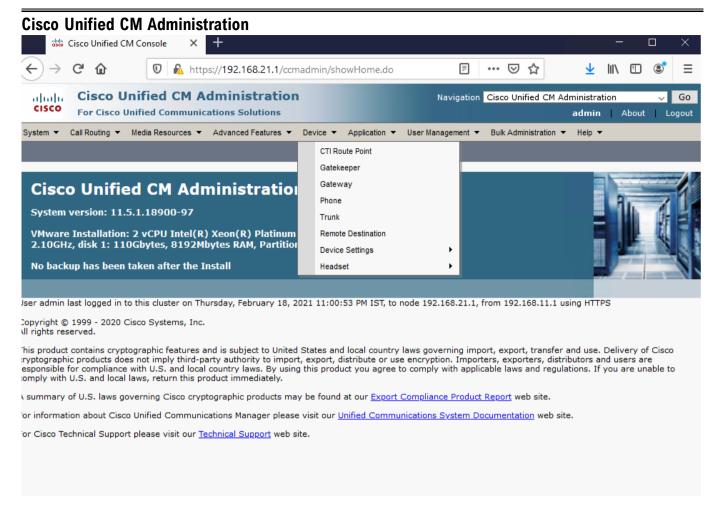
Web GUI and Six Consoles in CUCM Cluster



- There are 6 different web consoles available in CUCM to administrate
- We usually browse the IP Address or FQDN of CUCM PUB to get access to the web interface
- Cisco Tomcat Service is responsible for delivering the web interface over HTTPS
- All the CUCM nodes will have the web interface but all of them connects to the DB of CUCM PUB if PUB is running
- The only time the CUCM SUB talks to its own DB via HTTPS web GUI, when the CUCM PUB is down
- During such scenario, we will not be able to change any configuration on the cluster, we just can see the things (read only mode) since CUCM PUB is the read write copy of DB



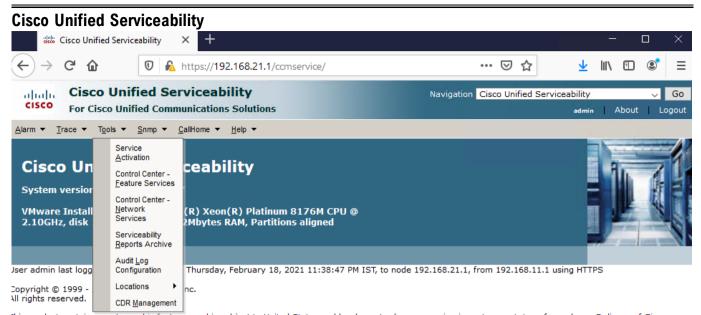
- Gives the details and reports about the cluster like Unified CM Cluster Overview, Database Status,
 Device Counts Summary and much more
- This interface can be used for auditing purposes
- While troubleshooting, we may consider looking at Reporting interface
- We use Application Credentials to login to Cisco Unified Reporting interface



- This is the heart of CUCM Administration
- Core day to day configurations like Adding Phone, Deleting Phones, Call Routing Configurations,
 etc. done from Cisco Unified CM Administration interface
- We will be here most of the time while dealing with CUCM
- We use Application User Credentials to login to Cisco Unified CM Administration interface



- Used to configure backup-device and perform manual and scheduled auto backup of the CUCM cluster
- Also used to restore the DB in case of major database corruption or failures
- Cluster backup is stored to SFTP server as flat files
- It is not recommended to use the vmware native backup option like vmware Snapshot for CUCM
 VM as gives bad performance and IO delay
- vmware image backup solutions like Veem is also not recommended to take CUCM backup
- Also, it is not recommended to enable vmware vmotion and HA for CUCM nodes
- The only recommended backup is via SFTP server
- We use Platform User Credentials (OS Admin credentials) to login to Cisco Unified Disaster
 Recovery System interface



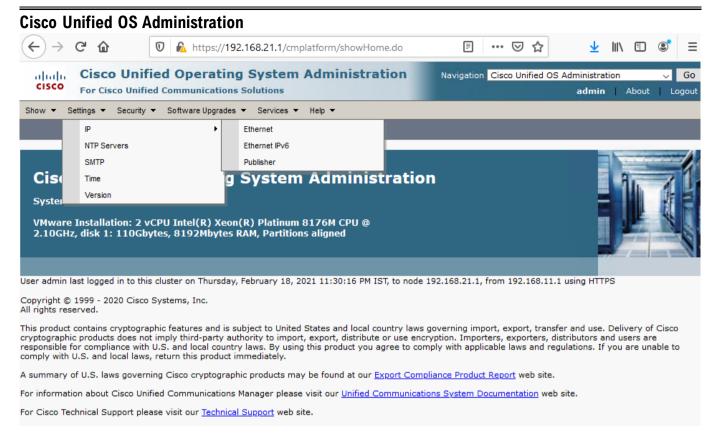
his product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco ryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible or compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

\ summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.

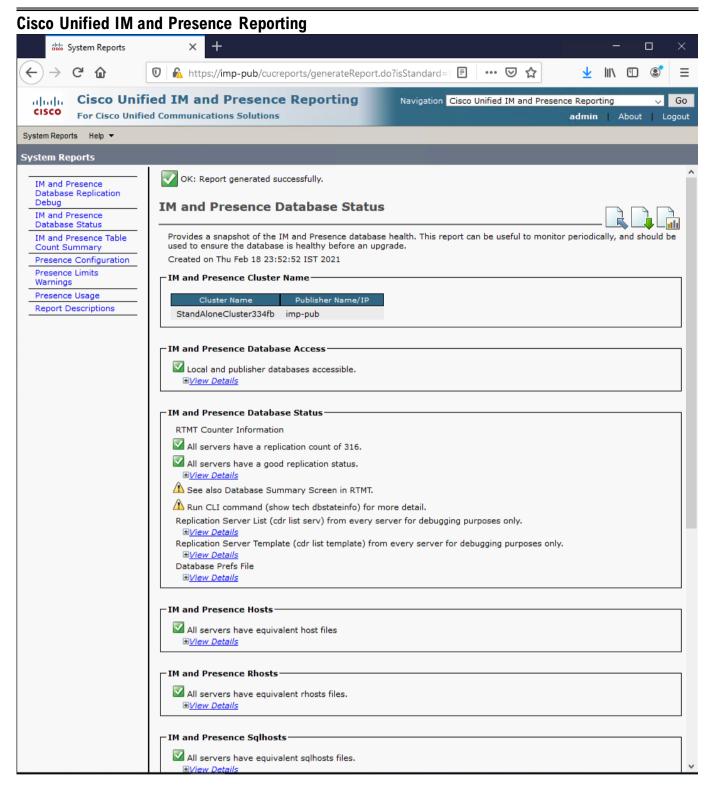
or information about Cisco Unified Communications Manager please visit our <u>Unified Communications System Documentation</u> web site.

for Cisco Technical Support please visit our Technical Support web site.

- Different services can be activated, started, stopped, and restarted from here
- We can also check the service up-time from this interface
- Trace levels (debugs log level) can be tuned from this interface
- Monitoring alarms are configured here, CUCM can sent events to external syslog servers
- We use **Application User Credentials** to login to Cisco Unified Service Ability interface



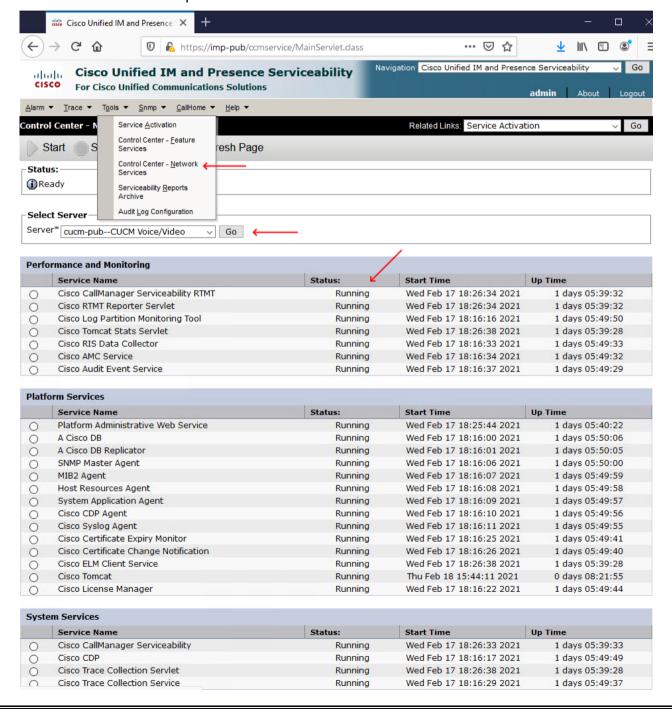
- Here we can interact to the Linux OS of CUCM like configuring IP Address, Changing DNS hostnames, Rebooting the node, Ping and much more
- CUCM Node can be upgraded using this interface
- Some system status can be verified from here
- This is the same interface we access via SSH CLI
- We use Platform User Credentials (OS Admin credentials) to login to Cisco Unified OS Administration interface



- Though you see this on the CUCM web GUI, it redirects to IMP Server if you have IMP in the cluster
- You can see that the web URL changes the moment you click here, this is like the reporting interface of CUCM for IMP servers
- We use Application User Credentials to login to Cisco Unified IM and Presence Reporting interface

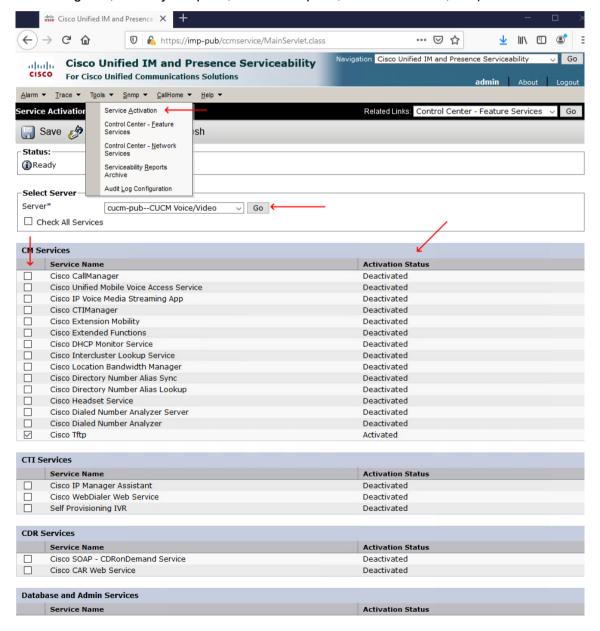
Network Services in CUCM

- The services that are running by default are called Network Services.
- These services are responsible for making sure the CUCM cluster operates smoothly (for example DB Services, Tomcat Service, DRS Backup service, etc.)
- All the nodes in the cluster runs these services by default
- We usually deal with these services during troubleshooting purpose only
- Network services can be started or stopped from the Service Ability >> Tools >> Control Center Network Services page
- We can also see the Up Time of each services here

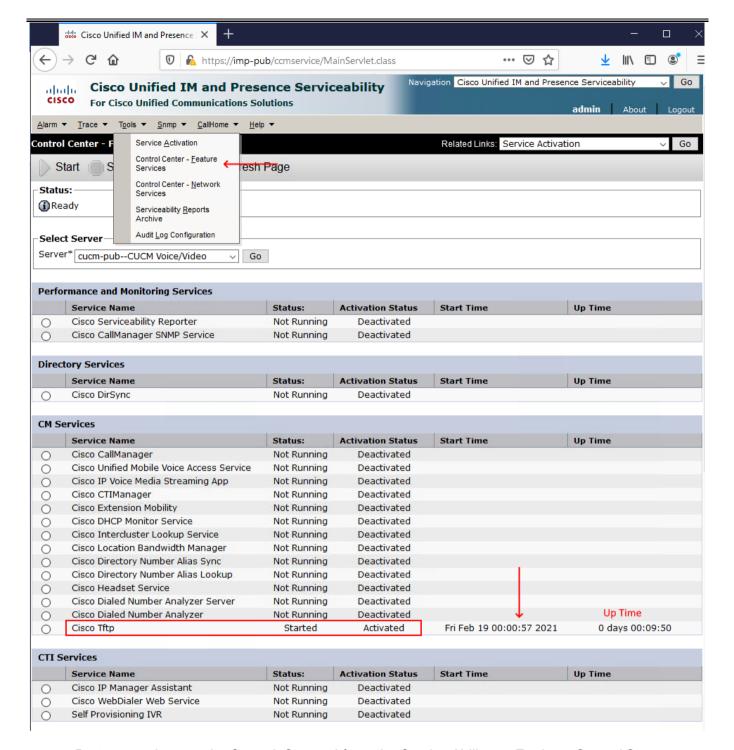


Feature Services in CUCM

- The services that are not running by default are called Feature Services.
- These services are manually activated for enabling the cluster to perform different features (device registration, TFTP, Extension Mobility, etc.)
- For example, to register a Phone or Telepresence endpoint in CUCM cluster, we need 2 services to be activated. 'Cisco TFTP Service' (responsible for delivering the configuration files to devices when they request registration) and 'CCM Service' (responsible for device registration and call processing)
- This configuration file contains necessary information for a device to get register (like which CUCM node to register, soft key template, button template, firmware details, etc.)



Feature services can be Activated from the Service Ability >> Tools >> Service Activation page

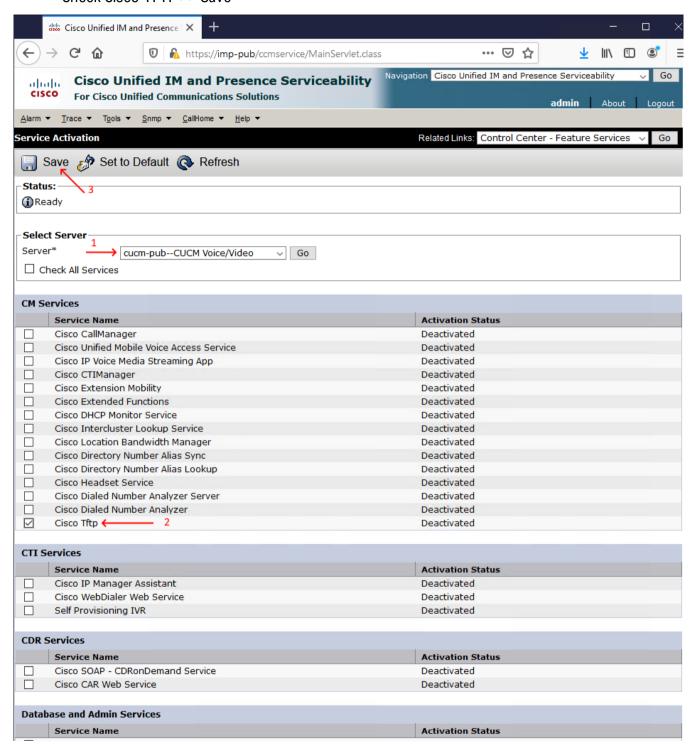


- Feature services can be Started, Stopped from the Service Ability >> Tools >> Control Center Feature Services page
- We can also see the Activation time and Up Time of each services here
- Since CUCM is a distributed cluster of many CUCM nodes, we activate services by considering load lancing in mind
- Hence, I have activated 'Cisco TFTP Service' on the CUCM PUB and 'CCM Service' on CUCM SUB01 and CUCM SUB02
- CUCM SUB01 and CUCM SUB02 can act as a redundant registration server for devices

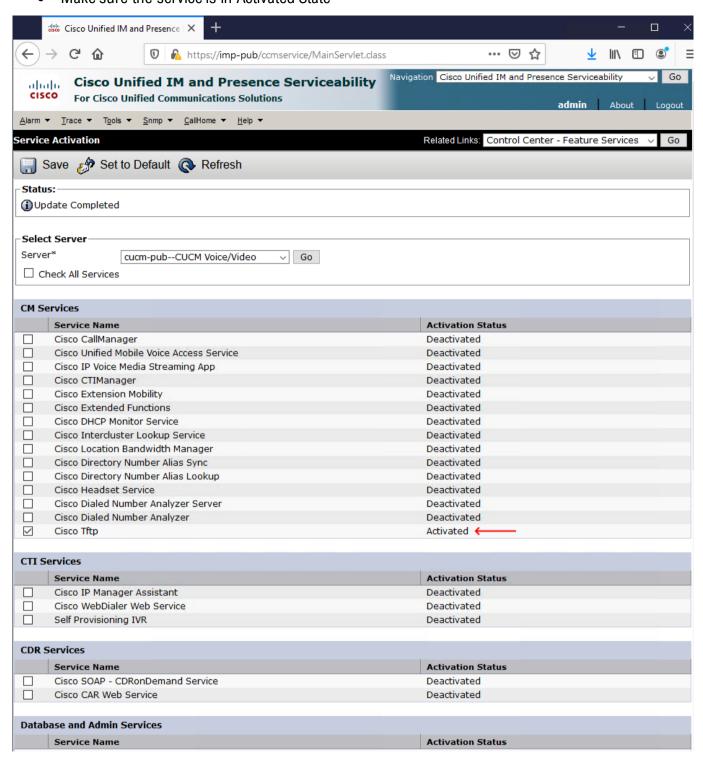
•	Here, I haven't activated 'CCM Service' on CUCM PUB because CUCM PUB has primary				
	responsibility to maintain and replicate the database to other nodes.				
•	Giving extra load on CUCM PUB by activating 'CCM Service' adversely affect the DB Replication				
	process. Hence it is not recommended activate 'CCM Service' on CUCM PUB				

[Lab] Service Activation in CUCM Cluster

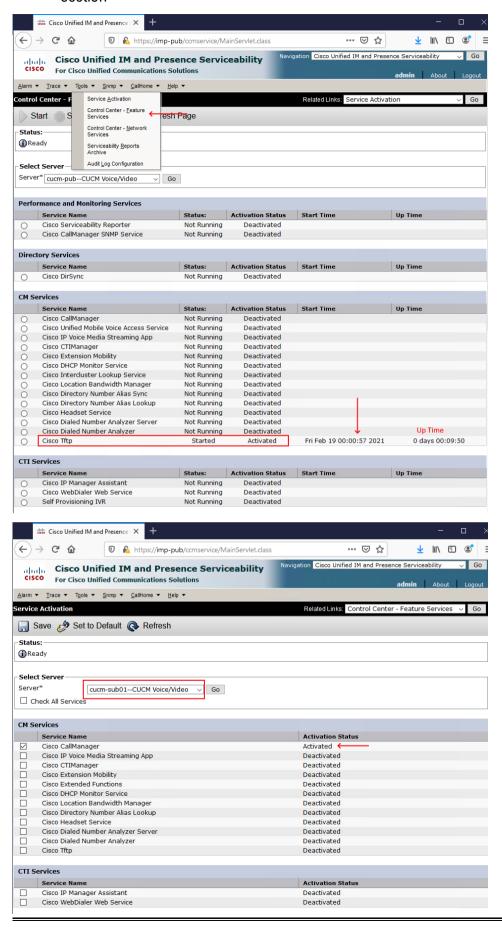
Cisco Unified Service Ability >> Tools >> Service Activation >> Select the CUCM PUB >> Go >>
Check Cisco TFTP >> Save

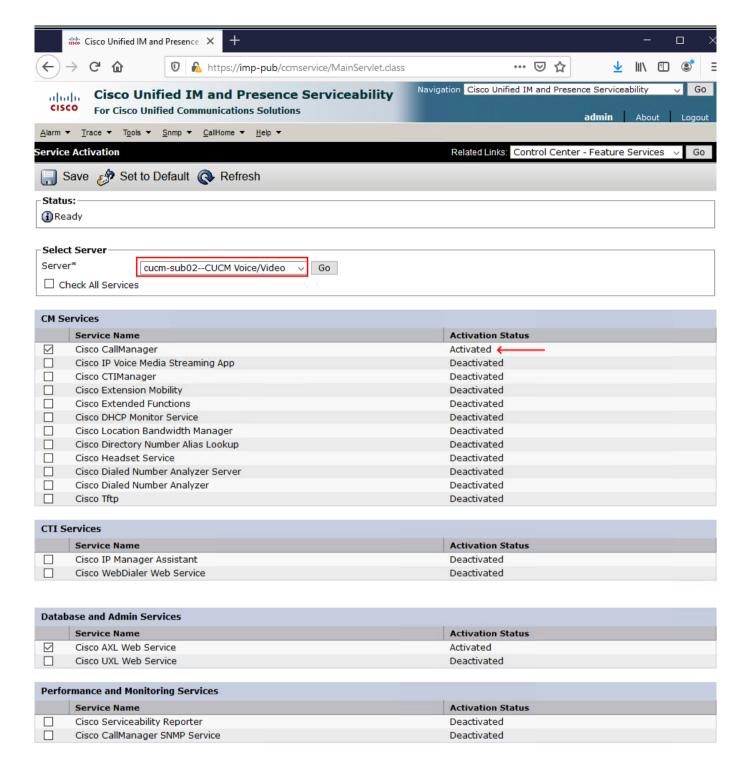


Make sure the service is in Activated State



 We can monitor the status, up time, activation time from the Control Center - Feature Services section





- In CUCM SUB01 and SUB02, activate Cisco CallManager service
- Now our cluster is ready to accept device registration, we will see how to register a Device to CUCM soon

Understanding Cisco IP Phone 8865



- The Cisco IP Phone 8865 is advanced, high-quality, full-featured VoIP communications and affordable entry into HD video (720p) communications
- Flexible deployment options include Cisco on-premises (CUCM), hosted, Mobile Remote Access (MRA) and WebEx Calling
- Read More: Cisco IP Phone 8865

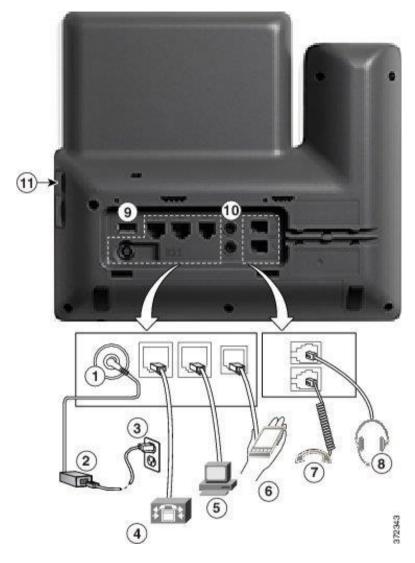
Cisco IP Phone 8865 Front Panel



1	Handset and Handset light strip	Indicates whether you have an incoming call (flashing red) or a new voice message (steady red).	
2	Camera	Use the camera for video calls.	
3	Programmable feature buttons and line buttons	Access your phone lines, features, and call sessions.	
4	Softkey buttons	Access to functions and services.	
5	Back, Navigation cluster, and Release	Back Return to the previous screen or menu. If you press and hold the back button for more than 0.5 secs (long press), you return to the main screen or the call screen. When you are in the settings screens, the long press takes you to the main screen. If you are in one of the call screens, the long press takes you to the call screen. Navigation cluster Navigation ring and Select button—Scroll through menus, highlight items and select the highlighted item. Release End a connected call or session.	
6	Hold/Resume, Conference, and Transfer	Hold/Resume Place an active call on hold and resume the held call. Conference Create a conference call. Transfer Transfer a call.	
7	Speakerphone, Mute, and Headset	Speakerphone Toggle the speakerphone on or off. When the speakerphone is on, the button is lit. Mute Toggle the microphone on or off. When the microphone is muted, the button is lit. Headset Toggle the headset on or off. When the headset is on, the button is lit.	
8	Contacts, Applications, and Messages	Contacts Access personal and corporate directories. Applications Access call history, user preferences, phone settings, and phone model information. Messages Autodial your voice messaging system.	

Volume button Adjust the handset, headset, and speakerphone volume (off hook) and the ringer volume (on hook).

Cisco IP Phone 8865 Back Panel



1	DC adaptor port	48V DC Power (If there is no PoE)
2	AC-to-DC power supply (optional)	AC to 48V DC power adapter
3	AC power wall plug (optional)	Power adapter wall plug
4	Network port (10/100/1000 SW) connection	IEEE 802.3at power enabled, to connect IP Phone to the switch (Network).
5	PC port (10/100/1000 PC) connection	To connect co-located PC (IP Phone acts as a mini switch). We can configure different VLANs for Network and PC ports.
6	Auxiliary port	To connect expansion module
7	Handset connection	To connect receiver handset
8	Analog headset connection (optional)	To connect Cisco Headset (RJ11)
9	USB port	USB devices can be connecte3d, e.g. USB wireless headset

10	Audio In/Out ports	To feed audio in and out
11	USB port	USB devices can be connecte3d, e.g. USB wireless headset

Understanding Cisco Telepresence Endpoint DX70



- The Cisco DX70 desk series 14-inch HD video and audio device
- A multi touch capacitive touch screen that provides an elegant and powerful user interface
- Flexible registration models on-premises, MRA and in the cloud
- Internal 2-port Cisco Ethernet switch allows for a direct connection to a 10/100/1000BASE-T Ethernet network (IEEE802.3i/802.3u/802.3ab) through an RJ-45 interface with single LAN connectivity for both the DX70 and a co-located PC
- Supports wireless network connection
- Firmware is either Cisco CE or Android, default comes with Cisco CE software
- DX70 is bit outdated Telepresence Endpoint, we have Cisco WebEx Desk Series

Cisco DX 70 Front Panel

Integrated collaboration at the desktop

Compact size for limited spaces

Features

- 802.11 a/b/g/n wireless and Gigabit Ethernet connectivity
- Bluetooth and USB (including a highcurrent side port for charging tablets and smart phones)
- Peripheral support offers many options for personalization
- Android 4.1 and Google Play Store access for third-party applications
- HDMI Input for PC/Mac Display and Content Sharing
- · Tiltable Document Camera
- · VESA (100 and 75) mount support

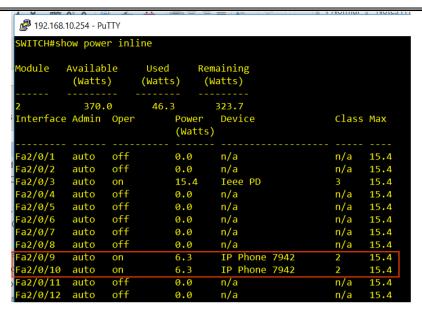


Cisco DX 70 Back Panel



Protocols and Services Used by Cisco IP Phones / Telepresence Endpoints

- NTP (Network Time Protocol): Used to synchronize time with a central server to enable standard time on all the IP Phones
- CDP (Cisco Discovery Protocol): Cisco Switches will identify connected IP Phone via CDP. This will
 make sure specific power delivery over PoE enabled switches
- **DHCP (Dynamic Host Configuration Protocol)**: DHCP is faster, easier, widely accepted method to distribute IP information to the clients, here IP Phones. DHCP can be provided by an existing dedicated DHCP server, Router, L3 Switch, other software or natively from CUCM itself. It provides following information to the IP Phones.
 - o IP Address
 - Subnet Mask
 - Default Gateway
 - DNS Server
 - o TFTP Server IP
- TFTP (Trivial File Transfer Protocol): IP Phones utilize TFTP to download their configuration files, firmware images etc. Normal TFTP server can't fulfill IP Phones requirement, hence we must have CUCM TFTP server. CUCM runs TFTP service at port number 6970
- **PoE (Power Over Ethernet)**: Provides DC power over Ethernet cabling. No extra wiring, no external power supply needed. Cisco Switch send FLP (Fast Line Pulse 147KHz), phone receive and send back the FLP to the switch. By this way switch will realize the device need power. Then switch provide required power. From CDP message switch can be able to analyze how much power needed for an IP Phone. 3rd party phone will take full power. Use show power inline command to verify the PoE



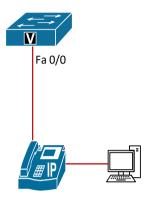
- DNS (Domain Name System): DNS provides IP to Domain mapping and vice versa. It is not critical
 to IP Phones. DNS is not a service that CUCM can offer
- SIP (Session Initiation Protocol): Used to resister the phone with CUCM and facilitates in call control. Open standard protocols
- SCCP (Skinny Client Control Protocol): Used to register the phone with CUCM and enables in call control. Cisco proprietary protocol, almost outdated. New IP phones don't support SCCP

DHCP Configuration for Cisco IP Phones and Telepresence Endpoints

- DHCP can be configured on a L3 switch, Router or dedicated DHCP servers (Windows, Linux, etc.)
- Option 150 is used to mention CUCM TFTP Server IP address
- Option 66 is used to point CUCM TFTP Server DNS name

[LAB] DHCP on L3 Switch

 In the below topology, the phones will get VLAN 11 and the respective IPs and PC will get VLAN 12 and corresponding IPs.



Creating a logical Interface for Voice VLAN

SWITCH(config)# interface Vlan10 SWITCH(config-if)# ip address 192.168.10.254 255.255.255.0

Creating a logical Interface for Data VLAN

SWITCH(config)# interface Vlan11 SWITCH(config-if)# ip address 192.168.20.254 255.255.255.0

Excluding IP Address from the pool

SWITCH(config)# ip dhcp excluded-address 192.168.10.1 192.168.10.10 SWITCH(config)# ip dhcp excluded-address 192.168.10.245 192.168.10.254

SWITCH(config)# ip dhcp excluded-address 192.168.20.1 192.168.20.10 SWITCH(config)# ip dhcp excluded-address 192.168.20.245 192.168.20.254

Creating DHCP Pool for phones with TFTP Options

SWITCH(dhcp-config)# ip dhcp pool HQ_PHONES SWITCH(dhcp-config)# network 192.168.10.0 255.255.255.0 SWITCH(dhcp-config)# default-router 192.168.10.254 SWITCH(dhcp-config)# dns-server 192.168.10.11 SWITCH(dhcp-config)# option 150 ip 192.168.10.253 192.168.10.13 SWITCH(dhcp-config)# option 66 ip cucm-pub.ajcollab.com SWITCH(dhcp-config)# domain-name ajcollab.com

Creating DHCP Pool for PCs

SWITCH(dhcp-config)# ip dhcp pool HQ_DATA SWITCH(dhcp-config)# network 192.168.20.0 255.255.255.0 SWITCH(dhcp-config)# default-router 192.168.20.254 SWITCH(dhcp-config)# dns-server 192.168.20.11

Assign Dual VLAN to the Switch Port

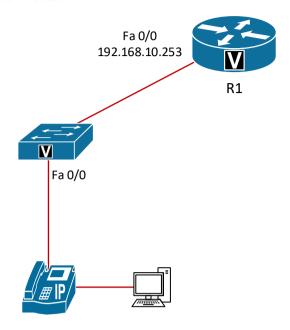
SWITCH(dhcp-config)# interface range fastEthernet 0/1-12 Switch(config-if-range)#switchport mode access Switch(config-if-range)#switchport voice vlan 10 Switch(config-if-range)#switchport access vlan 11

In this case the switch port is intelligent enough to assign Data VLAN to the PC port on the IP Phone. Hence, when you connect a PC on the PC Port, that device will be in Data VLAN 11 and IP Phone will be in Voice VLAN 10



[LAB] DHCP on Local Router

 In this situation, make sure there is no interface VLAN configured on the switch for the VLAN where the phones are connected



Assigning IP Address to router interface

Router(config)# interface fa 0/0
Router(config-if)# ip address 192.168.10.253 255.255.255.0
Router(config-if)# no shutdown

Excluding IP Address from the pool

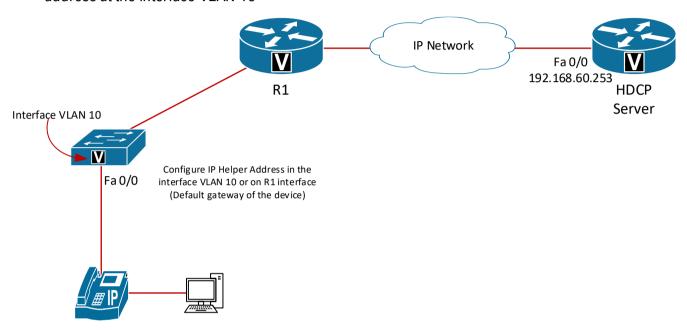
Router(config)# ip dhcp excluded-address 192.168.10.1 192.168.20.10 Router(config)# ip dhcp excluded-address 192.168.10.245 192.168.20.254

Creating DHCP Pool for phones with TFTP Options

Router(dhcp-config)# ip dhcp pool HQ_PHONES
Router(dhcp-config)# network 192.168.10.0 255.255.255.0
Router(dhcp-config)# default-router 192.168.10.253
Router(dhcp-config)# dns-server 192.168.10.11
Router(dhcp-config)# option 150 ip 192.168.10.253 192.168.10.13
Router(dhcp-config)# option 66 ip cucm-pub.ajcollab.com
Router(dhcp-config)# domain-name ajcollab.com

[Lab] DHCP Configuration on Remote Router or Device

- If DHCP server is located remotely (not in the LAN), the DHCP broadcast packets will drop at the default gateway of the device
- Hence, we would configure IP Helper address in the default gateway interface to reach a remote
 DHCP server
- In below figure, the DHCP traffic will drop at the interface VLAN 10. Hence configure IP Helper address at the interface VLAN 10

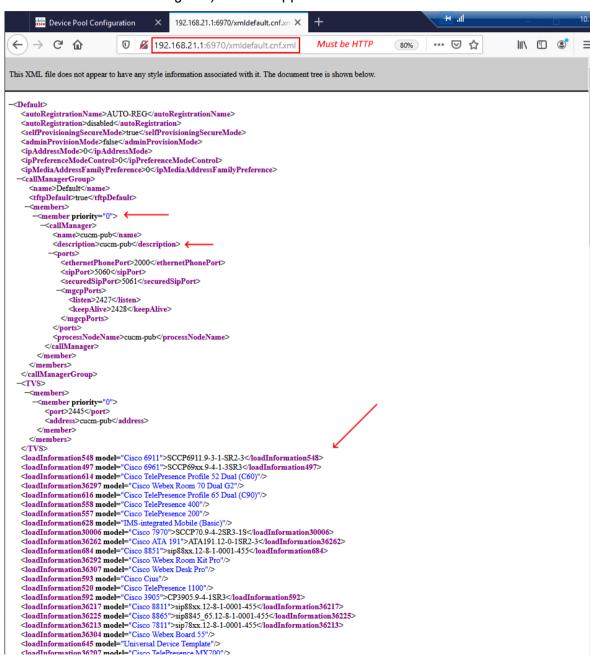


```
SWITCH(config)# interface Vlan11
SWITCH(config-if)# ip address 192.168.10.254 255.255.255.0
SWITCH(config-if)# ip helper-address 192.168.60.253
```

Note: Make sure, the Remote DHCP server should contains a DHCP pool with a network similar to the interface VLAN 10.

Default Phone Configuration File - xmldefault.cnf.xml

- As part of Phone registration process, the phone requests a configuration file from the CUCM
 TFTP Server over HTTP port 6970
- Phone will always request SEPMAC_ADDRESS.cnf.xml file, if the phone is not present in the
 CUCM database (not added before), this specific SEPMAC_ADDRESS.cnf.xml won't be available
- During such scenario (phone is not added in CUCM before), the phone requests the default configuration file xmldefault.cnf.xml from TFTP server
- To access this file, http://CUCM_TFTP_SERVER:6970/xmldefault.cnf.xml
- The file contains whether the Auto registration is Enabled or Disabled, CUCM Group (where we
 have the CUCM node to register) and all supported Phone models and available firmware



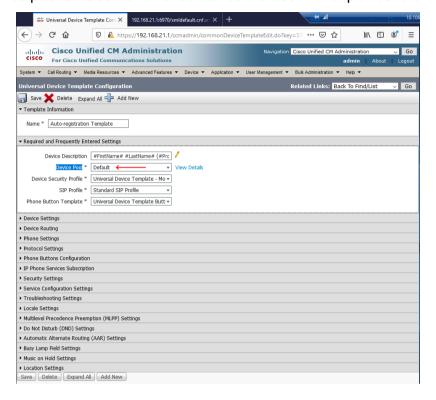
Auto Phone Firmware Upgrade Situations

- A phone will go for a firmware upgrade if its firmware is lower than the one available on xmldefault.cnf.xml file
- If the phone firmware is higher than the one available on xmldefault.cnf.xml, there won't be aby firmware upgrade

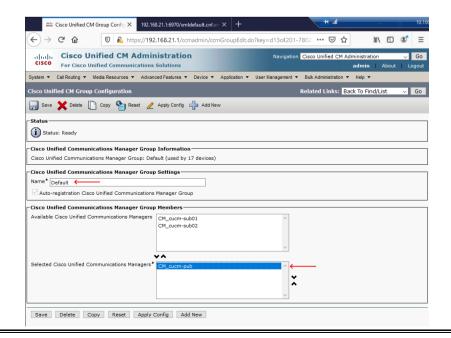


[Lab] Auto Registration of Cisco Endpoint in CUCM

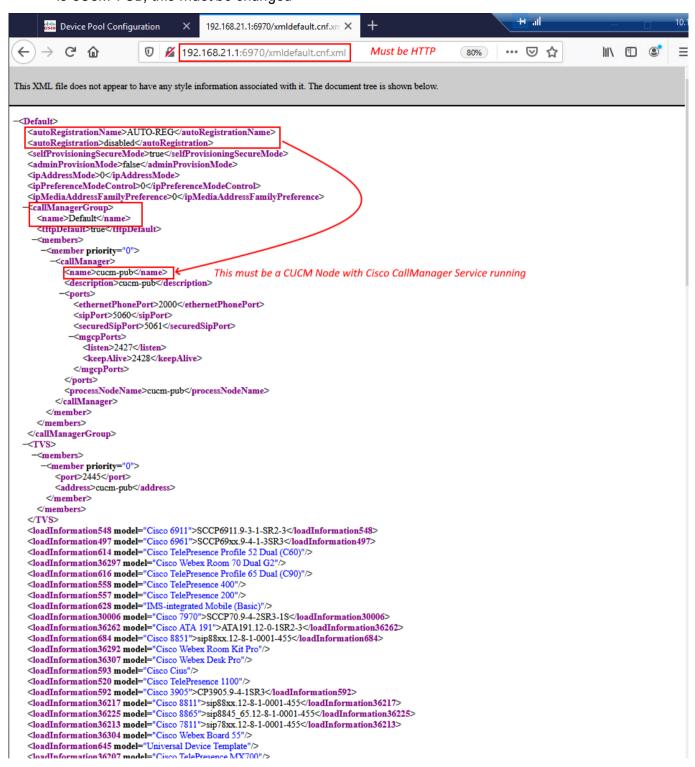
- Go to User Management >> User/Phone Add >> Universal Device Template
- Check the device pool there, phones will be taking this information to get register
- Here the device pool is Default and hence it takes the CUCM Group under the Default device pool



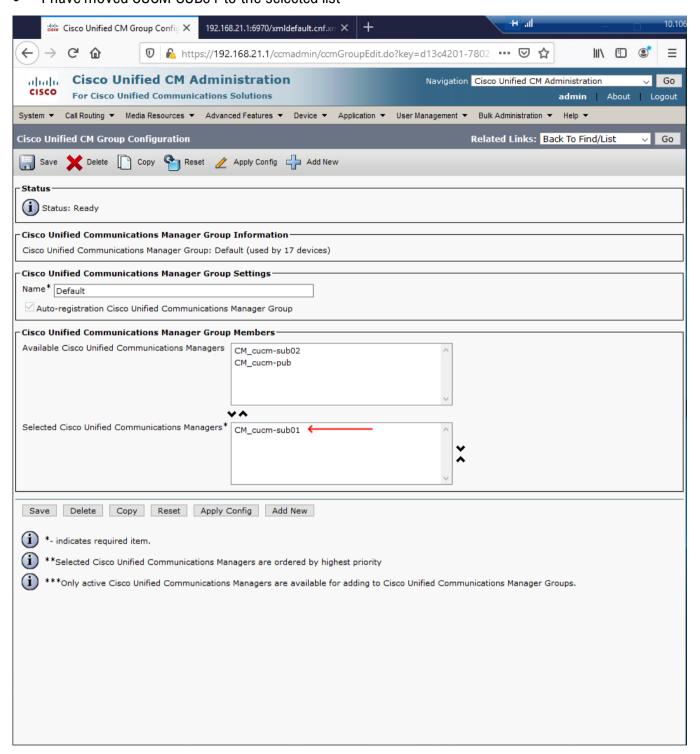
- Go to System >> Cisco Unified CM Groups and check the Default CUCM Group
- Here we see CUCM PUB in the Selected Cisco Unified Communications Managers list
- We haven't activated Cisco CallManager service on CUCM PUB
- Hence no registration will happen by default



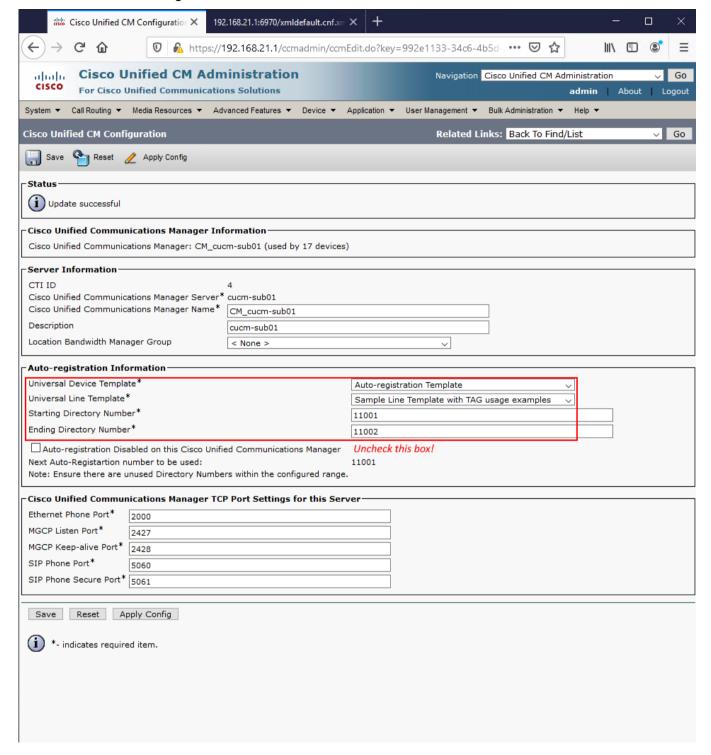
• If we look at the xmldefault.cnf.xml file, we see that Auto Registration is disabled and primary node is CUCM PUB, this must be changed



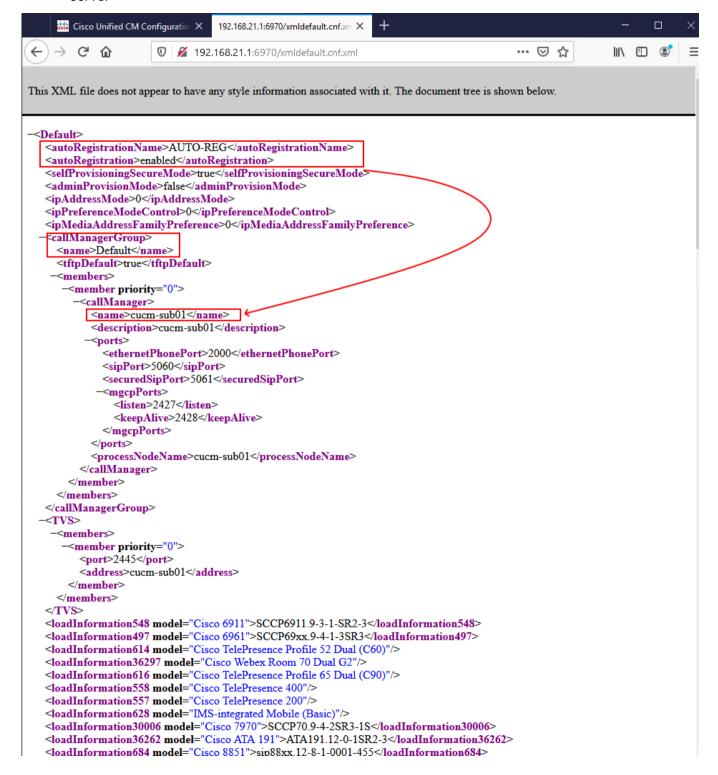
I have moved CUCM SUB01 to the selected list



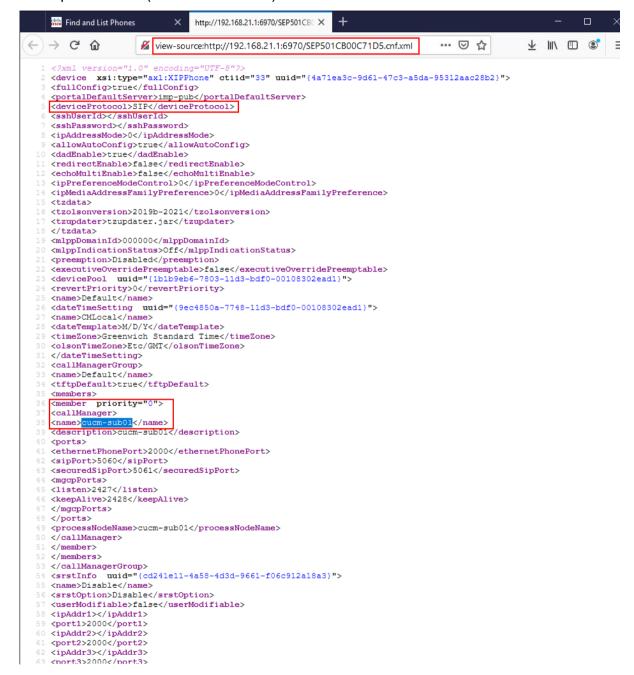
- Go to System >> Unified CM >> Select CUCM SUB01
- Set Universal Device and Line Template
- Configure starting DN and Ending DN
- Uncheck Auto-registration Disabled box



- Now the xmldefault.ncf.xml file shows that the auto registration is enabled
- The phone that gets this configuration file, will send a SIP Register Request to CUCM SUB01
- After receiving the Register message from the phone, the CUCM SUB01 will create a database entry and corresponding phone specific configuration file SEPMAC_ADDRESS.cnf.xml on the TFTP server

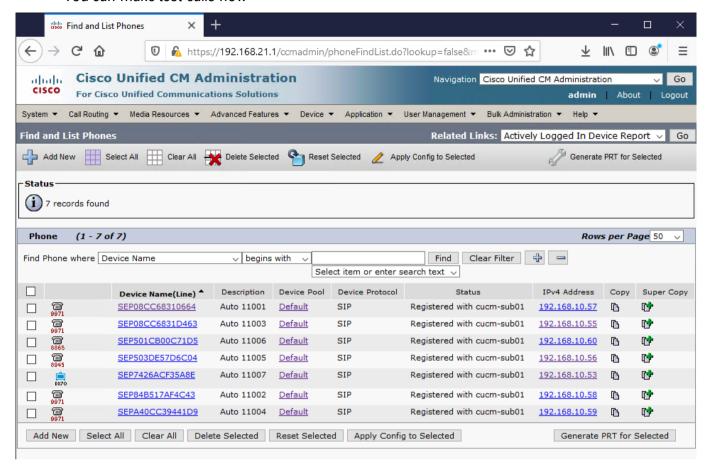


- Now the phone will go for a reboot, during next registration attempt, the phone will get its own configuration file SEPMAC ADDRESS.cnf.xml
- Basically, the Auto reg enabled xmldefault.cnf.xml helps to create SEPMAC_ADDRESS.cnf.xml
- Based on the information available on SEPMAC_ADDRESS.cnf.xml, the phone will get register to the respective node (here CUCM SUB01)

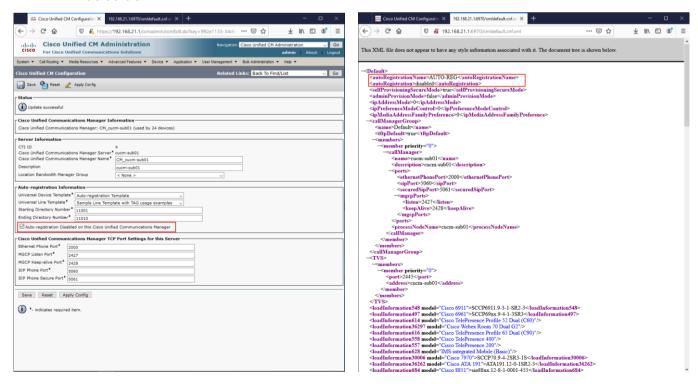


- Please note, here the CallManager node name is the host name 'cucm-sub01', phones should be able to resolve this to corresponding IP address to successfully register. So, make sure your phones DHCP pool has proper DNS server configured
- If the phone doesn't have DNS server, then change the CUCM node names to IP Address by going to System >> Server

- Device >> Phones, All the Phones are registered automatically with Default Device Pool
- You can make test calls now



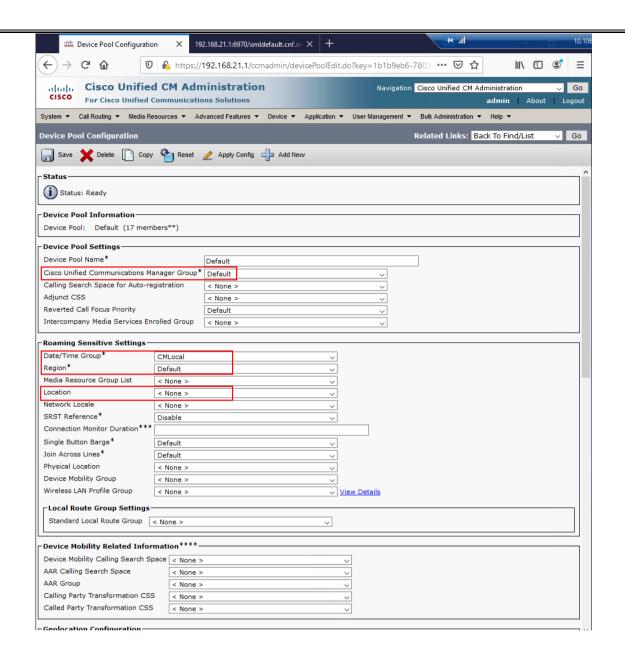
 After the lab, make sure you disable the Auto registration on CUCM SUB01, verify that on the xmldefault.cnf.xml



Device Pool in CUCM



- It is a set of common configurations for a group of devices. We usually create Device Pools to group similar devices (e.g. Phones in a specific location have same device pool)
- The device pool contains Registrar Server, Codec Settings, and many more information that is useful for a device
- To configure Device Pool, System >> Device Pool
- Let's evaluate some of the key components of a Device Pool



Cisco Unified Communications Manager Group

- It specifies a prioritized list of up to 3 Cisco Unified Communications Managers (CUCM nodes with Cisco CallManager service running)
- The first CUCM node in the list serves as the primary registration server for that group, and the other members of the group serve as backup for redundancy
- When primary CUCM node goes down, devices automatically fall back to secondary CUCM nodes
- This provides redundancy in Phone registration and call routing

Date/time group

- The date/time group specifies the time zone and the display formats for date and time for the devices
- CUCM Node time zone is given during the installation of CUCM server, Date/time group is solely
 used for setting the proper time zone for devices
- You can have you CUCM node in one geographical location and devices in entirely different location if network connectivity in place (Phones can reach CUCM nodes IP)
- Date/Time Group offsets the correct time learned via NTP to match local time zone where the
 device is located. Also, we can specify the format of time

Region

- Region is used to specify voice codec (bit rate) per calls within a region and between other regions
- Devices in same Region uses G.711 or G.722 whereas one Region to another Region uses G.729
 by default
- These relations can be modified according to our needs

Location

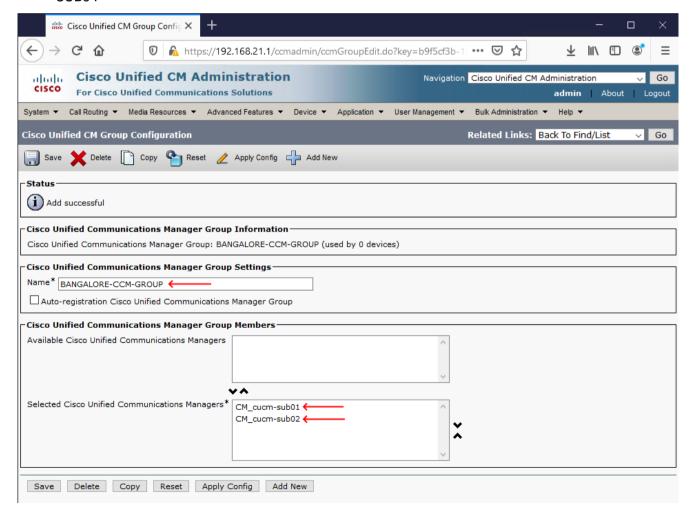
- The location specifies the total bandwidth that is available for calls to and from this location
- Use locations to implement call admission control (CAC) in a centralized call-processing system
- CAC enables you to limit the total amount of bandwidth that is available for audio and video calls between locations
- While Region specified per call bit rate, Location sets the maximum calls bitrate from that location

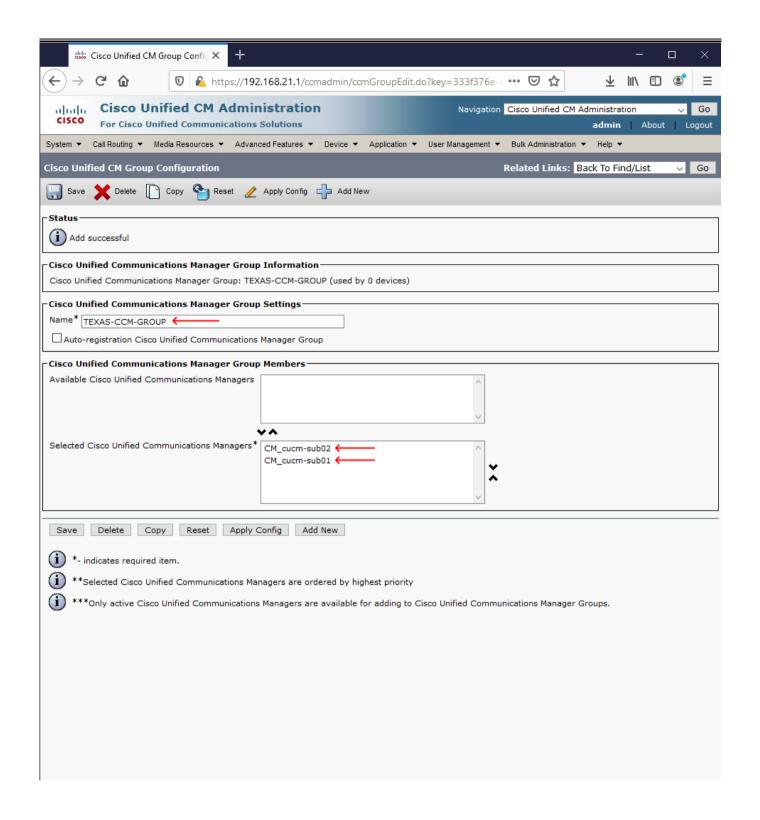
[Lab] Create a Device Pool

- In production cluster, we do not use the default device pool, instead we create specific device pools based on circumstances
- In our case, we have 2 geographical location (Bangalore and Texas), Bangalore is the headquarters and Texas is the remote branch office

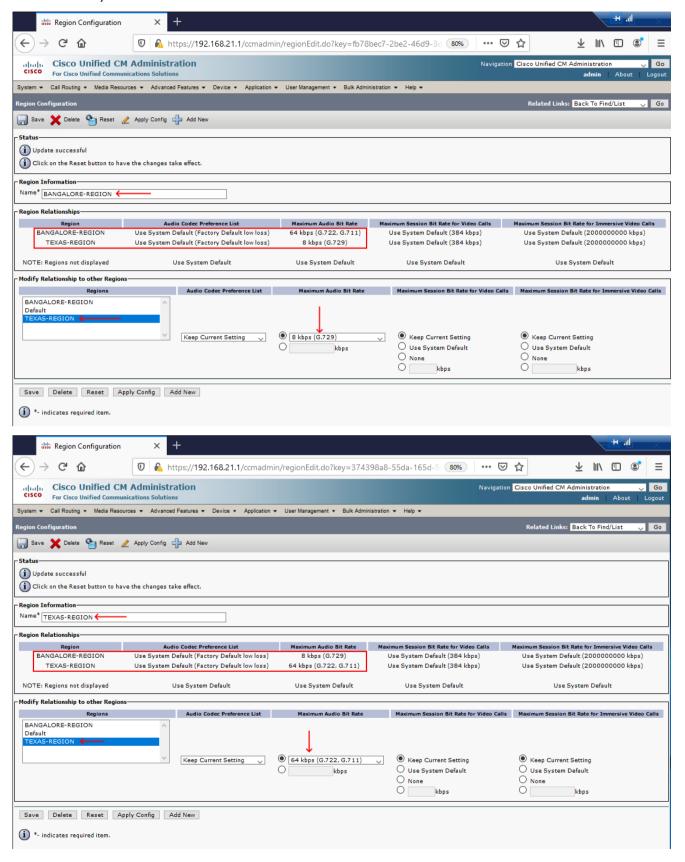
	Device Pool Component	BANGALORE-DEVICE-POOL	TEXAS-DEVICE-POOL
1	CUCM Group	BANGALORE-CCM-GROUP	TEXAS-CCM-GROUP
		Order = SUB01, SUB02	Order = SUB02, SUB01
2	Region	BANGALORE-REGION	TEXAS-REGION
3	Date/time Group	IST-DATE-TIME-GROUP	CST-DATE-TIME-GROUP
		(GMT+5:30 offset)	(GMT-6 offset)

- We set the Region relation between these 2 areas to G.729 (8 kbps) and with in the region to G.711 (64kbps)
- System >> Cisco Unified CM Groups >> Add New >>
- Add 2 CUCM groups with node priority, BANGALORE = SUB01 & SUB02; TEXAS = SUB02 & SUB01

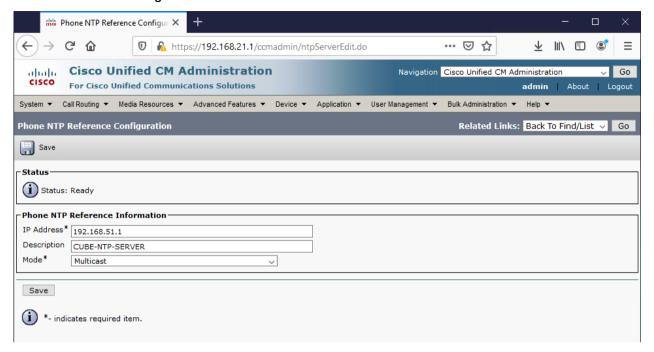




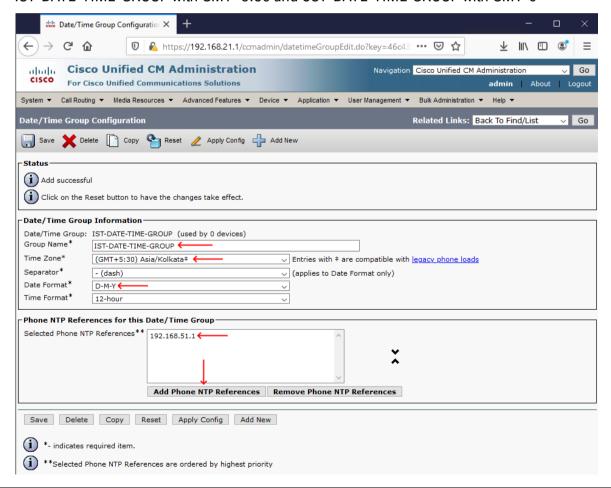
- System >> Region Information >> Region >> Add New >>
- Add 2 regions and set the relation (SAME REGION = G.711; between BANGALORE & TEXAS = G.729)

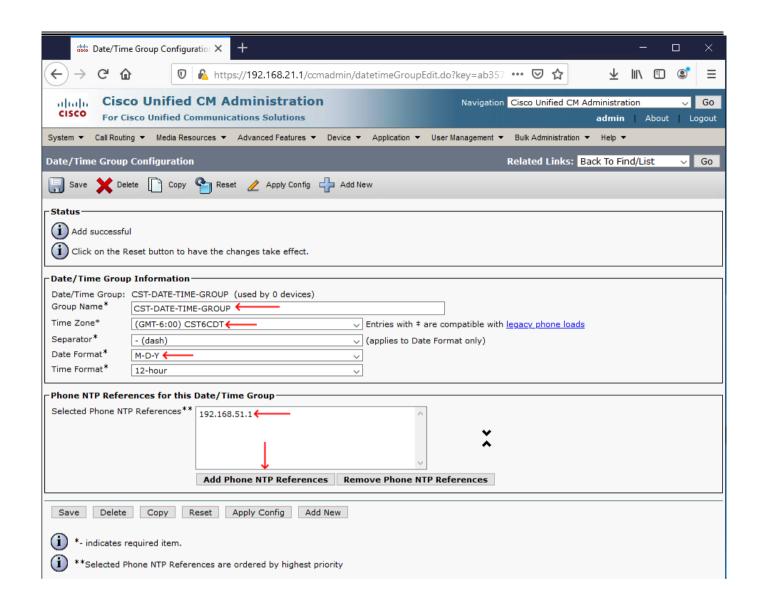


- System >> Phone NTP References >> Add New
- SIP Phones take the reference time from NTP Reference configuration
- SCCP Phones will ignore this

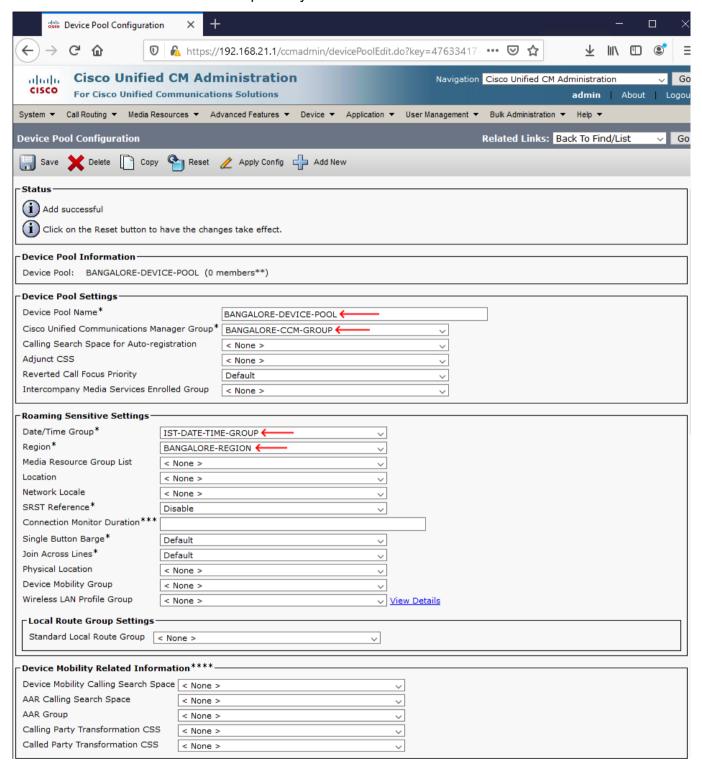


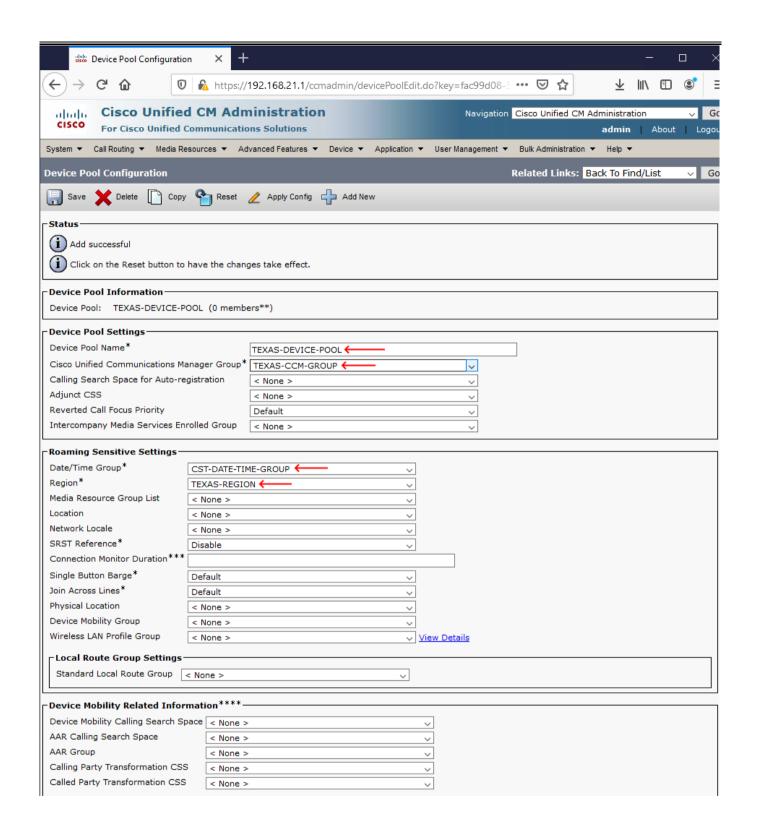
- System >> Date/Time Group >> Add New
- IST-DATE-TIME-GROUP with GMT+5:30 and CST-DATE-TIME-GROUP with GMT-6





- System >> Device Pool >> Add New
- Associate corresponding CUCM Group, Date/Time Group and Region to BAGALORE-DEVICE-POOL and TEXAS-DEVICE-POOL respectively

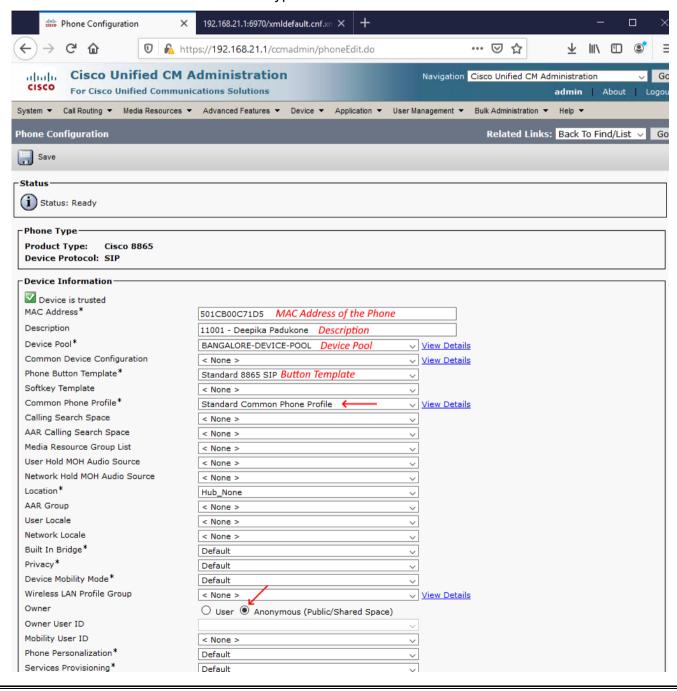


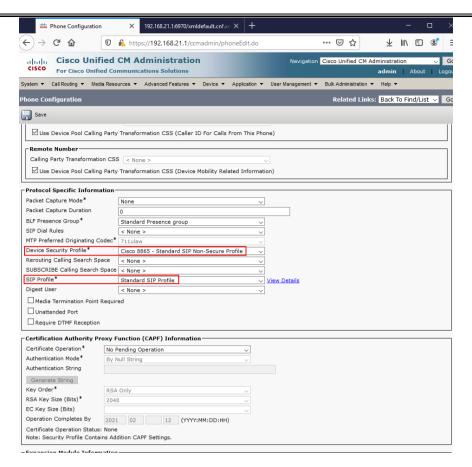


[Lab] Manual Registration of Cisco Endpoint in CUCM

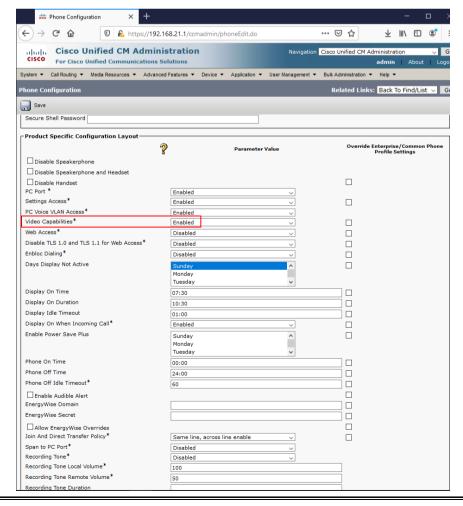
- As we know that any Phones will initially look for SEPMAC_ADDRESS.cnf.xml, if they are unable to find it, then go for xmldefault.cnf.xml
- Based on the auto registration information present in the xmldefault.cnf.xml file, the phone may register or stay without registering
- Auto registration is disabled in in most of the enterprise companies since we can't control the
 Extension number and other parameters for the phones via auto registration
- Manually adding the Phone MAC details and tuning other parameters in CUCM Database is called manual registration

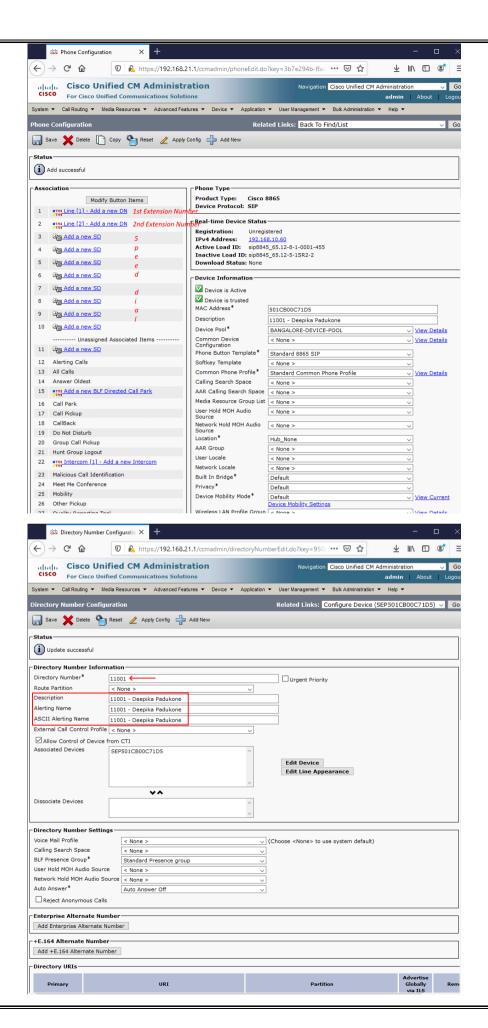
Device >> Phones >> Add New >> Phone Type: Cisco 8865 >>

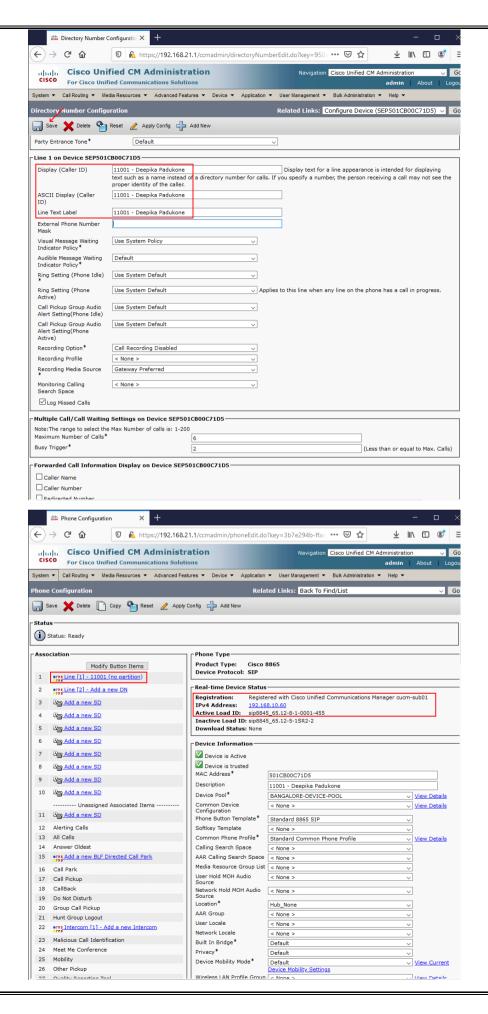




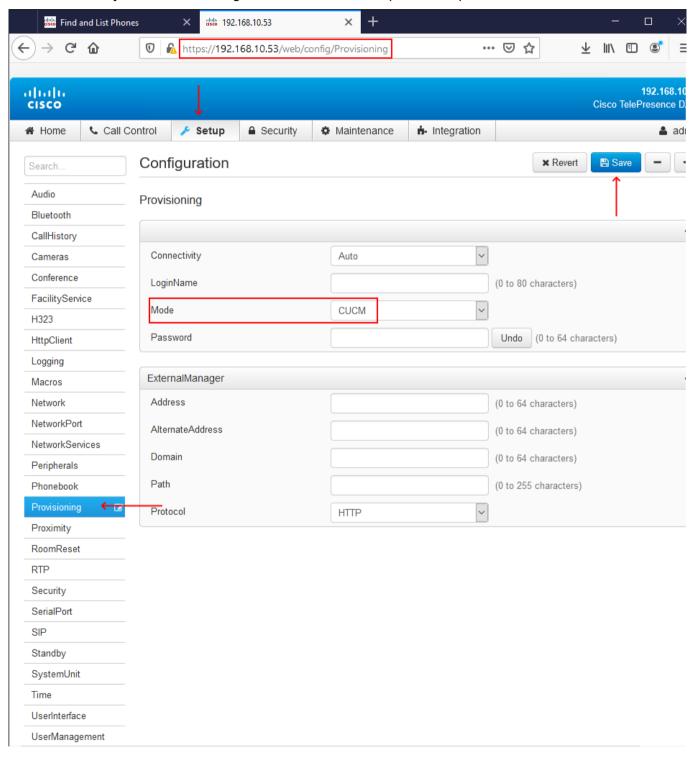
Video Capabilities must be enabled if we need video calls feature



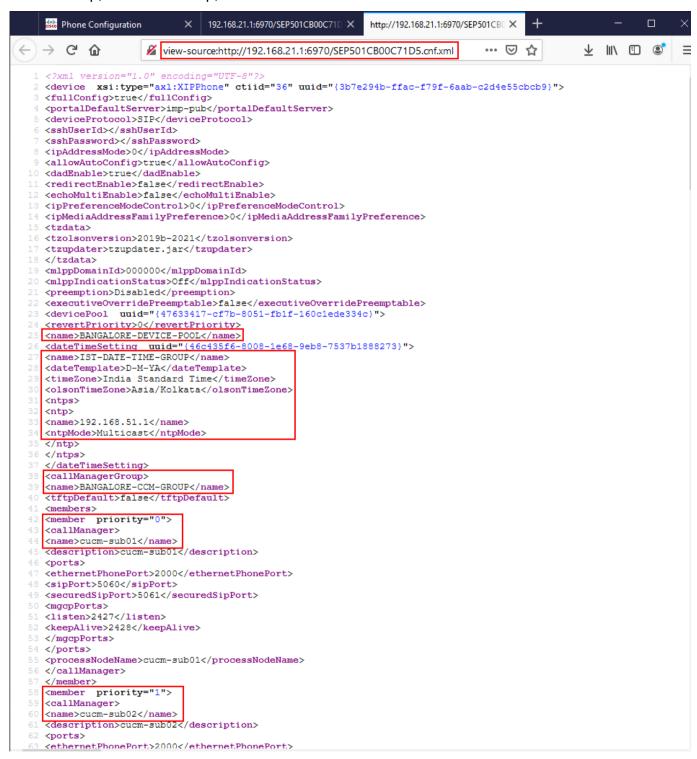




- Telepresence endpoints do not register even if they added to CUCM manually, we must perform additional step on the DX70 web interface to get that registered to CUCM
- Access the DX70 web interface by browsing the device IP Address
- Setup >> Configuration >> Provisioning >> Mode: CUCM >> Save
- Then only DX70 will auto register to CUCM, This step is not required for IP Phones

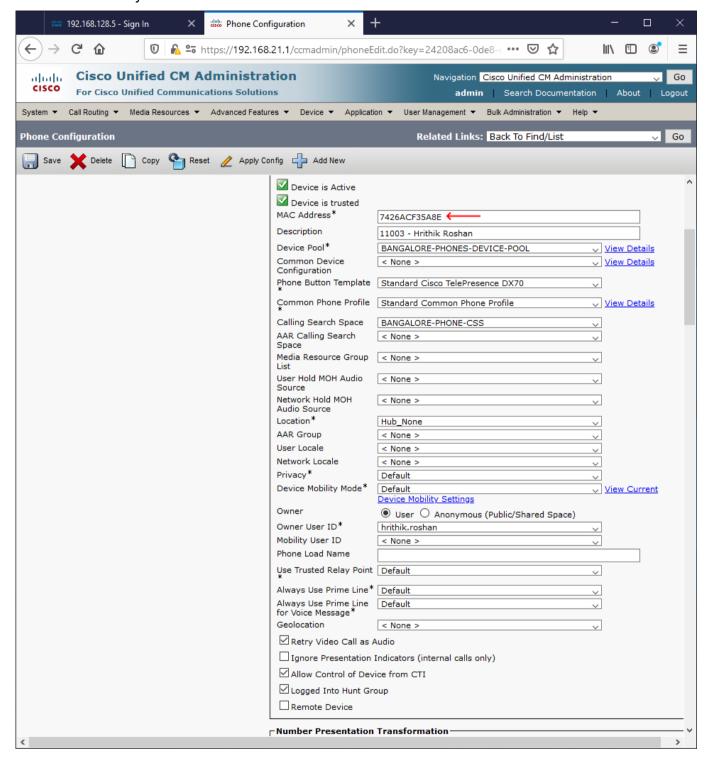


 If we look at the SEP501CB00C71D5.cnf.xml file, we would be able to see the Device Pool, CUCM Group, Date/Time Group, etc.

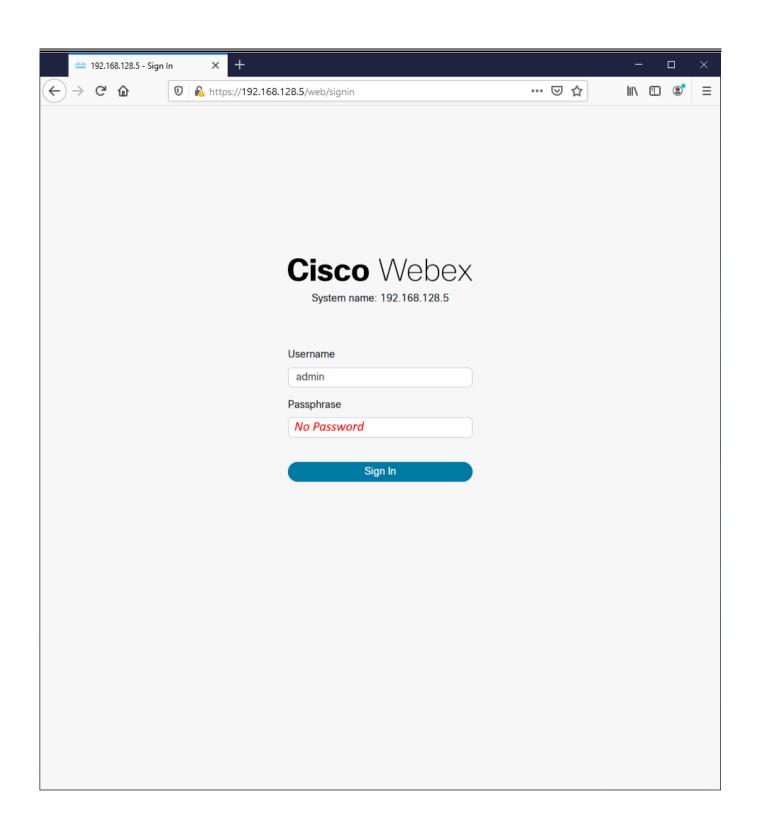


[Lab] Telepresence Endpoint Registration

- Connect network cable to DX70 and Perform factory reset of the unit
- Make sure you have added the unit in CUCM



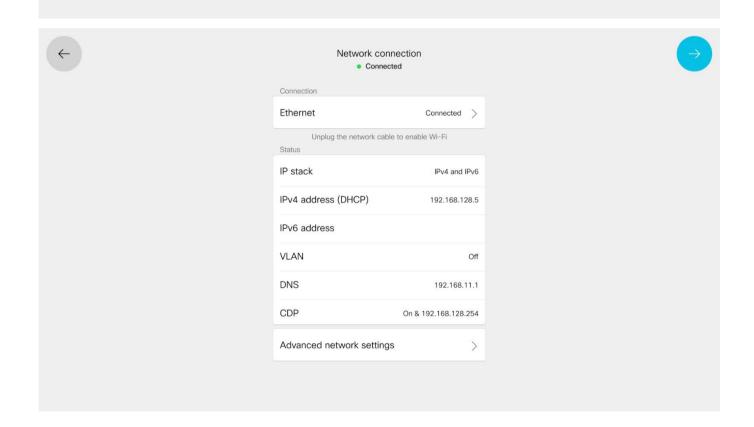
· Browse the device IP Address to access the web interface

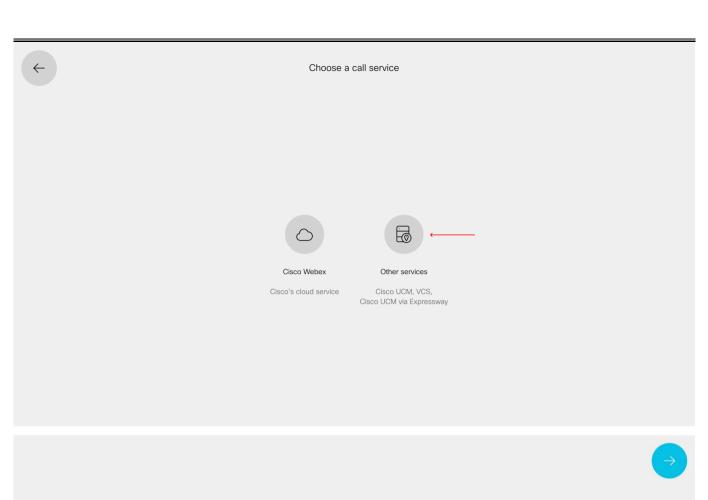


Serial number: FOC1846N3S3 IPv4: 192.168.128.5

Welcome

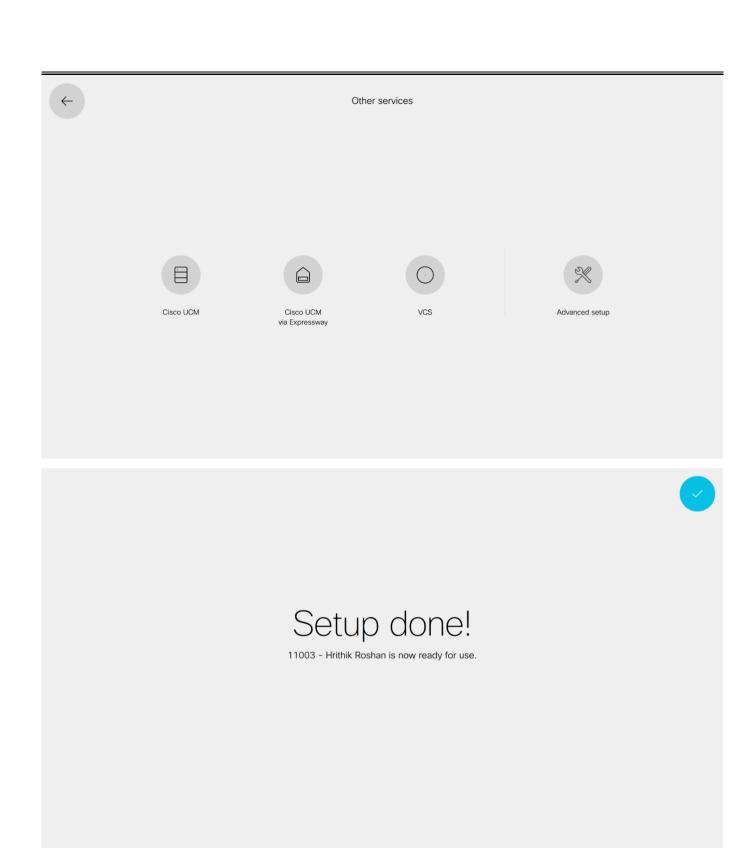


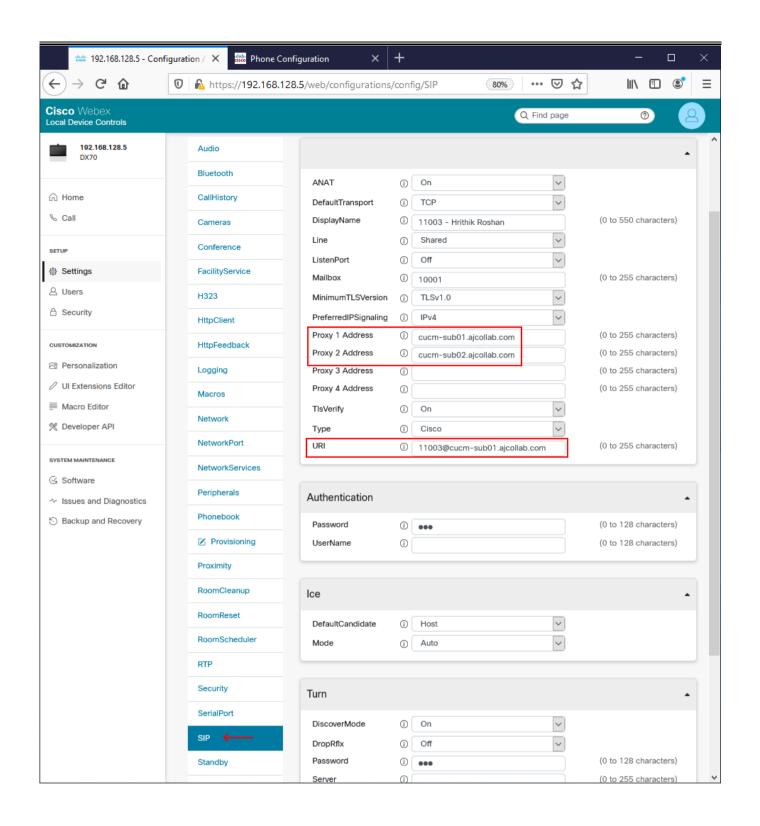




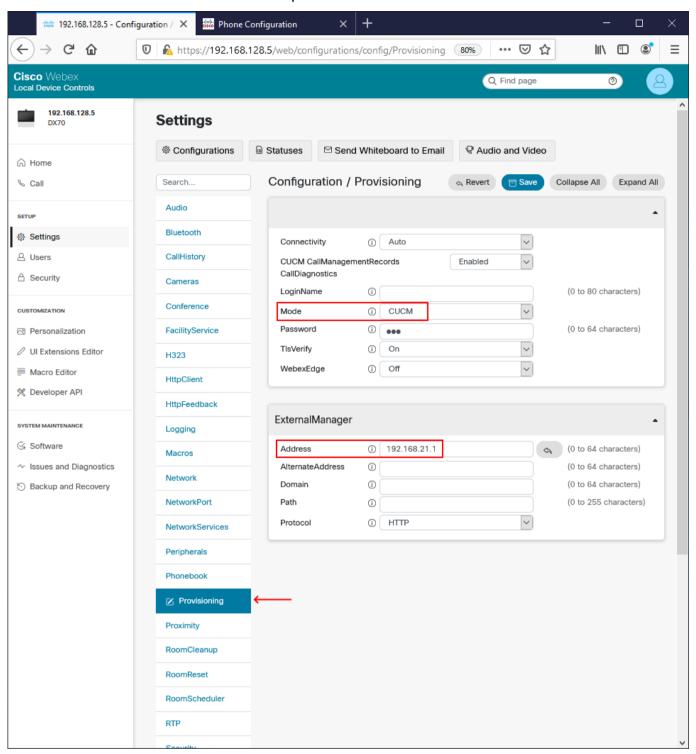
A Cisco UCM service at 192.168.21.1 192.168.21.2 is available in your network. Continuing will activate this device to that service. Alternatively, choose another service.

Change service





 If you do not have a physical access to the device, then you can configure the provisioning server as the TFTP Server in the DX70 web portal



[Lab] Softkey Template

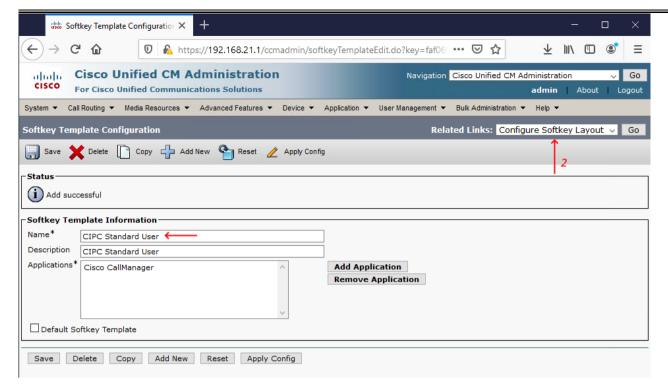
- Softkey are used to perform different tasks from the phone (e.g. Redial, New Call, CFwdAll, etc.)
- Softkeys will change based on the state of the phone (on-hook, off-hook, connected states, etc.)



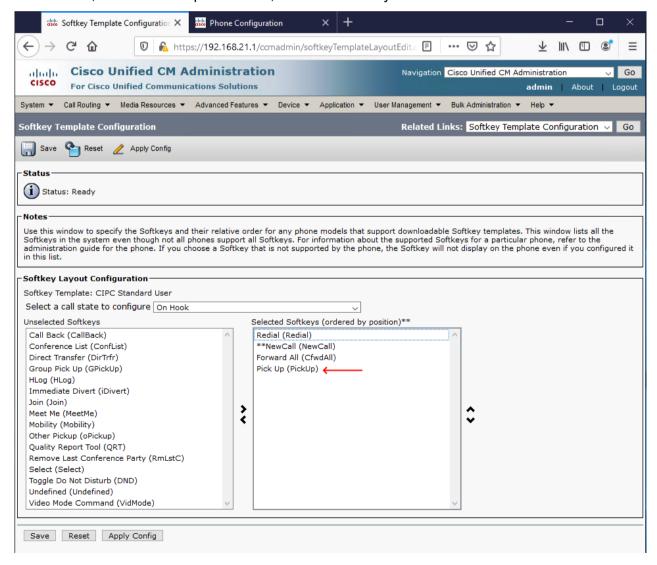




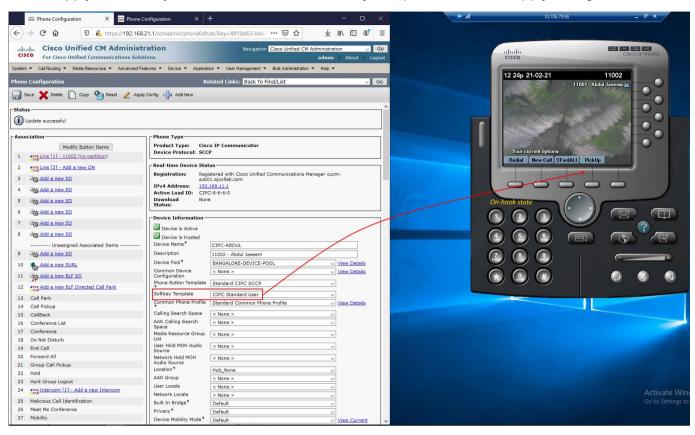
- Standard keys will be available in the default Softkey template, but we can customize it
- Device >> Device Settings >> Softkey Template >> Copy the Standard User template



Now, based on the specific state, select the softkeys and save it



• Apply this softkey in Device >> Phone >> Softkey Template >> Save >> Apply Config

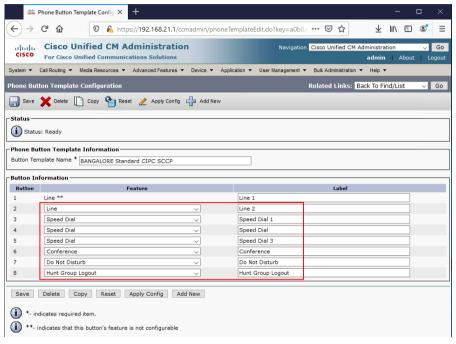


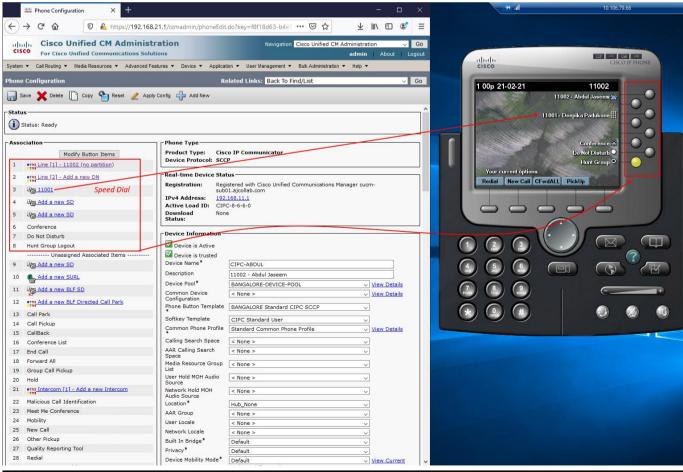
Latest phones like Cisco 88XX, Cisco 99XX have dedicated hard keys for special functions like
 Hold, Conference, Transfer. Hence, adding those options in softkeys will not reflect on the phone



[Lab] Phone Button (Line & Feature) Template

- Defines the behavior of Phone Buttons. First buttons dedicated for Directory Number (DN), we can customize rest of the buttons
- Special features (option that we have seen in softkeys) can also be added to these buttons
- Device >> Device Settings >> Phone Button Template >> Copy Standard CIPC SCCP >>





User Management in CUCM



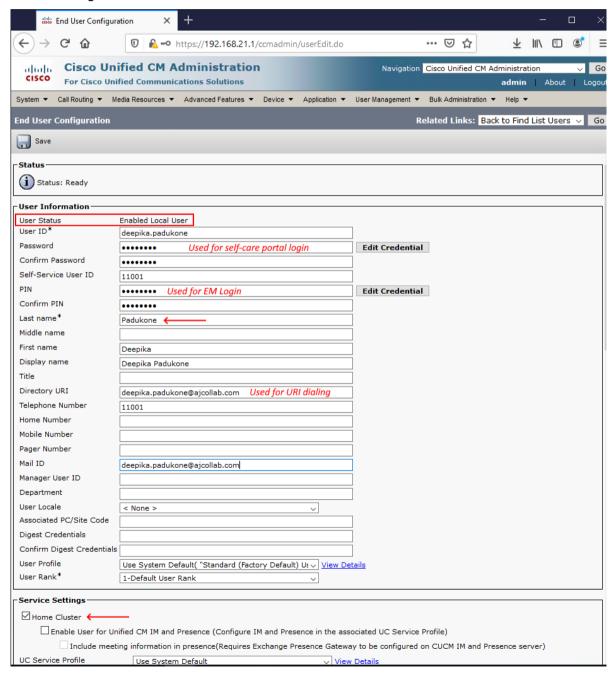
- Creating additional users in CUCM database extends the capabilities of Unified Communications network
- There are 2 types of users available in CUCM End Users and Application Users
- The Administrator user that we created at the time of installation was an Application user, that is used to login to CUCM Administration page after installing
- We have created this user only during CUCM PUB installation
- After the CUCM evolved, there are not much difference in End users and application users

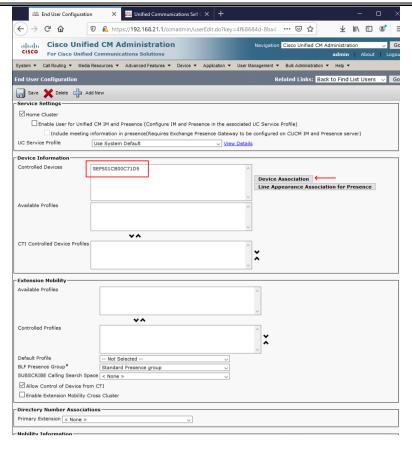
End Users

- Most of the time we deal with End users
- Third party SIP phones require end user to authenticate the registration
- Jabber IM Presence requires End Users
- End user plays a critical role in Licensing since we use user-based license these days
- End users can control phones after you associate a phone in the End User Configuration window
- Self-Care portal is designed for end users to login and manage their phone and telephony features
- We can have end users created locally on the CUCM or integrate via LDAP

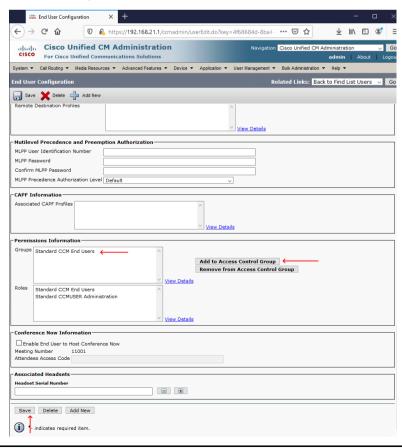
[Lab] Configure Local End User

User Management >> End User >> Add New

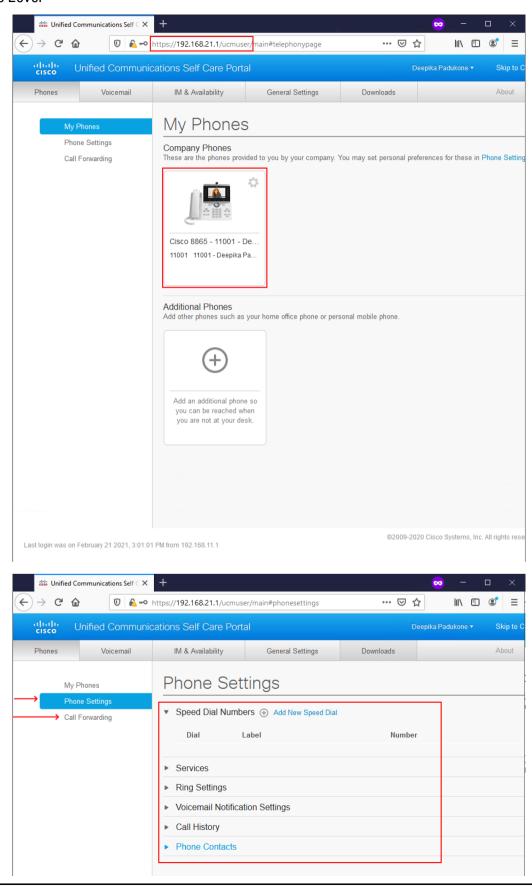


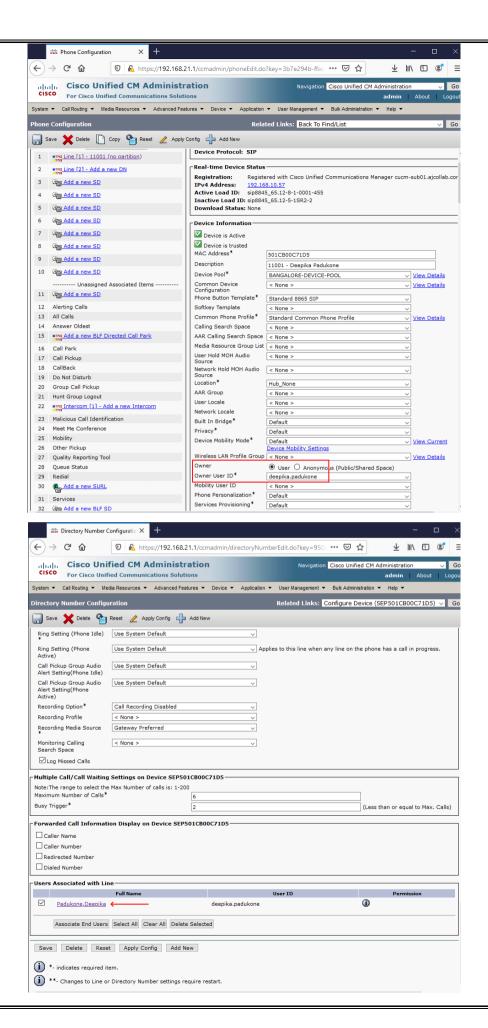


- User Group contains one or more User Roles, each User Roles will have set of privileges
- With Standard CCM End Users Group, the user can get access to Self-care portal (https://CUCM_IP/ucmuser/)

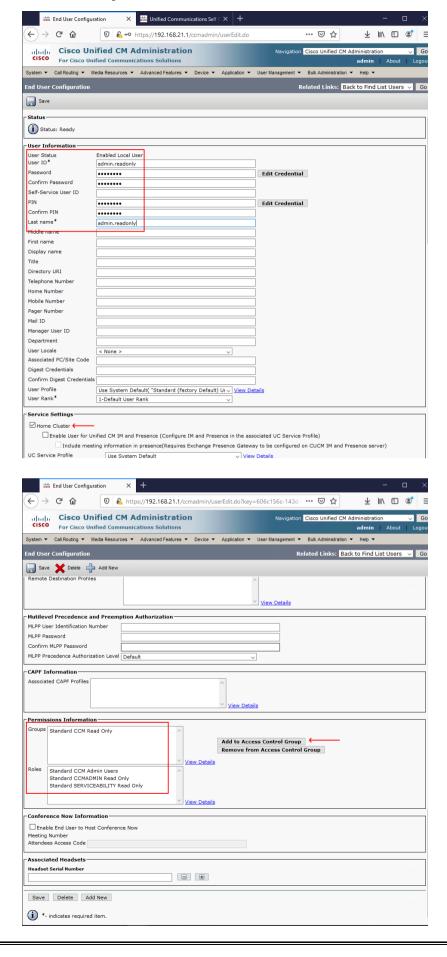


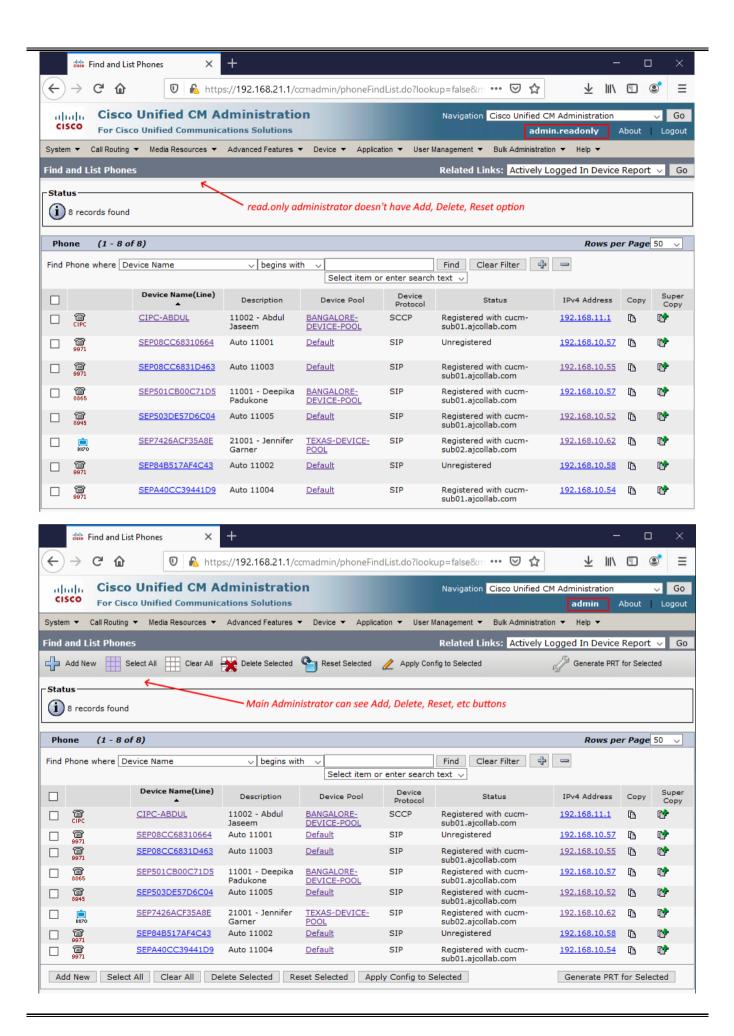
- End user can perform basic telephony features like Call Forward, Speed Dials, etc.
- Once we have the end user, it is recommended to associate the end user in the Device Level and Line Level





[Lab] Configure CUCM Read Only Administrator





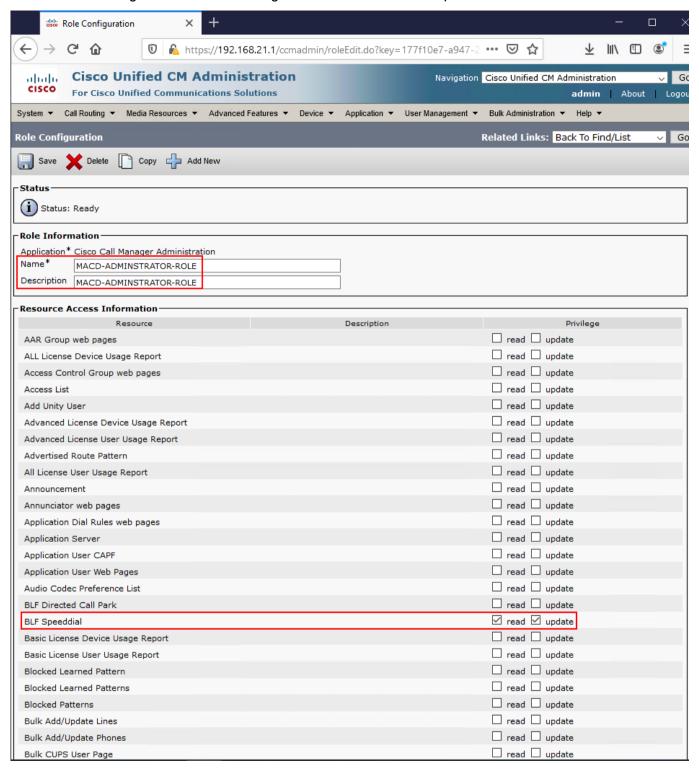
[Lab] Configure CUCM MACD Administrator

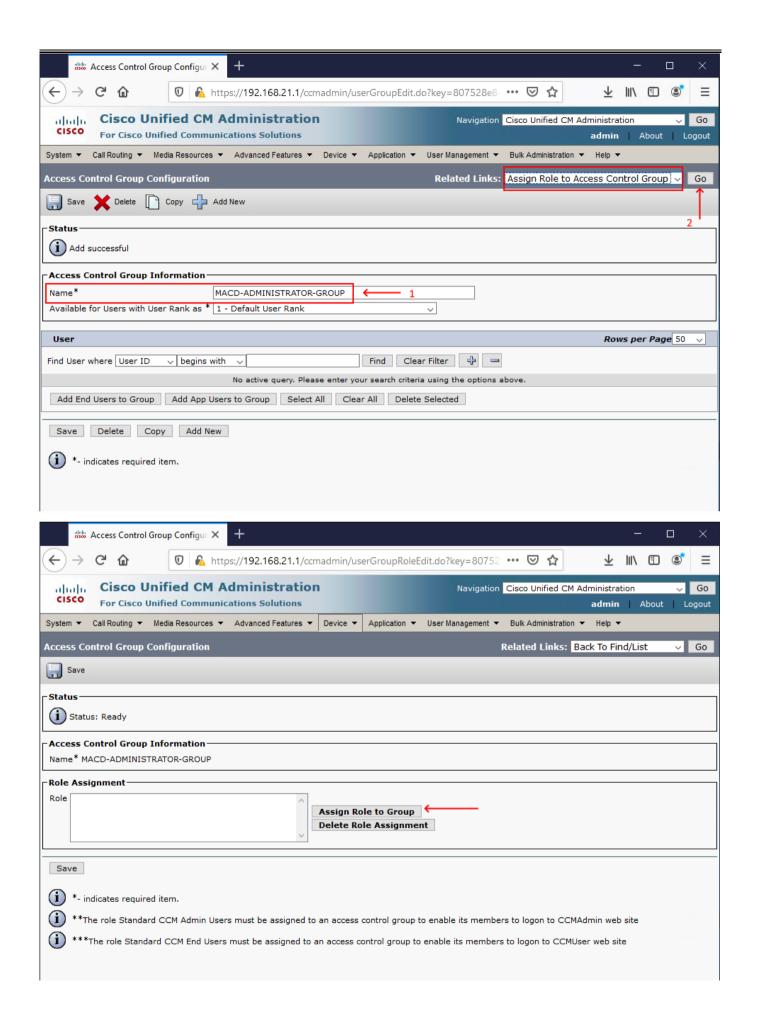
- When you start working with a company's UC infrastructure, based on your job role, you may get different access control to CUCM cluster
- If you are an L1 or MACD Engineer, you can only perform Move, Add, Change, Delete (MACD)
 roles
- This can be done via custom Access Control Group and Roles
- User Management >> User Settings >> Role >> Add New >> Application: Cisco Call Manager
 Administration >> Name: MACD-ADMINSTRATOR-ROLE

Add below roles with read and update,

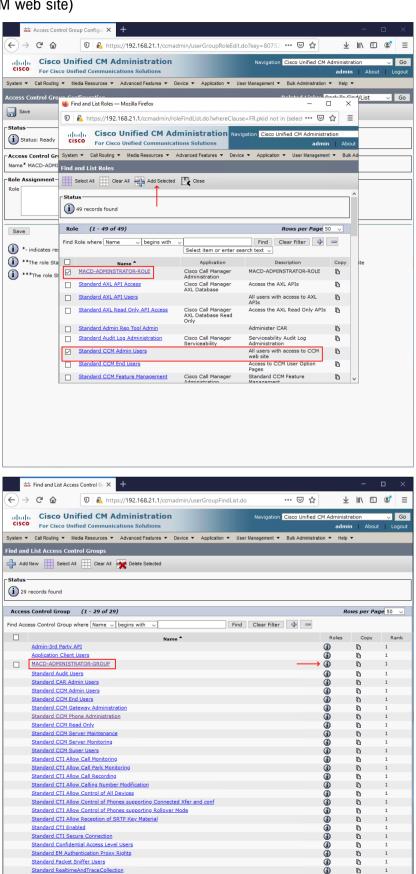
- o BLF Speeddial
- Bulk Export Phones
- o Bulk Insert UDP
- CTI Route Point web pages
- o Default Device Profile web pages
- Device Profile web pages
- Directory Number web pages
- o Firmware Load web pages
- Line Appearance web pages
- Phone Button Template web pages
- o Phone web pages
- o Reorder Info
- Softkey Template web pages

User Management >> User Settings >> Access Control Group >> Add New





Select MACD-ADMINISTRATOR-ROLE and Standard CCM Admin Users (system role provides access to CCM web site)



0

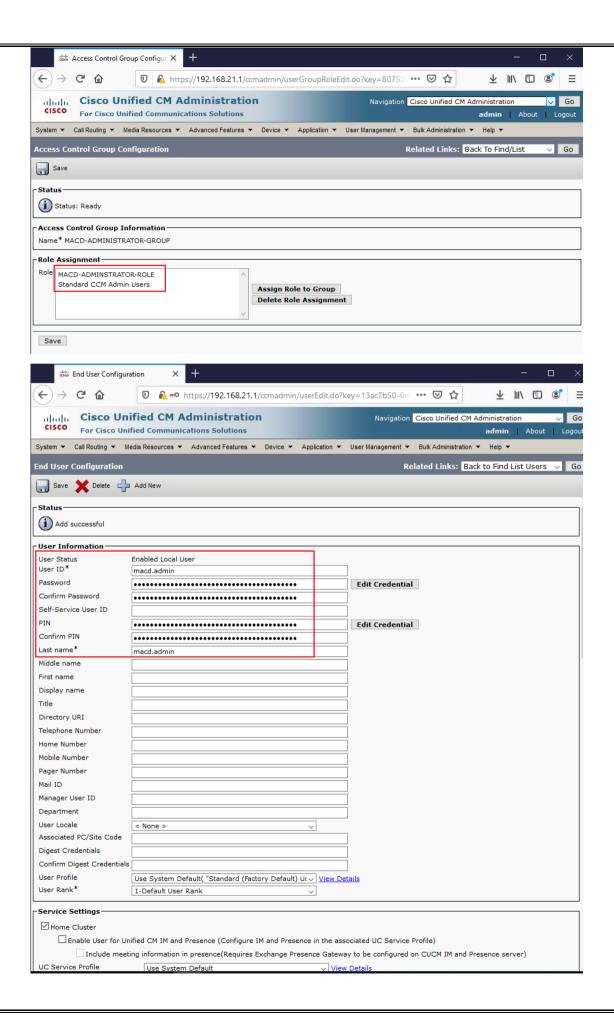
0

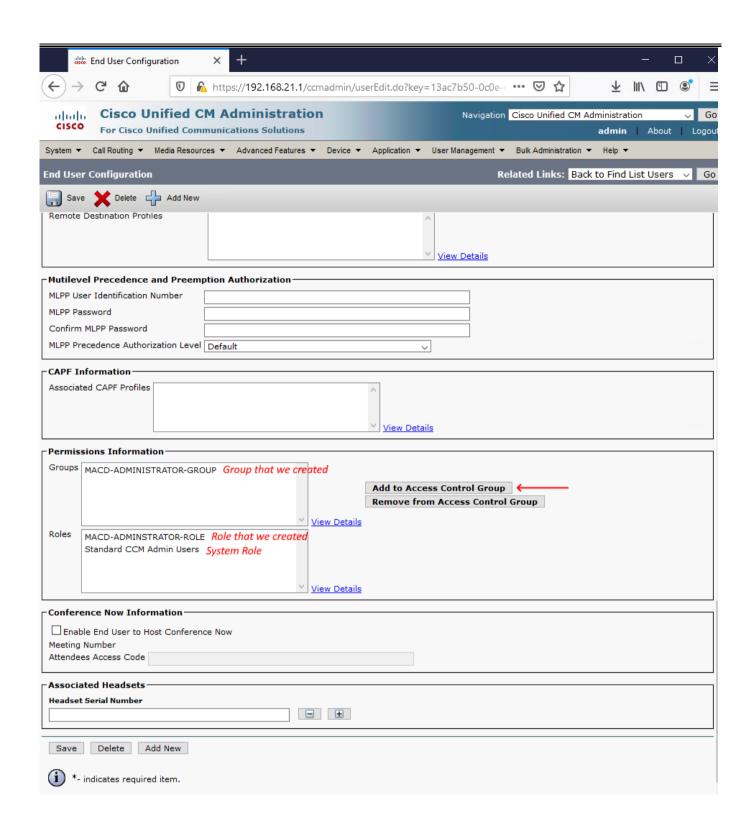
Standard CTI Allow Reception of SRTP Key Material

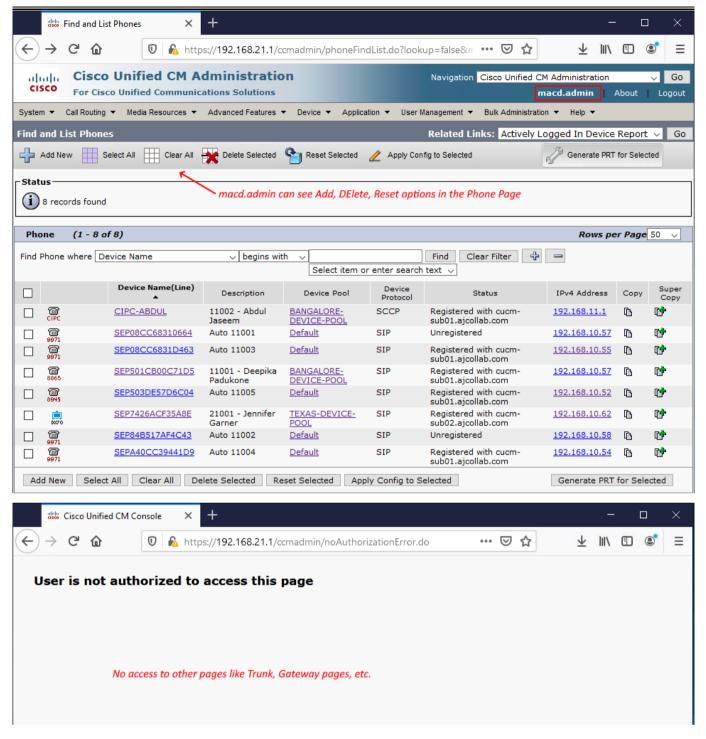
Standard CTI Enabled Standard CTI Secure Connection

Standard TabSync User
Third Party Application Users Add New | Select All | Clear All | Delete Selected

Standard Confidential Access Level Use Standard EM Authentication Proxy Rights
Standard Packet Sniffer Users
Standard RealtimeAndTraceCollection







 The end user macd.admin can only access Phone configurations but not any other options in CUCM

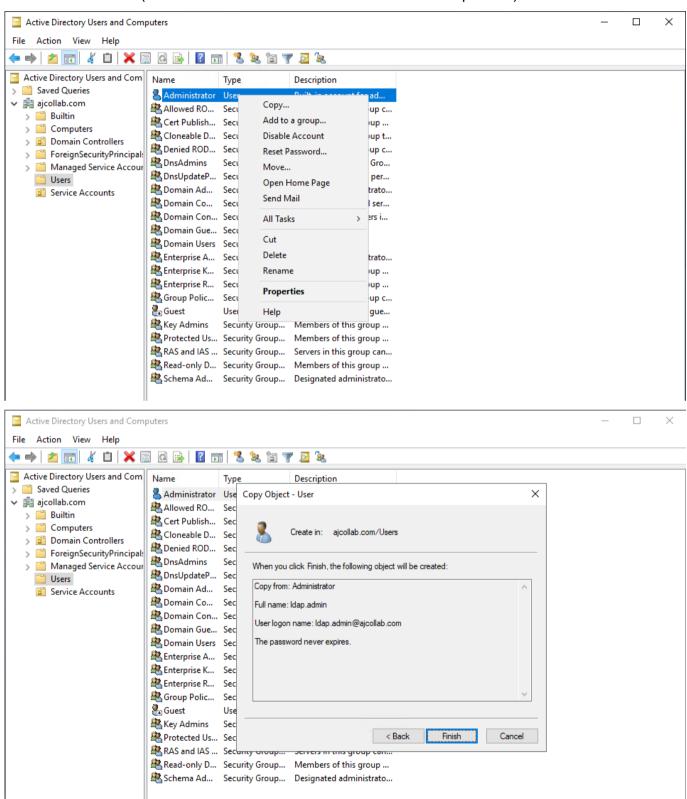
LDAP (Light Weight Directory Access Protocol)

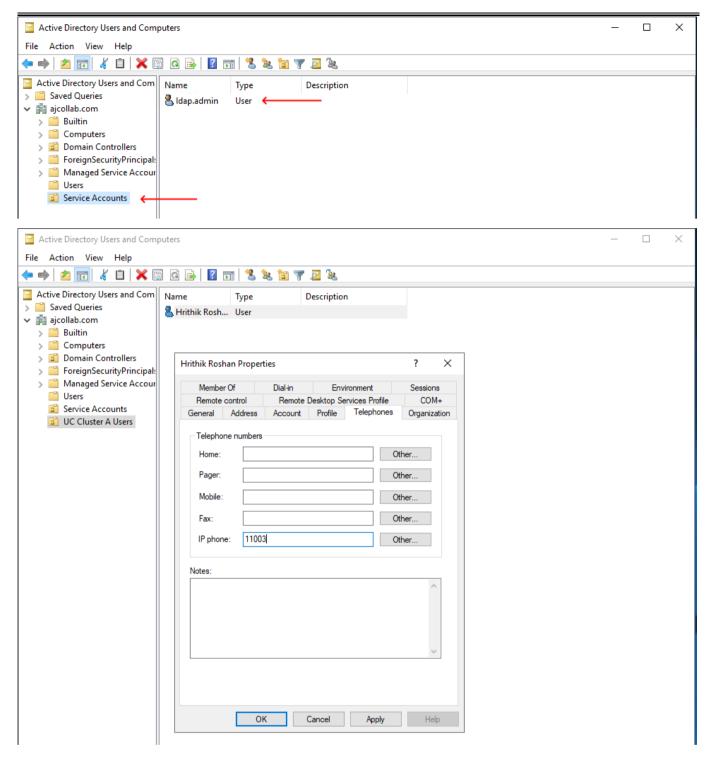


- End users can be synchronized to CUCM database from LDAP server
- Microsoft Active Directory Domain Services (ADDS) provides standard LDAP services to CUCM
- Application users cannot be synced using LDAP, it is always local to the CUCM DB
- We can enable LDAP Authentication so that the user authentication is handled by LDAP server
- Authorization is always provided from CUCM (using Group and Roles)

[LAB] Configuring LDAP

- In the Active Directory, create an OU called 'Service Accounts' and create a user inside it
- We can copy the built-in Administrator account and create 'ldap.admin' and drag to the Service
 Accounts OU (there is no defined rule like this but this is a best practice)

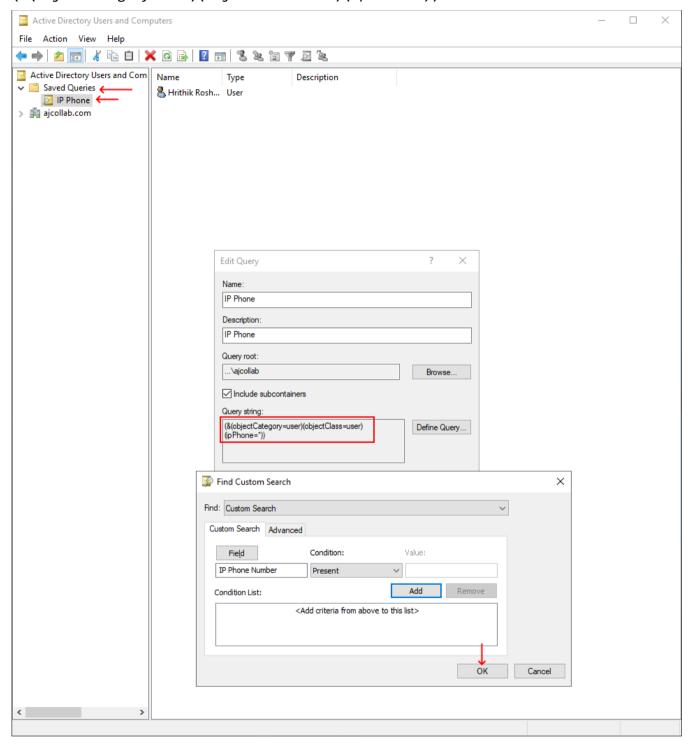




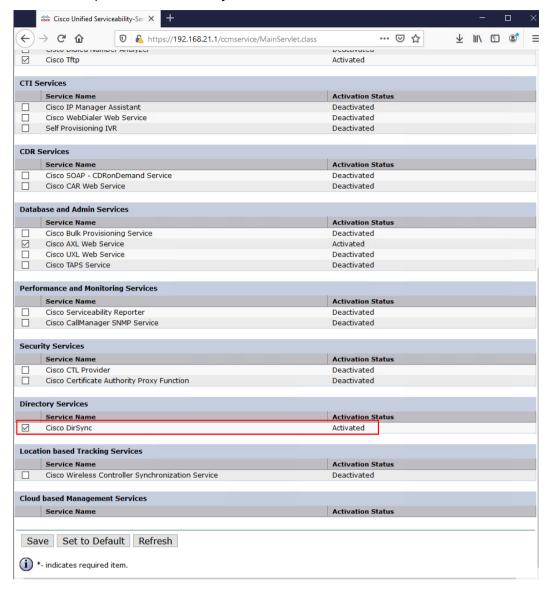
- I have created another OU UC Cluster A Users and added a user 'hrithik.roshan' there
- In the Telephone tab, I have added IP Phone field = 11003
- When I sync CUCM with LDAP, there is a chance that every user in AD will get synchronized. To avoid this, we create an LDAP Custom Filter with IP Phone field

• Custom filter for listing only users with IP Phone field is below,

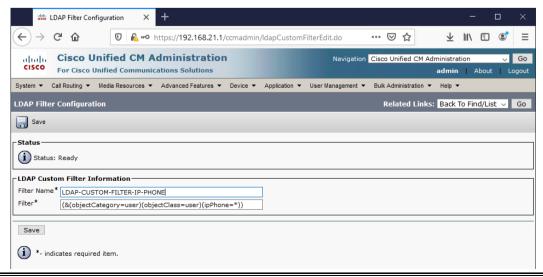
(&(objectCategory=user)(objectClass=user)(ipPhone=*))



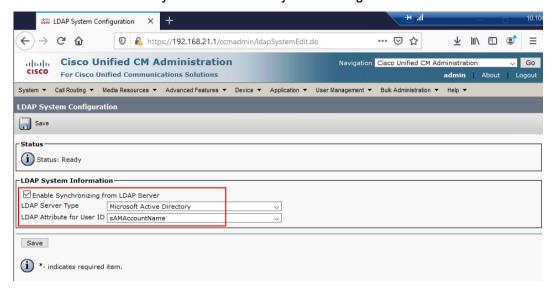
- Serviceability >> Tools >> Service Activation >> Select CUCM PUB >> Cisco DirSync
- Cisco DirSync only available in CUCM PUB
- This service is responsible for LDAP sync



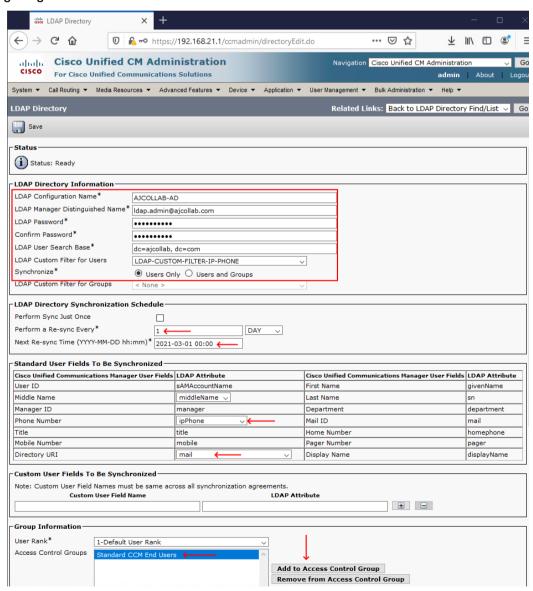
System >> LDAP >> LDAP Custom Filter >> Add New

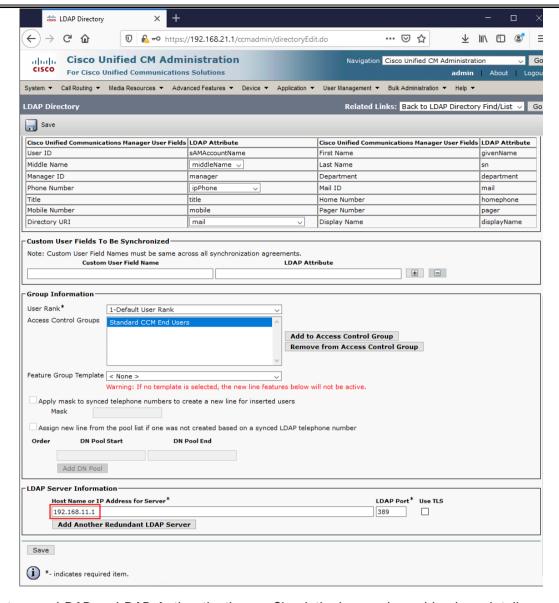


System >> LDAP >> LDAP System >> Enable Synchronizing from LDAP Server >> Save

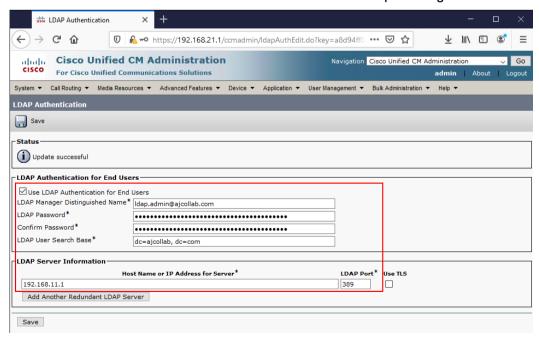


- System >> LDAP >> LDAP System >> LDAP Directory >> Add New
- Configure given details and add AD Server IP

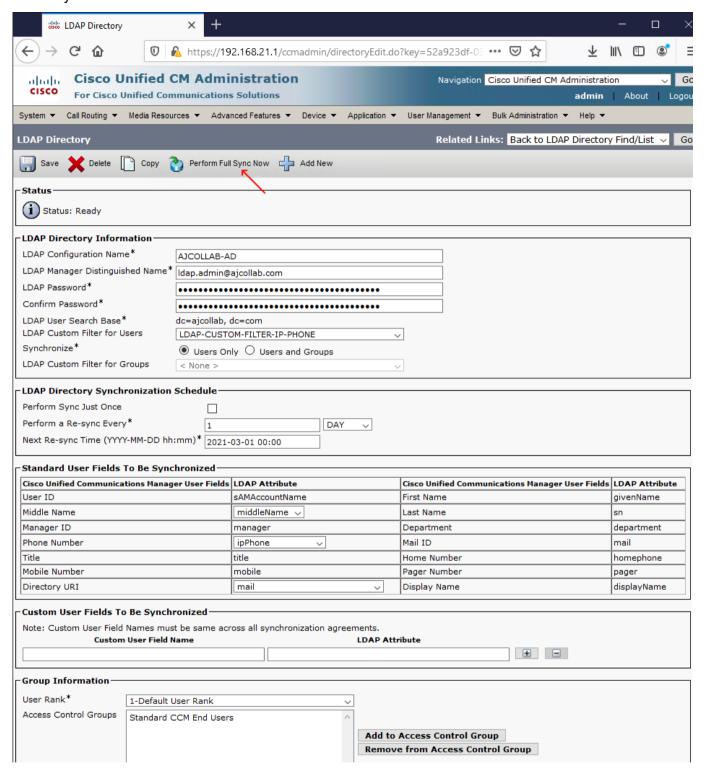




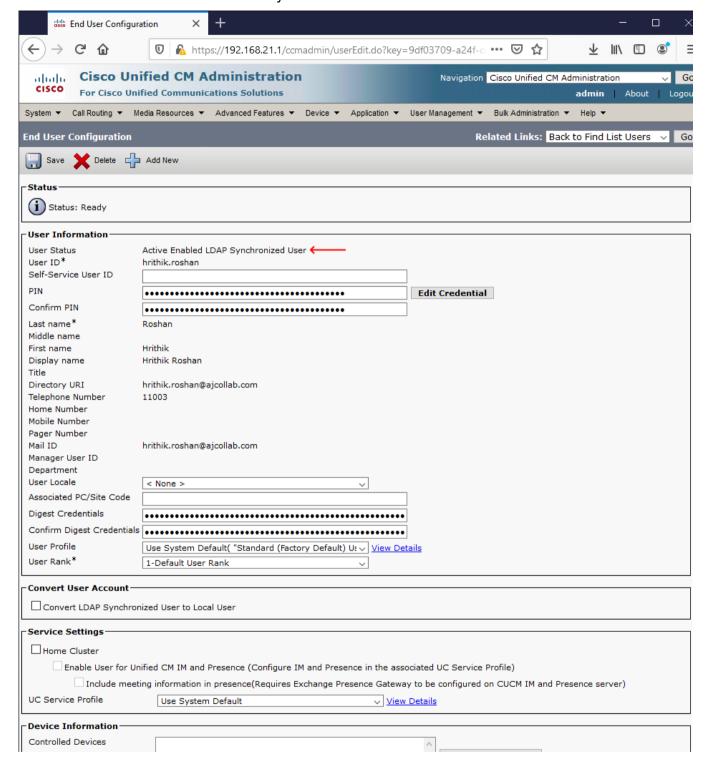
System >> LDAP >> LDAP Authentication >> Check the box and provide given details



 System >> LDAP >> LDAP System >> LDAP Directory >> Select AJCOLLAB-AD >> Perform Full Sync Now

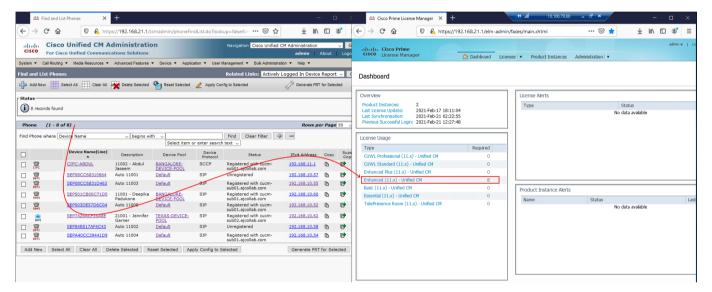


- User Management >> End User >> Find
- You can see new user has been synced from LDAP server



License Consumption

- After the installation of CUCM, we integrated it to Cisco Prime License Manager in previous class
- Now let's see how the licenses are consumed
- When you add a Cisco 8865 Phone to CUCM, it picks 'Enhanced' license
- It provides rights to: One device, including all Basic features, plus advanced (voice and video) call control features including desktop and mobile clients.
- Examples include Cisco 3911, Cisco 3951, Cisco 6941, Cisco 6945, Cisco 6961, Cisco 79xx, Cisco 89xx, Cisco 99xx, Cisco E20, Cisco TelePresence EX60, Cisco TelePresence EX90 and third-party SIP

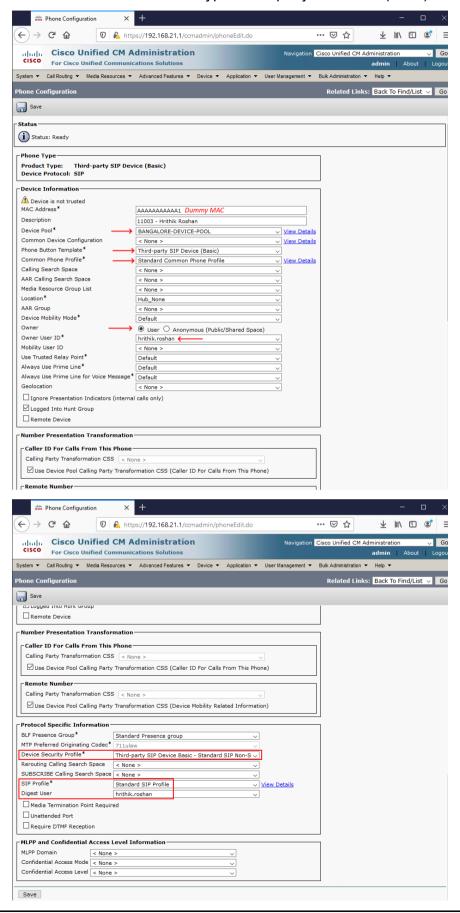


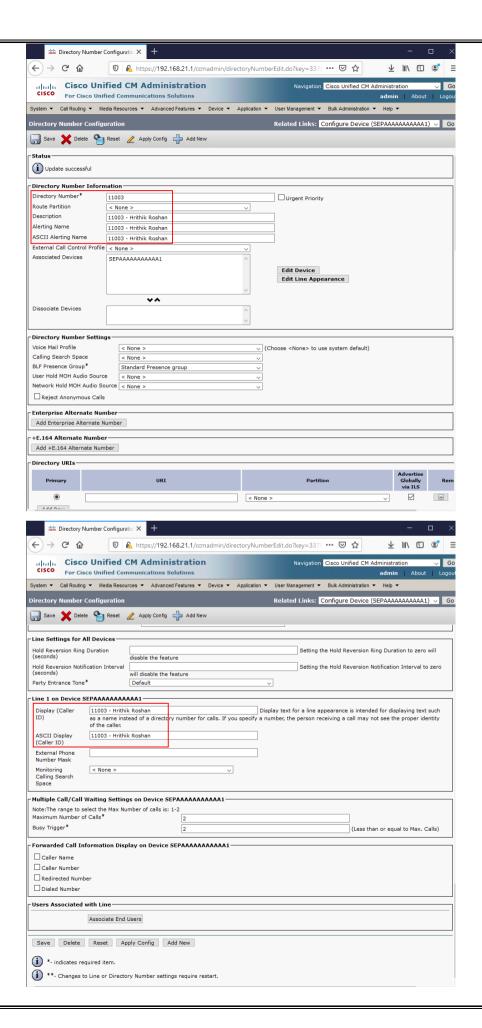
- Licenses are categorized in two Cisco User Connect Licensing (UCL) and Cisco Unified Workspace
 Licensing (UWL)
- Detailed description of these two are given in below table
- When you create a new end user and associate the phone to the user, then the Enhanced license will be free and one

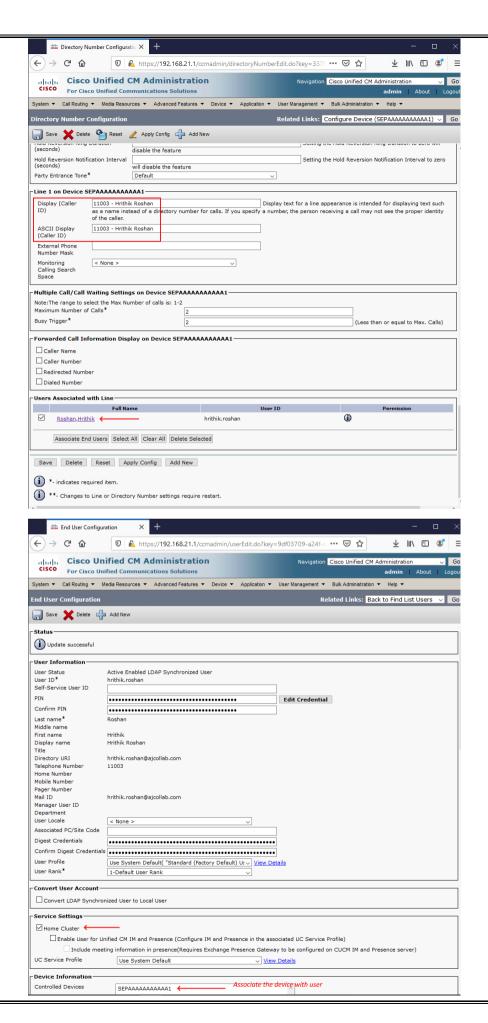
Cisco Unified Workspace Cisco User Connect Licensing (UCL) Licensing (UWL)	Essential	One device providing basic voice via analog device (phone or fax). ATA 100 ATA 107 OF
		 Examples include analog phones, ATA 186, ATA 187, Cisco 3905 and Cisco 6901
	Basic	 One device, including all Essential features, plus basic (voice and video) call control features.
		Examples include Cisco 6911 and Cisco 6921
	Enhanced	 One device, including all Basic features, plus advanced (voice and video) call control features including desktop and mobile clients.
		 Examples include Cisco 3911, Cisco 3951, Cisco 6941, Cisco 6945, Cisco 6961, Cisco 79xx, Cisco 89xx, Cisco 99xx, Cisco E20, Cisco TelePresence EX60, Cisco TelePresence EX90 and third-party SIP
	Enhanced Plus	 Advanced voice, video call control Up to 2 devices and including all Enhanced features (e.g. Desk Phone, Jabber Soft Phone) This is based on End user phone association
	Cisco Unified Workspace	 Advanced voice, video call control
	(CUWL) Standard	 Up to 10 devices per user (e.g. Desk Phone, Jabber Soft Phone, Jabber iPhone, Jabber Android, etc.)
	Cisco Unified Workspace	Advanced voice, video call control
	(CUWL) Professional	 Up to 10 devices per user (e.g. Desk Phone, Jabber Soft Phone, Jabber iPhone, Jabber Android, etc.)
		Professional collaboration workspace application features

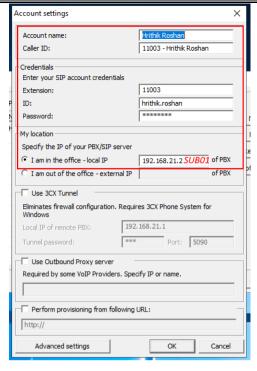
[Lab] Third Party SIP Phone Registration

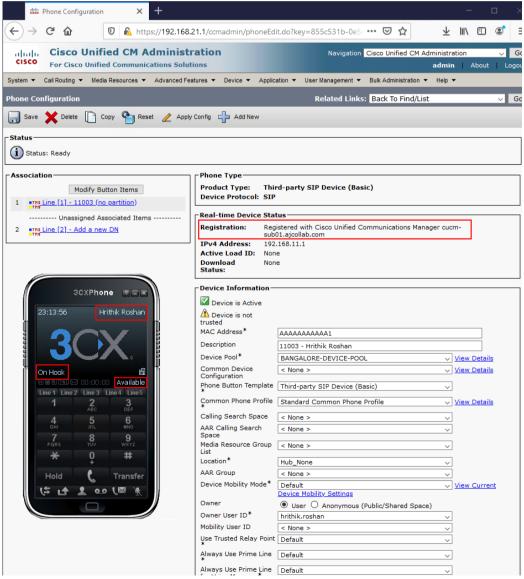
Device >> Phone >> Add New >> Phone Type: Third-party SIP Device (Basic)





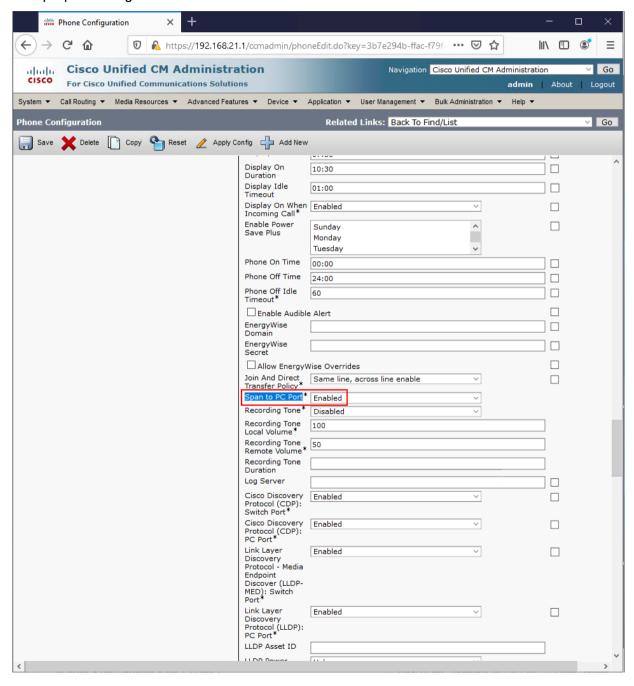




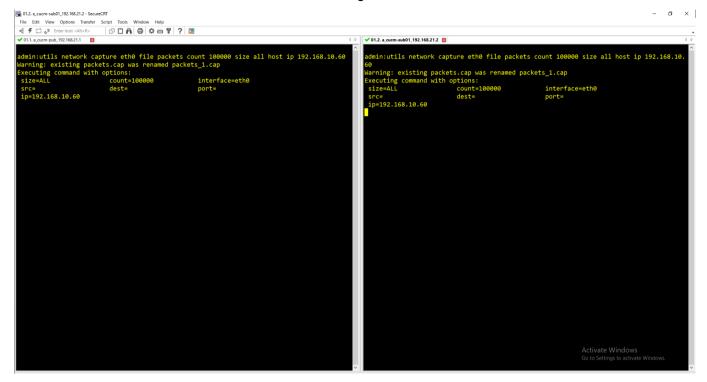


IP Phone Packet Capture Procedure

- We can take PCAPs either from Phone PC Port locally or from CUCM Node remotely
- If we want to take PCAPs from Phone directly, connect a Laptop or PC (With Wireshark installed) on the PC Port of the Phone
- Enable 'Span to PC Port' option in the Phone configuration page and the start capturing the NIC of Laptop/PC using Wireshark



- To enable PCAPS from CUCM, use below command utils network capture eth0 file packets count 100000 size all host ip 192.168.10.60
 - This will start capturing Packets till you hit Control + C
 - You need to do this on TFTP as well as the target CUCM Subscriber node



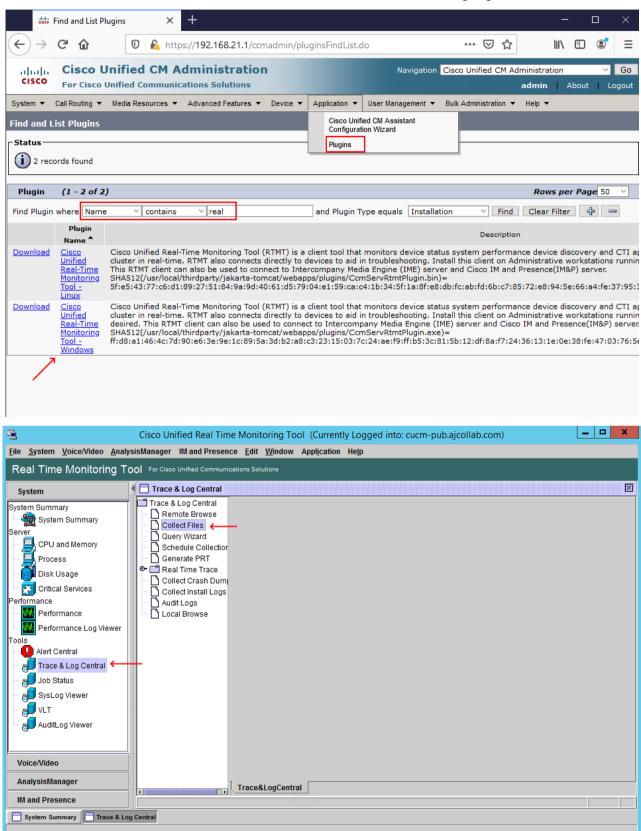
- If you are not sure about the Phone IP, then you can collect all packets using below command utils network capture eth0 file packets count 100000 size all
 - This will have some serious CPU utilization on the node

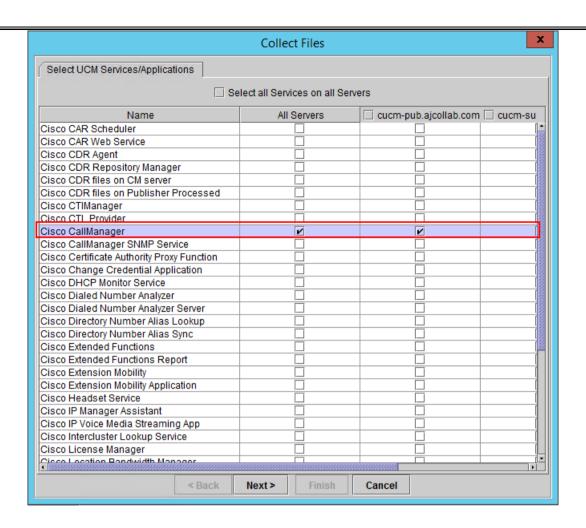
- PCAPs will be available at platform/cli/ location
- You can collect the PCAP either from CLI to an SFTP Server or from RTMT

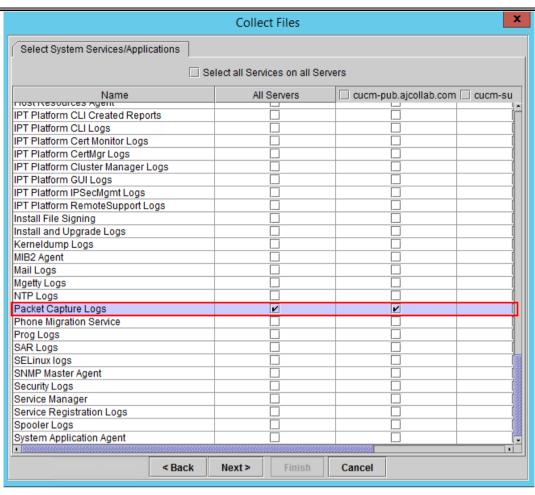
```
admin:utils network capture eth0 file packets count 100000 size all
Executing command with options:
size=ALL
                           count=100000
                                                     interface=eth0
                           dest=
src=
                                                      port=
ip=
Control-C pressed \leftarrow
admin:file list activelog platform/cli/
packets.cap
dir count = 0, file count = 1
admin:
admin:file get activelog platform/cli/packets.cap
Please wait while the system is gathering files info ... Get file: active/platform/cli/packets.cap
done.
Sub-directories were not traversed.
Number of files affected: 1
Total size in Bytes: 54112785
Total size in Kbytes: 52844.516
Would you like to proceed [y/n]? y
SFTP server IP: 192.168.11.5
SFTP server port [22]:
User ID: sftp.admin
Password: *******
Download directory: /
```

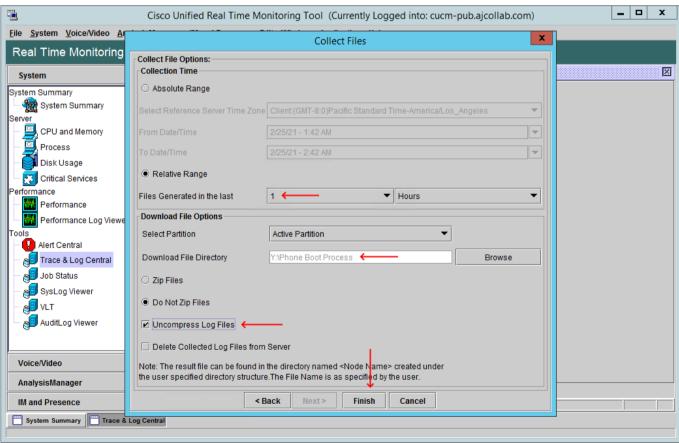
RTMT To Collect Logs and PCAPs from CUCM Cluster

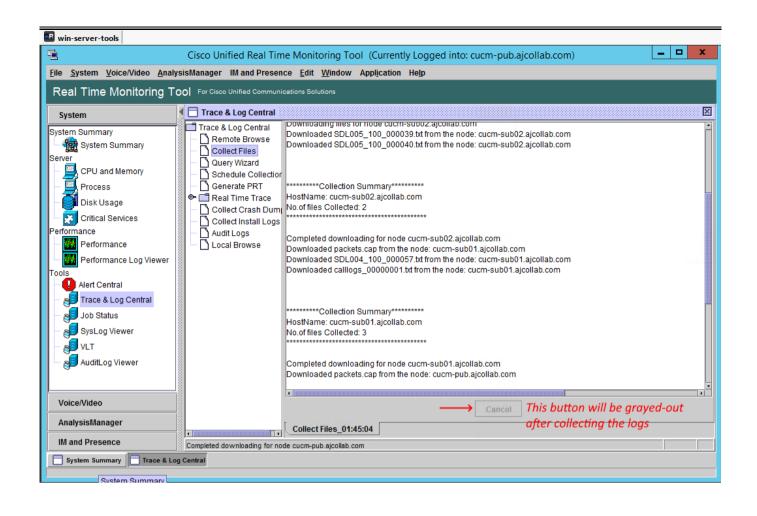
- We can download RTMT from Applications >> Plugins Menu
- RTMT can be used to Monitor the CUCM cluster as well as collecting log files





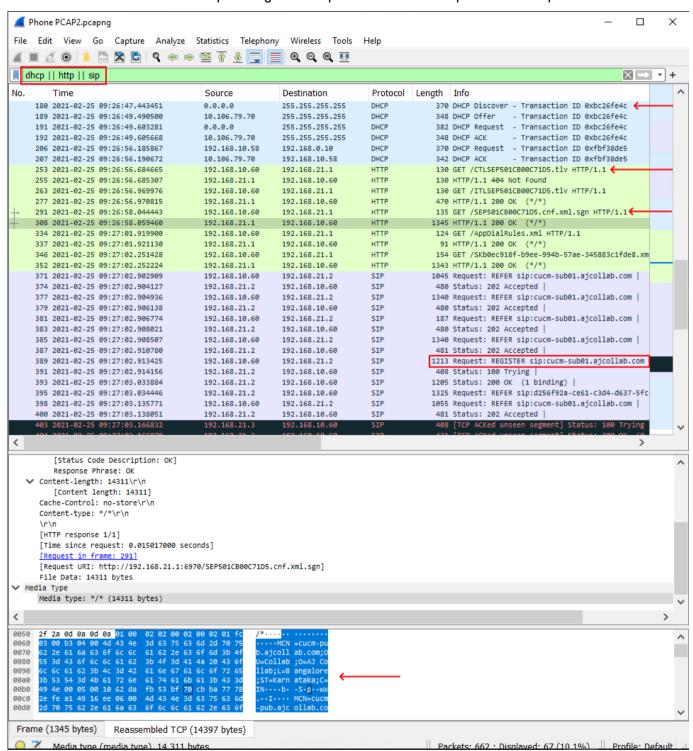






IP Phone Bootup Process and Registration

- When you connect an IP Phone to a network, there are some key steps that the IP Phone will go through to get register with CUCM
- We will now see the complete registration process with the help of Packet Captures



Note: This is the PCAP from Phone Side. CUCM PCAPs won't have the DHCP part since DHCP is provided by the Lab switch.

Sample PCAPs can be downloaded from here: IP Phone Packet Capture from Phone Side

1. Obtain Power

- IP Phone gets power either from PoE Switch or from external power adapter
- When IP Phones are connected to PoE Switch, Phones will send Fast Link Pulse (FLP 147K Hz) to the Switch
- Up on receiving the FLP, switch will realize that the device requesting power and it delivers the maximum power over Ethernet cable
- Once the IP Phone boots-up, with the help of CDP messages, switch can understand that the device doesn't require maximum power and eventually reduces to the required level based on the Phone model

2. Run Bootstrap Loader

 IP Phone initializes the firmware image and boots from it, this will initialize the hardware and software.

3. Obtain VLAN Information

- IP Phone acts as a mini switch, meaning we can connect a co-located PC on IP Phone
- PC will get Data VLAN and Phone will get Voice VLAN
- With the help of CDP messages, Phone can understand the configured Voice and Data VLANs

4. Obtain IP Address from DHCP Server

- Phone uses port number 68 and DHCP Server use 67
- Phone will broadcast DHCP DISCOVER message
- Only the DHCP server will reply to that message by DHCP OFFER (L2 Unicast) contains Pool of IP Address, Subnet Mask, Default Gateway, TFTP Server IP, DNS Server, Domain Name and lease duration
- Now the Phone will send a broadcast **REQUEST** for one IP from this pool
- DHCP Server will ACK the message (L2 Unicast) saying that it accepts the request and mapped the
 IP to the client
- This process is known as DORA process in DHCP

5. Request CTL File

- IP Phone firmware is programmed to request CTL file from the TFTP Server after getting TFTP server IP from DHCP Server
- CTL file will be available only on secure clusters, hence most of the time Phone gets 404 Not Found from the TFTP Server
- CTL is used for encrypted Signaling and Media

6. Request ITL File

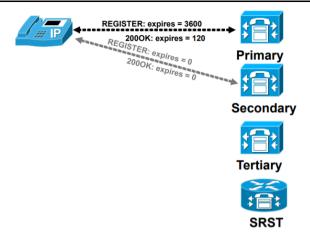
- IP Phone requests ITL file from the TFTP. CUCM V8 or later will provide ITL file by default
- The Phone configuration file will be signed by default and Phone uses ITL to verify the identity of TFTP server and decrypt the configuration file
- When an IP Phone downloads ITL file, then, it will not trust any other TFTP server. That is the reason we clear ITL file while connecting phone from one cluster to another cluster
- Signed configuration file will have '.sgn' at the end

7. Download Configuration File

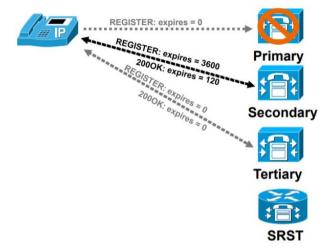
- Now the IP Phone send a request for SEP<MAC_ADDRESS>.cnf.xml.sgn
- This file will have the CUCM Node information to get register. This will also have details like firmware image, Date / Time format, etc.

8. Register with Primary CUCM Node

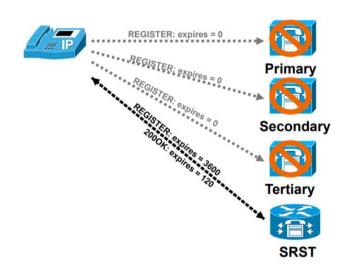
- Phone will send SIP Register message to the primary CUCM listed in the phone configuration file
- Phone will send a keep-alive every 120 seconds to the primary CUCM server
- In the initial REGISTER message from Phone, the EXPIRE timer set to 3600 seconds. As a response to this CUCM sends ACK by modifying the EXPIRE timer to 120 seconds
- Therefore, the phone sends keep-alive every 120 seconds (actually 115 seconds which is 120 minus the delta value configured in SIP profile, which is 5 seconds by default)
- In this case, the phone sends keep-alive every 115 seconds



- Phone exchanges the Register message to Backup CUCM with Expires field set to 0
- If primary goes down, phone switches over to secondary and establishes keepalive connection,
 begins polling primary periodically in order to fall back when it becomes available again

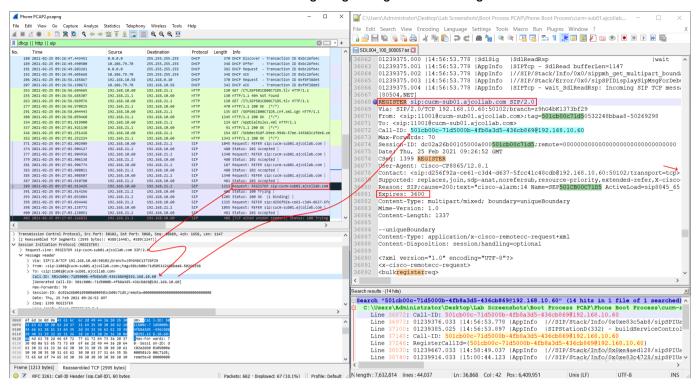


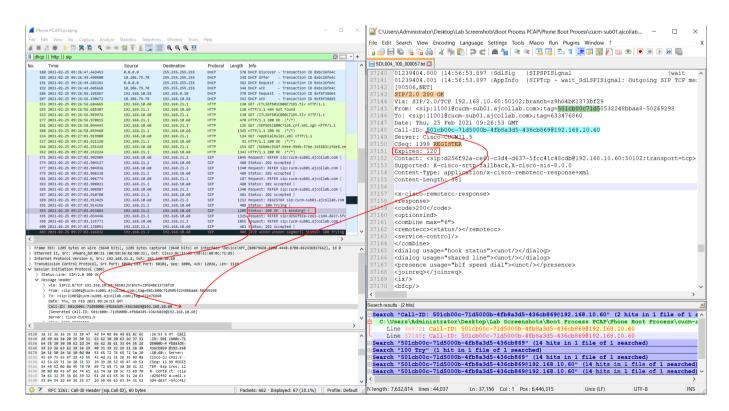
- When Phone is registered with Secondary CUCM, it maintains REGISTER messages with Primary and tertiary CUCM nodes
- When all the three nodes are down, Phone will try to register with SRST



IP Phone Registration CCM Logs

 We have collected the Cisco CallManager logs while collecting the PCAPs, let's see what the information are available in the CCM Logs regarding the Phone registration





Auto Registration CCM Logs

- I have taken an example of a Phone Auto Registration
- Phone MAC: 08CC6831D463; Phone IP 192.168.129.1; PUB+TFTP: 192.168.21.1; SUB01: 192.168.21.2; SUB02: 192.168.21.3
- Phone Sends Register Message to the SUB01 CUCM where we have enabled Auto Registration

```
00008705.001 |23:22:11.941 |AppInfo |//SIP/SIPUdp/wait SdlDataInd: Incoming SIP UDP
message size 1049 from 192.168.129.1:[5060]:
[71, NET]
REGISTER sip: 192.168.21.2 SIP/2.0
Via: SIP/2.0/UDP 192.168.129.1:5060; branch=z9hG4bK6763daf2
From: <sip: AUTO-REG@192.168.21.2>; tag=08cc6831d46300045c438d60-6315c0fe
To: <sip:AUTO-REG@192.168.21.2>
Call-ID: 08cc6831-d4630003-44a0964c-54e6773c@192.168.129.1
Max-Forwards: 70
Date: Fri, 01 Jan 1982 00:18:23 GMT
CSeq: 101 REGISTER
User-Agent: Cisco-CP9971/9.4.2
Contact: <sip:b5677e67-31dc-43ca-86f5-
3ee19ef2345e@192.168.129.1:5060;transport=udp>;+sip.instance="<urn:uuid:00000000-
0000-0000-0000-
08cc6831d463>";+u.sip!devicename.ccm.cisco.com="SEP08CC6831D463";+u.sip!model.ccm.ci
sco.com="493";video
Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-
cisco-callinfo, X-cisco-serviceuri, X-cisco-escapecodes, X-cisco-service-control, X-
cisco-srtp-fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-
8.0.1
Content-Length: 0
Reason: SIP;cause=200;text="cisco-alarm:14 Name=SEP08CC6831D463
ActiveLoad=sip9971.9-4-2SR4-1.loads InactiveLoad=sip9971.9-4-2SR2-2.loads Last=cm-
closed-tcp"
Expires: 3600
```

CUCM Creates an Entry for the Phone in its DB and allocates a free DN

CUCM Sends a Notify Message to IP Phone to go for a restart

00009179.001 |23:23:06.566 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 192.168.129.1:[5060]:

[101,NET]

NOTIFY sip:AUTO-REG@192.168.129.1:5060 SIP/2.0

Via: SIP/2.0/UDP 192.168.21.2:5060;branch=z9hG4bK3743b8bac

From: <sip:192.168.21.2>;tag=1645647548

To: <sip:AUTO-REG@192.168.129.1>

Call-ID: f1946280-dc1af62-4-215a8c0@192.168.21.2

CSeq: 101 NOTIFY Max-Forwards: 70

Date: Wed, 30 Jun 2021 17:52:34 GMT

User-Agent: Cisco-CUCM11.5 Event: service-control Subscription-State: active

Contact: <sip:192.168.21.2:5060>

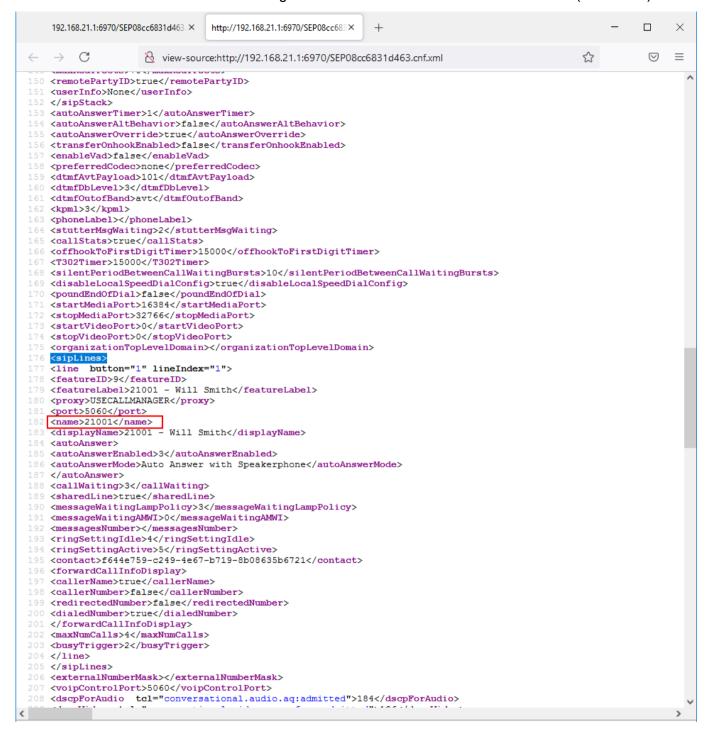
Content-Type: text/plain

Content-Length: 84

action=restart

RegisterCallId={ 08cc6831-d4630003-44a0964c-54e6773c@192.168.129.1}

❖ At this point CUCM DB has an entry for the particular phone and its Configuration file has been created in TFTP as well. The configuration file contains the SIP Line information (DN 21001)



During the rebooting stage, now the phone gets this configuration file and sends another Register message to CUCM with extension 21001 00009278.004 |23:23:47.172 |AppInfo |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 192.168.129.1 on port 51584 index 3 with 2366 bytes: [106,NET] REGISTER sip:192.168.21.2 SIP/2.0 Via: SIP/2.0/TCP 192.168.129.1:51584;branch=z9hG4bK581b5ee2 From: <sip: 21001@192.168.21.2>; tag=08cc6831d463000414fa4229-7106803b To: <sip:21001@192.168.21.2> Call-ID: 08cc6831-d4630003-33899d07-6ebc2d9b@192.168.129.1 Max-Forwards: 70 Date: Fri, 01 Jan 1982 00:00:49 GMT CSeq: 101 REGISTER User-Agent: Cisco-CP9971/9.4.2 Contact: <sip:f644e759-c249-4e67-b719-8b08635b6721@192.168.129.1:51584; transport=tcp>; +sip.instance="<urn:uuid:00000000 -0000-0000-0000-08cc6831d463>";+u.sip!devicename.ccm.cisco.com="SEP08CC6831D463";+u.sip!model.ccm.ci sco.com="493" Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, Xcisco-callinfo, X-cisco-serviceuri, X-cisco-escapecodes, X-cisco-service-control, Xcisco-srtp-fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-8.0.1 Reason: SIP;cause=200;text="cisco-alarm:14 Name=SEP08CC6831D463 ActiveLoad=sip9971.9-4-2SR2-2.loads InactiveLoad=sip9971.9-4-2SR4-1.loads Last=cmclosed-tcp" Expires: 3600 Content-Type: multipart/mixed; boundary=uniqueBoundary Mime-Version: 1.0 Content-Length: 1249 CUCM SUB01 will confirm the registration with 2000K. All the logs are available in the CUCM SUB01 server 00009301.001 |23:23:47.286 |AppInfo |SIPTcp - wait SdlSPISignal: Outgoing SIP TCP message to 192.168.129.1 on port 51584 index 3 [108,NET] SIP/2.0 200 OK Via: SIP/2.0/TCP 192.168.129.1:51584;branch=z9hG4bK581b5ee2 From: <sip:21001@192.168.21.2>;tag=08cc6831d463000414fa4229-7106803b To: <sip:21001@192.168.21.2>;tag=1491078936 Date: Wed, 30 Jun 2021 17:53:47 GMT Call-ID: 08cc6831-d4630003-33899d07-6ebc2d9b@192.168.129.1 Server: Cisco-CUCM11.5 CSea: 101 REGISTER Expires: 120 Contact: <sip:f644e759-c249-4e67-b719-8b08635b6721@192.168.129.1:51584; transport=tcp>; +sip.instance="<urn:uuid:00000000 -0000-0000-0000-08cc6831d463>";+u.sip!devicename.ccm.cisco.com="SEP08CC6831D463";+u.sip!model.ccm.ci sco.com="493";x-cisco-newreg

Supported: X-cisco-srtp-fallback, X-cisco-sis-8.0.0 Content-Type: application/x-cisco-remotecc-response+xml

Content-Length: 367

❖ If we take a look at SUB02 Logs, we will see another backup Register request with Expire set to 0 00008660.002 | 23:23:48.898 | AppInfo | SIPTcp - wait SdlReadRsp: Incoming SIP TCP message from 192.168.129.1 on port 52141 index 3 with 904 bytes: [61, NET] REGISTER sip:192.168.21.3 SIP/2.0 Via: SIP/2.0/TCP 192.168.129.1:52141;branch=z9hG4bK70a3b429 From: <sip: 21001@192.168.21.3>; tag=08cc6831d46300060bacb5a2-41cc1749 To: <sip:21001@192.168.21.3> Call-ID: 08cc6831-d4630002-73b16e1e-656be1fe@192.168.129.1 Max-Forwards: 70 Date: Wed, 30 Jun 2021 17:53:48 GMT CSeq: 101 REGISTER User-Agent: Cisco-CP9971/9.4.2 Contact: <sip:f644e759-c249-4e67-b719-8b08635b6721@192.168.129.1:52141; transport=tcp>; +sip.instance="<urn:uuid:00000000 -0000-0000-0000-08cc6831d463>";+u.sip!devicename.ccm.cisco.com="SEP08CC6831D463";+u.sip!model.ccm.ci sco.com="493";expires=0;cisco-keep-alive Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, Xcisco-callinfo, X-cisco-serviceuri, X-cisco-escapecodes, X-cisco-service-control, Xcisco-srtp-fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-8.0.1

Content-Length: 0

Expires: 0

SUB02 will respond with 2000K

00008664.001 |23:23:48.899 |AppInfo |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 192.168.129.1 on port 52141 index 3 [63,NET]

SIP/2.0 200 OK

Via: SIP/2.0/TCP 192.168.129.1:52141;branch=z9hG4bK70a3b429

From: <sip:21001@192.168.21.3>;tag=08cc6831d46300060bacb5a2-41cc1749

To: <sip:21001@192.168.21.3>;tag=116502652

Date: Wed, 30 Jun 2021 17:53:48 GMT

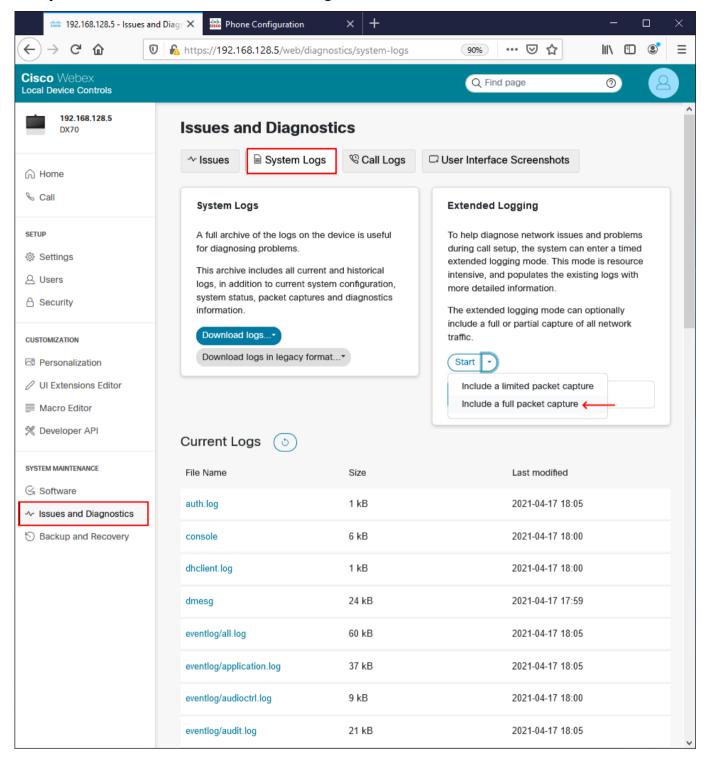
Call-ID: 08cc6831-d4630002-73b16e1e-656be1fe@192.168.129.1

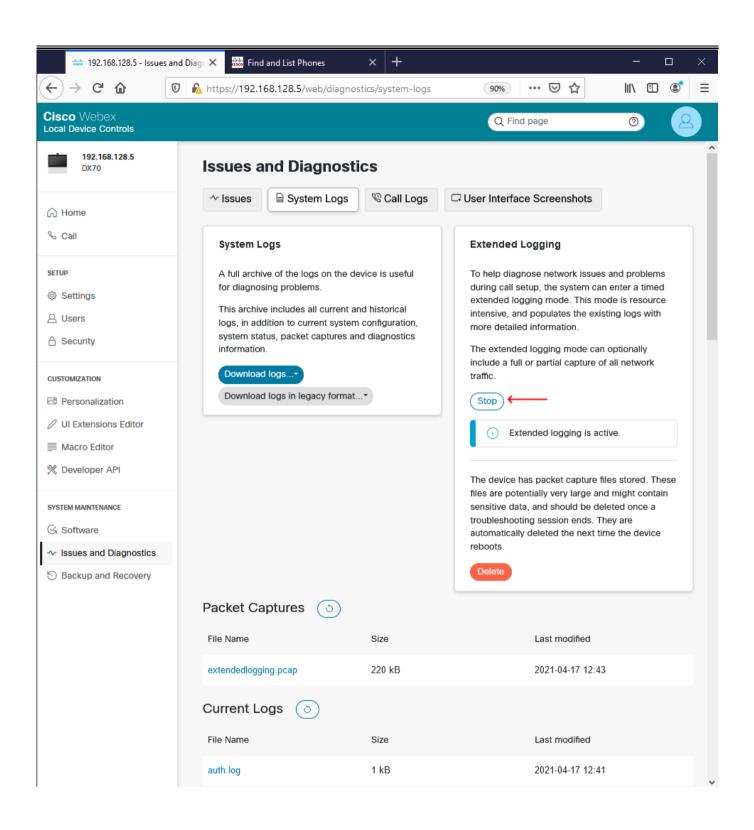
CSeq: 101 REGISTER

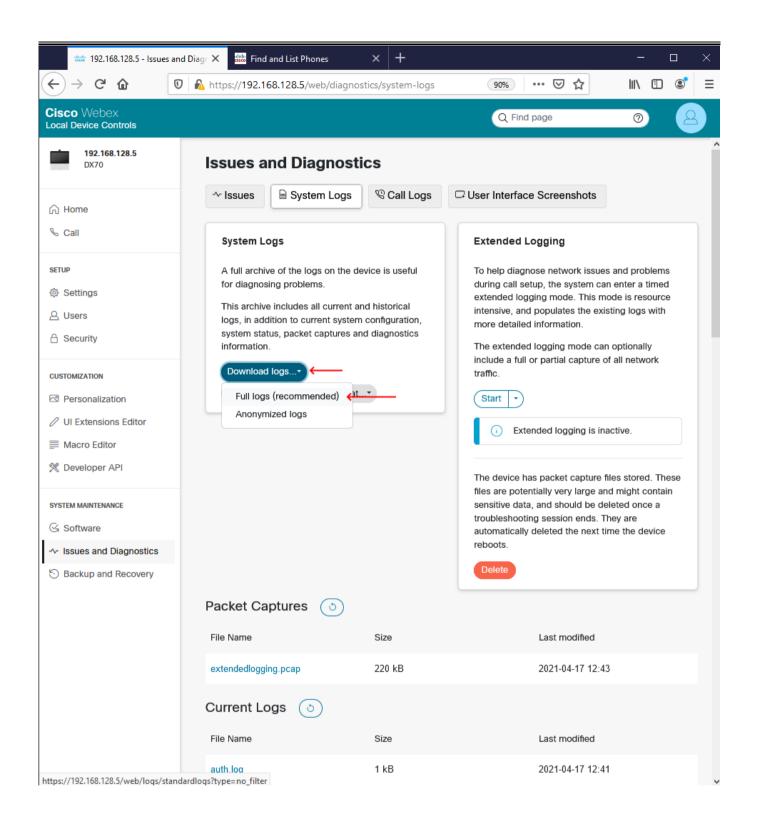
Expires: 0

Content-Length: 0

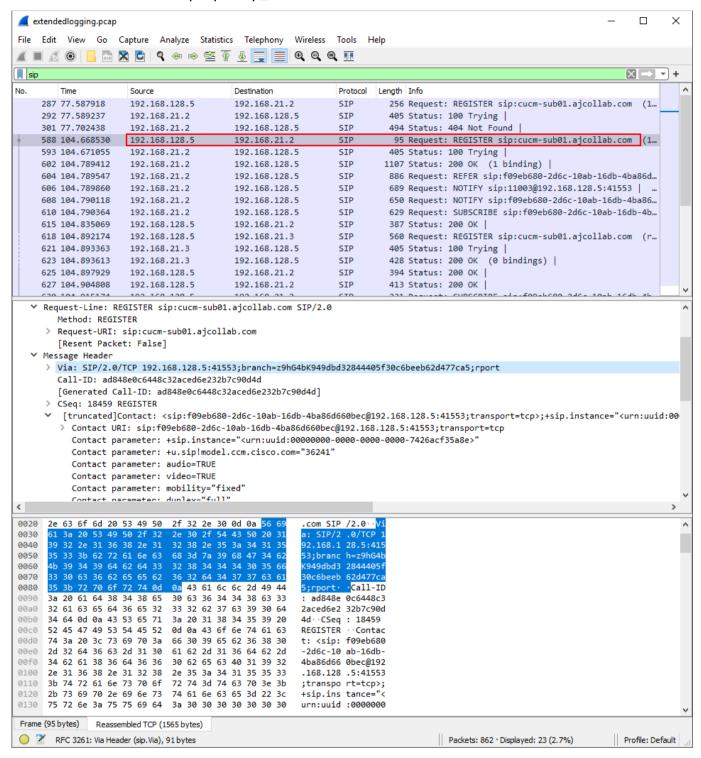
Telepresence Device DX70 PCAP and Logs







PCAPs will be in \tmp\tcpdump_files



- xconfig.txt: Configuration summary
- xstatus.txt: Device and peripheral status summary
- \var\log\eventlog\all.log: Complete device logs
- \tmp\tcpdump_files\extendedlogging.pcap: Complete Packet Capture

The above logs can be downloaded from here: DX70 Logs

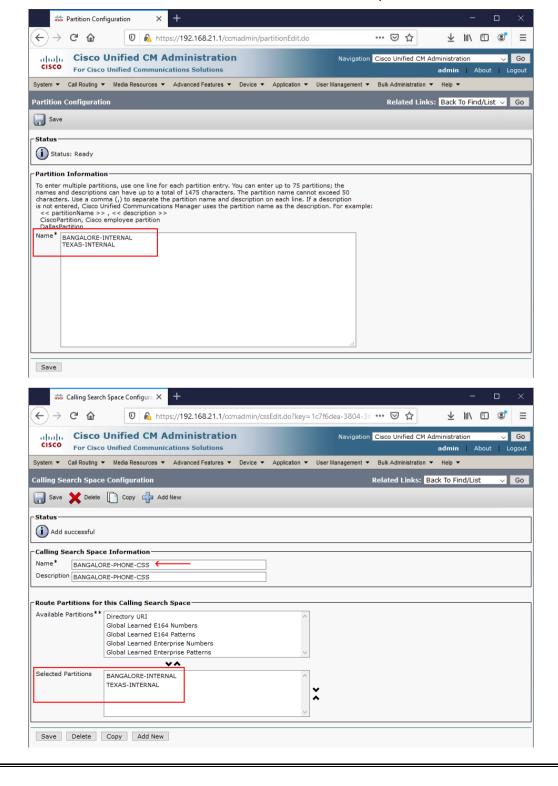
Calling Search Space (CSS) and Partition

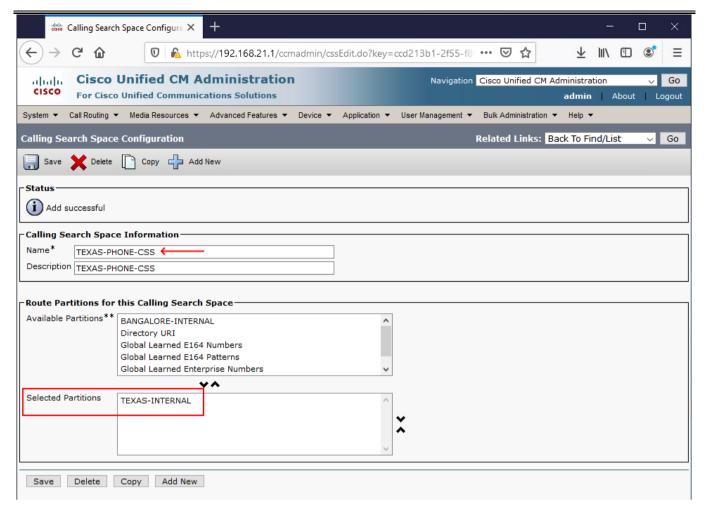


- Calling privileges (who can call what) in CUCM cluster can be implemented using CSS (Calling Search Space) and Partitions
- For example, some users can call Internal number only and some other users are allowed to call
 internal as well as Local external PSTN calls whereas other set of users are allowed to call Internal,
 Local, National and International numbers
- CSS and Partitions are also used to implement TEHO (Tail End Hop Off) that allows organizations
 to save money on PSTN toll charges by routing long distance and international call across the
 private IP WAN. TEHO is an application of LCR (Least Cost Routing)
- A partition is label attached to a directory number (DN) with similar reachability, for e.g. Internal numbers are labeled to INTERNAL-PT
- Partitions are a logical lock. All phone numbers are accessible by all devices by default. After
 partitions has been applied to a number, a lock has been placed on the phone number restricting
 who can dial it
- Every phone number, Route Pattern, Translation Patterns, on CUCM can be applied to a Partition
- All phone numbers are in null partition by default. All devices have access to Null Partition
- A CSS defines which partitions are accessible to a particular device
- CSS are assigned to device, Lines, Gateways, Trunks, Voice Mail Ports, etc.
- Restriction is placed by evaluating CSS of Calling party and Partition of called party (CSS is like a Phone book & Partition is Contacts)
- The CSS of calling party must contain the partition of called party
- Same number in different partitions are treated as two individual numbers

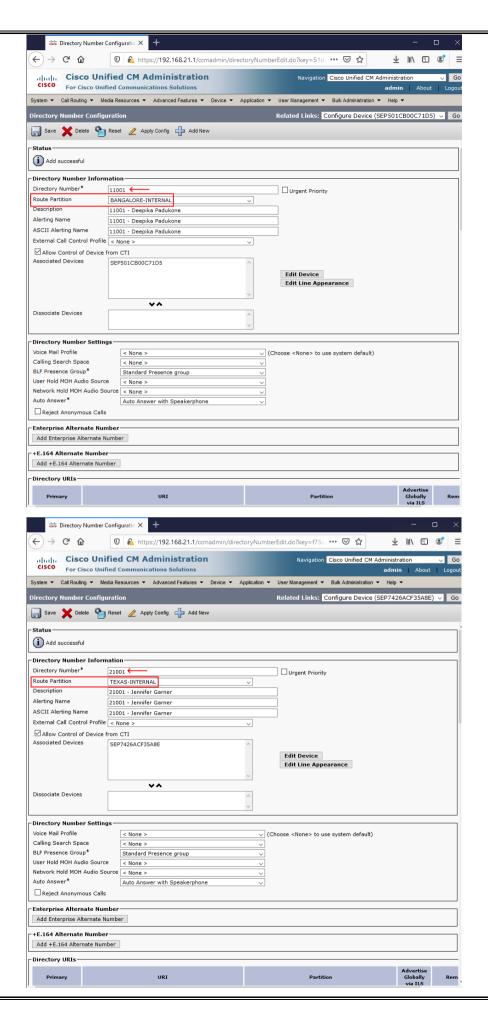
[Lab] CSS Partition Configuration

- Let's create partitions and CSS for the phones in Bangalore and Texas
- Our Aim: Bangalore Phones can call both Bangalore and Texas Phone whereas Texas Phones can call only Texas phones not Bangalore
- Call Routing >> Class of Control >> Partitions >> Add New
- We can add multiple Partitions in one go
- I have added BANGALORE-INTERNAL and TEXAS-INTERNAL partitions

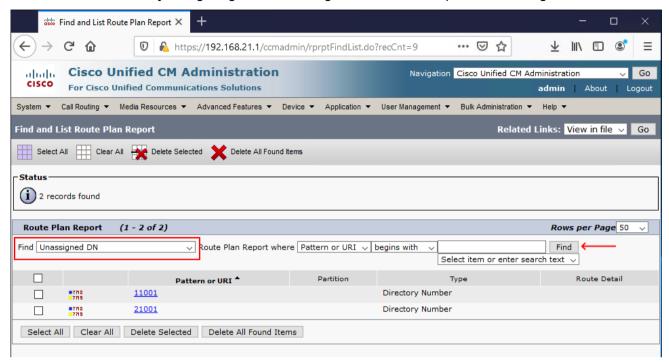




- In BANGALORE-PHONE-CSS, I have added BANGALORE-INTERNAL and TEXAS-INTERNAL Partitions hence Bangalore Phones can call both Bangalore as well as Texas phones
- In TEXAS-PHONE-CSS, I have added only TEXAS-INTERNAL Partition so that it can call only Texas
 Phones bit Bangalore
- 1XXXX range is used for Bangalore Phones and 2XXXX used for Texas phones
- As of now we added the directory numbers directly without partition, partition can be assigned while adding the DN or later
- When you add partition later, the number without partition will be left as unused in CUCM database, this must be deleted

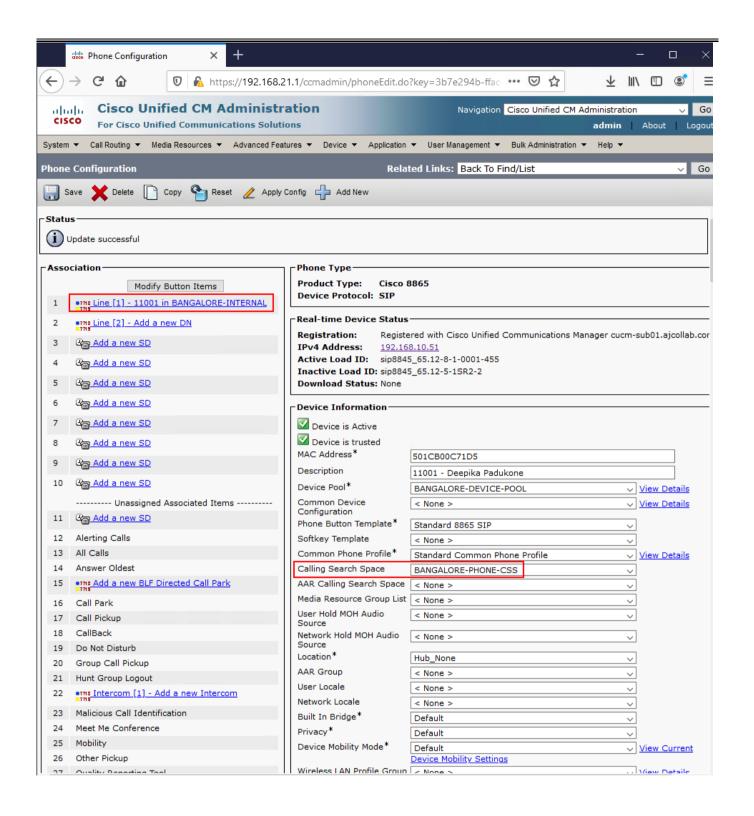


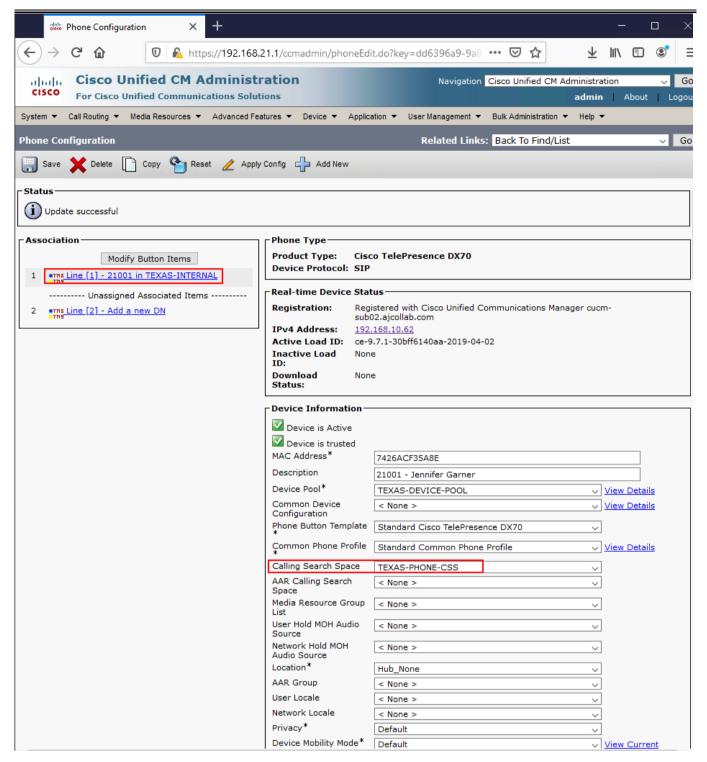
- When we assign Partition to 11001 and 21001 DN, in the backed CUCM created new numbers and the old DNs are unused
- You can see that by navigating to Call Routing >> Route Plan Report >> Unassigned DN >> Find



Now apply the CSS for the Phones. You can assign BANGALORE-PHONE-CSS to Bangalore phones and TEXAS-PHONE-CSS to Texas phones

- CSS can be applied to Phone level or Line (DN) level, if you apply on both, the Line CSS will take preference
- The total CSS will be the combination of Line and Device level CSS
- I'm applying CSS on the Phone level here, after the configuration, you can test the calls

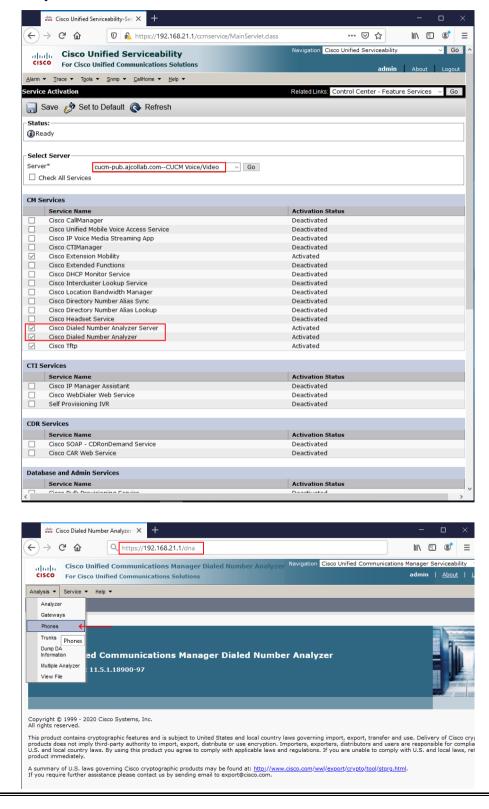


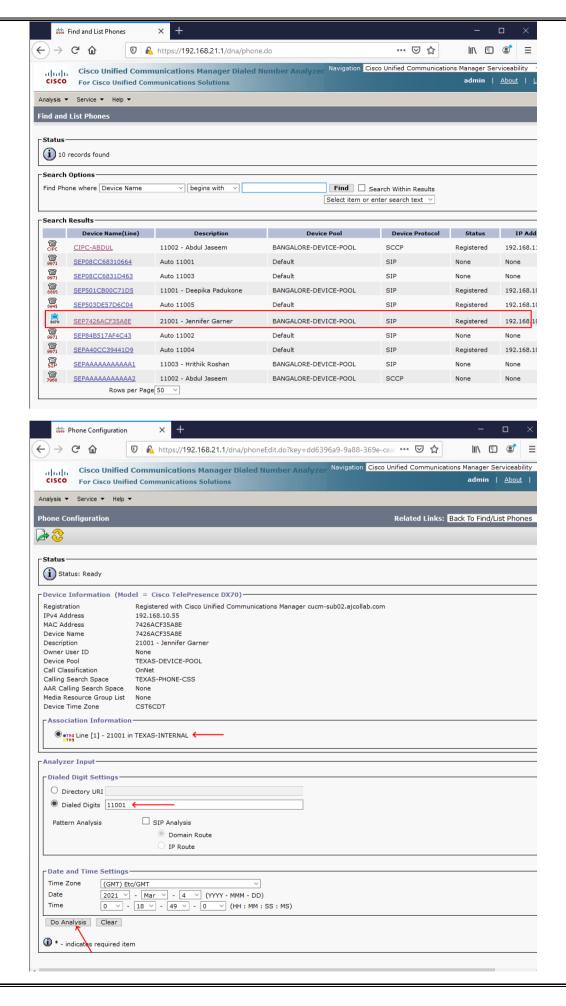


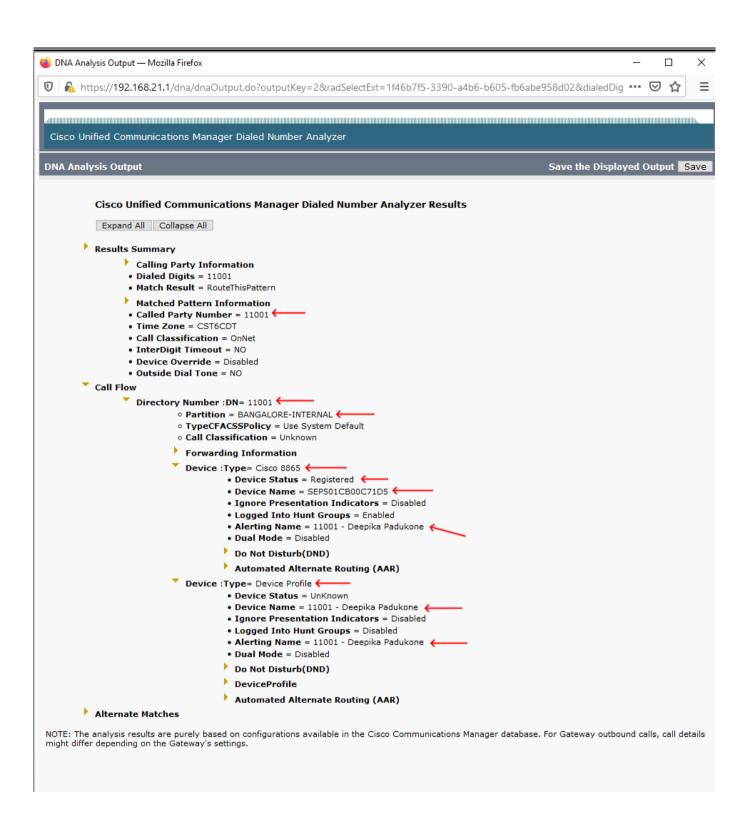
After the test, I have added BANGALORE-INTERNAL partition to TEXAS-PHONE-CSS just make sure
every phone can internally call every other phones irrespective of the location

Dialed Number Analyzer - DNA

- The tool allows you to test a Cisco CUCM dial plan configuration, this tool analyzes the dialed digits and shows details of the calls
- Because a dial plan can be complex, involving multiple devices, translation patterns, route patterns, route lists, route groups, calling and called party transformations, and device level transformations, a dial plan may contain errors

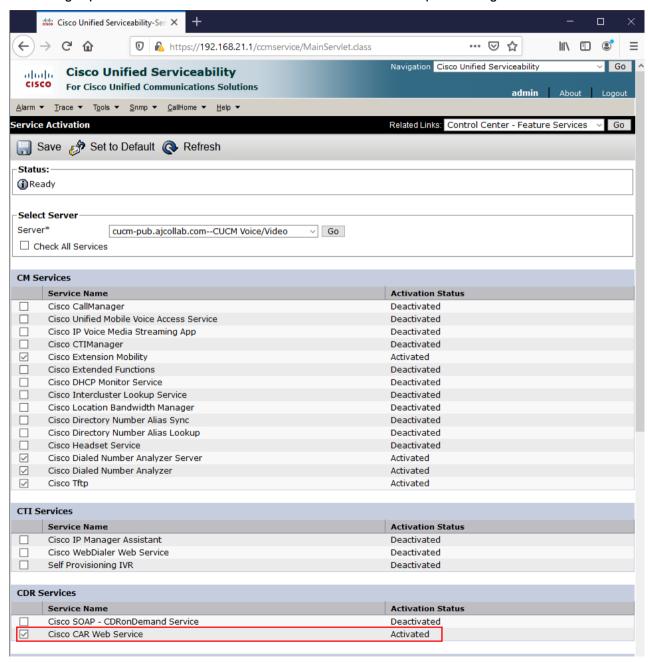


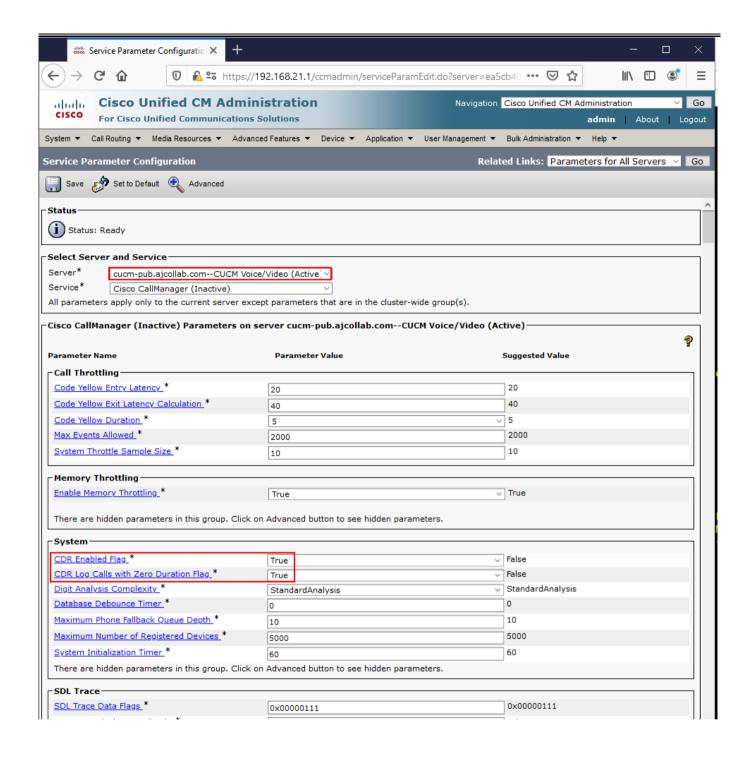


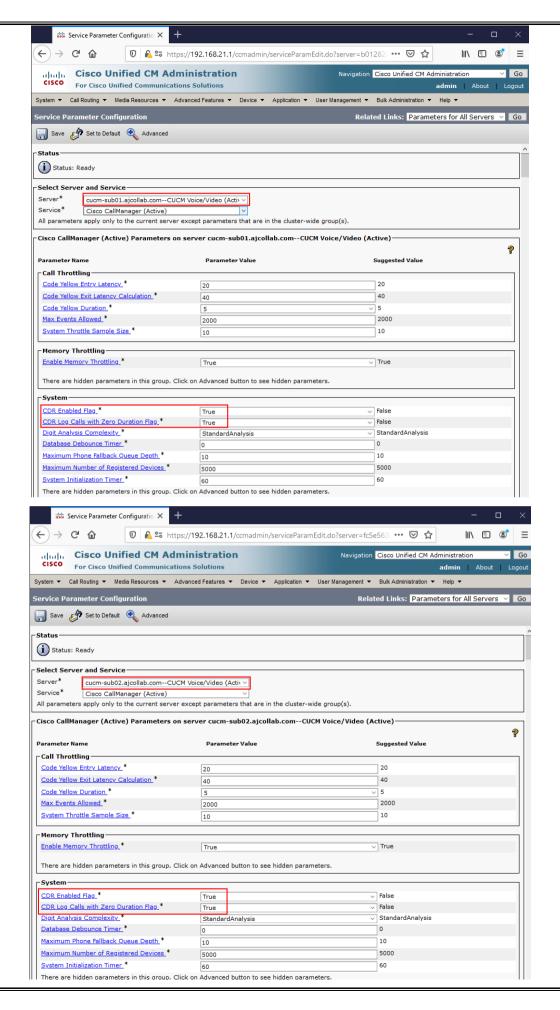


Call Detailed Record - CDR

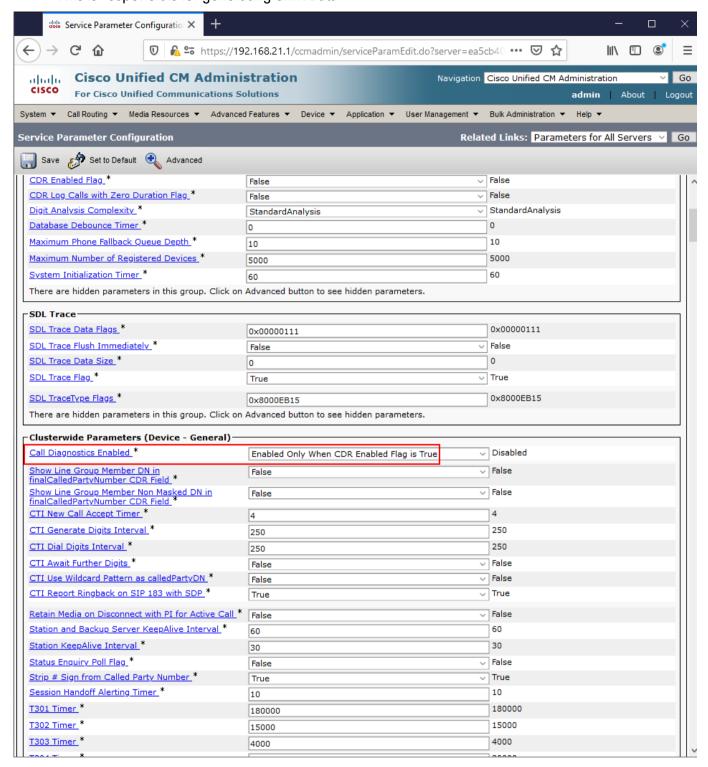
- Every call that CUCM processes can be logged. It contains information about Calling Number,
 Called Number, Originator device name, Destination device name, Originator IP, Destination IP,
 Duration of the call, etc. These logs are called CDR (Call Detailed Record)
- Call Diagnostic details like Packet sent Packet received, packet loss, jitter, etc. are called CMR (Call Management Record)
- CDRs stored in Subscribers & uploaded to CDR/CAR Database of Publisher Server at regular interval (this interval can be administratively set)
- CDR Database can be used by 3rd party billing application to prepare internal or external phone billing reports. We should activate CDR & CMR on all the call processing CUCM nodes

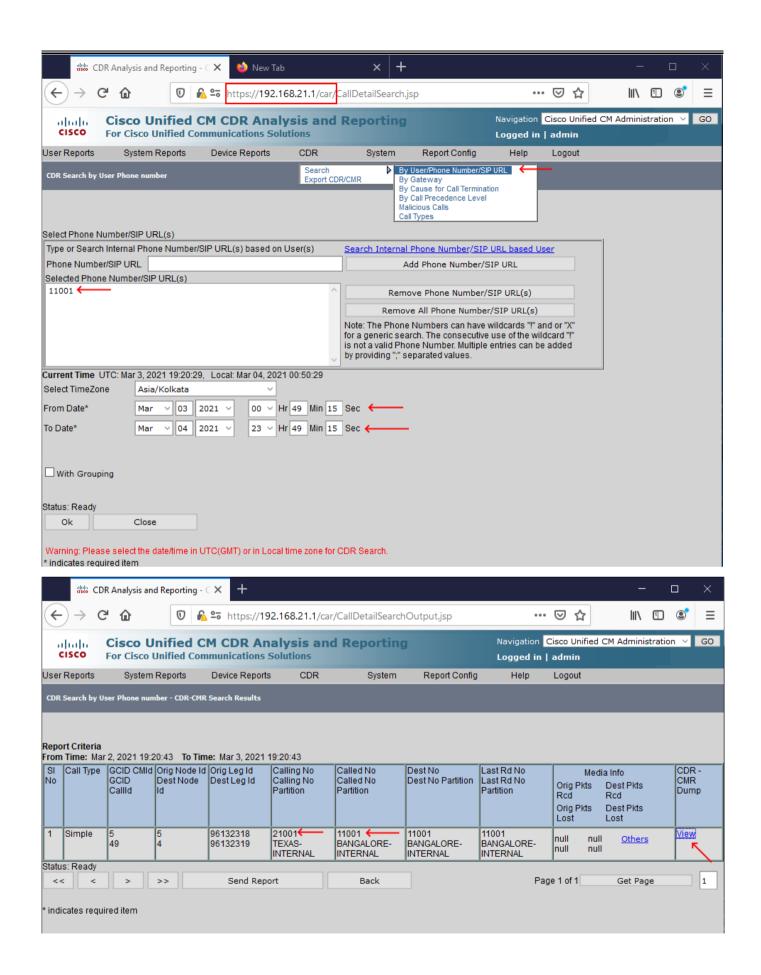


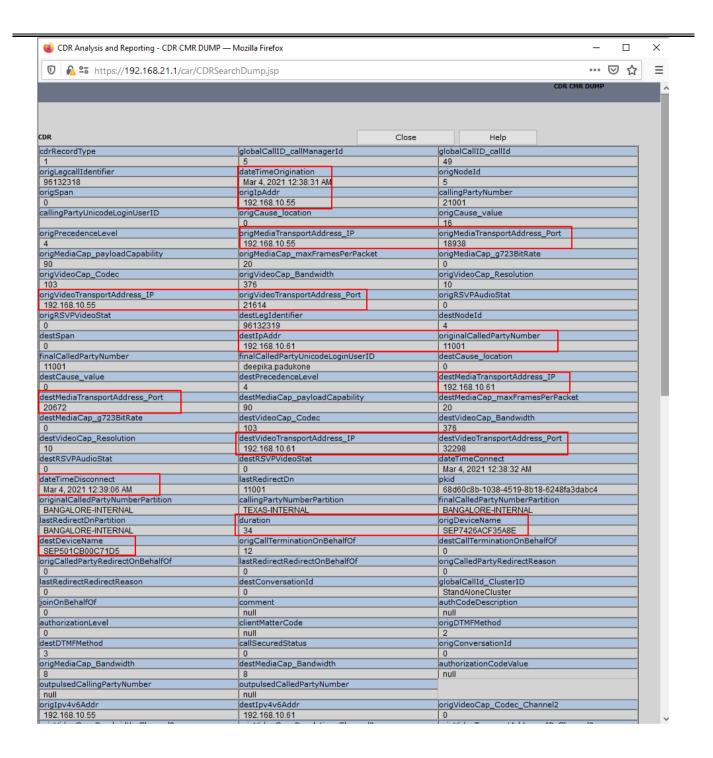


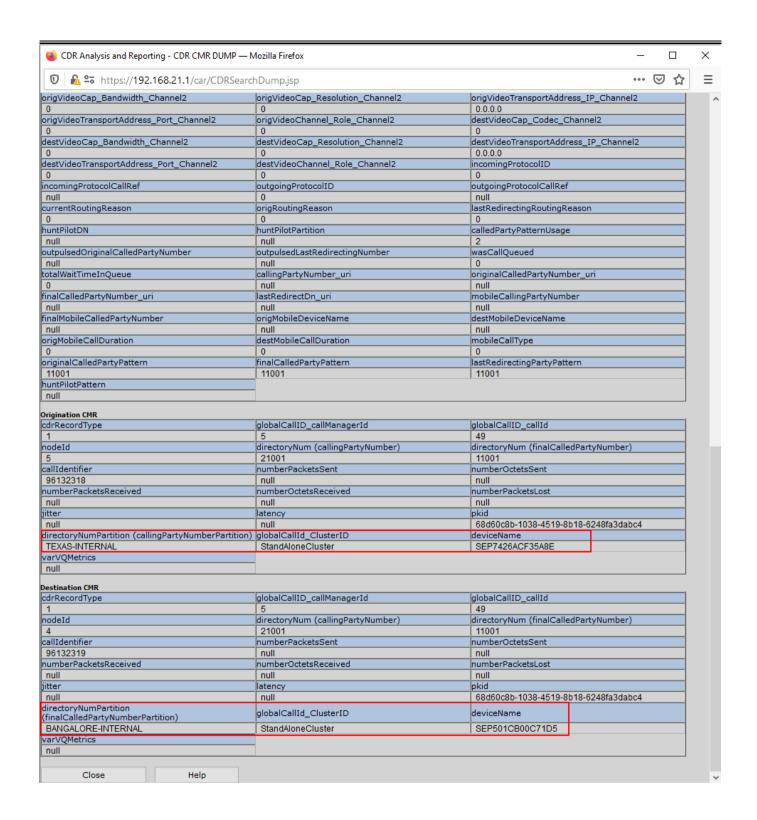


- Also enable 'Call Diagnostics' on all the nodes to get CMR data
- This is responsible for generating CMR Data

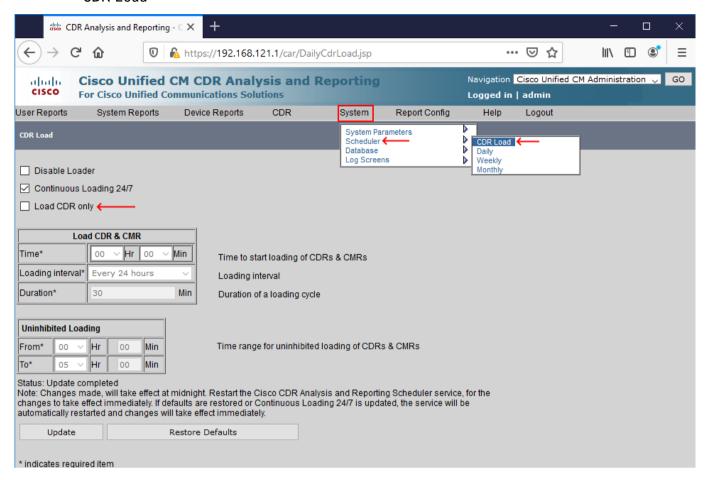








If the CMR data always shows 'null', then disable the 'Load CDR only' from System >> Scheduled
 >> CDR Load



- CDR RAW files are saved in '/cm/cdr_repository/'
- We can list and view those files in Raw format from CLI

```
01.1. a_cucm-pub_192.168.21.1 - SecureCRT
    File Edit View Options Transfer
                                                                                                                                                                                      Script Tools Window Help
   ■ F 🖾 💸 Enter host <Alt+R>
                                                                                                                                                                                               마 🖺 👸 🖨 🌣 📾 🕇 🤗 🍱

✓ 01.1. a_cucm-pub_192.168.21.1 

      dmin file list activelog /cm/cdr_repository/processed/20210303
 cdr_StandAloneCluster_05_202103031909_0 <
dir count = 0, file count = 1
  admin file view activelog /cm/cdr_repository/processed/20210303/cdr_StandAloneCluster_05_202103031909_0
    cdrRecordType", "globalCallID_callManagerId", "globalCallID_callId", "origLegCallIdentifier", "dateTimeOrigi.ocation", "origCause_value", "origPrecedenceLevel", "origMediaTransportAddress_IP", "origMediaTransportAddresleoCap_Codec", "origVideoCap_Bandwidth", "origVideoCap_Resolution", "origVideoTransportAddress_IP", "origVideotransportAddress_IP", "origVideotransportAddress_IP", "origVideotransportAddress_IP", "origVideotransportAddress_IP", "desettpAddr", "originalCalledPartyNumber", "finalCalledPartyNumber", "finalCalledPartyNumber", "desettpAddr", "originalCalledPartyNumber", "finalCalledPartyNumber", "finalCalledPartyNumber", "finalCalledPartyNumber", "finalCalledPartyNumber", "desettpAddr", "originalCalledPartyNumber", "finalCalledPartyNumber", "finalCalled
                              pAddr", "originalCalledPartyNumber", "finalCalledPartyNumber", "finalCalledPartyUnicodeLoginUserID", "d

_Port", "destMediaCap_payloadCapability", "destMediaCap_maxFramesPerPacket", "destMediaCap_g723BitRate

ransportAddress_Port", "destRSVPAudioStat", "destRSVPVideoStat", "dateTimeConnect", "dateTimeDisconnect

Partition", "lastRedirectDnPartition", "duration", "origDeviceName", "destDeviceName", "origCallTerminat

'origCalledPartyRedirectReason", "lastRedirectRedirectReason", "destConversationId", "globalCallId_CluestDMFMethod", "callSecuredStatus", "saidConversationId", "saidConversationId", "globalCallId_CluestDMFMethod", "callSecuredStatus", "saidConversationId", "globalCallId_CluestDMFMethod", "callSecuredStatus", "saidConversationId", "globalCallId_CluestDMFMETHOD "saidConversationId", "globalCallId_CluestDMFMETHOD "saidConversationId", "globalCallId_CluestDMFMETHOD "saidConversationId", "globalCallId_CluestDMFMETHOD "saidConversationId", "saidConversationId"
       f","origCalledPartyRedirectReason","lastRedirectRedirectReason","destConversationId","globalCallTerming","destDTMFMethod","callSecuredStatus","origConversationId","origMediaCap_Bandwidth","destMediaCap_BaIpv4v6Addr","origVideoCap_Codec_Channel2","origVideoCap_Bandwidth_Channel2","origVideoCap_Resolution_Channel2","destVideoCap_Bandwidth_Channel2","destVideoCap_Resolution_Channel2","destVideoCap_Resolution_Channel2","IncomingProtocolCallRef","OutgoingProtocolID","OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingICID","IncomingOrigIOI" "IncomingTeamTOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingOrigIOI" "IncomingOrigIOI" "IncomingOrigIOI" "IncomingOrigIOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingOrigIOI" "IncomingOrigIOI" "IncomingOrigIOI" "OutgoingProtocolCallFalledPartyPatternUsage","IncomingOrigIOI" "IncomingOrigIOI" "IncomingOrigI
   end of the file reached
        ptions: q=quit, n=next, p=prev, b=begin, e=end (lines 1 - 3 of 3) :
```

• To know more about CDR Architecture, please read Cisco CDR Documentation

IP Phone to IP Phone Call Flow

- Now let us see SIP call flow between an IP Phone to another IP Phone that is registered on a different node
- When you deal with any call flow issues or while working with Cisco TAC, call sample data is crucial to isolate the failure. Always try to get below information before troubleshooting any call.

These can be obtained from CDR

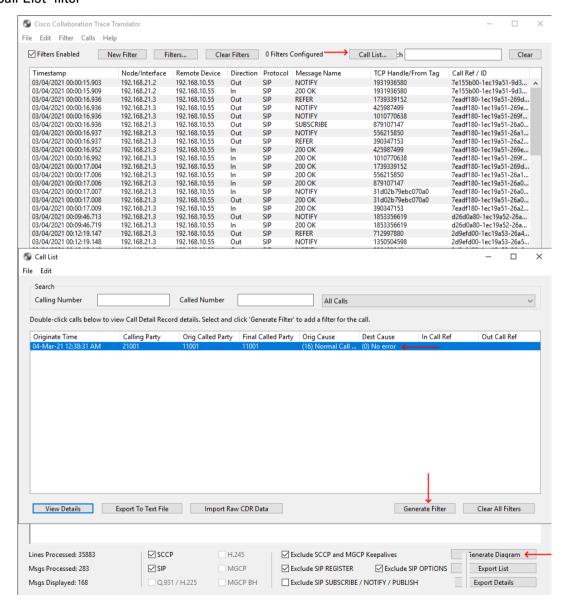
Calling Number: 21001; SEP7426ACF35A8E; DX70

Called Number: 11001; SEP501CB00C71D5; Cisco 8865

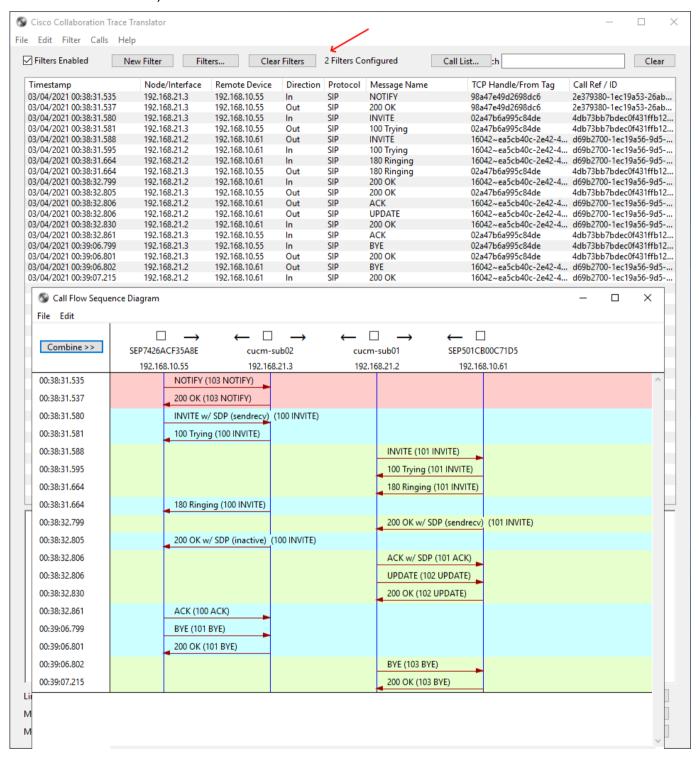
Time of the call: Mar/04/2021; 12:38:31 AM IST

Duration of the call: 34 Sec

Collect the CallManager logs, drag and drop the entire folder to Translator X and then generate a
 'Call List' filter



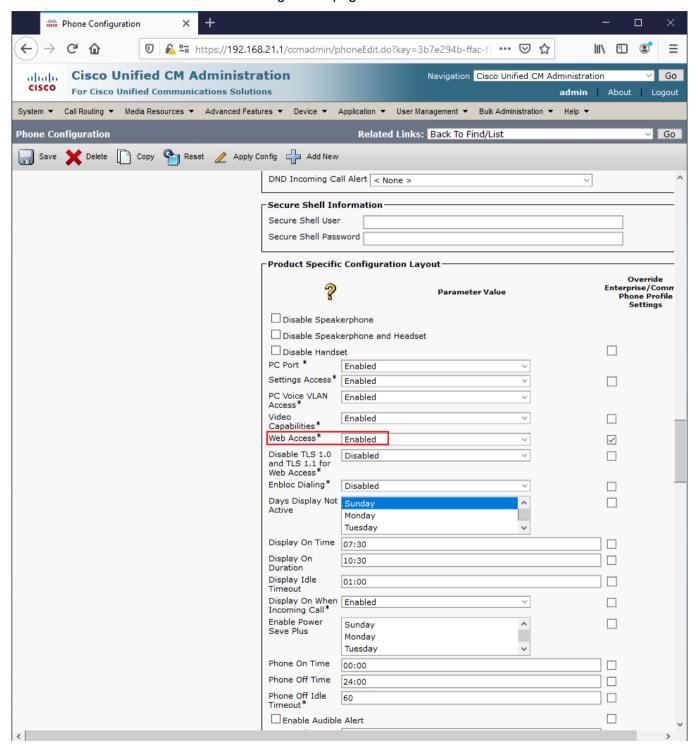
 This filter may not work all the time (situations like the log doesn't have the complete messages or some failed call) but most of the time it works

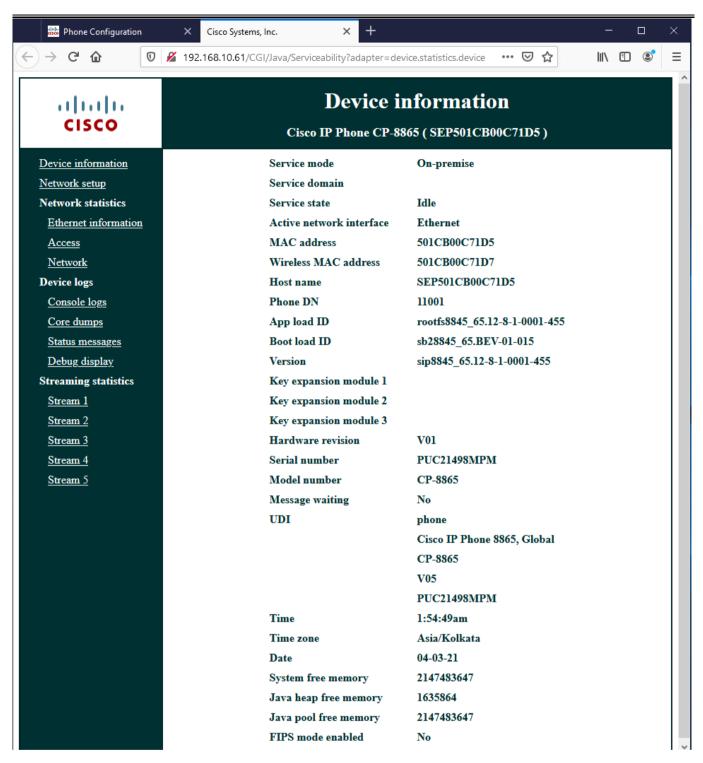


- IP Phone 1 (192.168.10.55) sends INVITE to CUCM-SUB2 where it is registered. This INVITE is called EARLY OFFER since the message contains SDP (Session Description Protocol) that talks about media attributes (IP, Port, Codec, etc.) of Phone 1
- CUCM-SUB2 sends 100 TRYING back to the IP Phone and sends SDL signal to CUCM-SUB01. SDL signals are not visible in Translator X. SDL signal used to communicate between CUCM Nodes
- Now CUCM-SUB01 passes the INVITE to IP Phone 2 (192.168.10.61), since CUCM is not a PROXY but B2BA (Back to Back User Agent), it alters the SIP Headers. This INVITE is DELAYED OFFER since it doesn't have SDP in it
- IP Phone 2 will respond with 100 Trying and 180 Ringing to CUCM-SUB01
- CUCM-SUB01 signals CUCM-SUB02 via SDL messages and 180 Ringing flows to IP Phone 1
- When Phone 2 is answered, it sends 200 OK with SDP where we have the media attributes of Phone 2
- CUCM picks a codec based on the initial INVITE from DX70 and 2000K from 8865 Phone
- CUCM-SUB2 sends 200 OK message with selected SDP attributes to DX70, hence DX70 got the media attributes of 8865 Phone
- CUCM-SUB01 sends ACK message 8865 Phone that contains the selected media attributes of DX70 that has been shared on the initial INVITE. Now 8865 got the media information
- DX70 sends ACK message to CUCM-SUB02.
- When DX70 disconnects the call, it sends BYE message to CUCM-SUB02 and CUCM-SUB2 responds with 200 OK
- The BYE message eventually forwarded to 8865 Phone from CUCM-SUB01 and it gets a 200 OK response from 8865

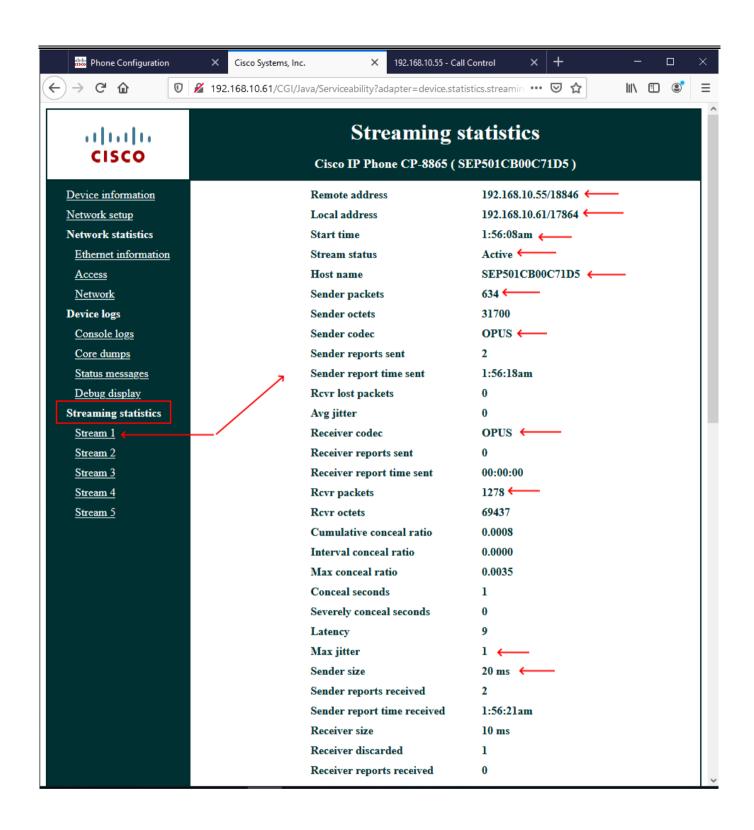
IP Phone Web Access

- It gives all device information (MAC, Host Name, Model, DN, Firmware, Network Configuration & statistics, logs, etc.)
- Device Page >> Web Access >> Enabled >> Reset
- Now click IP Phone IP Address to get web page



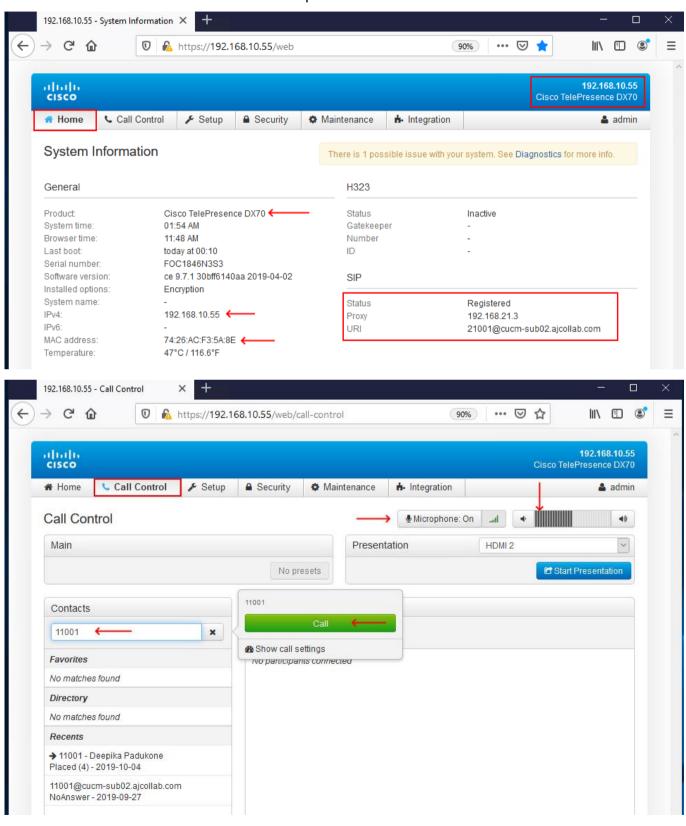


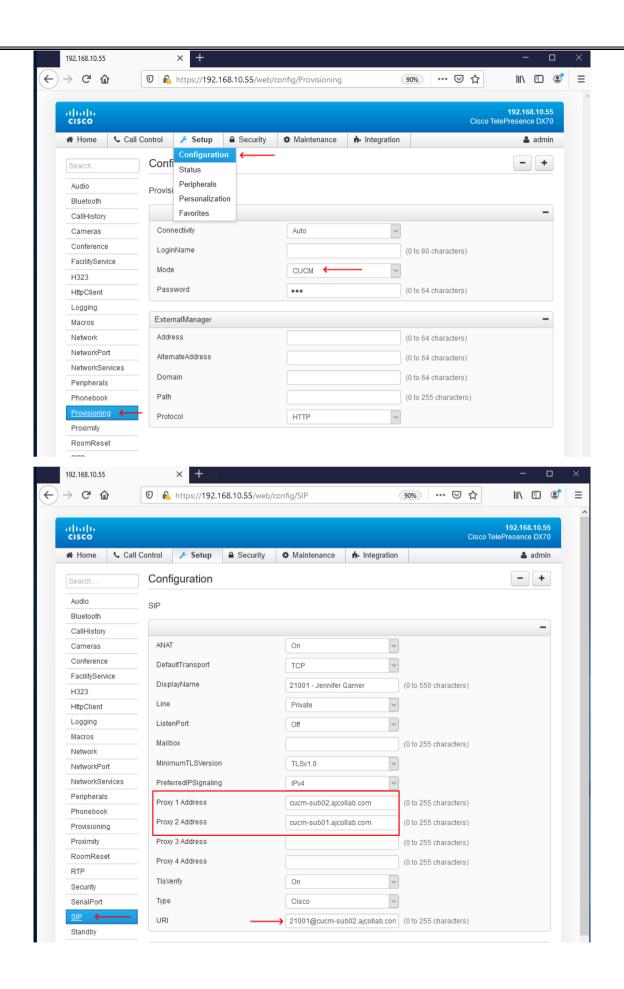
 This can be used to identify media streams status as well, this is helpful while troubleshooting 'one way or no way audio' issues

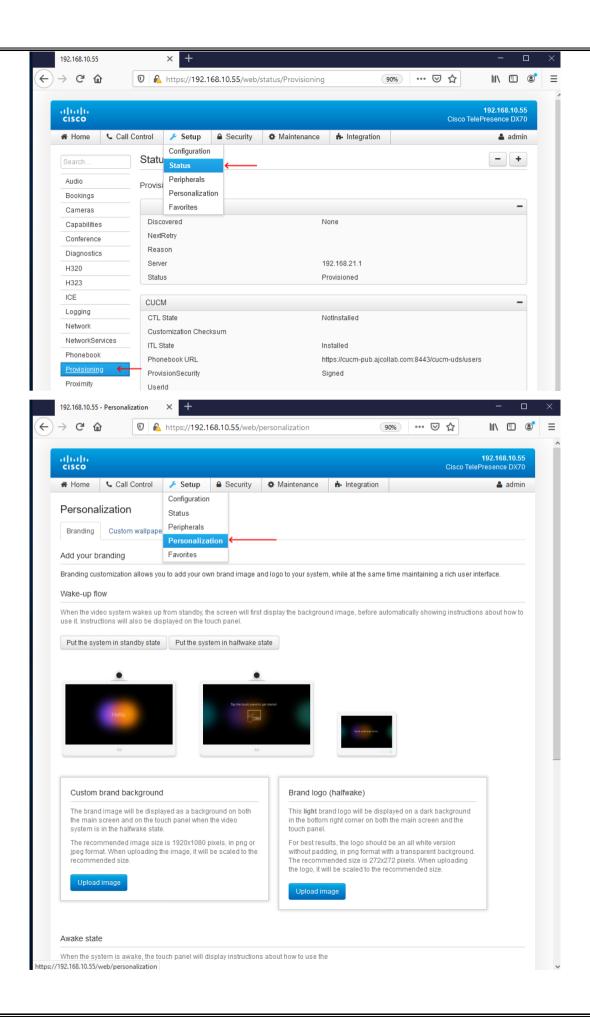


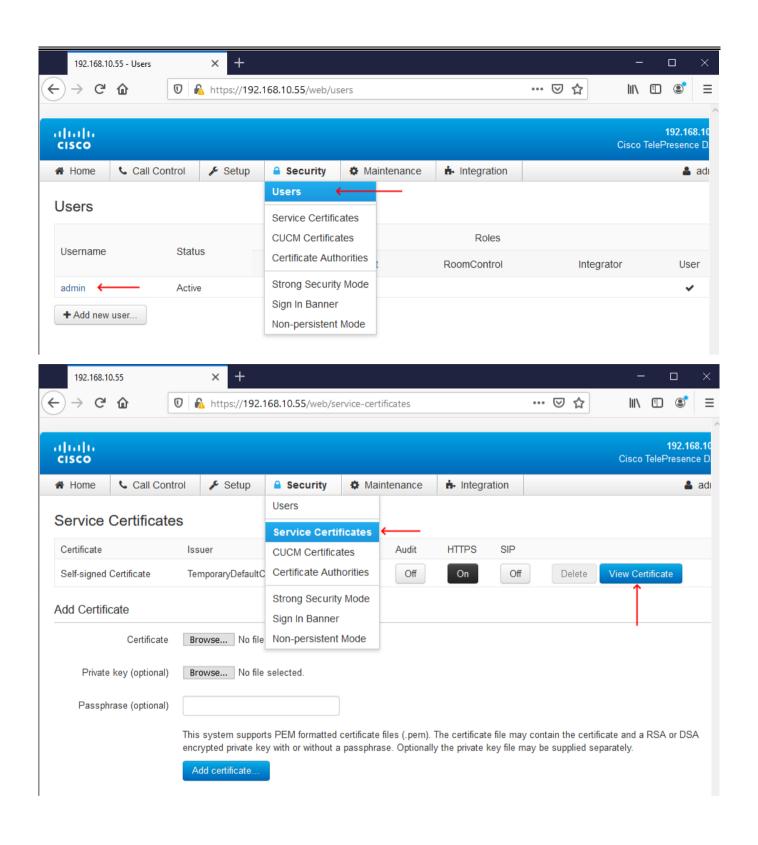
DX70 Web Management Console

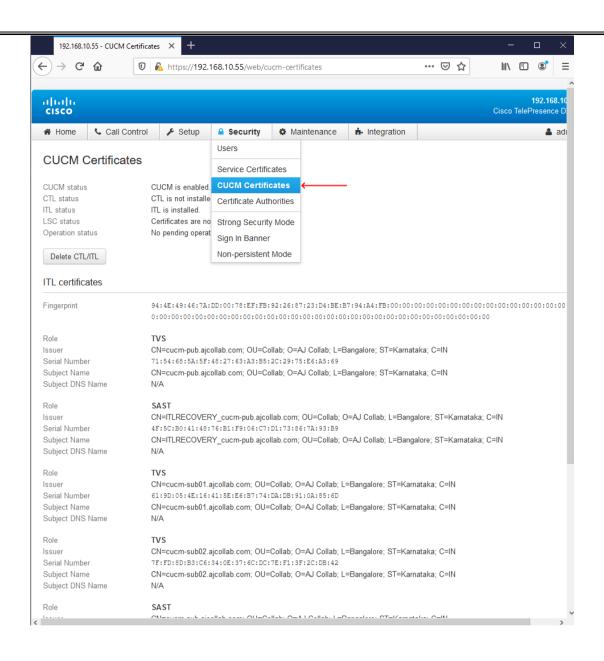
- Every telepresence endpoints have a dedicated Web interface to manage the device
- The default username is admin without password

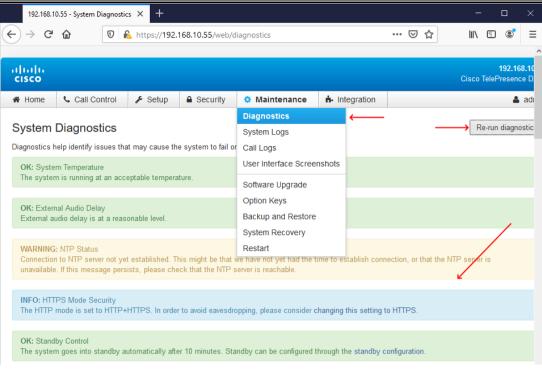


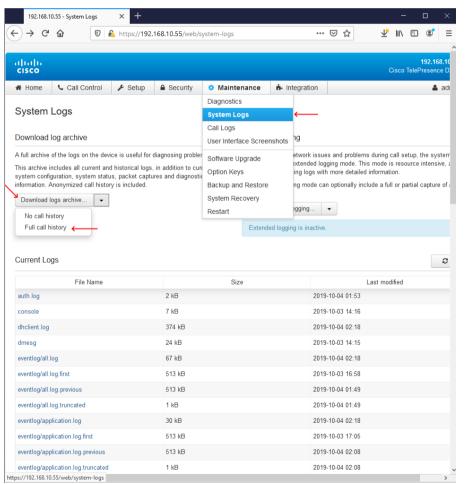


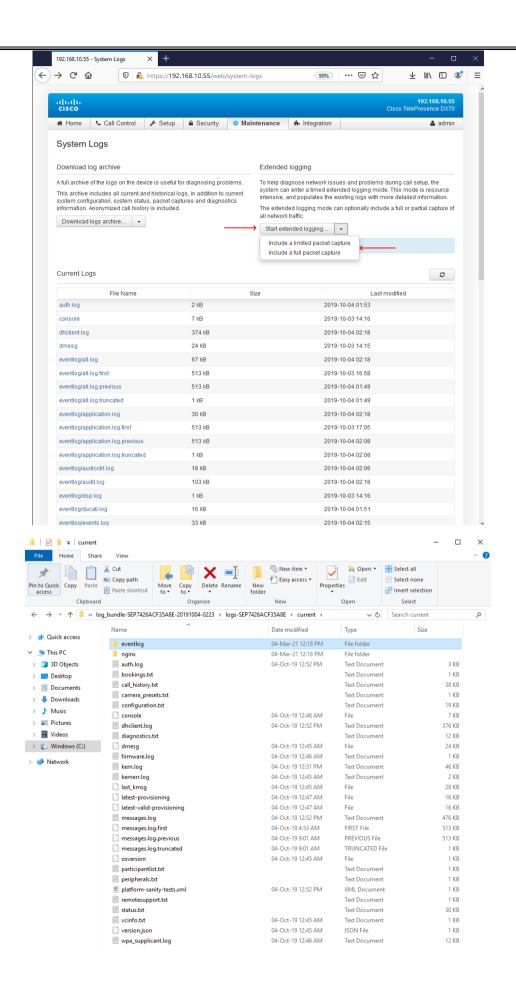


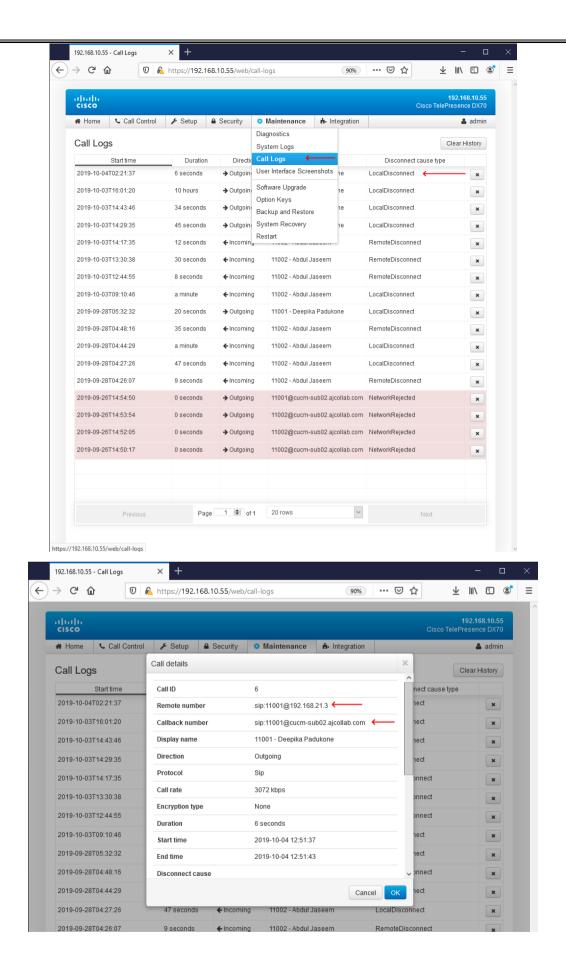


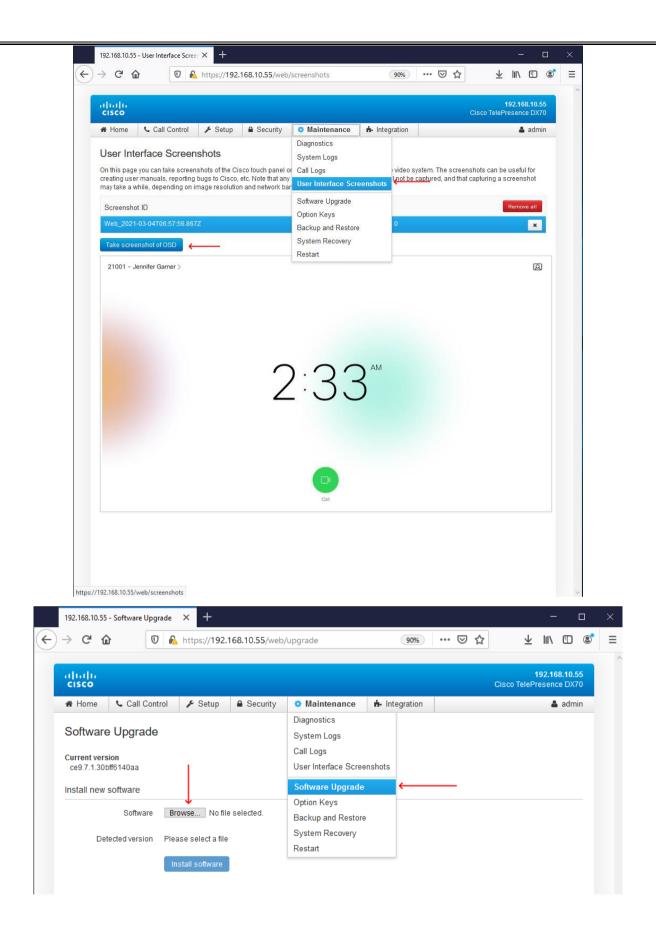


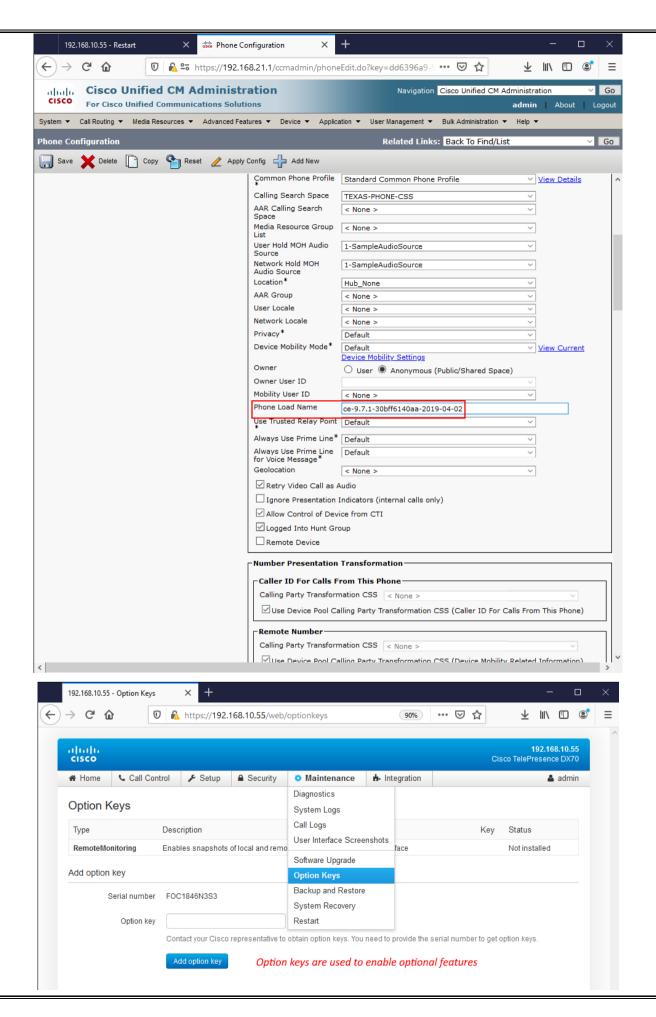


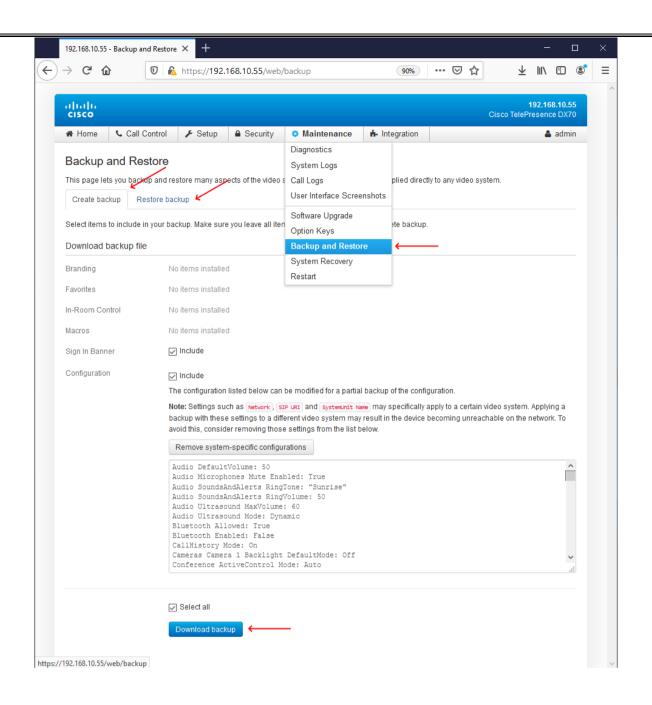


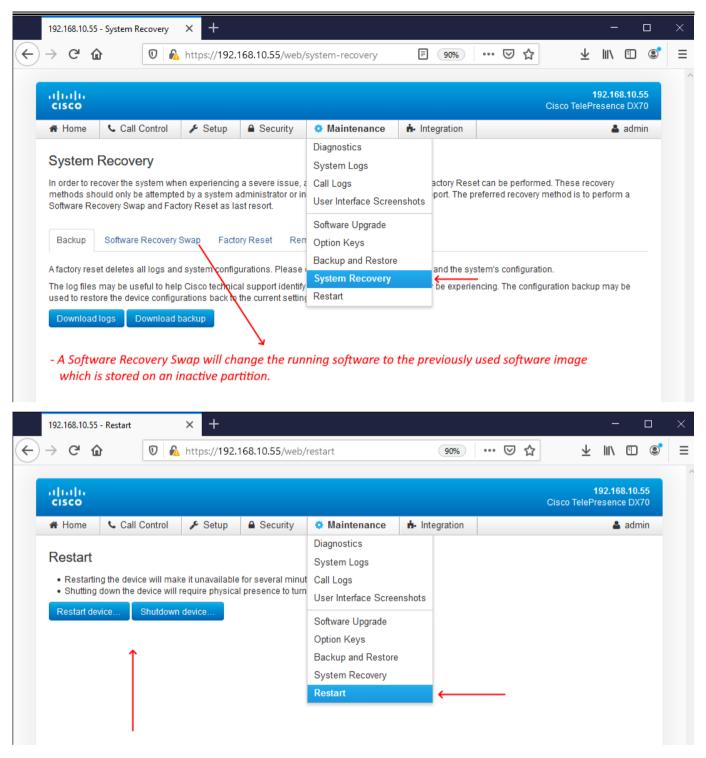












Please note, Updated devices will have slightly different look and feel in the web interface, but the options and functions are same as shown here.

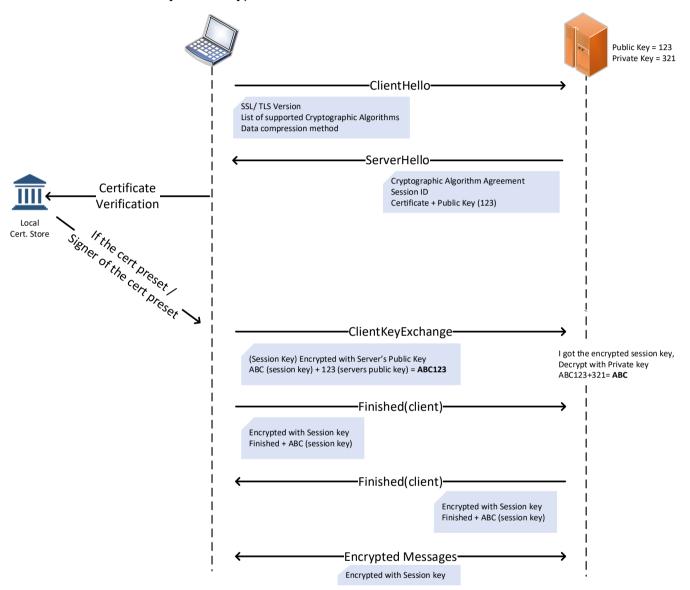
SSL Certificates



- SSL Certificates are used between end points to build a trust/authentication and encryption of data
- This confirms that the endpoints communicate with the intended device and have the option to encrypt the data between the two endpoints
- When systems communicate each other initially they will exchange SSL certificates. If one system trusts the other system's certificates, then the connection is established otherwise it terminated
- To trust a certificate, there are 2 ways, Remote system's certificate must be locally installed on the
 local system or there should be globally trusted authority who issued certificate for remote
 system. Since the authority is globally trusted, the local system trusts the authority as well as all
 the certificates' issues by the authority
- This authority is called CA (Certificate Authority). We can have internal CA or Public CA. Internal CA is trusted by all systems inside the enterprise and public CA is trusted by everyone in the world

SSL/TLS Handshake and PKI Infrastructure

- Secure Socket Layer (SSL) and its newer version Transport Layer Security (TLS) are cryptographic protocols that provide security in the internet communication
- The fundamental idea is that when a message is encrypted with a device's Public Key, only same device's Private Key can decrypt it

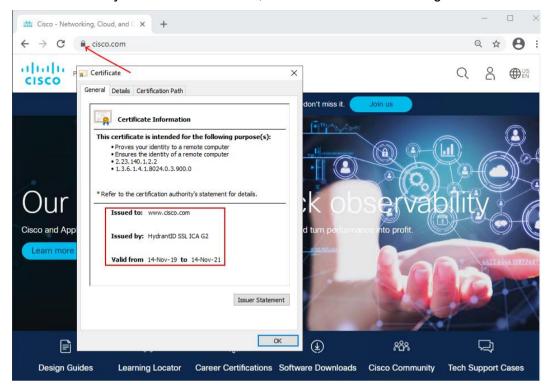


- Client sends CLIENT HELLO message to the server. CLIENT HELLO contains SSL or TLS Version, list of Cryptographic algorithms, Data compression method
- 2. Server responds with SERVER HELLO that contains Cryptographic algorithm agreement, Session ID, Server Certificate, Public key
- 3. Client verifies the certificate (either locally trusted or public CA) and trusts the server
- 4. Client sends CLIENT KEY; it is a shared secret key (also called session key) encrypted with server's public key. Now server got the session key, and it can decrypt using it's own private key

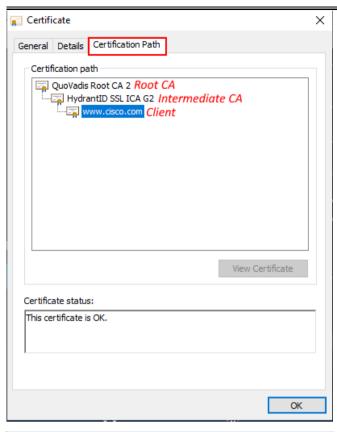
- 5. Client sends FINISHED (client) message encrypted with previous session key. Now the client part handshake is complete
- 6. Server responds with FINISHED (server) message encrypted with session key. Now the server part handshake is complete
- 7. Now client and server can exchange messages that are symmetrically encrypted with the session key
- Till we generate the Session key, the process is called Asymmetric Encryption
- Once we have the session key exchanged between Client and Server, then the message encryption
 is using the session key and this process is called Symmetric Encryption
- This complete process of key exchange and encryption is collectively called PKI (Public Key Infrastructure)

Understanding Public CA Signed Certificate

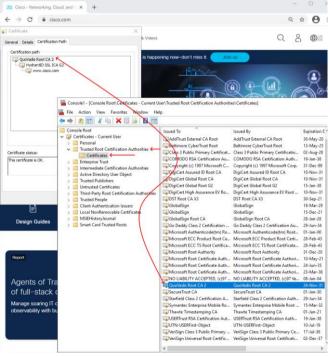
- These certificates are signed by a central Certificate Authority CA
- When we browse any SSL enabled website, we can click on the lock sign and view the certificate



- Issued to: Tells that this certificate has been issued to www.cisco.com
- Issued by: The globally trusted Certificate Authority CA who signed the certificate
- Validity: Is the date range that this certificate is usable

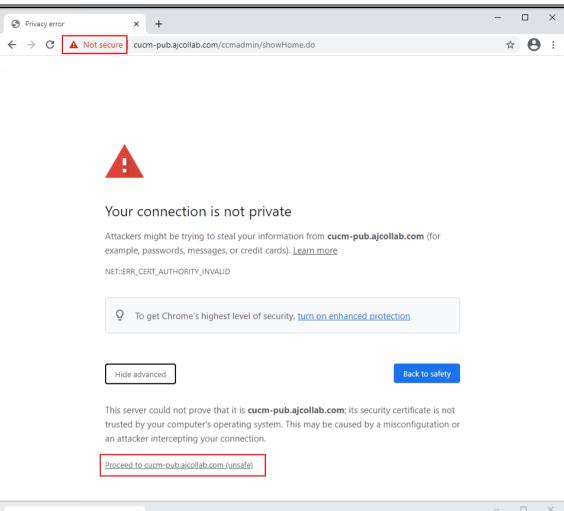


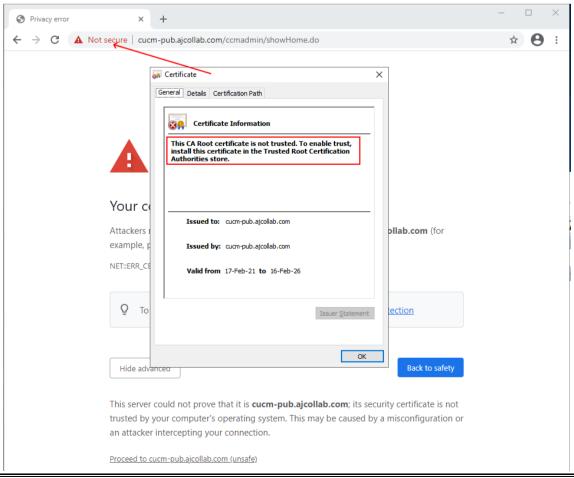
- If we go to the Certification Path, we could see the Authority Chain
- Here 'QuoVadis Root CA 2' is the Root Level
 CA and 'HydrantID SSL ICA G2' is the
 Intermediate CA who signs the certificate
- Here my system trusts the 'QuoVadis Root
 CA 2' authority since my OS Manufacture has
 installed 'QuoVadis Root CA 2' inside my
 operating system itself



You can verify this by going to MMC >> Add
 or Remove Snap-in >> Certificates >> Trusted
 Root Certification Authorities >> Certificates

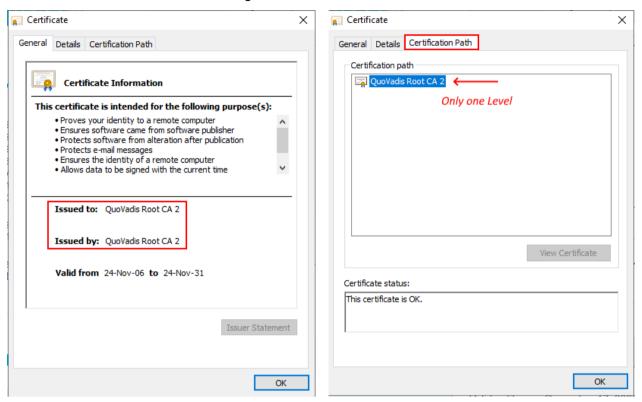
- Getting a public CA signed certificate involve some cost, it is not free
- If the signing authority is not trusted, we will get warning whether to accept or reject the certificate. When we accept the certificate, it will be added to the Trusted Root CA section of your OS or locally store on the browsers





Understanding Self Signed Certificate

- Who will sign the certificate of Root CA? We know that there is not entity above that. Hence the certificate of CA is signed by CA itself
- Such certificates are called Self Signed Certificates



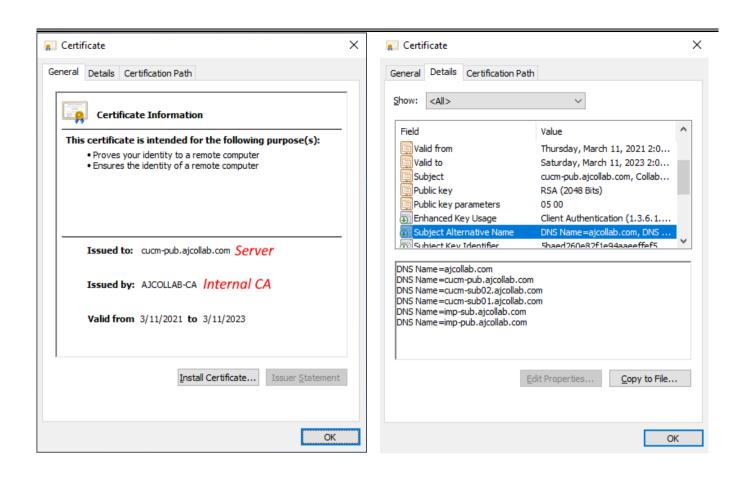
If the Issued to and Issued by are the same system, then the certificate is called Self Signed
 Certificate. Also, the Certification path will have only one level



- When we install CUCM, it will have some selfsigned certificates, means CUCM will sign its own certificate. CUCM is not a CA but it generates a certificate during the installation
- This certificate is not trusted by our OS; hence we get warning when we browse the CUCM web page

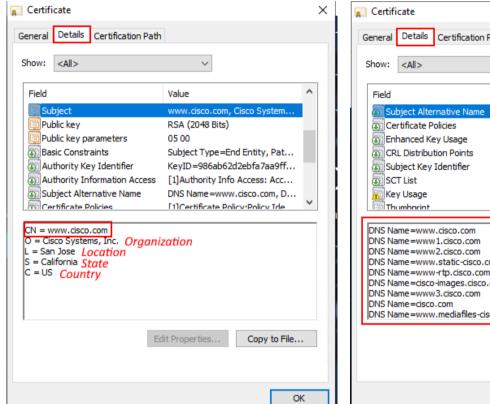
Internal or Enterprise CA Signed Certificate

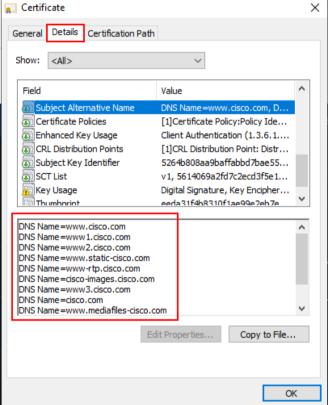
- Since getting a Public CA Signed certificate is not feasible for all the servers in an enterprise and manually accepting the certificate warning of self-signed certificate gives end user uncertainties, to address this issue, we use Enterprise or Internal CA
- The internal CA Certificate is trusted by all the entities in the organization (installed in Trusted Root
 CA) hence we can trust the connection, as well as we don't get the warning
- We can enable Microsoft Windows Server to be an Internal CA server, then use this server to issue certificates for all systems in the organization



Understanding Components of Certificates

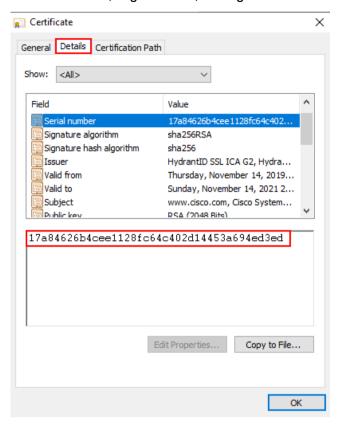
- Common Names (CN) references to the IP address or Fully Qualified Domain Name (FQDN) of the
 address that is requested. For instance, if you enter https://www.cisco.com, then the CN or SAN
 must have www.cisco.com in the header
- Subject Alternative Names (SAN) references other FQDNs that can use the same certificate. This
 will be an alternate FQDN of the or different server that offers same service



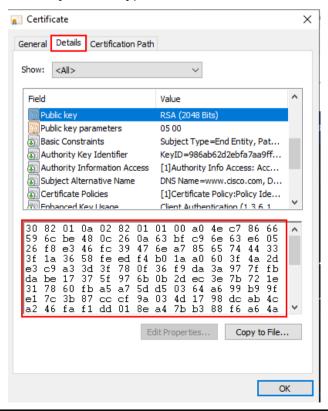


- · We can see the above information in the Details tab of the Certificate
- Here all other FQDNs can use the same certificate. These fields are added by CA after receiving our Certificate Signing Request (CSR)
- The URL request for www.cisco.com from the browser checks the URL FQDN against the information the CN or SAN of certificate.
- In this case, they match, and it shows the SSL handshake is successful. This website has been verified to be the correct website and communications are now encrypted between the desktop and the website
- In this case Certificates CN must be www.cisco.com or SAN must contain www.cisco.com
- In CUCM infrastructure, we can use all the Subscriber servers FQDNs as SAN field so that only one certificate is needed for the entire cluster

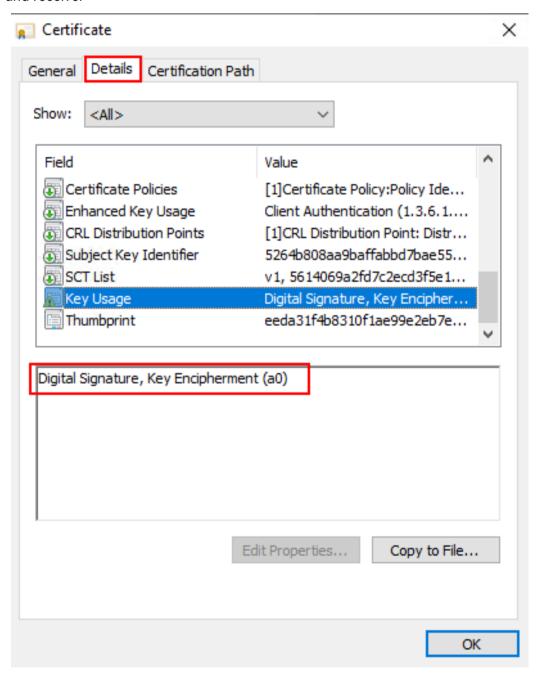
• **Serial Number**: All certificates have a unique serial number. You can use this to compare if the certificates are the same certificates, regenerated, or bogus



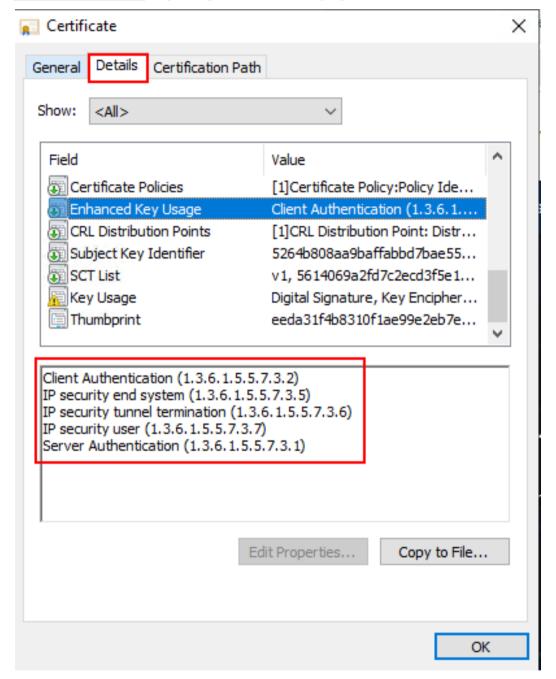
Public Key: Used to encrypt the data during communication. It is based on PKI infrastructure.
 Client uses server's Public Key to encrypt the data, Whatever data is encrypted by the Public Key, only same server's private key can decrypt



- Key Usage: This extension defines the purpose of the public key contained in a certificate
 - Key encipherment: Indicates that the certificate may be used to encrypt the symmetric key
 - <u>Digital signature</u>: When public key used to verify the authenticity
 - <u>Data Encipherment</u>: When the public key is used for encrypting user data, other than cryptographic keys
 - <u>Certificate Signing</u>: When public key is used to verify a signature on certificates. This extension can be used only in CA or self-signed certificates
 - <u>Key agreement</u>: Use when the sender and receiver of the public key need to derive the key
 without using encryption. This key can then be used to encrypt messages between the sender
 and receiver



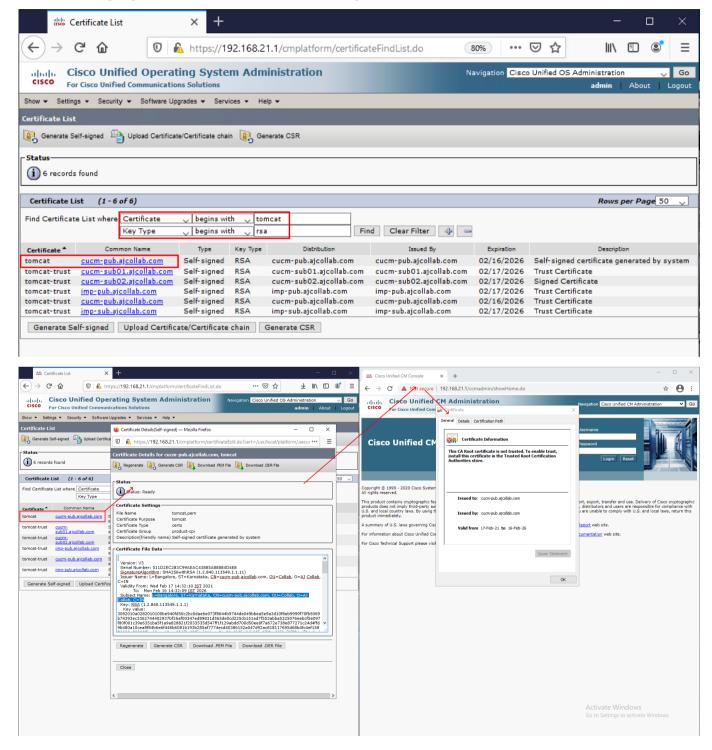
- Enhanced Key Usage: This is an extension of key usage, it used for non-critical purpose
 - o Server authentication: Digital signature, key encipherment or key agreement
 - o <u>Client authentication</u>: Digital signature and/or key agreement



CUCM SSL Certificates

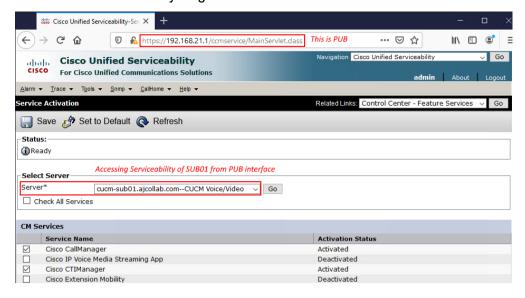
Tomcat

- CUCM WEB GUI uses Tomcat Certificate to encrypt HTTP communications to the server
- Phones will use this certificate when they access HTTPS services like Corporate Directory
- Secure Extension Mobility and Secure Extension Mobility Cross Cluster
- Below figure shows the Tomcat certificates in CUCM right after the cluster installation
- The highlighted one is the CUCM-PUB self-signed certificate

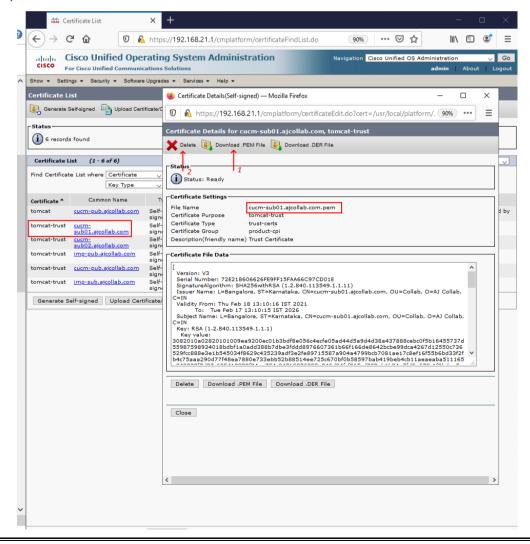


Tomcat Trust

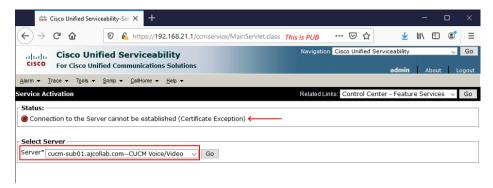
Tomcat trust store will have self-signed certificates of other nodes; hence we will be able to access
other nodes from Service Ability Page of CUCM



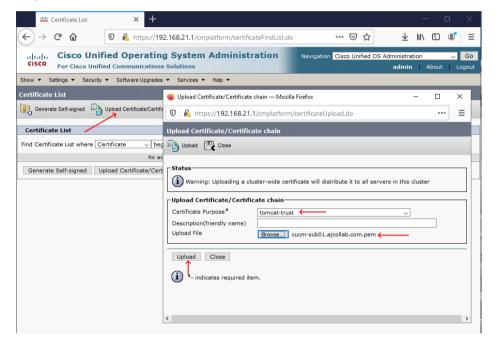
What if I delete the tomcat-trust of CUCM-SUB01 from PUB? I'm going to delete the tomcat-trust
of 'cucm-sub01.ajcollab.com' from CUCM-PUB after downloading it (so that I can upload it back
again)



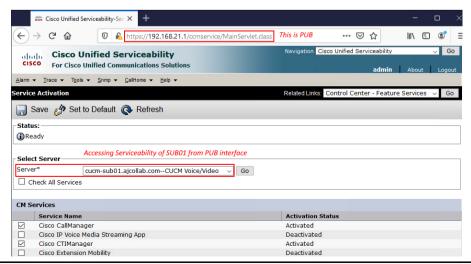
 Now if I access CUCM-SUB01 from serviceability of CUCM-PUB, I get "Connection to the Server cannot be established (Certificate Exception)" error



 Upload the CUCM-SUB01 certificate back to CUCM-PUB as tomcat-trust and then try to access the Serviceability of CUCM-SUB01 from CUCM-PUB

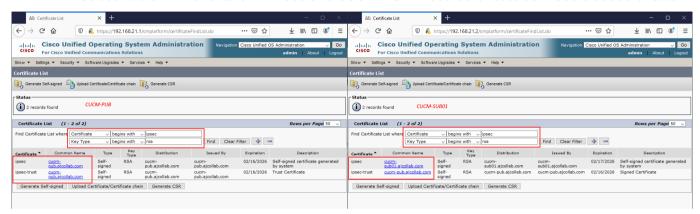


- We will be able to access the Serviceability of SUCM-SUB01 from CUCM-PUB.
- Though it asks for restarting Cisco Tomcat service after the trust certificate upload, it is not necessary, but it is recommended

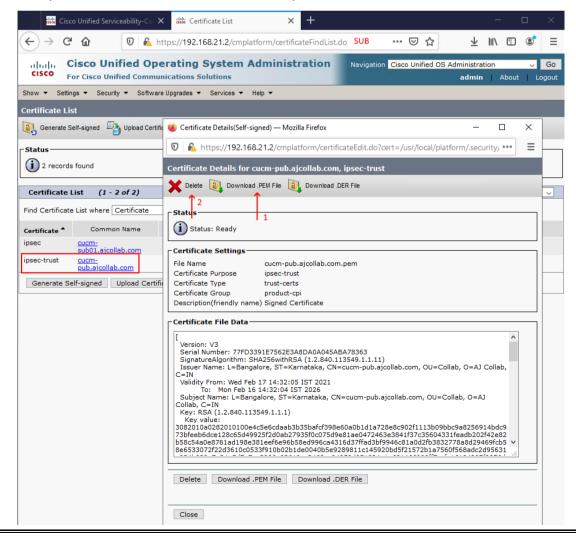


IPSec

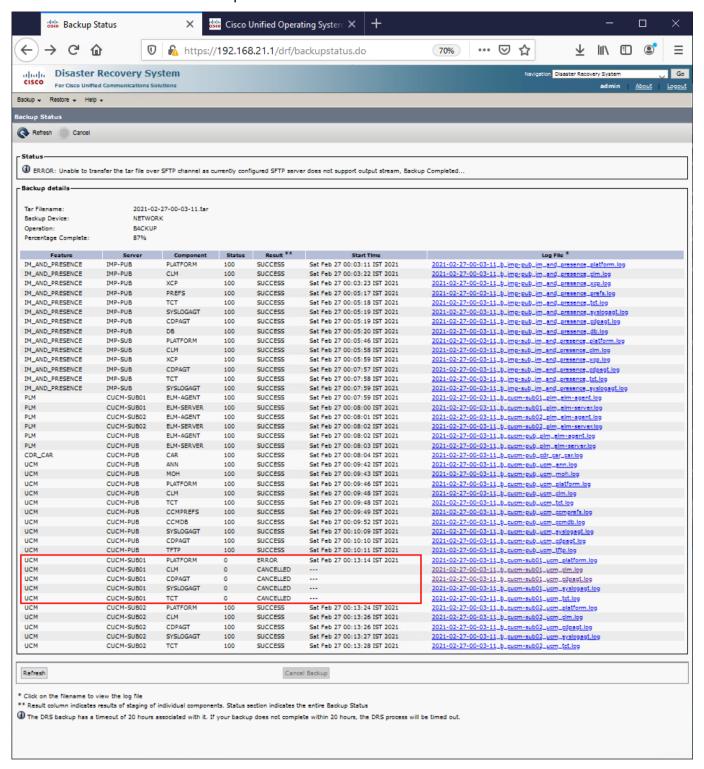
- Disaster Recovery System (DRS) or Disaster Recovery Framework (DRF) uses IPSec while communicating between DRF Master and DRF Local
- We use CUCM-PUB as the master server to take backup of other servers
- CUCM-PUB will have CUCM-PUB IPSec and CUCM-PUB as IPSec trust
- All Subscribers will have its own IPSec and CUCM-PUB IPSec certificate as IPSec trust certificate



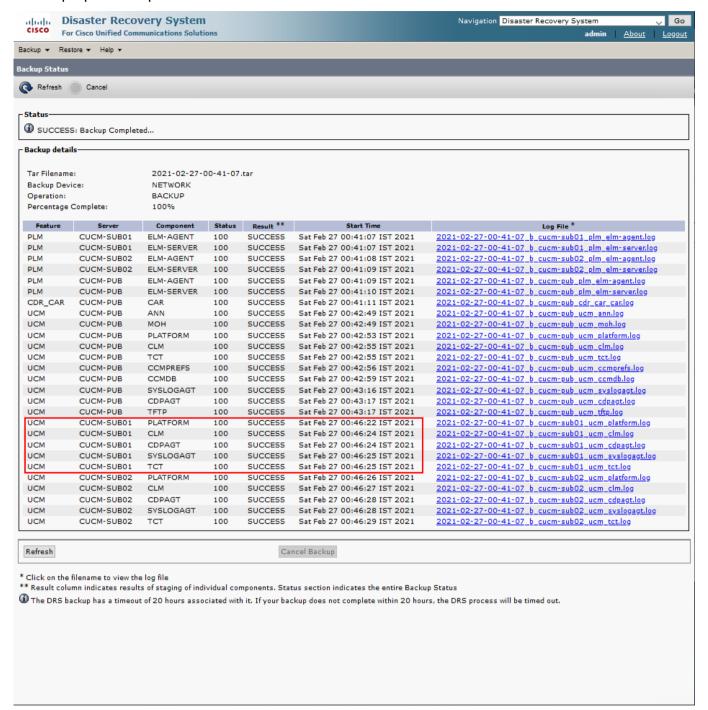
 To test the DRF failure, you can delete the IPSec Trust certificate from CUCM-SUB01, the backup will fail (download the certificate before delete)



We can see the DRS Backup failed for CUCM-SUB01



 You can re-upload the CUCM-PUB IPSec certificate to CUCM-SUB01 as IPSec trust to perform proper backup



CallManager

- Used to sign the Phone configuration file
- Secure Session Initiation Protocol (SIP) trunks or media resources (Conference bridges, Media
 Termination Point (MTP), Xcoders, and so on) uses CallManager Certificate, any issues on these
 certificates will affect respective services
- The AXL request also uses CallManager certificate
- CUCM-PUB will have its CUCM-PUB CallManager certificate and CallManager certificates of other
 CUCM nodes as CallManager-trust

Trust Verification Service (TVS)

- IP Phone won't have enough memory to store certificates, hence it keeps TVS certificate
- Phone will have the TVS Certificate of the node where it is currently registered to
- During HTTPS request from IP Phone, the certificate is verified by CUCM node where it is registered since phone doesn't have trust store
- TVS Certificate also used to authenticate configuration files

CAPF Certificate

- Used for encryption of call signaling and media
- This enables the functionalities of CTL File
- By default, CAPF is not enabled and hence the cluster is called Non-secure cluster
- When CAPF is enabled, then the cluster is called Mixed mode cluster

Note: Cluster security is vast area to study, and it is not the scope of this article

Table Showing Certificate and Trust Store

CUCM-PUB		CUCM-SUB01	
tomcat	tomcat-trust	tomcat	tomcat-trust
CUCM-PUB	CUCM-PUB	CUCM-SUB01	CUCM-PUB
	CUCM-SUB01		CUCM-SUB01
	CUCM-SUB02		CUCM-SUB02
	IMP-PUB		IMP-PUB
	IMP-PUB		IMP-PUB
ipsec	ipsec-trust	ipsec	ipsec-trust
CUCM-PUB	CUCM-PUB	CUCM-SUB01	CUCM-PUB
callmanager	callmanager-trust	callmanager	callmanager-trust
CUCM-PUB	CUCM-SUB01	CUCM-SUB01	CUCM-PUB
	CUCM-SUB02		CUCM-SUB02
tvs	tvs-trust	tvs	tvs-trust
CUCM-PUB		CUCM-SUB01	

Note: When certificates are expired, we must regenerate it.	. For certificate regeneration process, please

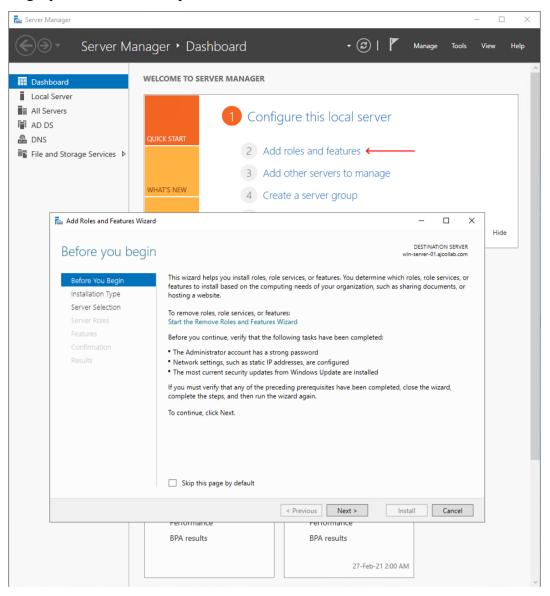
Certificate Signing Request - CSR

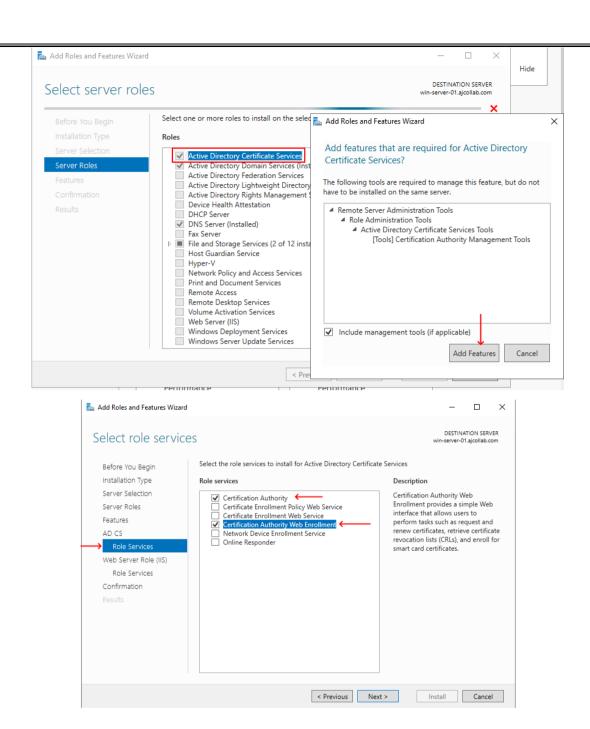
- Many of the enterprises never use self-signed certificate for Cisco Tomcat, rest of the certificates stays as self-signed most of the time (unless specific compliance requirements are in place)
- We generate CSR and pass it to CA; they will sign the certificate and provide identity certificate. We can then upload this to our server
- To create a third-party certificate for a CUCM server, you need a CSR to present to the CA
- Every time this option is used, a new private key and CSR is generated
- Cisco Unified Operating System Administration > Security > Certificate Management > Generate
 CSR > choose the service you want to create the certificate > then Generate CSR

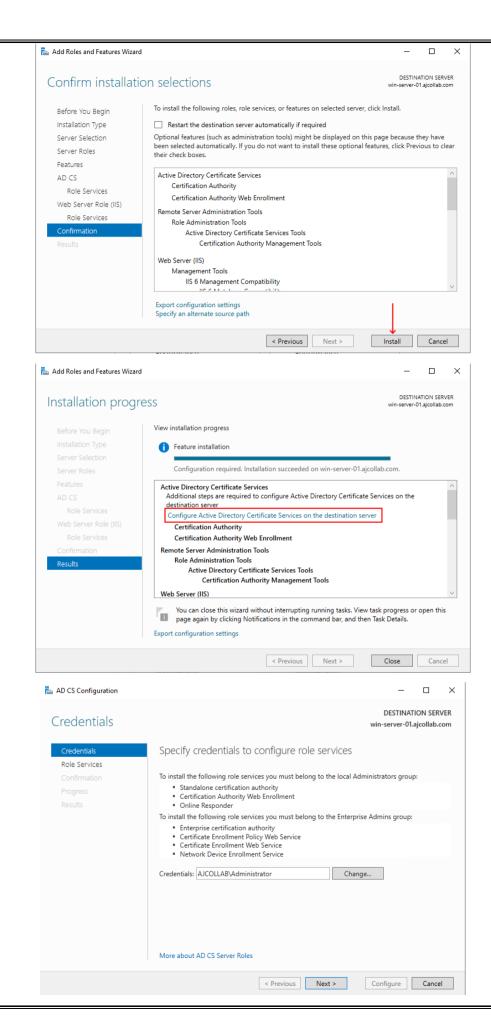
CUCM SAN (Multi-Server Subject Alternate Name) Certificate

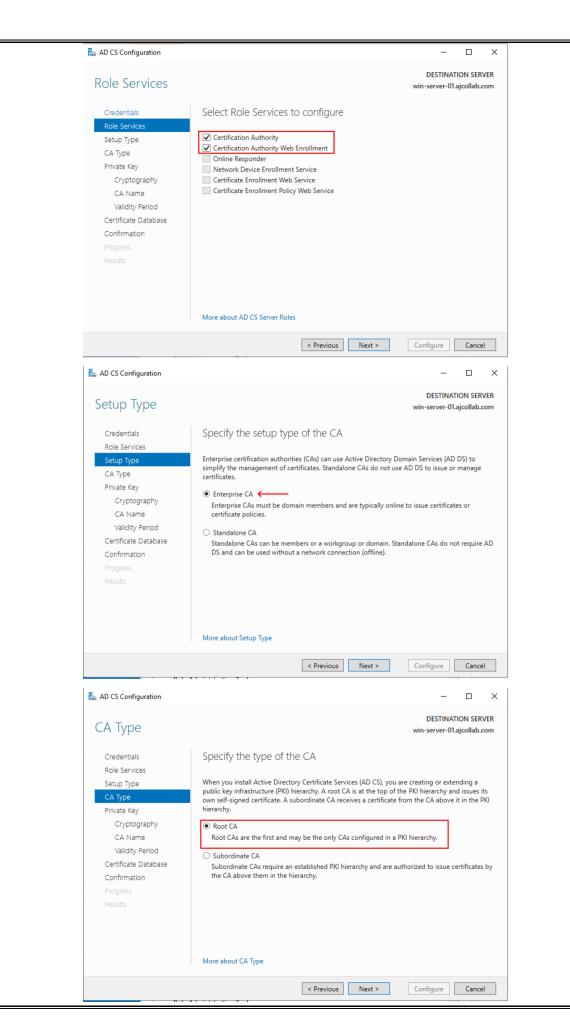
- The Subject Alternative Name field lets you specify additional host names (sites, IP addresses, common names, etc.) to be protected by a single SSL Certificate
- This can be used as a cluster wide certificate; we do not need separate certificates for each node
- From I experience, I have seen most of the enterprises are using Multi SAN certificate for Cisco Tomcat Service

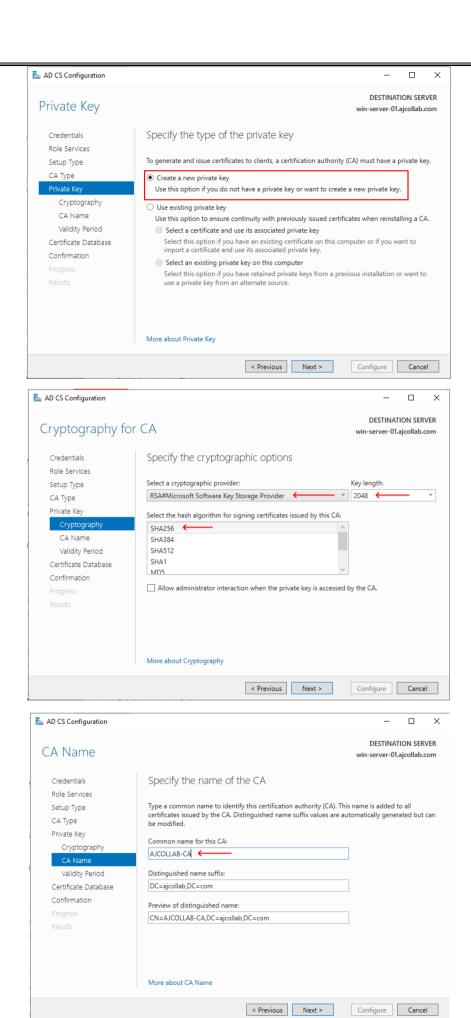
[Lab] Setting up Internal / Enterprise CA in Windows Server 2019

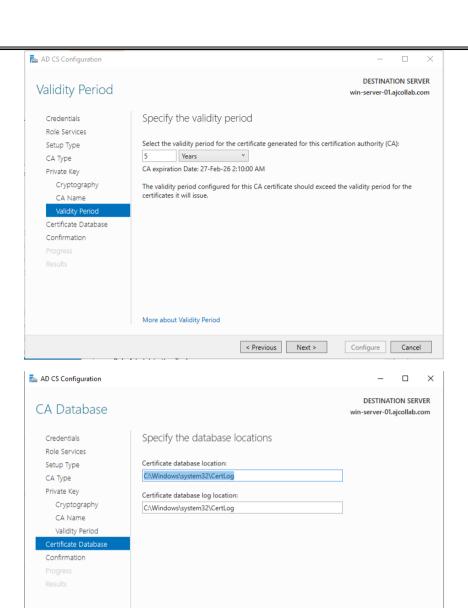






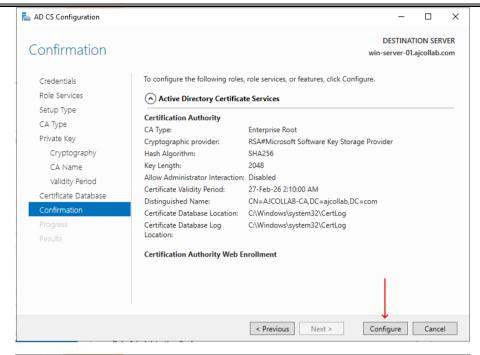


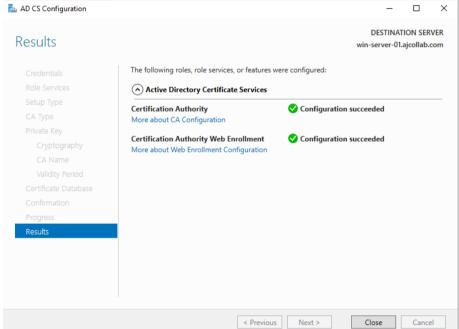




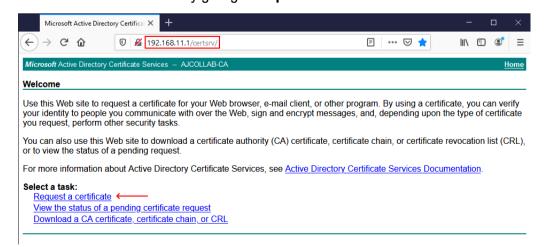
More about CA Database

< Previous Next > Configure Cancel

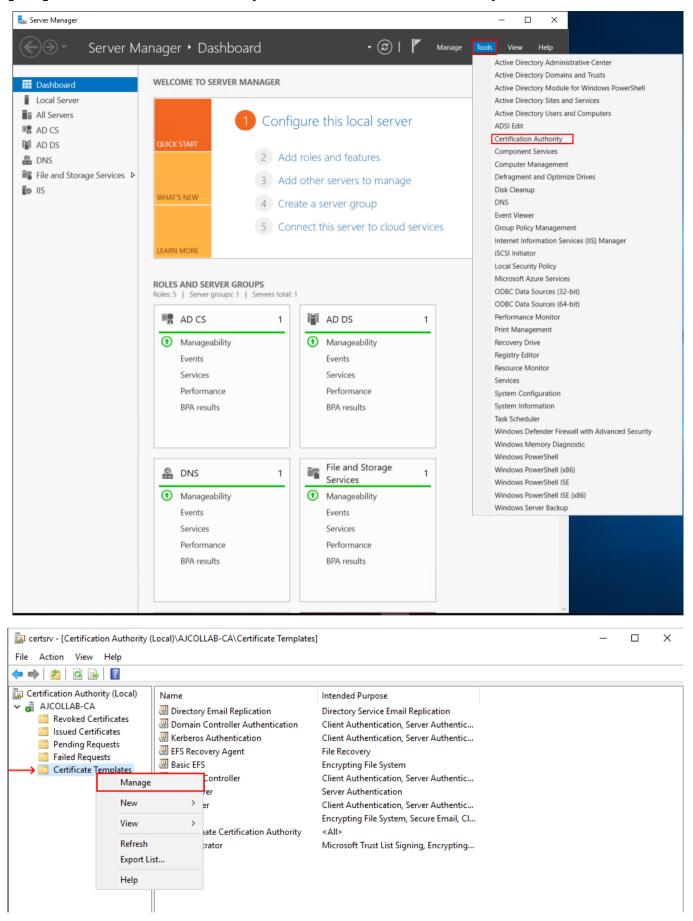


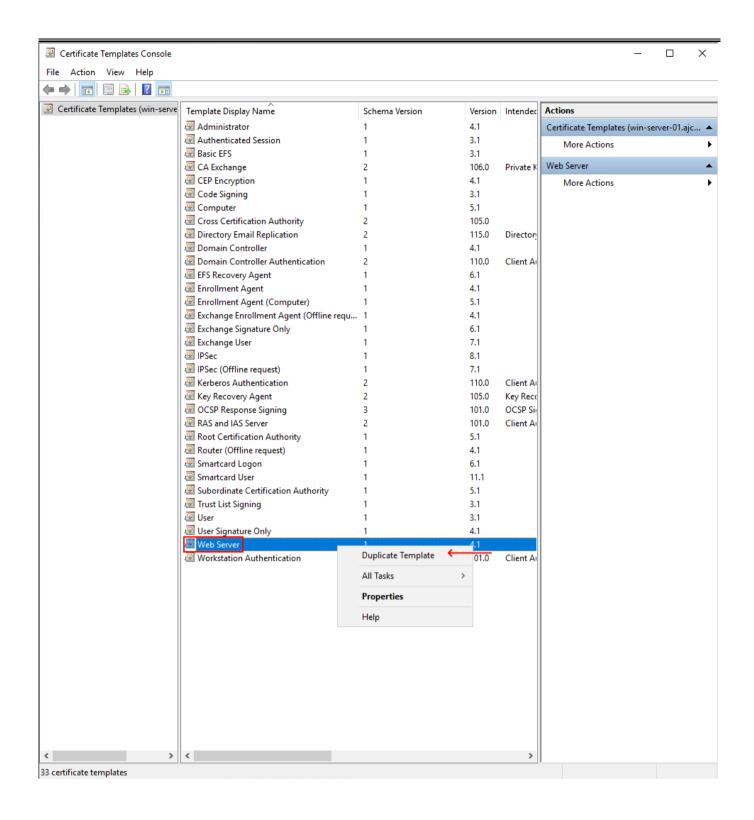


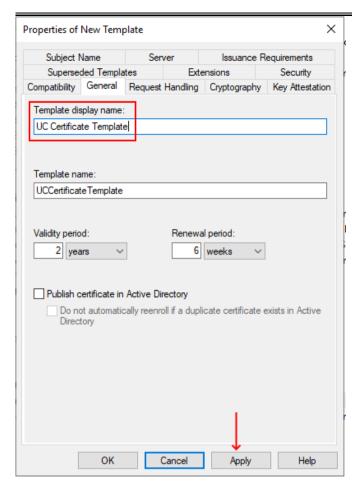
Access the Certificate Services by going to http://WINDOWS-SERVER-IP/certsrv/

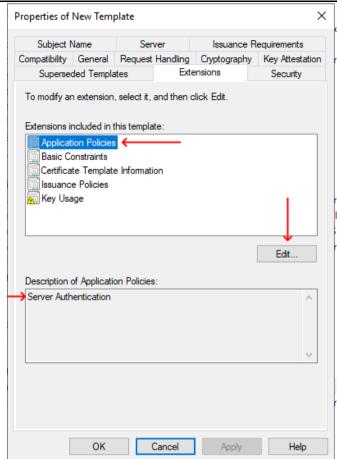


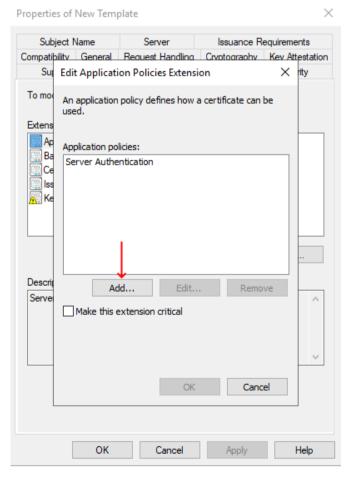
[Lab] Create UC Certificate Templates in Windows Server Enterprise CA

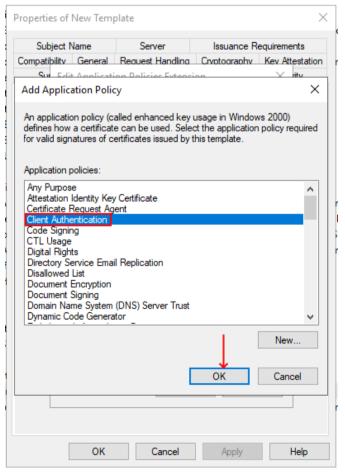


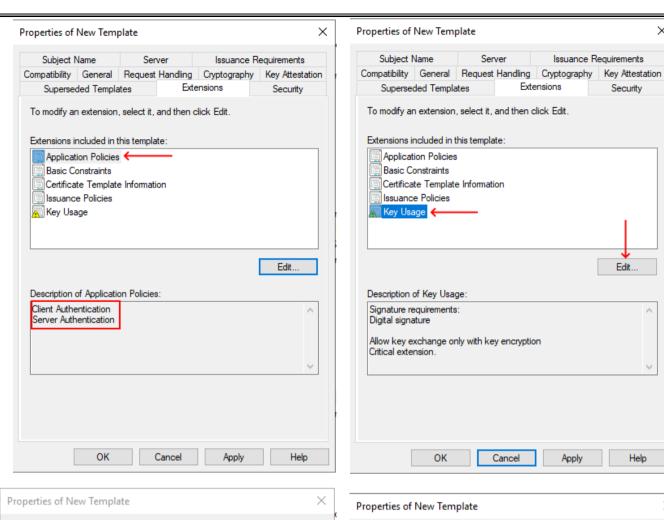


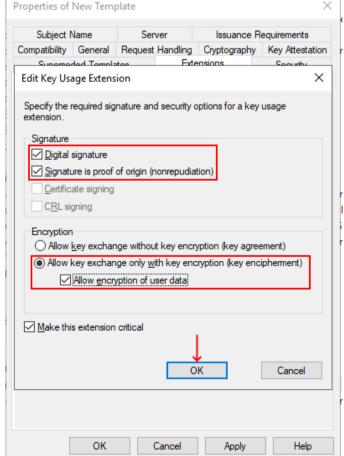


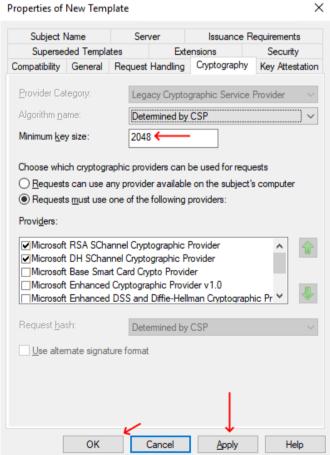








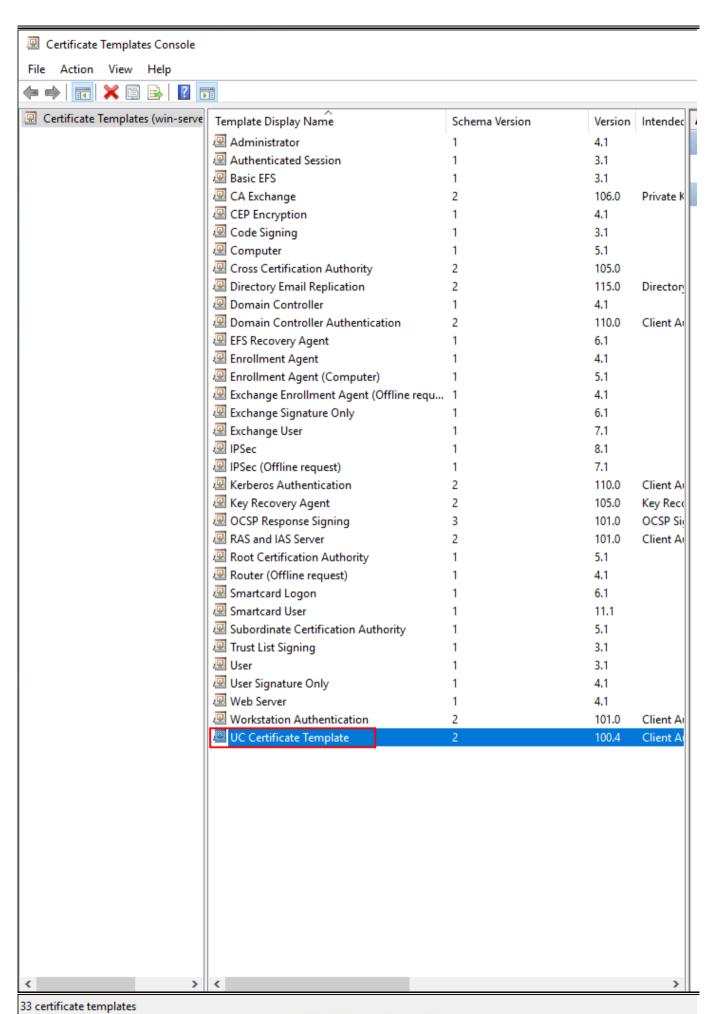


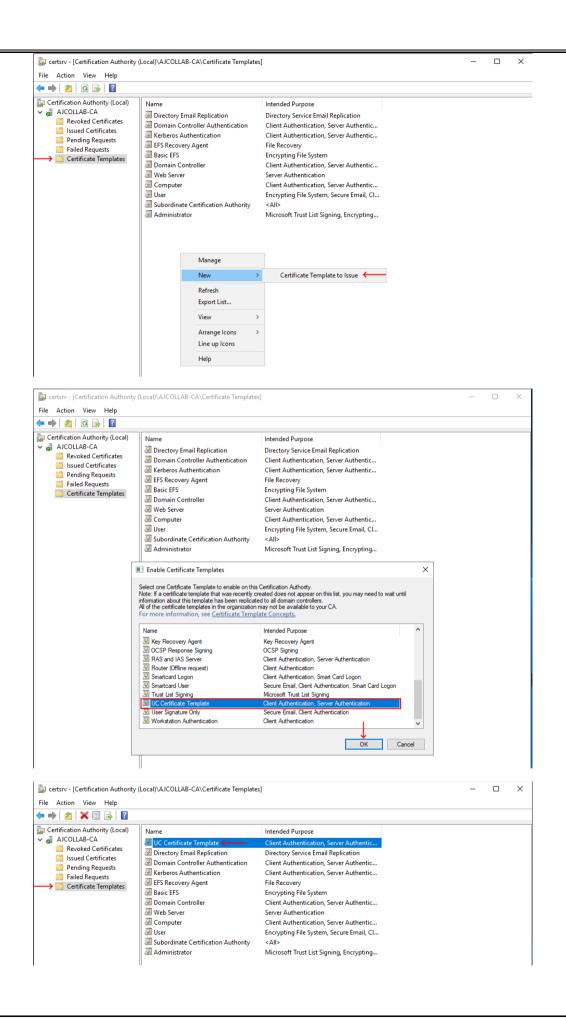


Security

Edit.

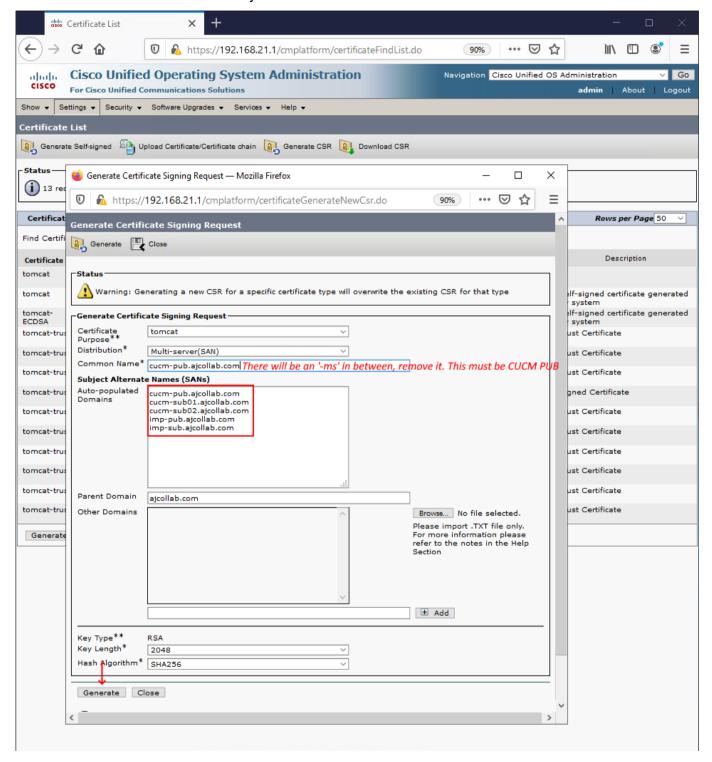
Help

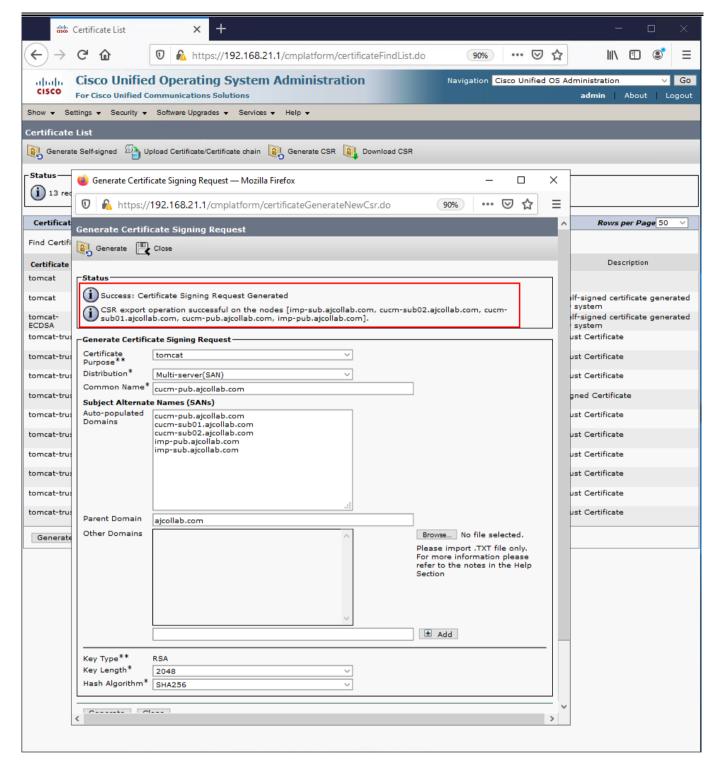




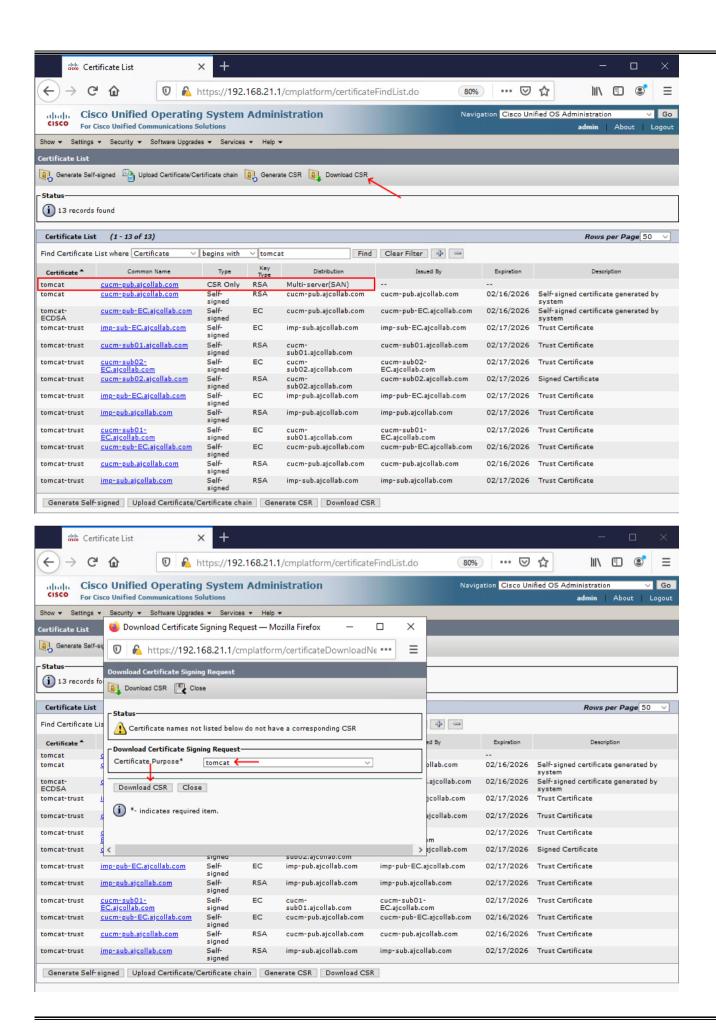
[Lab] Install Multi SAN Certificate for CUCM Cluster Tomcat Service

- On System >> Server >> Set all the server names to FQDNs
- OS Administration >> Security >> Generate CSR

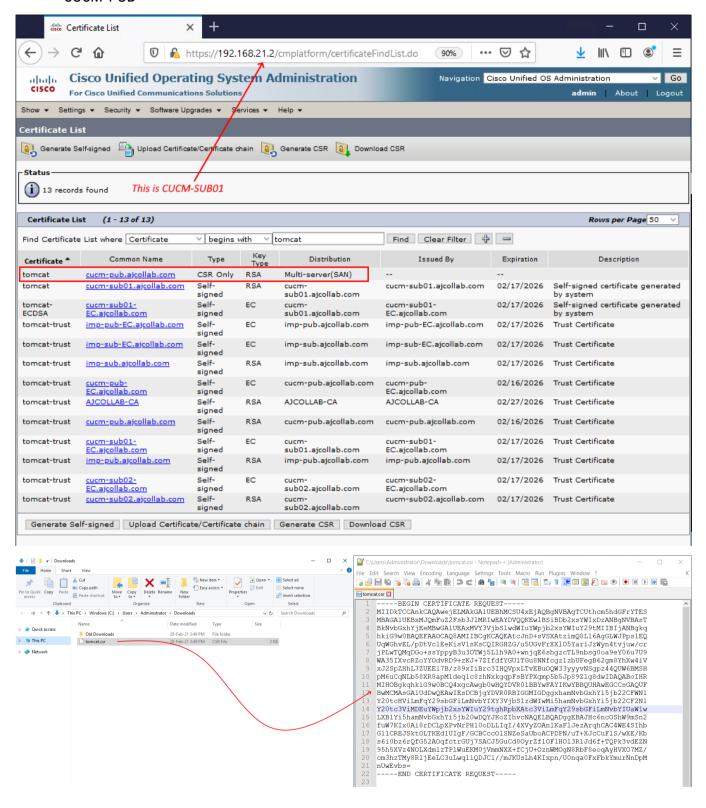


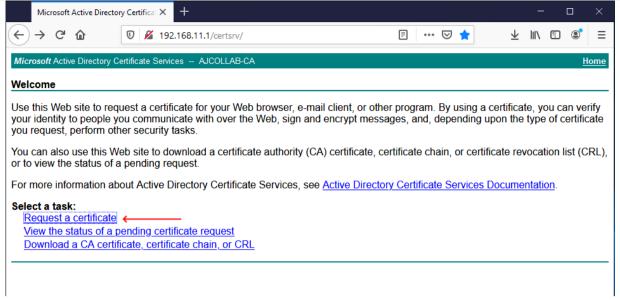


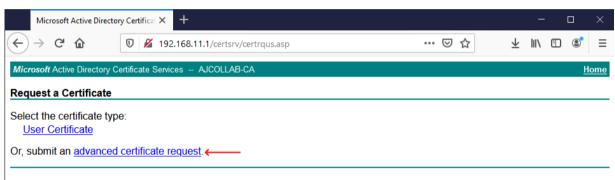
- The same CSR has been pushed to all other nodes in the cluster
- Respective private keys for each node will be created now even if we are performing this task on CUCM-PUB

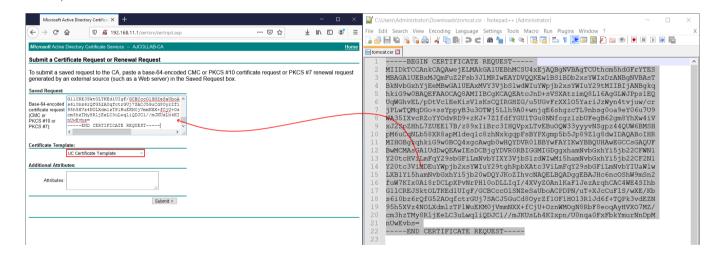


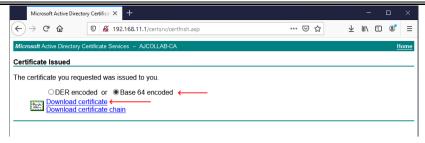
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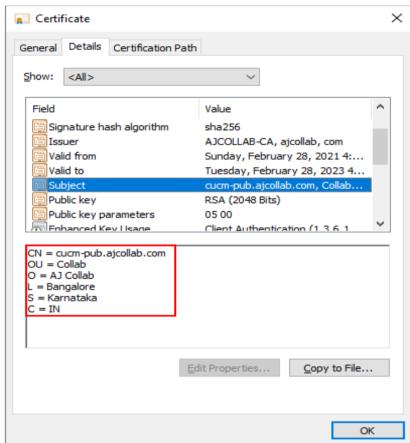


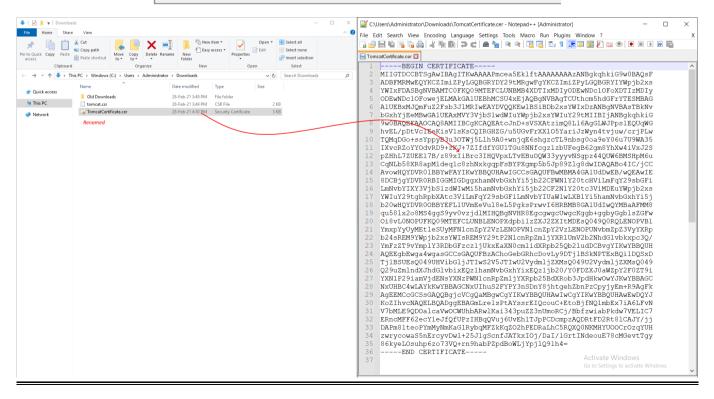


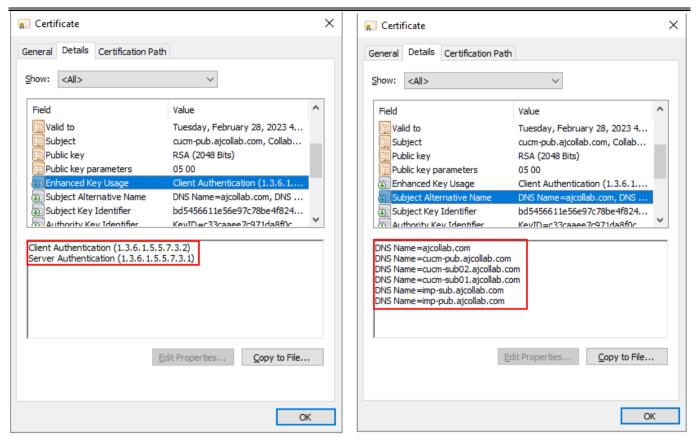


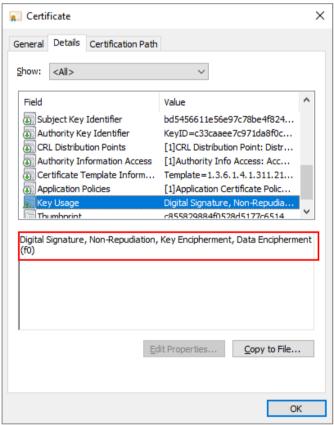














Use this Web site to request a certificate for your Web browser, e-mail client, or other program. By using a certificate, you can verify your identity to people you communicate with over the Web, sign and encrypt messages, and, depending upon the type of certificate you request, perform other security tasks.

You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CRL), or to view the status of a pending request.

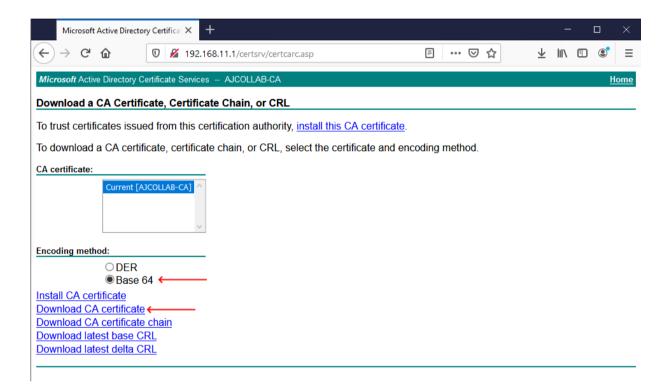
For more information about Active Directory Certificate Services, see Active Directory Certificate Services Documentation.

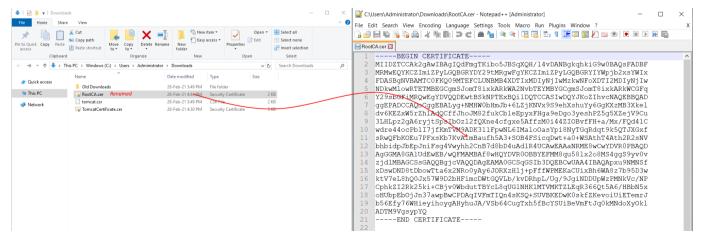
Select a task:

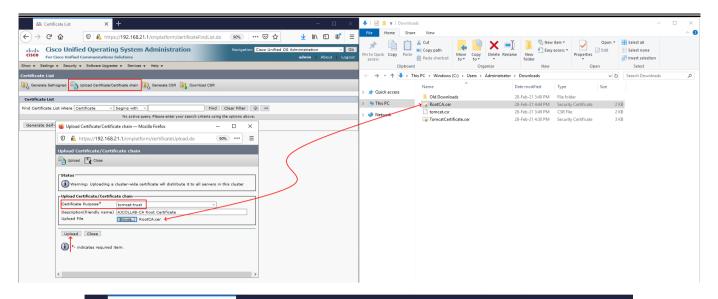
Request a certificate

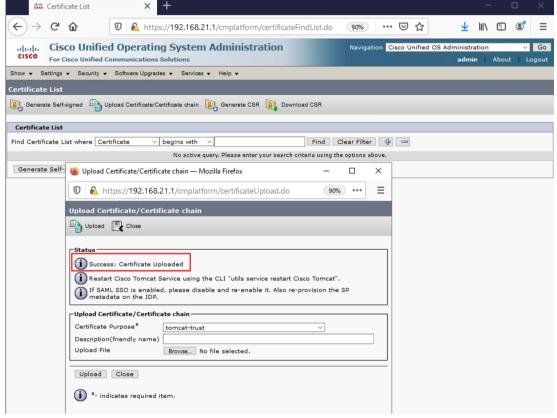
View the status of a pending certificate request

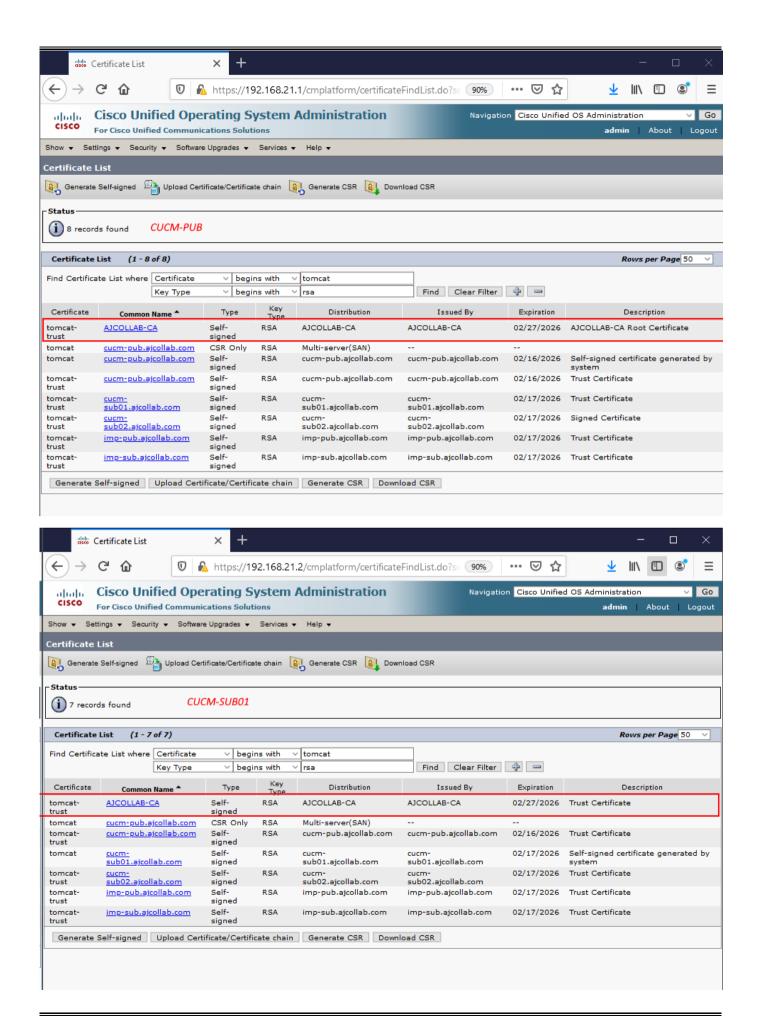
Download a CA certificate, certificate chain, or CRL

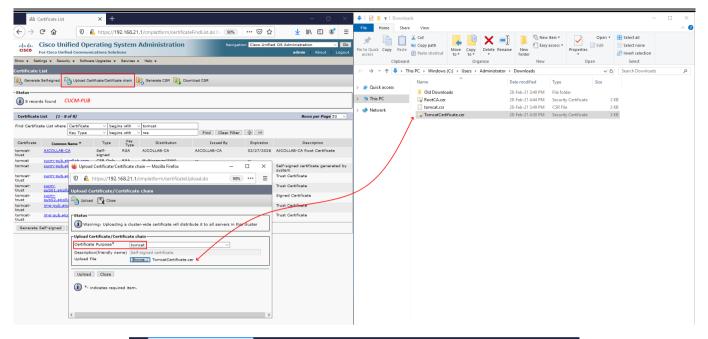


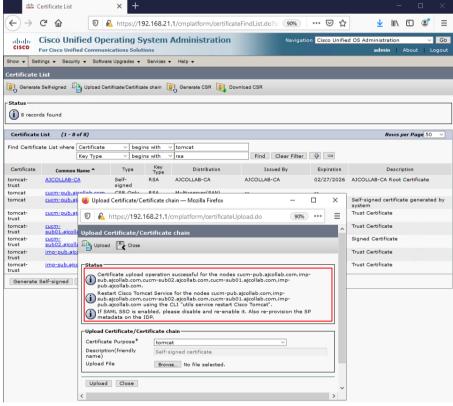


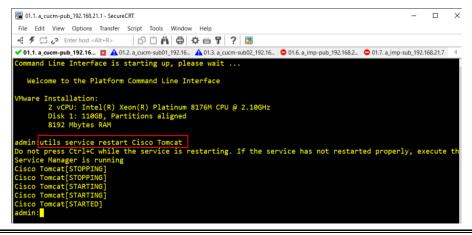


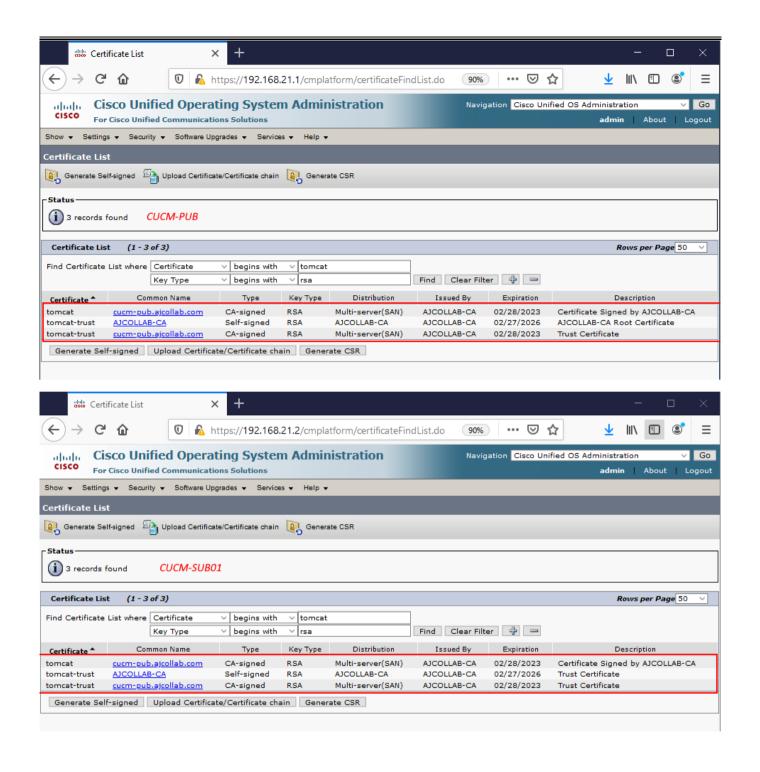




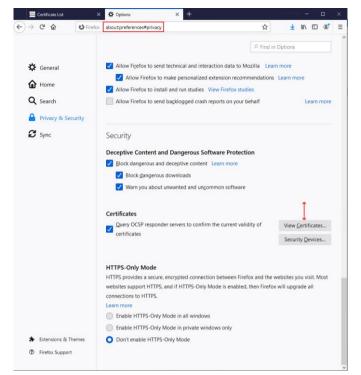


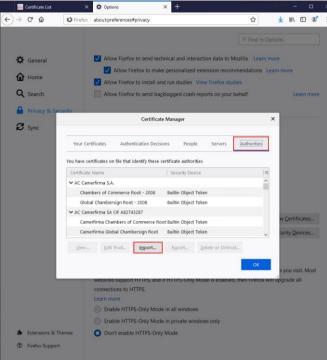


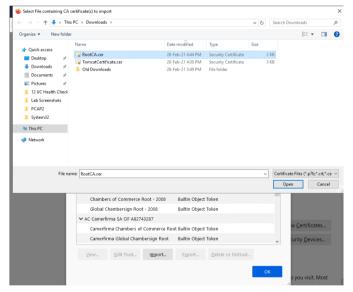


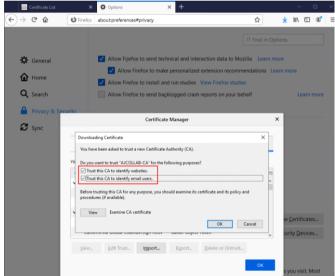


· Adding the Root-CA as trust in local Firefox Browser certificate store

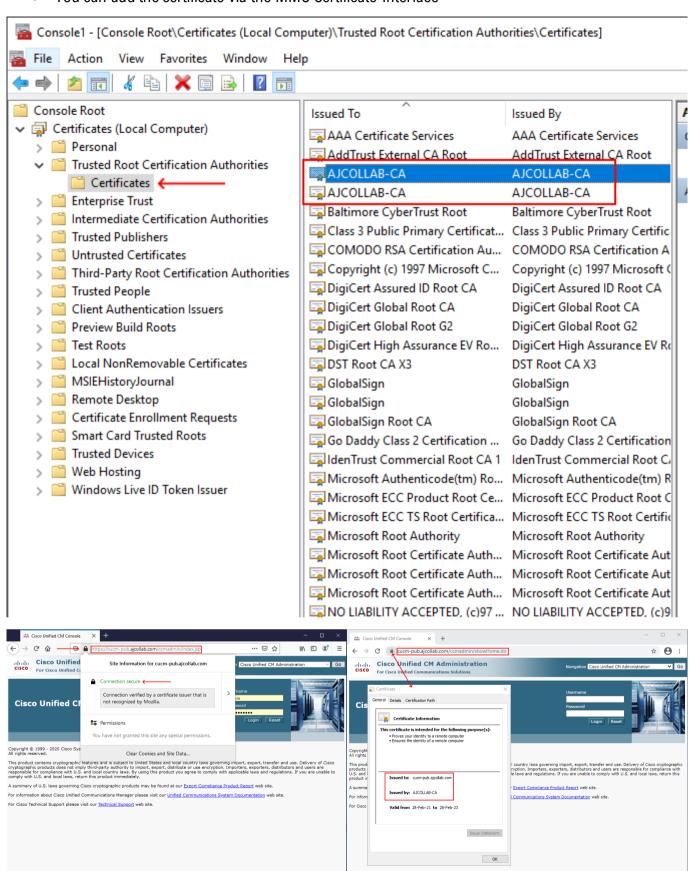








You can add the certificate via the MMC Certificate Interface



Cisco IP Phone Services



- Cisco IP Phones services are applications that use the web client, web server and XML capabilities
 of Cisco IP Phone
- IP Phone firmware contains a micro web browser that enables limited web browsing capability
- For example, IP Phone can be used to turn ON and OFF the lights in a conference hall, IP Phones can be used to order food from a hotel room, etc.
- These services are subscribed to the specific IP Phones so that those phones can get these features. Enterprise subscription will make sure all the IP Phones will get that service
- Following list represents some of the configuration parameters related to IP Phone services and XML operations

To tune the IP Phone service URLs, go to System >> Enterprise Parameters)

URL Authentication: http://ANY-CUCM-NODE-IP:8080/ccmcip/authenticate.jsp

This URL gets used to validate requests made directly to the phone. This URL is automatically configured at install time. If the URL is removed, the push capabilities to the Cisco IP Phones will be disabled.

URL Directories: http://ANY-CUCM-NODE-IP:8080/ccmcip/xmldirectory.jsp

This parameter specifies the URL when the Directory button is pressed (missed calls, dialed calls, etc.).

URL Idle:

Points to a service that provides text or image to be displayed on the phone screen when the phone is idle

URL Idle Time:

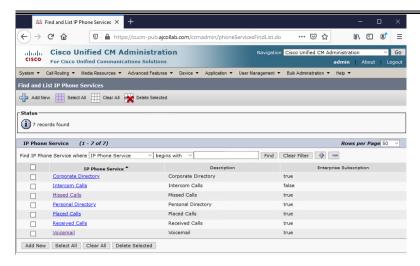
Parameter indicates the time in seconds that a phone wait before initiating URL idle service

URL Information: http://ANY-CUCM-NODE-IP:8080/ccmcip/GetTelecasterHelpText.jsp
It points GetTelecasterHelpText.jsp service in CUCM. It provides help or call statistics when user pushes '?/ i' button

URL Services: http://ANY-CUCM-NODE-IP:8080/ccmcip/getservicesmenu.jsp

It points to getservicesmenu.jsp services. It returns a CiscoIPPhoneMenu object with a list of services that are subscribed to the device when the user presses the service (Globe) button

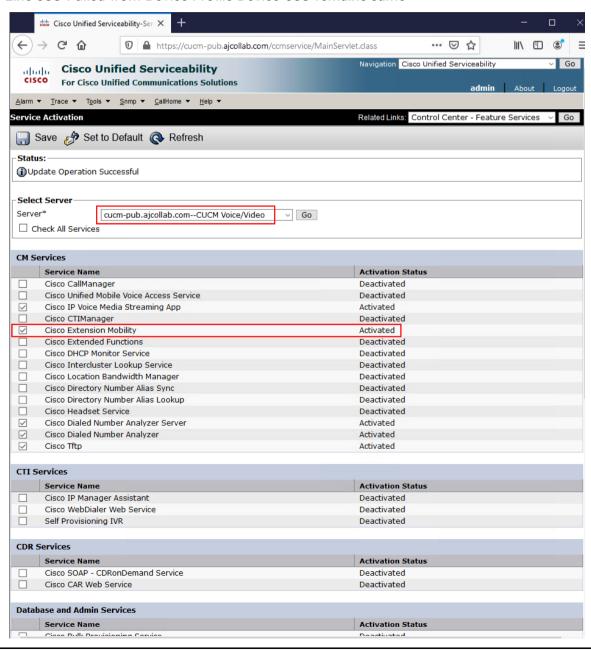
Note: If your Phones doesn't have proper DNS server to resolve these DNS host names, the services will fail. Such scenarios, you can change these services to IP Address based instead of FQDN.

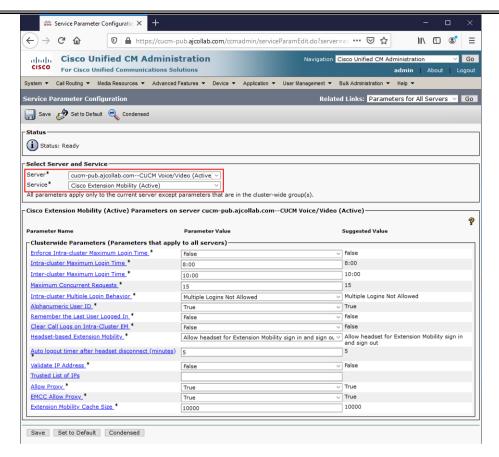


- There are few default build in services available
- Device >> Device Settings >> IP
 Phone Services

[Lab] Extension Mobility

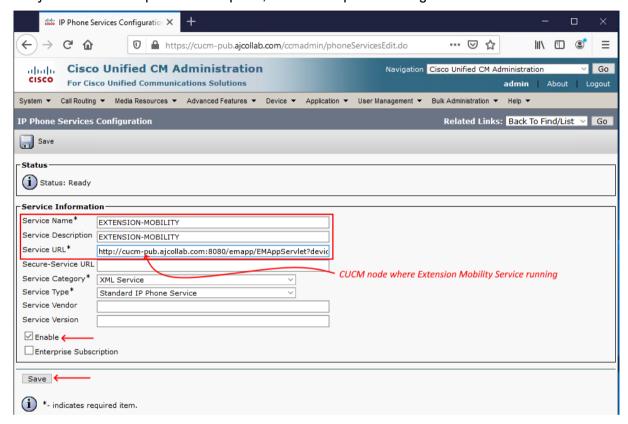
- Allows roaming users to login to any device and get their personal settings such as Line number,
 Speed dial, Forward settings, calling privileges, Music on Hold source etc.
- Device specific parameter remain the same
- Instead of configuring phones for users, we just create Device Profile. It is a virtual phone profile that can move around to whatever phones the user logs into
- The configuration changes are triggered by a user login with a user ID & password, when the user stops using the phone, he / she logs out and default configuration reapplied
- It is implemented as a phone service and works on single cluster. From CUCM V8 onwards
 Extension Mobility Cross Cluster (EMCC) can be implemented
- Line CSS Pulled from Device Profile Device CSS remains same

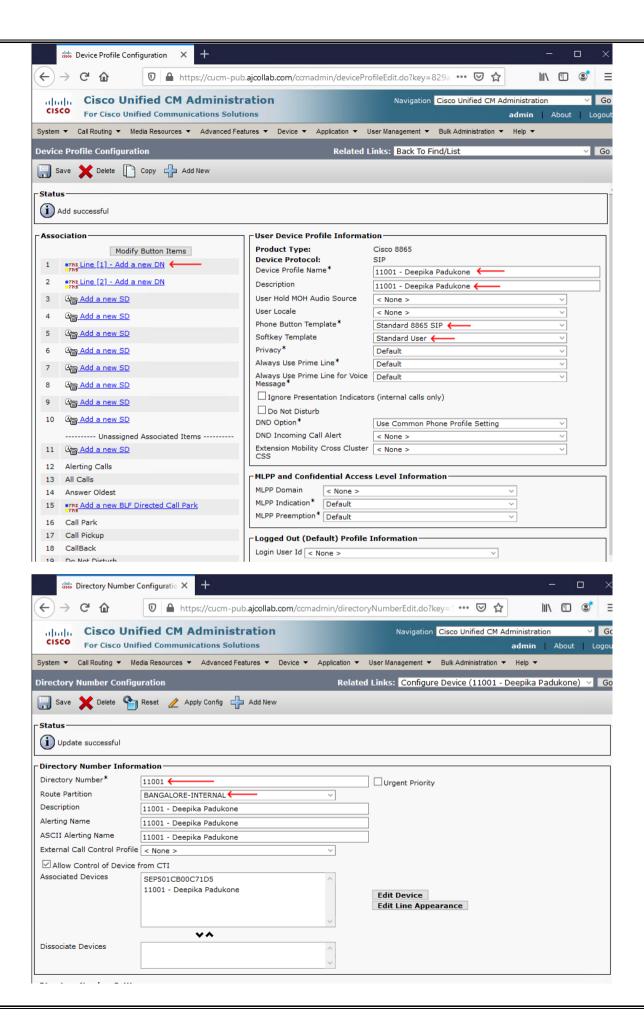


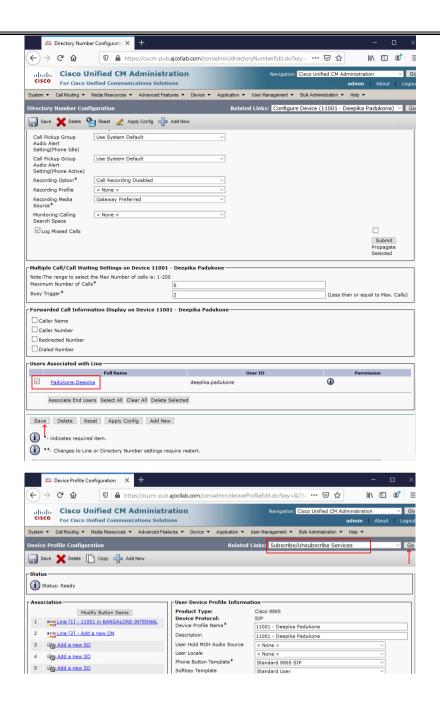


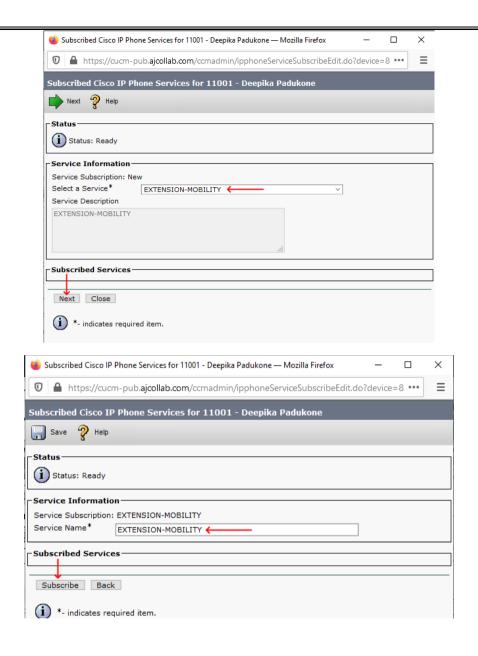
- **Enforce Intra-cluster Maximum Login Time**: This parameter determines whether a maximum login time is enforced for local login.
- Intra-cluster Maximum Login Time: This parameter specifies the maximum time that a user is allowed to be locally logged in to a device. (Default 8 hours). After 8 hours the user automatically logged out. The system ignores this parameter if the Enforce Maximum Login Time parameter is set to False.
- Inter-cluster Maximum Login Time: This parameter specifies the maximum time that a user is allowed to be remotely logged in to a device in EMCC mode. EMCC always enforce auto logout based on this value irrespective of the value of Enforce Maximum Login Time parameter (Default is 10 Hours)
- Intra-cluster Multiple Login Behavior: This parameter specifies the behavior for multiple
 attempted logins by the same user on different devices within the same cluster allowed or not. For
 EMCC, multiple logins are always allowed
- Alphanumeric User ID: This parameter specifies whether the user ID to be used is alphanumeric
 or numeric.
- Remember the Last User Logged In: This parameter specifies whether the user ID of the last user logged in on a phone is remembered by the extension mobility application.

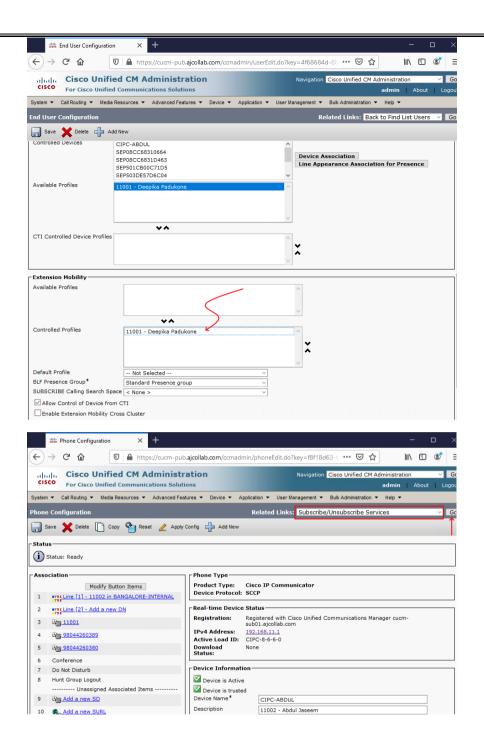
- Clear Call Logs on Intra-Cluster EM: This parameter determines whether the call information stored on the phone directory (missed calls, placed calls, received calls) is cleared when a user manually logs in or out of a phone in the same cluster. For Extension Mobility Cross-Cluster (EMCC), the call log is always cleared when the user logs in or out of a phone)
- Device >> Device Settings >> Phone Services >> Add New >>
- http://cucm-pub.ajcollab.com:8080/emapp/EMAppServlet?device=#DEVICENAME#
- Here CUCM-PUB runs Extension Mobility Service
- If you enable Enterprise Subscription, then all the phones will get the service

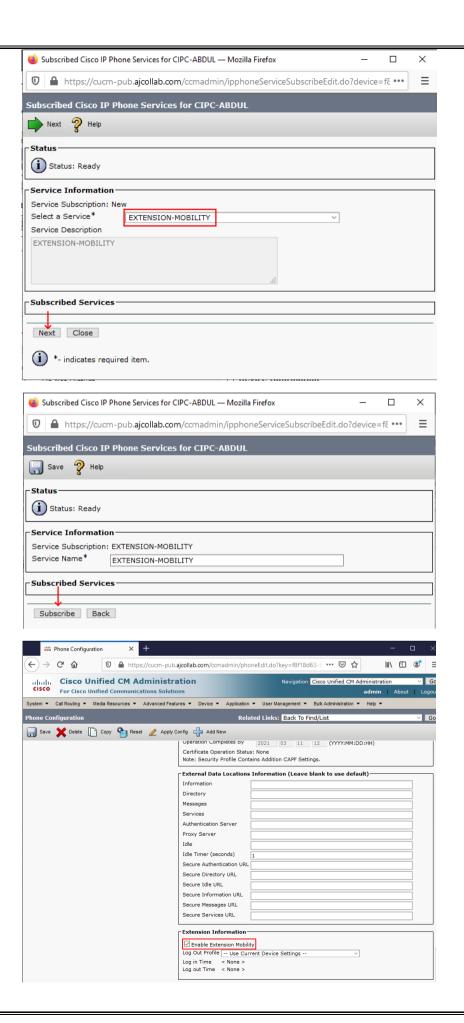




















Troubleshooting Extension Mobility and Common Error Codes

While working in any enterprise UC infrastructure we usually face some common issues associated with extension mobility. I have added most familiar issues that I have dealt with in the past.

Error: Host not found

- Make sure IP Phone can reach the service URL FQDN
- Make sure IP Phone can resolve the service URL FQDN
- Check Tomcat service is running on the Extension Mobility Enabled node
- If service URL is updated, please update the subscriptions in the phone

You can't see the EM feature after hitting the services button

- Verify that you have configured the Extension Mobility service
- Verify the service URL is correct
- Start/Restart the EM services on the node

You can't log in/out of the EM feature, but you can see it after pressing the services button

- This error comes when you haven't enabled the extension mobility on the Phone
- Subscribed the service to the phones/device profiles but haven't associated user to a device profile

Error: -After performing a login or logout, the user finds that the phone resets instead of restarting

- Locale change may provide the basis for reset.
- If the User Locale that is associated with the login user or profile is not the same as the locale or device, after a successful login, the phone will perform a restart that is followed by a reset.
- This occurs because the phone configuration file is being rebuilt

Error [201] - Authentication error

The user should check that the correct User ID and PIN were entered

Error [22] - Dev.logon disabled

 Make sure that you have chosen "Enable Extension Mobility" check box on the Phone Configuration window.

Error [205] - User Profile Absent

Make sure that you have associated a Device Profile to the user

Error [208] - EMService Conn. error

 Verify that the Cisco Extension Mobility service is running on the node where service URL is pointed to

Error [25] - User logged in elsewhere

- Check whether the user is logged in to another phone.
- If multiple logins need to be allowed, ensure the Multiple Login Behavior service parameter is set to Multiple Logins Allowed or Auto logout

Error [503] - Http Error

Check that the Cisco Extension Mobility Application service is running on the node where service
 URL is pointed to

Error [202] - Blank user ID or Pin

Make sure that you enter a valid user ID and Pin

Error [26] - Busy, please try again

- Check whether the number of concurrent login/logout requests is greater than the Maximum Concurrent requests service parameter. If so, lower the number of concurrent requests
- To verify the number of concurrent login/logout requests, use Cisco Unified Communications
 Manager Cisco Unified Real-Time Monitoring Tool to view the Requests in Progress counter in the
 Extension Mobility object.

Error [6] - Database Error

- Check whether a large number of requests exists
- If large number of requests exists, the Requests in Progress counter in the Extension Mobility object counter specifies a high value.
- If the requests are rejected due to large number of concurrent requests, the Requests Throttled counter also specifies a high value.

Error [207] - Device Name Empty or Error: - XML Error [4] Parse Error

 Check that the URL that is configured for Cisco Extension Mobility is correct and there shouldn't be any space in between

Error - Cisco 8945 IP Phone does not show EM service

Set service provisioning to default or internal. Refer Bug CSCtx70127

Error [http-8080-9]

EMX509TrustManager - checkServerTrusted: BSCUCM001.blocksolutions.local Certificate not found in the keystore: the certificate chain is not trusted, Could not validate path.

- This happens when we have HTTPS in the service URL
- TVS Service failed to verify the service URL certificate, usually happens when there was a certificate related activity (regenerate, install new, etc.)
- You can try restarting TVS service of the node where phone is registered
- Make sure certificate activities are done in proper way

Error [213] - Login is unavailable

- This error comes when the device or phone load does not support EMCC (e.g. non-supported phone models, supported phone models with older phone load)
- It could also be the incorrect service URL and/or secure Service URL

Error - Untrusted IP Error

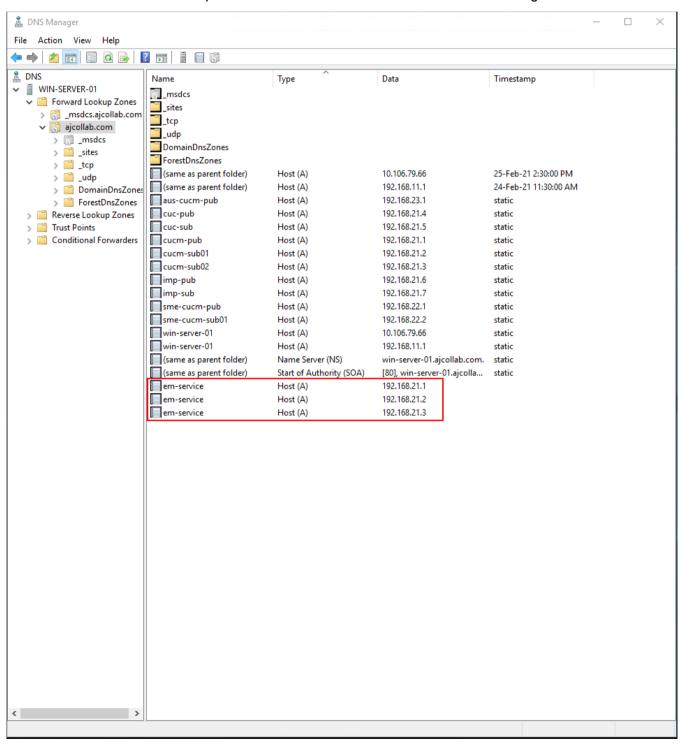
 This error comes when "Validate IP Address" service parameter is set to true and user tries to login/logout from a machine whose IP address is not trusted i.e. not listed in Trusted List of Ips service parameter

Error - 79XX phones cannot access certain SURLs when running firmware 9-0-3+

Upgrade or Downgrade to the supported firmware

DNS Based Redundancy in Extension Mobility

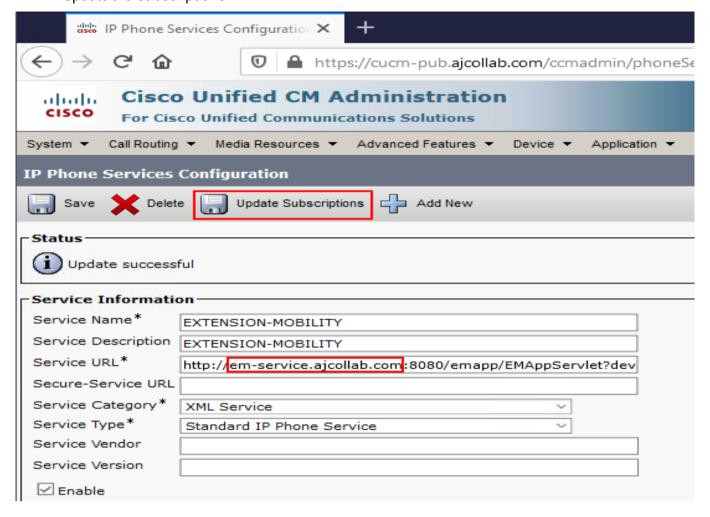
- As we know that we can configure only one service URL per service in Extension Mobility that is http://cucm-pub.ajcollab.com:8080/emapp/EMAppServlet?device=#DEVICENAME#
- Here if EM service cucm-pub.ajcollab.com is down or the node itself down, then the future fails for the users
- To provide redundancy on EM Future, we can enable Extension Mobility Service on other nodes, then
 create a DNS A Record that points to IP Address of the nodes those are running EM Service



 The FQDN resolves to each IP address alternatively, this is a kind of stateless load balancing and uses default behavior of DNS Server

```
Administrator: Command Prompt
                                                                                                                                                                                                                                                                                                                                                                                                                                             П
                                                                                                                                                                                                                                                                                                                                                                                                                                                                    ×
   Microsoft Windows [Version 10.0.17763.107]
(c) 2018 Microsoft Corporation. All rights reserved.
responsible mail addr = (rot
serial = 0
refresh = 28800 (8 hours)
retry = 7200 (2 hours)
expire = 604800 (7 days)
default TTL = 86400 (1 day)
Server: UnKnown
 refresh = 28800 (8 hours)
retry = 7200 (2 hours)
expire = 604800 (7 days)
default TTL = 86400 (1 day)
    Server: UnKnown
   Name: em-service.ajcollab.com
Addresses: 192.168.21.2 (
                                      192.168.21.3
192.168.21.1
responsible mail addition (responsible mail addi
   Server: UnKnown
   Name:
                                        192,168,21,2
   C:\Users\Administrator>_
```

- Update the service URL with new FQDN
- Update the Subscriptions



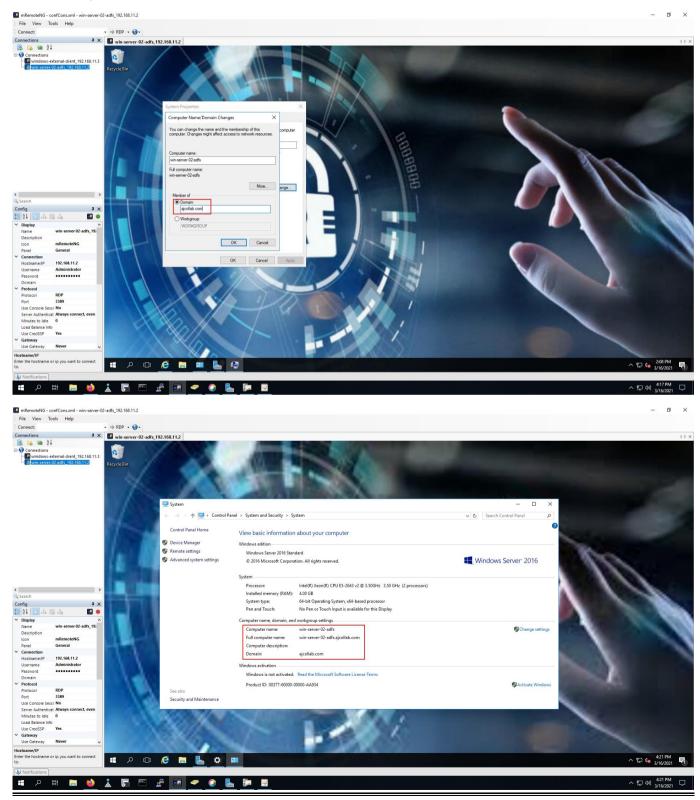
Single sign-on (SSO)

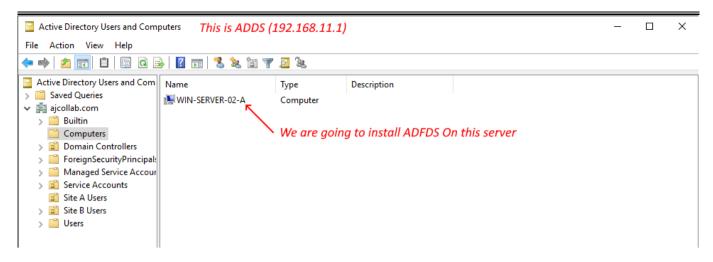


- Single sign-on (SSO) is an authentication method that enables users to securely authenticate with multiple applications and websites by using just one set of credentials
- One set of credentials to access multiple different services
- Authentication handled by IdP (Identity Provider) server, there are 2 types of SSO
- Intra-Organizational SSO: Access resources within the organization
- Inter-Organizational SSO: Also known as federated SSO, establish trust between multiple orgs to authenticate users
- Identity information formatted using Security Assertion Markup Language (SAML) 2.0, XML based open standard
- Service Provider (SP): Provides the service that being utilized (application or system that user logs in to. e.g. CUCM, CUC, IMP, etc.)
- Identity Provider (IdP): System that challenges a user for their credentials and tells the SP if the login was successful or not. Microsoft Azure and okta are cloud based IdPs Microsoft Windows ADFDS, PingiD etc. are other IdPs
- Claim: An IdP configuration that determines what information is to send to SP. It's basically taking some attributes (uid as SAM-AccountName)
- In the CUCM SSO, the SP is CUCM and the IdP can be a Windows Server with ADFDS installed

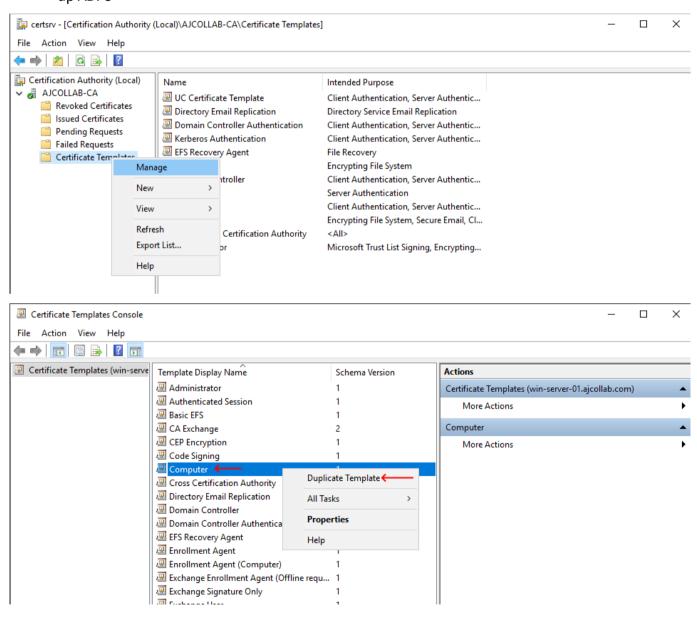
[Lab] Configure Active Directory Federation Services (ADFS)

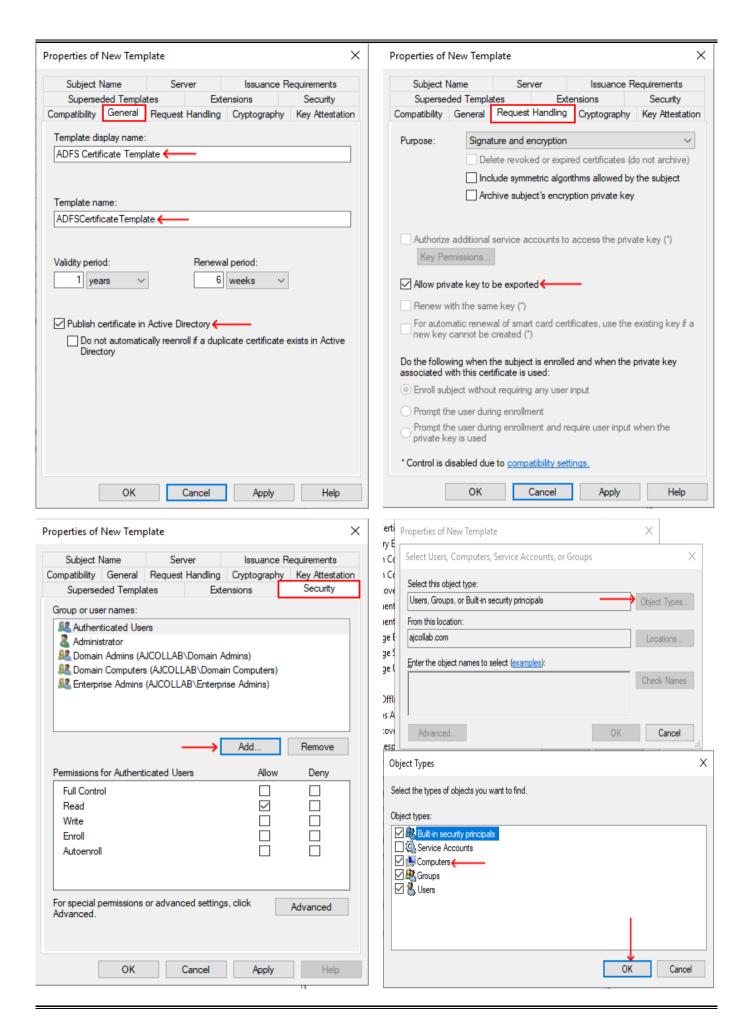
- I have installed a Windows Server 2016 for the ADFS Service, again, installing Windows Server is out of the scope of this article
- 192.168.11.1 AD DS, CA
- 192.168.11.2 AD FDS
- Add your ADFS Server to the Domain

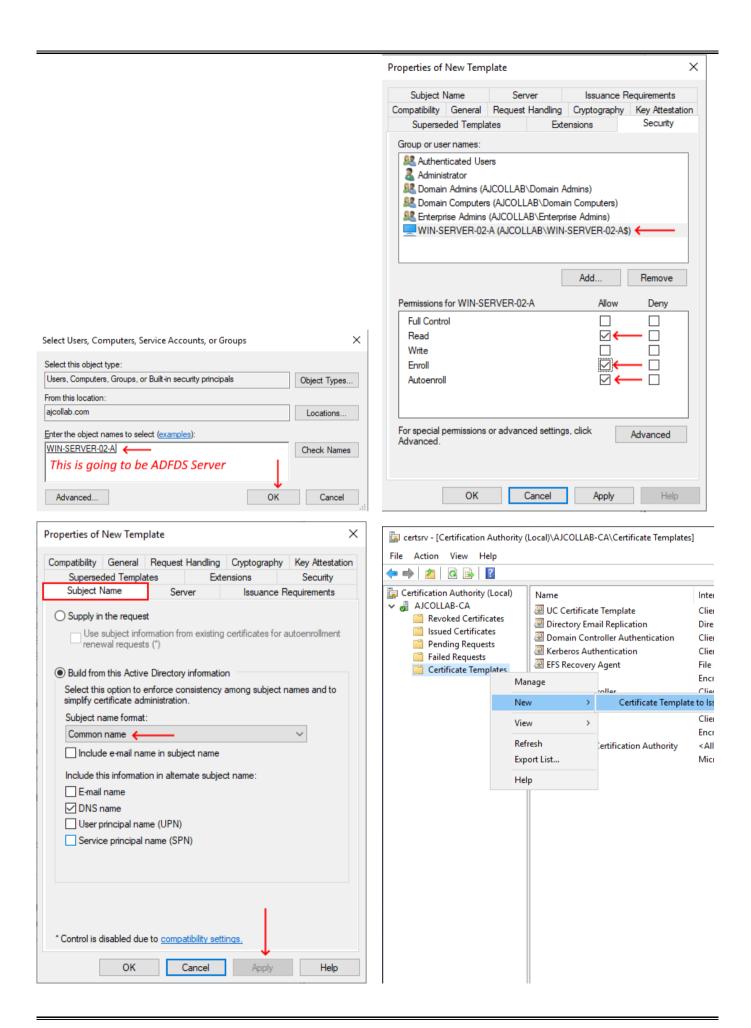


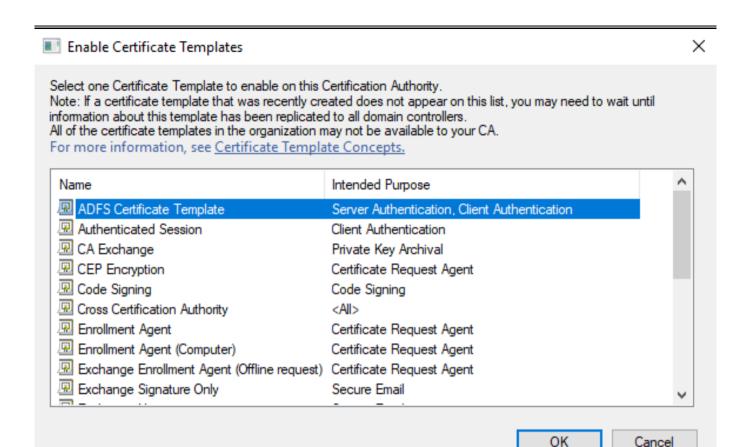


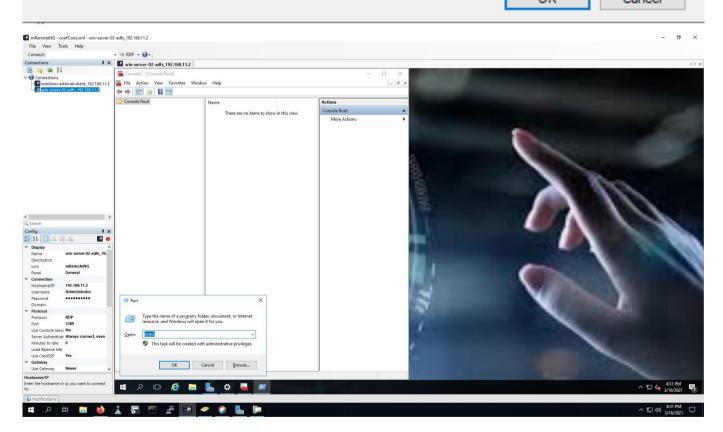
 On your CA Server Create a Certificate Template that will be used by ADFS machine after setting up ADFS

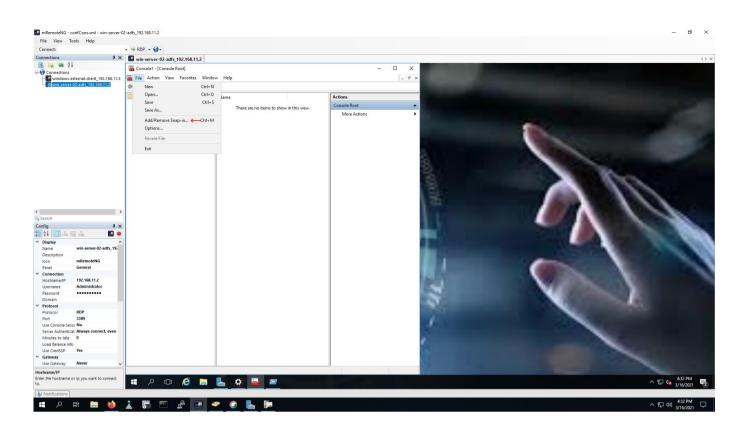


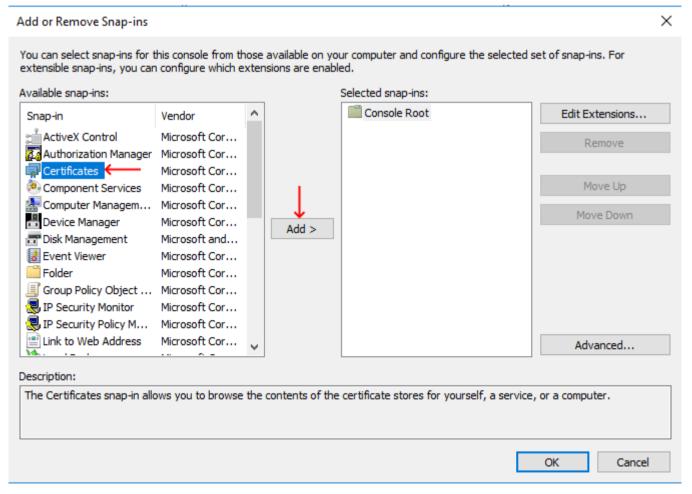


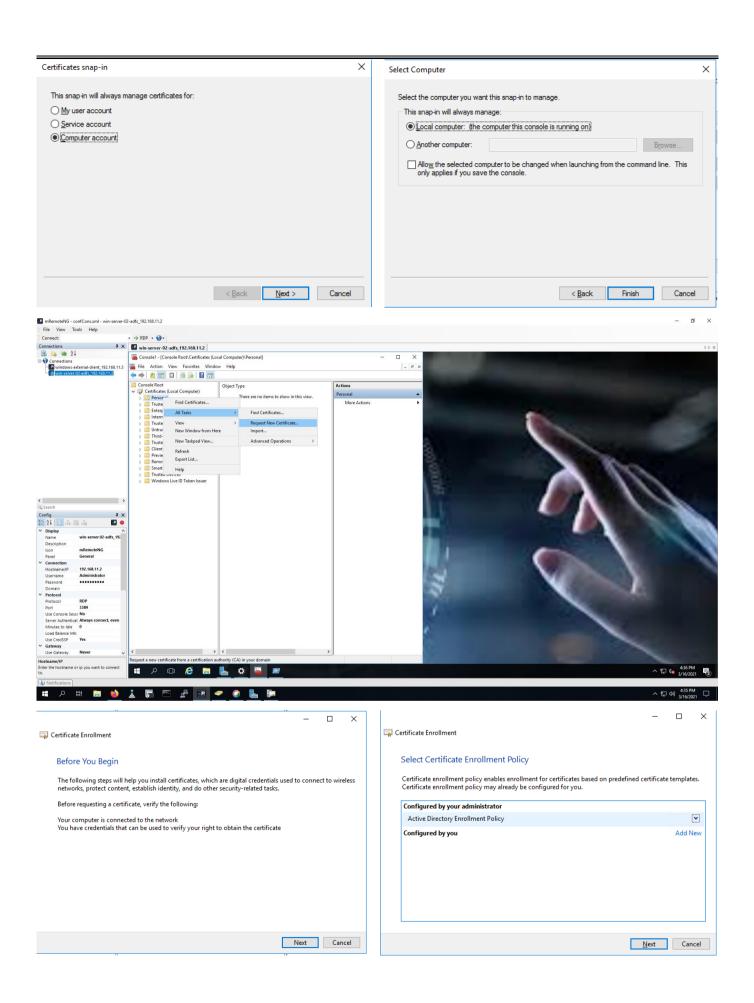


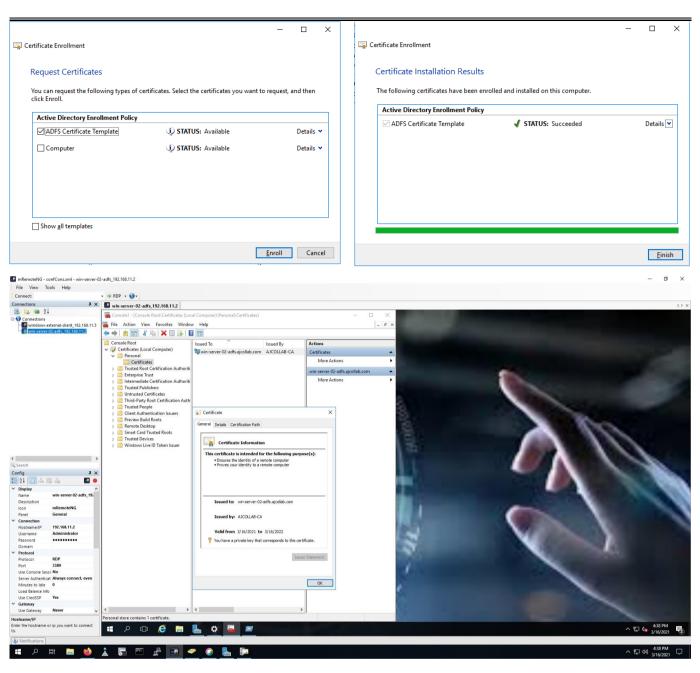




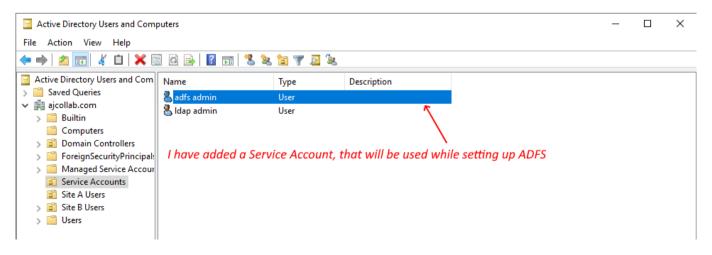




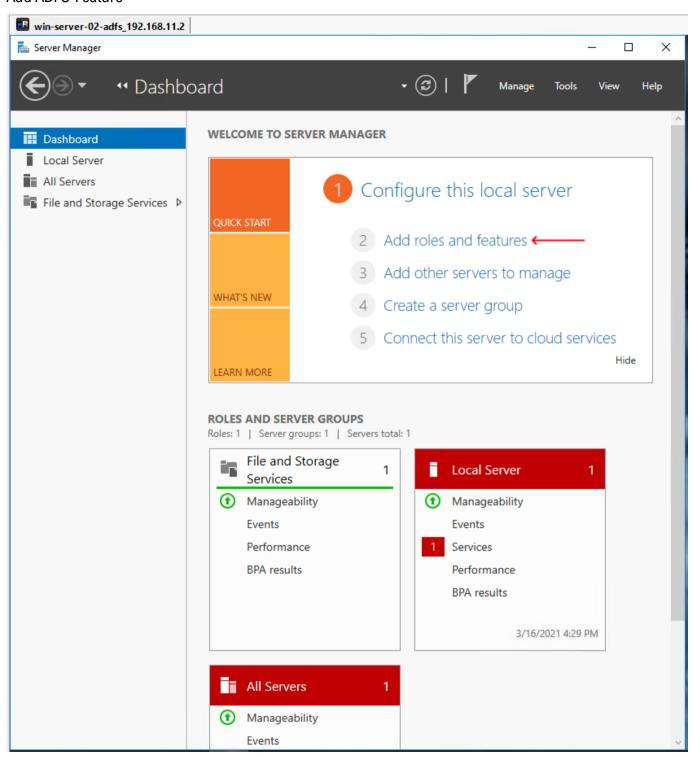


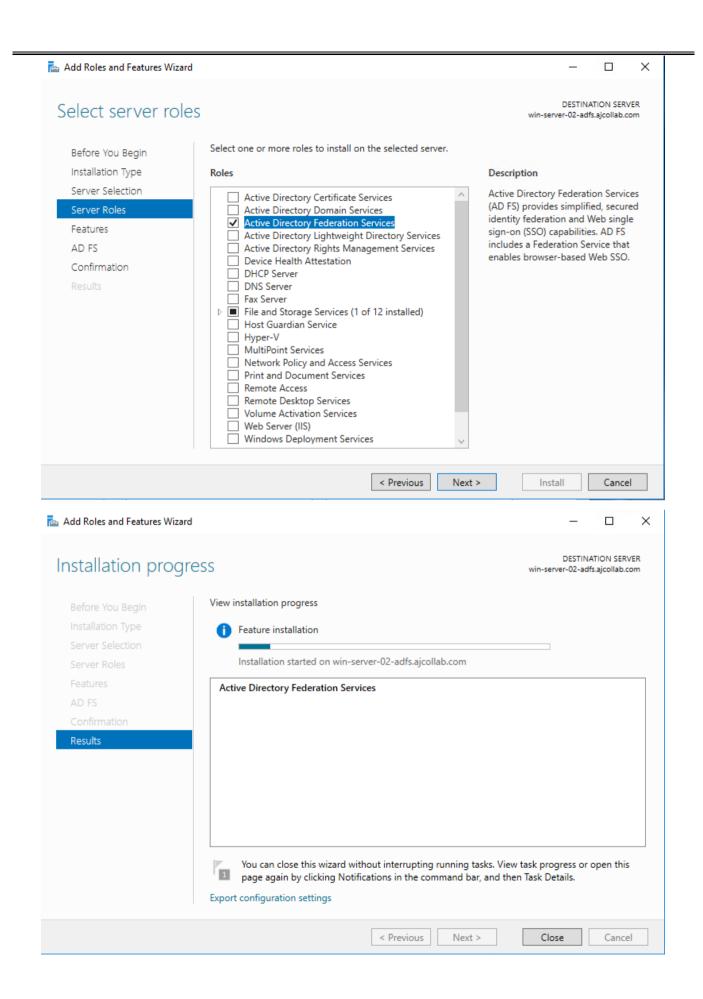


Add a Service Account in AD

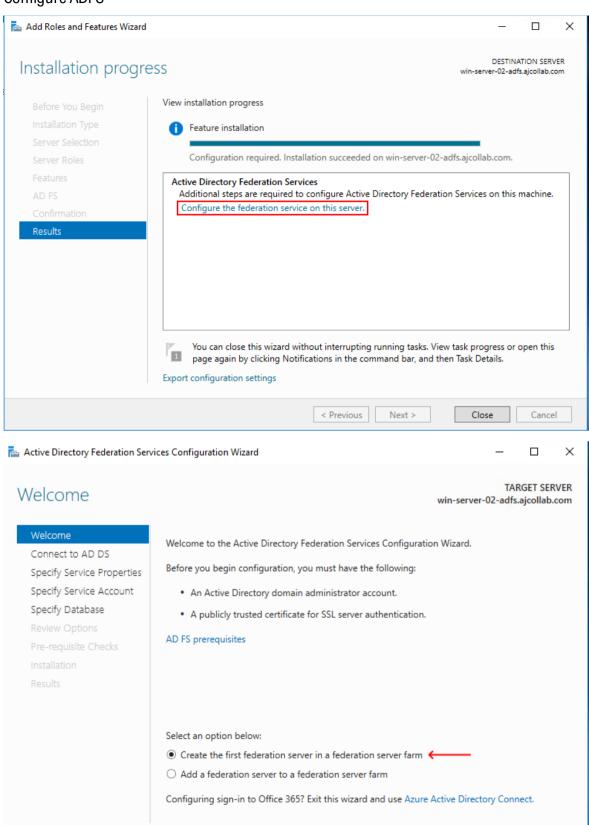


Add ADFS Feature





Configure ADFS

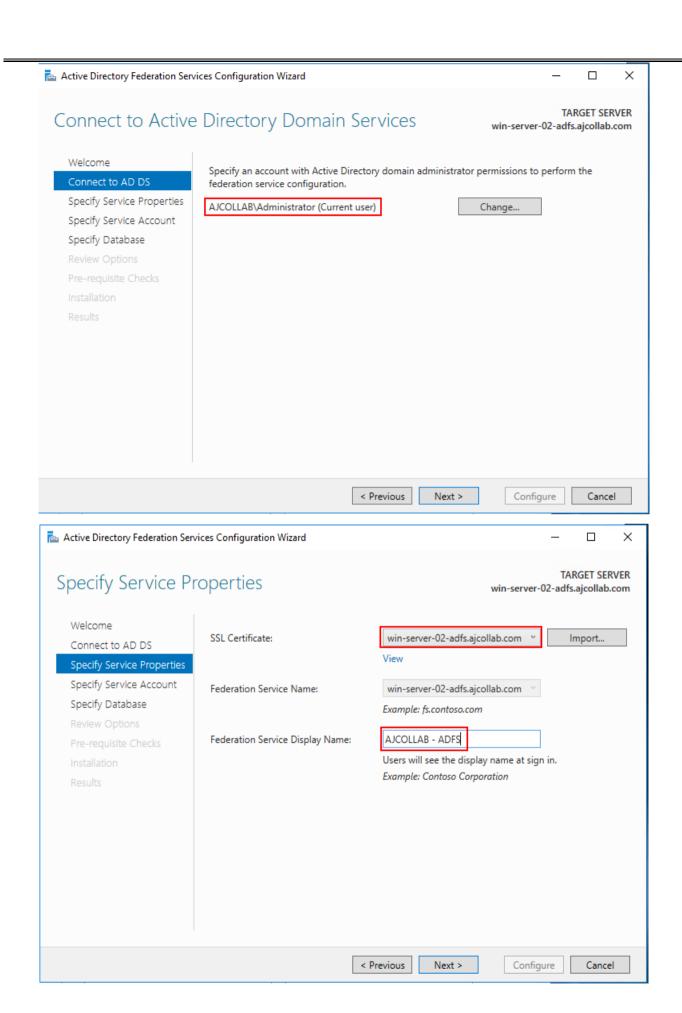


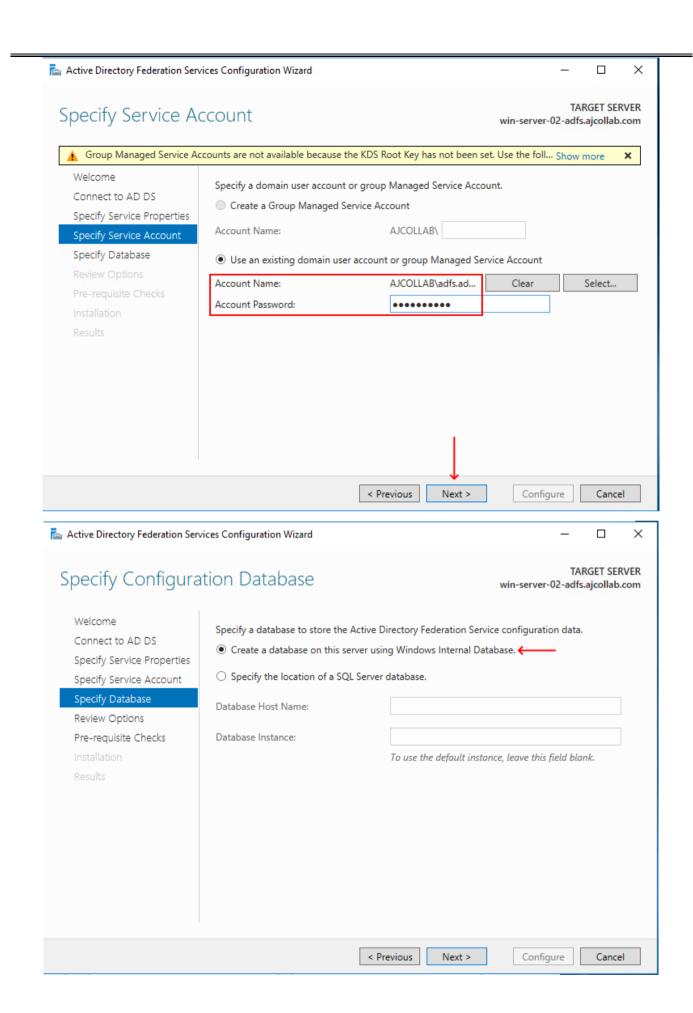
< Previous

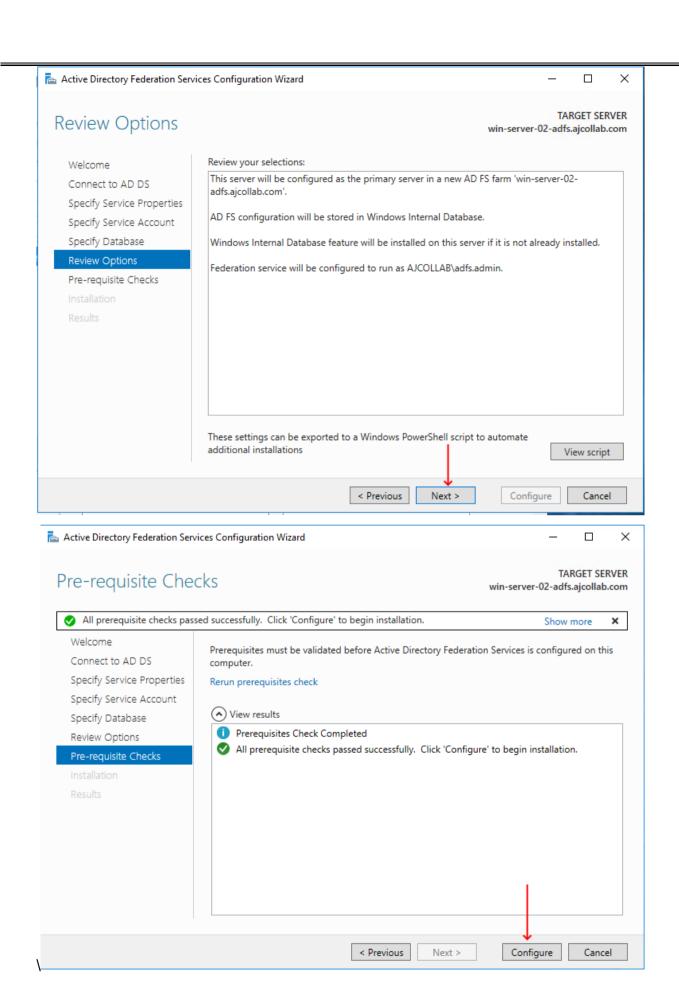
Next >

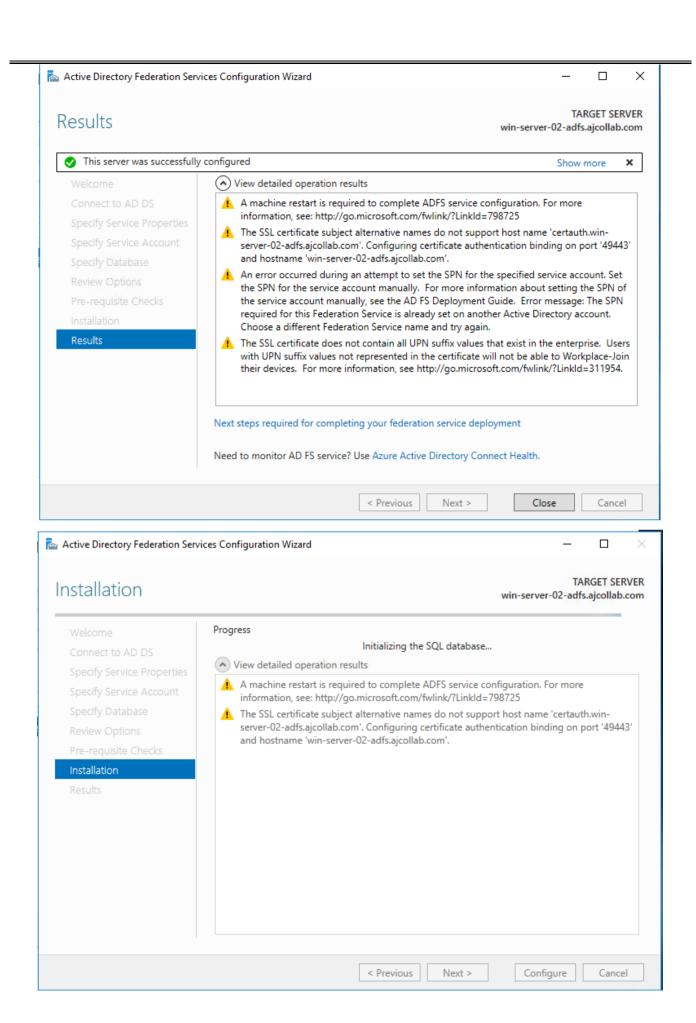
Configure

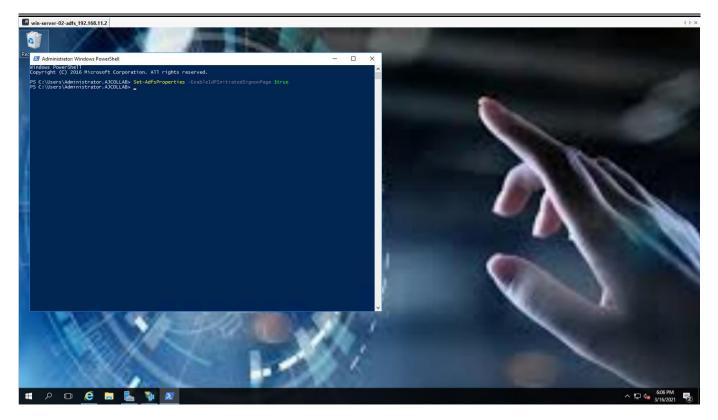
Cancel



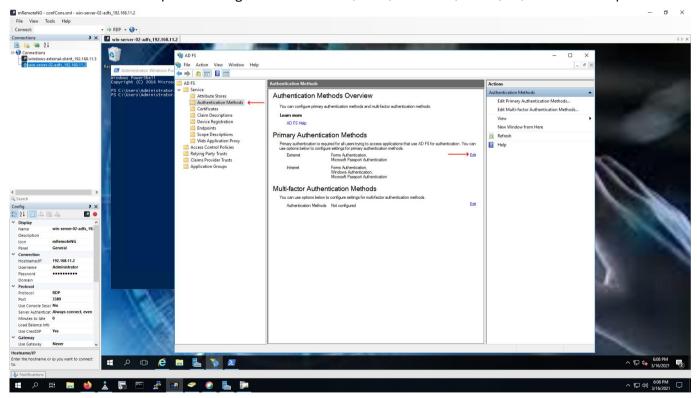


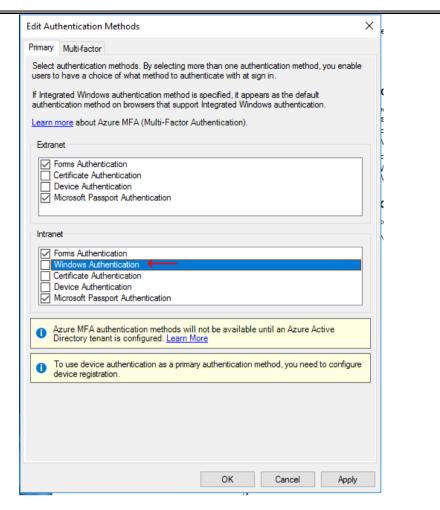




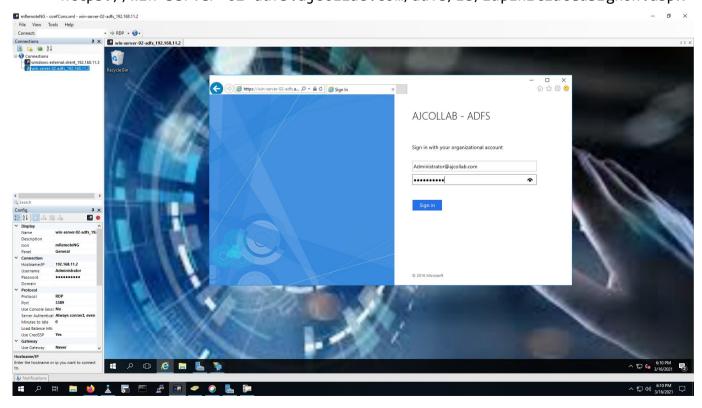


- Run below commands,
- Set-AdfsProperties -EnableIdPInitiatedSignonPage \$true
- Enable-AdfsEndpoint -TargetAddressPath "/adfs/services/trust/13/windowstransport"





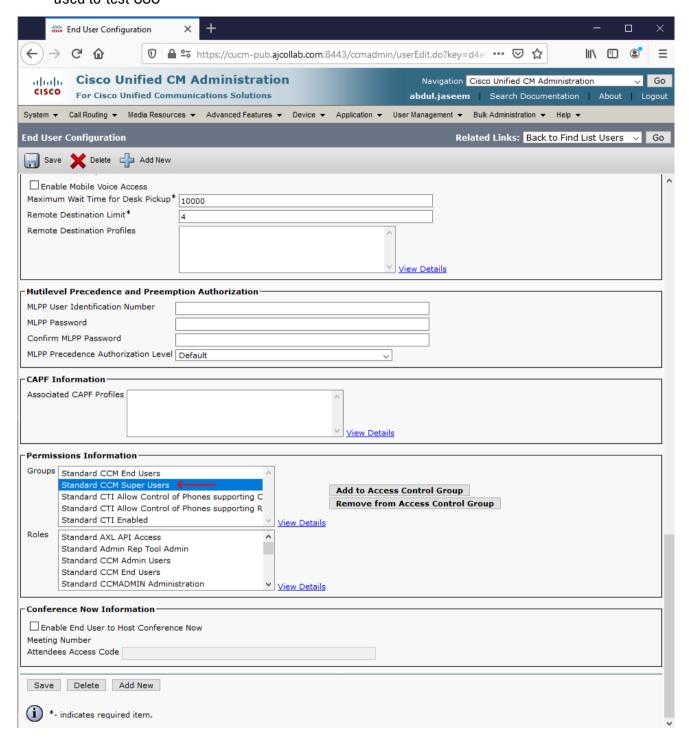
- Access the sample SSO login page
- https://win-server-02-adfs.ajcollab.com/adfs/ls/idpinitiatedsignon.aspx



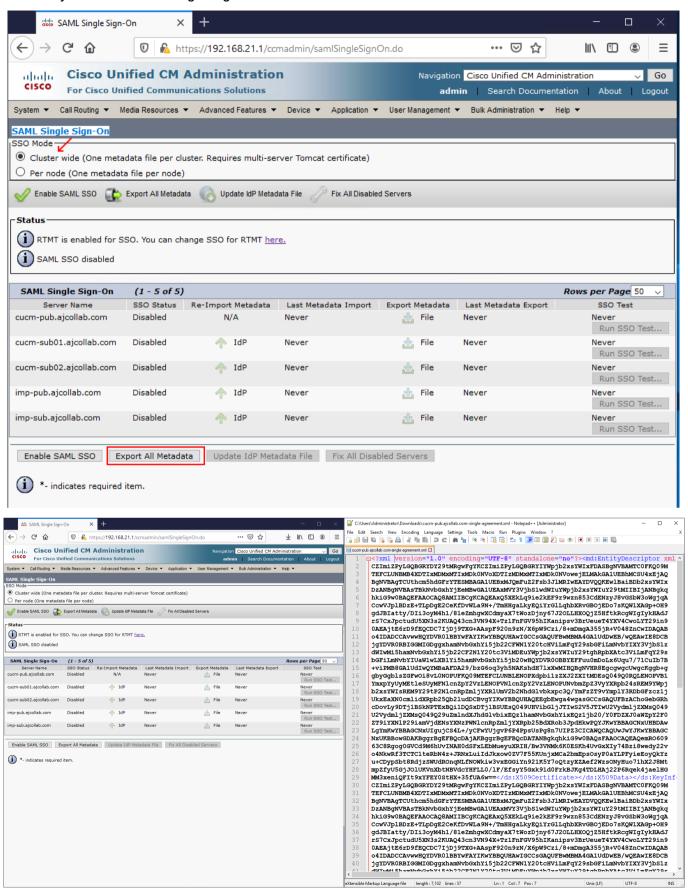


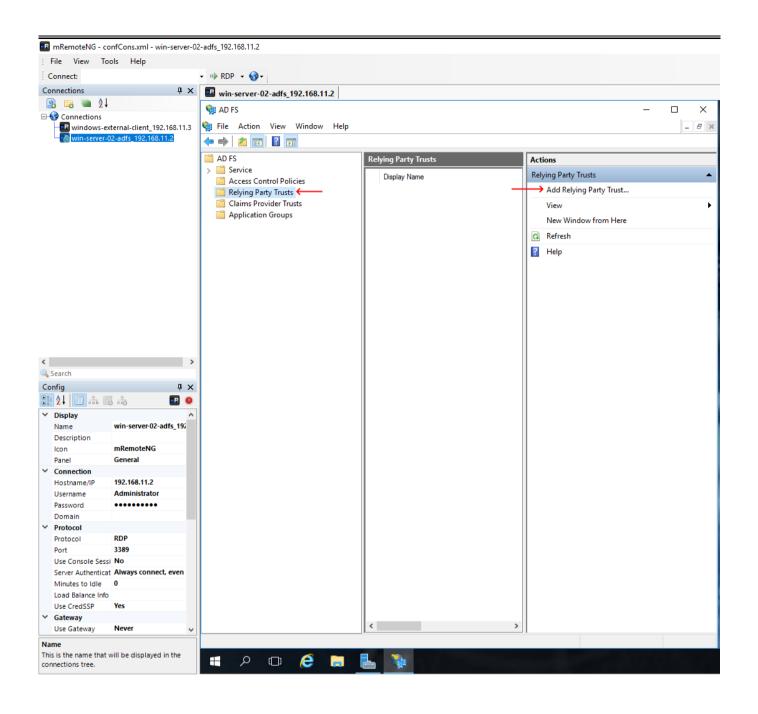
[Lab] Configure SSO in CUCM

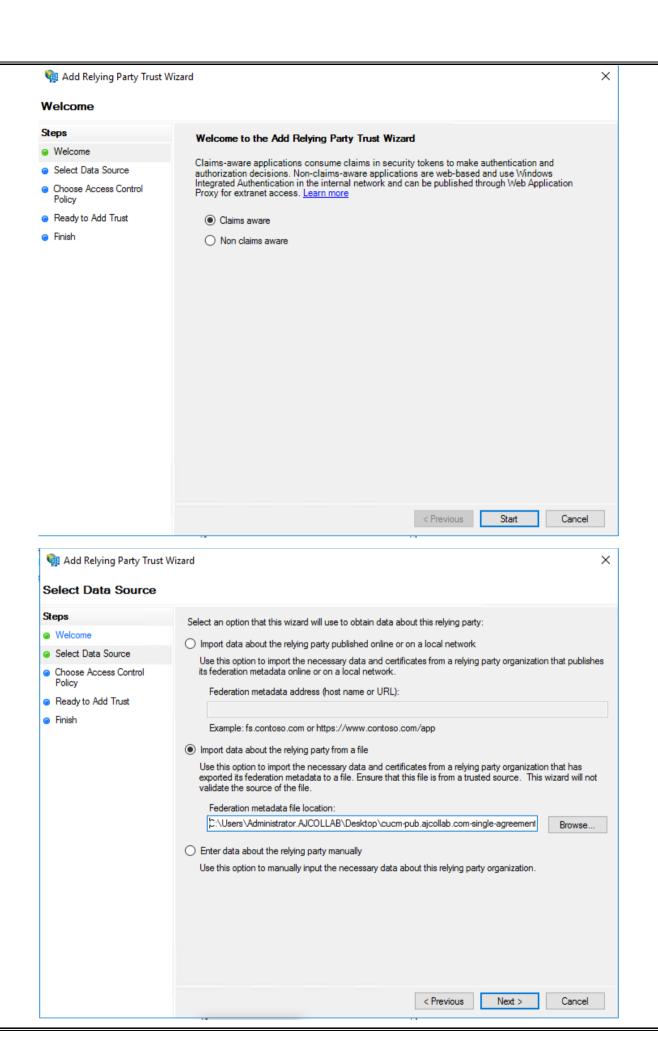
 Provide Standard CCM Super Users role to any LDAP synced user in CUCM, this account will be used to test SSO

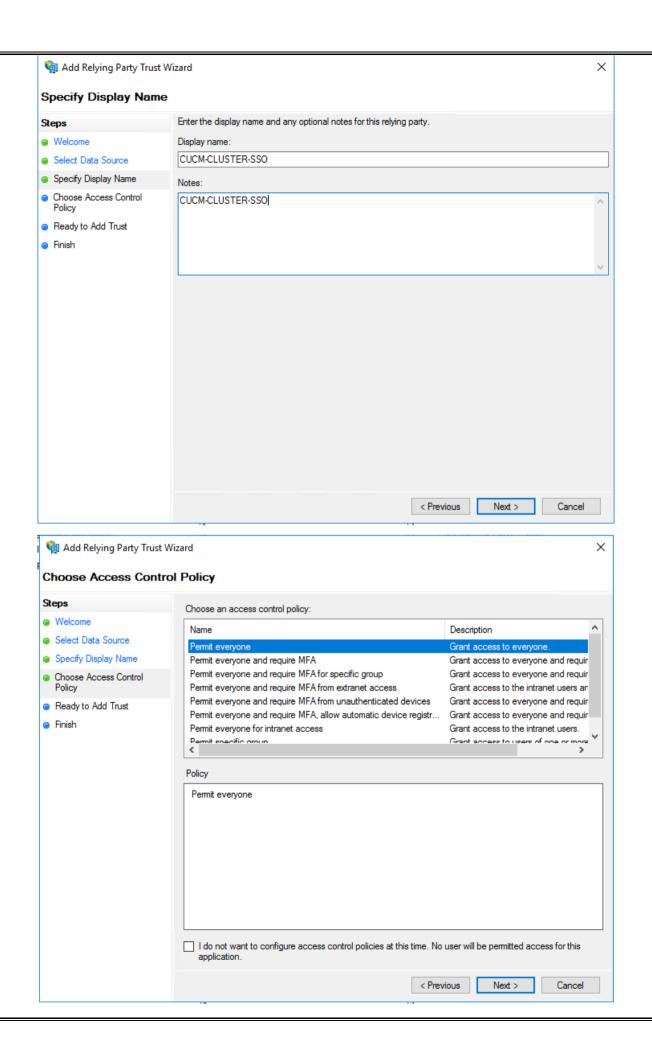


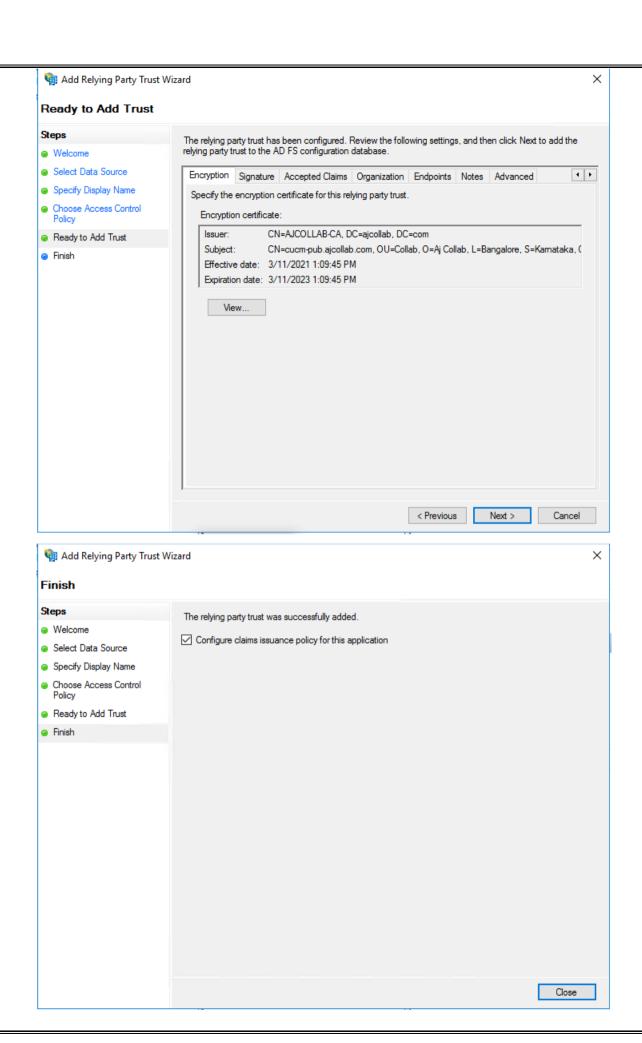
System >> SAML Single Sign-On

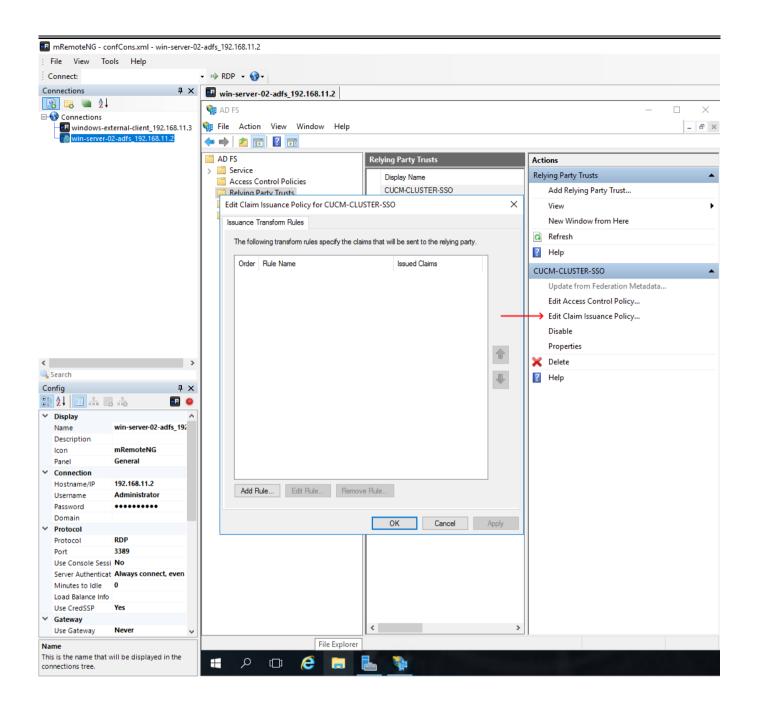


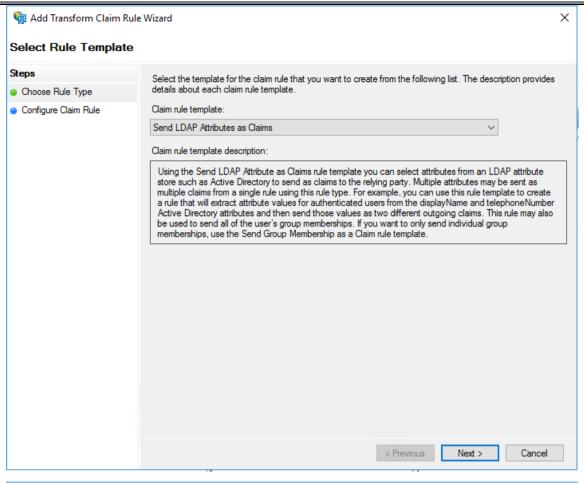


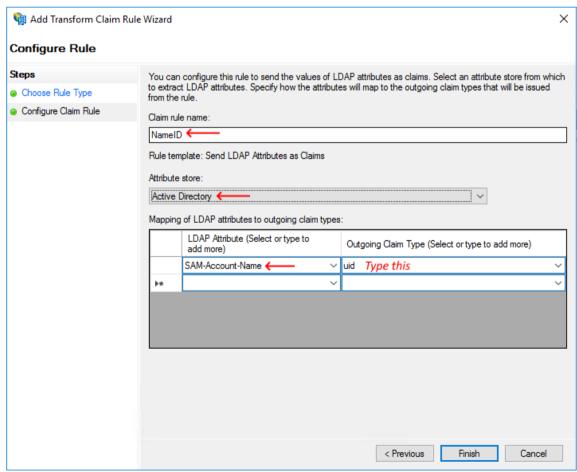


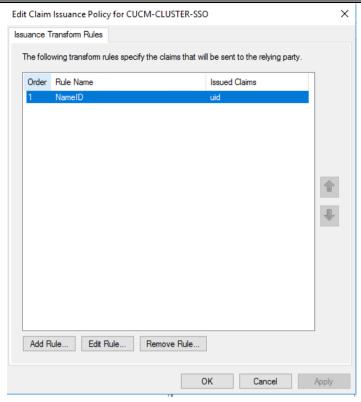


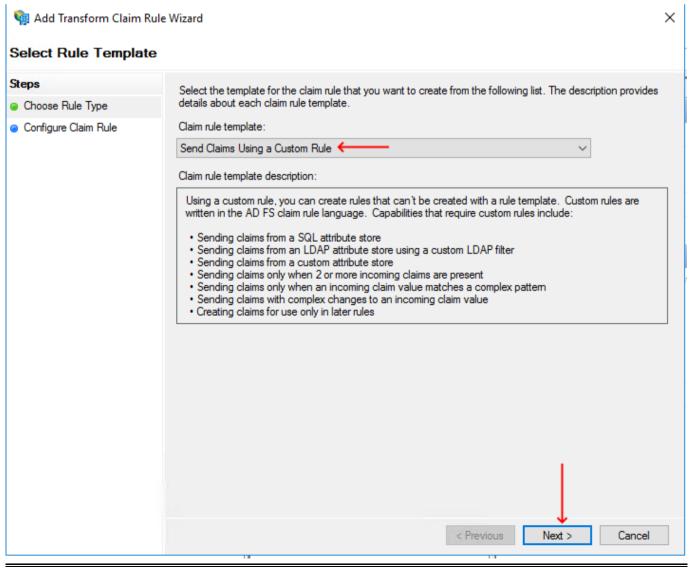


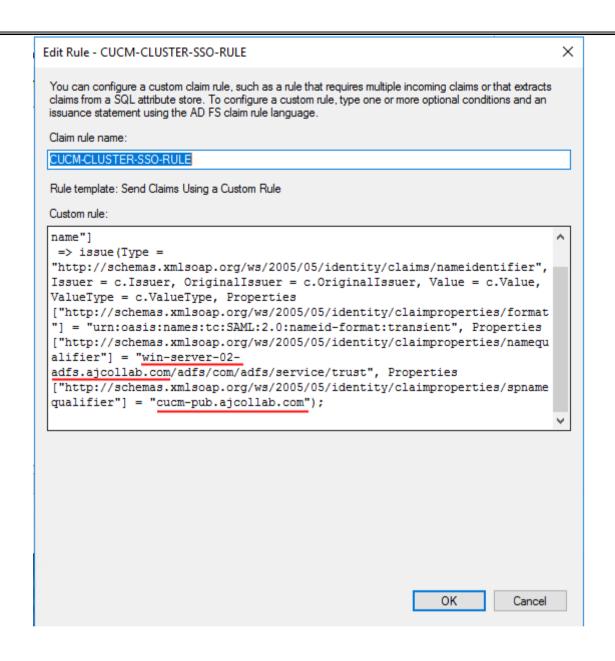




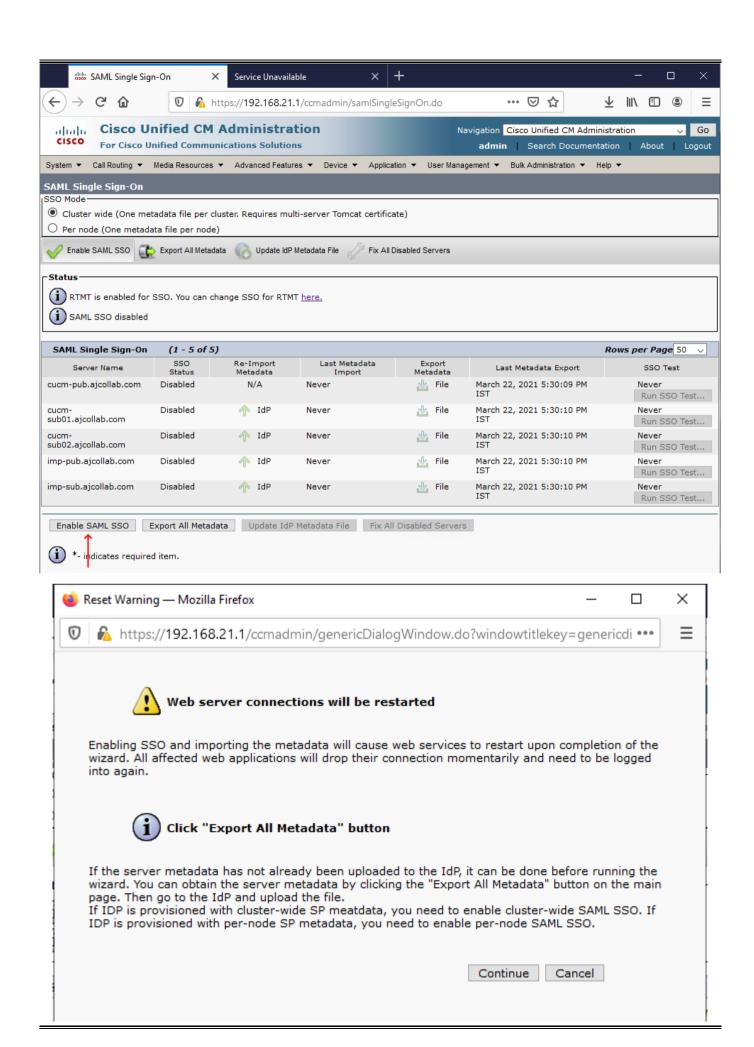


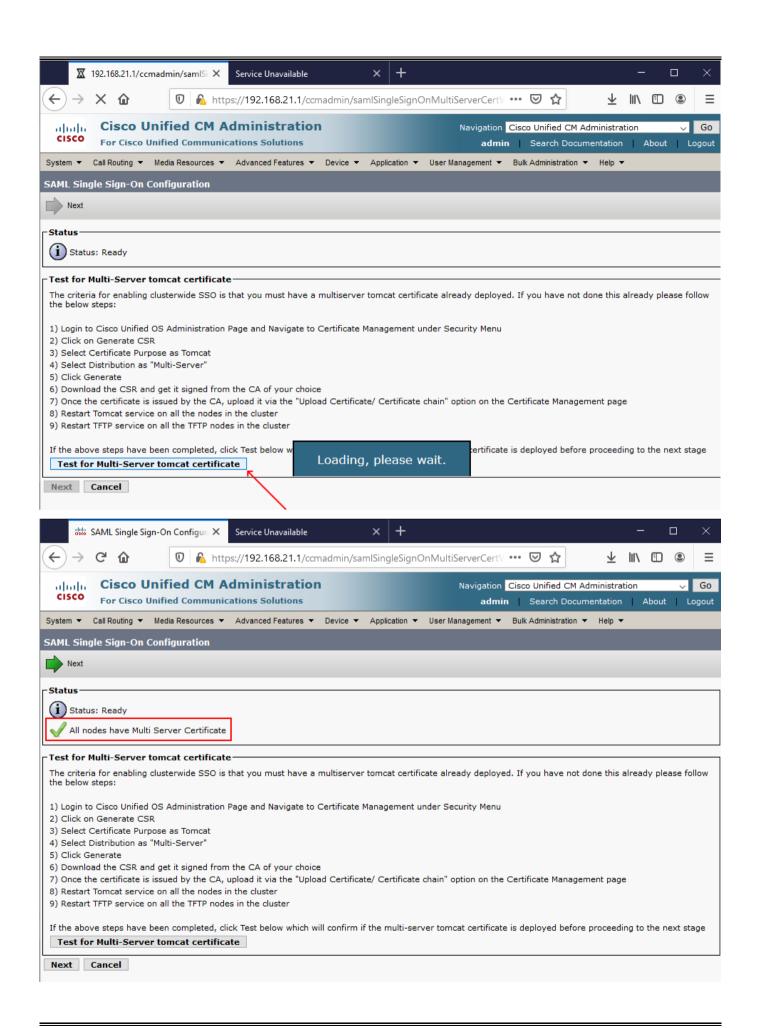


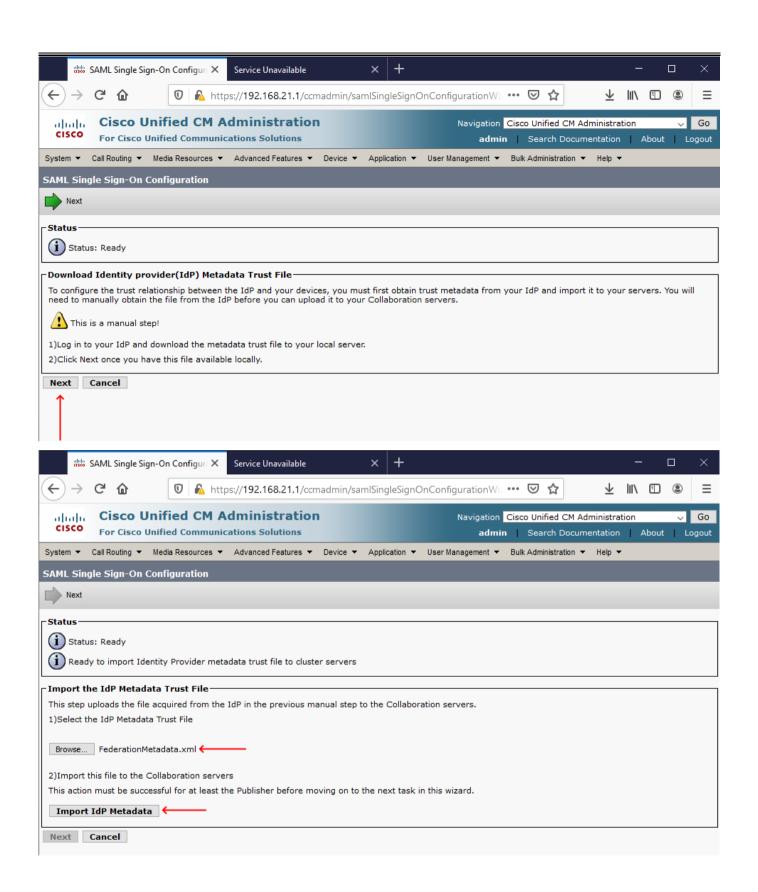


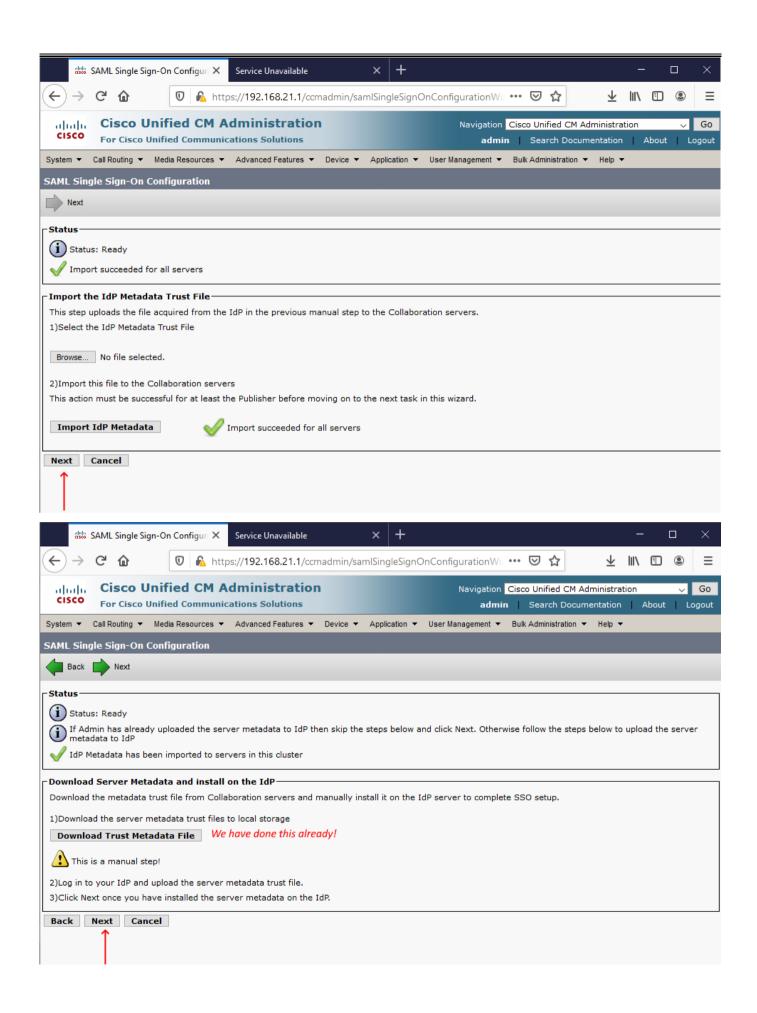


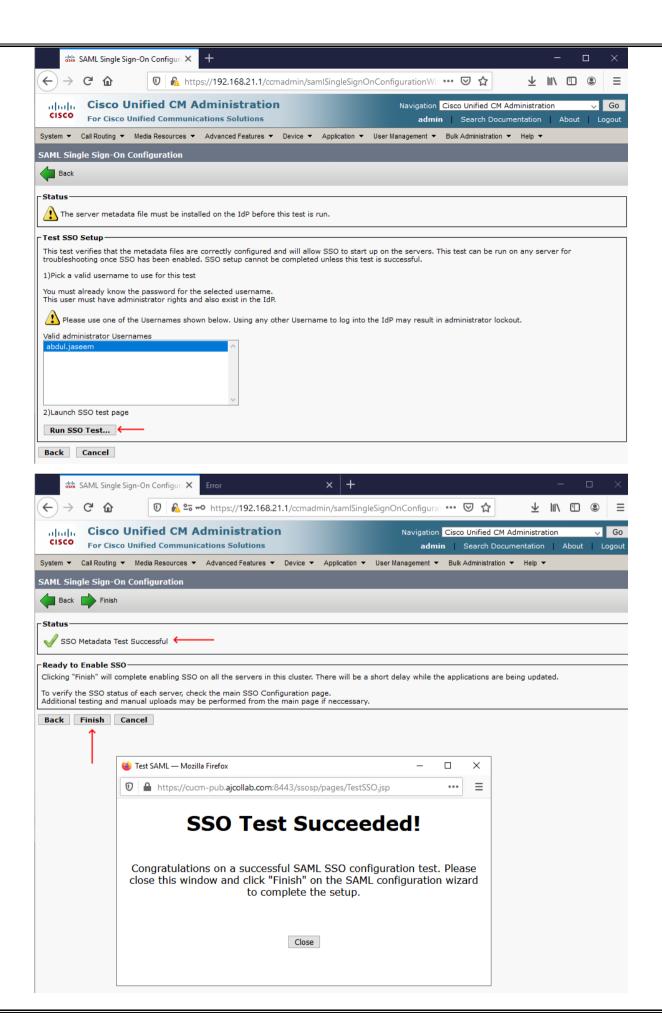
- Download ADFS Metadata by following this link from Mozilla or Google Chrome browser
- https://ADFS-SERVER-FQDN/FederationMetadata/2007-06/FederationMetadata.xml

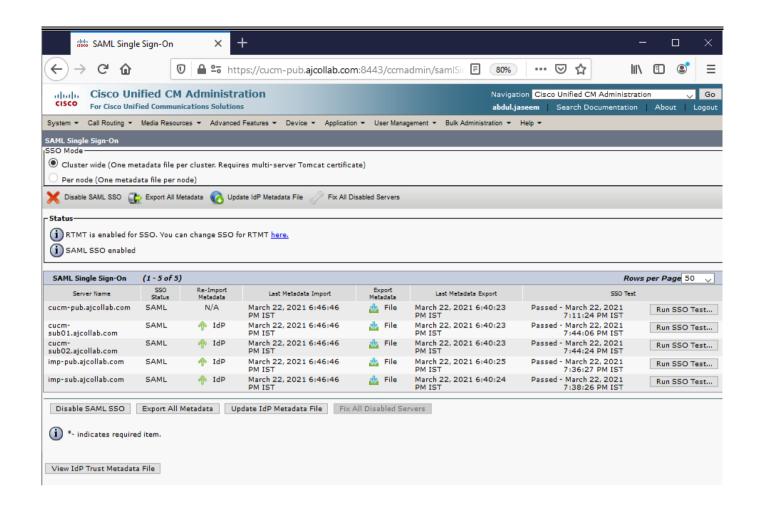








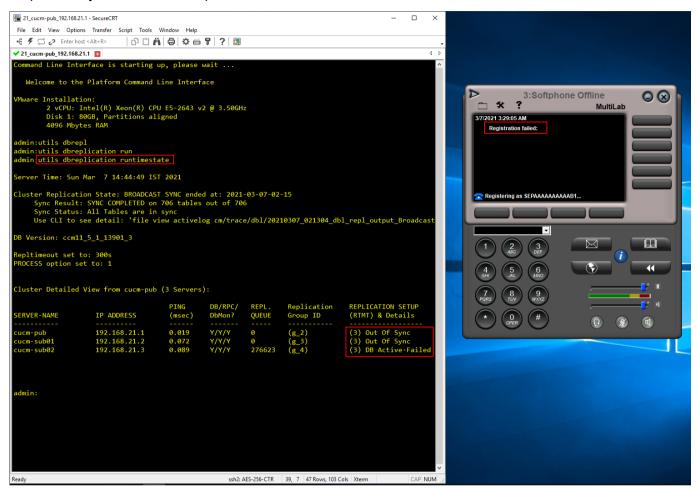




CUCM DB Replication

 When DB Replication failed between nodes, so many weird issues will happen like Phone Registration Issues, Call Routing Issues, and other service-related issues

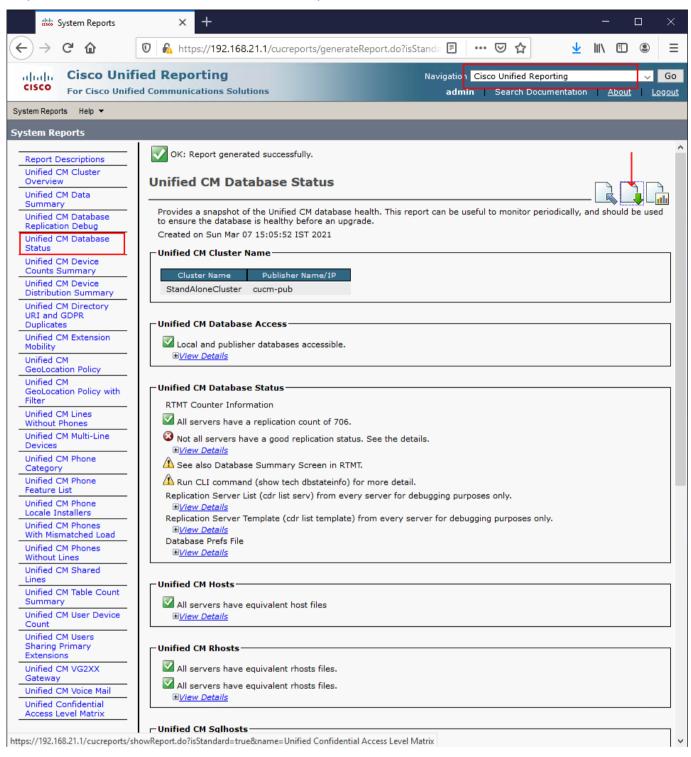
Step 1: Verify the Database Replication



Below table shows the meaning of Replication states

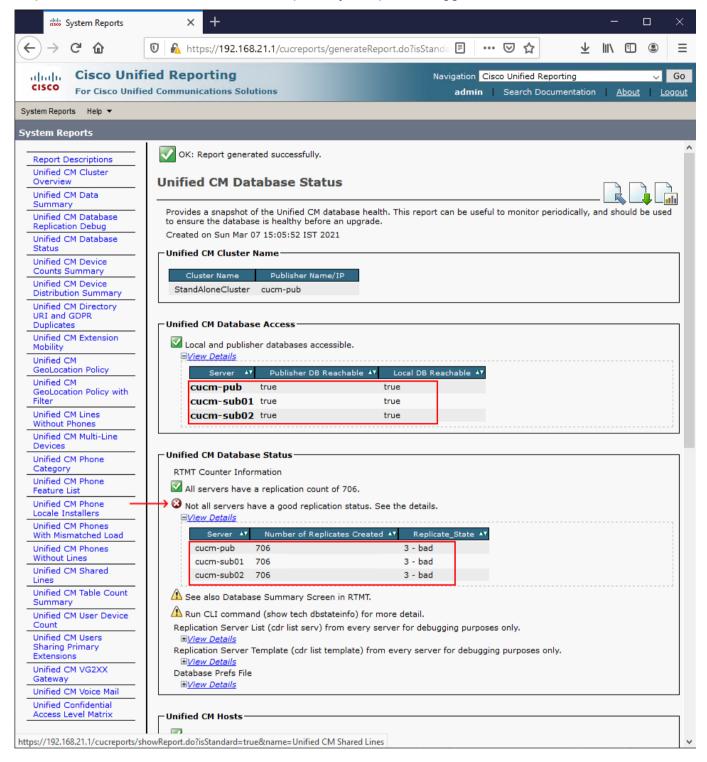
Value	Meaning	Description
0	Initialization State	Replication is in the process of setting up. A setup failure might have occurred if replication is in this state for more than an hour.
1	The Number of replicates is incorrect	Set up is still in progress. This state is rarely seen
2	Replication is good	Logical connections are established, and the tables are matched with the other servers on the cluster.
3	Mismatched tables	Logical connections are established but there is an unsurety whether the tables match
4	Setup Failed/Dropped	Server no longer has an active logical connection in order to receive any database table across the network. No replication occurs in this state

Step 2: Generate Unified CM Database Status Report



Download the report for future reference

Step 3: Review the Unified CM Database Report any component flagged as an error



- Ensure the Local and the Publisher databases are accessible
- Verify Unified CM Database Status
- In case of an error, check for the network connectivity between the nodes. Verify if the A Cisco DB service is running from the CLI of the node using the utils service list command

Step 4: Check Network Connectivity to other nodes from Publisher

```
21_cucm-pub_192,168,21.1 - SecureCRT
                                                                                                                                                                                  П
 File Edit View Options Transfer Script Tools Window Help
 - | 라 🖺 M | 🖶 🌣 🖮 🎖 | ? | 🝱
✓ 21_cucm-pub_192.168.21.1 🗵 ✓ 22_cucm-sub01_192.168.21.2 🗸 23_cucm-sub02_192.168.21.3
                                                                                                                                                                                           4
 admin:utils network connectivity cucm-sub01 <
This command can take up to 3 minutes to complete.
Continue (y/n)?y
 Running test, please wait ...
 Network connectivity test with cucm-sub01 completed successfully. \leftarrow
admin:utils network connectivity cucm-sub02
This command can take up to 3 minutes to complete.
Continue (y/n)?y
 Running test, please wait ...
Network connectivity test with cucm-sub02 completed successfully. \leftarrow
admin:
 admin:
 admin:
21_cucm-pub_192.168.21.1 - SecureCRT
File Edit View Options Transfer Script Tools Window Help
                                     D A A B A A A B P 3 8
 ■ # □ C Enter host < Alt+R>
✓ 21_cucm-pub_192.168.21.1 🗵 ✓ 22_cucm-sub01_192.168.21.2 ✓ 23_cucm-sub02_192.168.21.3
amp-sub
admin:show network cluster
192.168.21.6 imp-pub.ajcollab.com imp-pub Subscriber cups DBPub not authenticated - INITIATOR since Sun Mar 7 13:15:47 2021
192.168.21.2 cucm-sub01.ajcollab.com cucm-sub01 Subscriber callmanager DBSub authenticated using TCP since Sun Mar 7 15:44:27 2021
192.168.21.3 cucm-sub02.ajcollab.com cucm-sub02 Subscriber callmanager DBSub authenticated using TCP since Sun Mar 7 15:44:54 2021
192.168.21.7 imp-sub.ajcollab.com imp-sub Subscriber cups DBSub not authenticated - INITIATOR since Sun Mar 7 13:15:47 2021
192.168.21.1 cucm-pub.ajcollab.com cucm-pub Publisher callmanager DBPub authenticated
 Server Table (processnode) Entries
 mp-pub
                                                               ■View Details
                                                               ■View Details
                                                                 Server ≜₹
                                                                                    dbl rpchello 'nodename'
                                                                  cucm-pub DBL XML-RPC Server Version 1.1 [19779]
[OK]
                                                                 DBL RPCHELLO
cucm-sub01
cucm-sub01 DBL XMI-RPC Server Version 1.1 [18887]
[OK]
                                                                 DBL RPCHELLO
cucm-sub02
CUCM-Sub02
DBL XML-RPC Server Version 1.1 [19096]
[OK]
                                                             Connectivity Success for cucm-pub
                                                              Connectivity Success for cucm-sub01
                                                             Connectivity Success for cucm-sub02
```

Step 5: Verify DB Services are Started in the Problematic Nodes

```
21_cucm-pub_192.168.21.1 - SecureCRT
                                                                                                                                                                                                                                                                                                                                                    File Edit View Options Transfer Script Tools Window Help
                                                                                               | D D A | B | $ € T | ? | 8
  € 🗲 🚅 💸 Enter host <Alt+R>
min:utils service list sequesting service status, please wait...

stem SSH [STARTED]

uster Manager [STARTED]

mme Service Cache [STARTED]

tropy Monitoring Daemon [STARTED]

sco SCSI Watchdog [STARTED]

rvice Manager [STARTED]

TPS Configuration Download [STARTED]

rvice Manager is running

tting list of all services

Return code = 0

Cisco DB [STARTED]

Cisco DB Replicator[STARTED]

isco AMC Service[STARTED]
                    CO UN REPLICATION STARTED]
AMC Service[STARTED]
AXL Web Service[STARTED]
Audit Event Service[STARTED]
CAR DB[STARTED]
CAR DS STARTED]
CAR Scheduler[STARTED]
                   CAR Scheduler[STARTED]
CDP[STARTED]
CDP Agent[STARTED]
CDP Agent[STARTED]
CDR Agent[STARTED]
CDR Repository Manager[STARTED]
CallManager Admin[STARTED]
CallManager Serviceability[STARTED]
CallManager Serviceability[STARTED]
CallManager Serviceability[STARTED]
Certificate Change Notification[STARTED]
Certificate Expiry Monitor[STARTED]
Change Credential Application[STARTED]
DRF Decal[STARTED]
DRF Master[STARTED]
Database Layer Monitor[STARTED]
Dialed Number Analyzer[STARTED]
Dialed Number Analyzer Server[STARTED]
DirSync[STARTED]
E911[STARTED]
E911[STARTED]
                     E911[STARTED]
ELM Client Service[STARTED]
                                                                                                                                                                                        ssh2: AES-256-CTR 47, 7 47 Rows, 103 Cols Xterm
                                                                                                                                                                                                                                                                                                                                                      CAP NUM ...
22_cucm-sub01_192.168.21.2 - SecureCRT
 min:show myself ← 
chine Name : cucm-sub01←
count name : admin
ivilege level : 4
mmand count : disabled
gging setting : disabled
          in:utils service list←
              mesting service status, please wait...

em SSH [STARTED]

ter Manager [STARTED]

opy Monitoring Daemon [STARTED]

op SCSI Watchdog [STARTED]

ice Manager [STARTED]

is Configuration Download [STARTED]

ice Manager is running

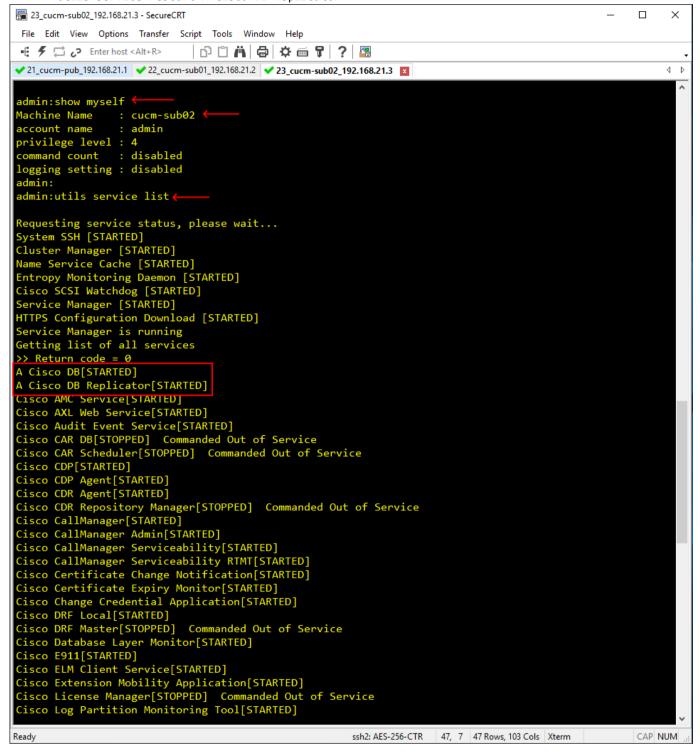
ing list of all services

teturn code = 0

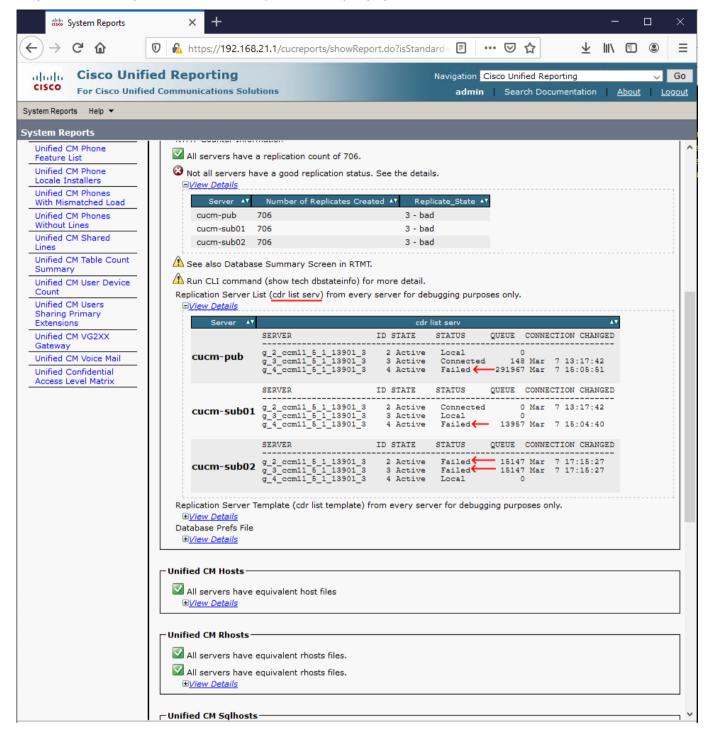
sco DBISTARTED]
                    co DB Replicator(STARTED)
AMC Service(STARTED)
AXL Web Service(STARTED)
Audit Event Service(STARTED)
Audit Event Service(STARTED)
CAR DB(STOPPED) Commanded Out of Service
CAR Scheduler(STOPPED) Commanded Out of Service
CDP(STARTED)
CDP Agent(STARTED)
CDR Agent(STARTED)
CDR Agent(STARTED)
CDR Agent(STARTED)
CDR Sanosiatory Manager(STOPPED) Commanded Out of
                         DR Agent[STARTED]

DR Repository Manager[STOPPED] Commanded Out
TIManager[STARTED]
allManager[STARTED]
allManager Admin[STARTED]
allManager Serviceability[STARTED]
allManager Serviceability[STARTED]
allManager Serviceability RTMT[STARTED]
ertificate Change Notification[STARTED]
ertificate Expiry Monitor[STARTED]
hange Credential Application[STARTED]
BRF Master[STOPPED] Commanded Out of Service
atabase Layer Monitor[STARTED]
ialed Number Analyzer[STARTED]
ialed Number Analyzer Server[STARTED]
911[STARTED]
                                                                                         ger[STOPPED] Commanded Out of Service
                                                                                                                                                                                        ssh2: AES-256-CTR 47, 7 47 Rows, 103 Cols Xterm
                                                                                                                                                                                                                                                                                                                                                 CAP NUM
```

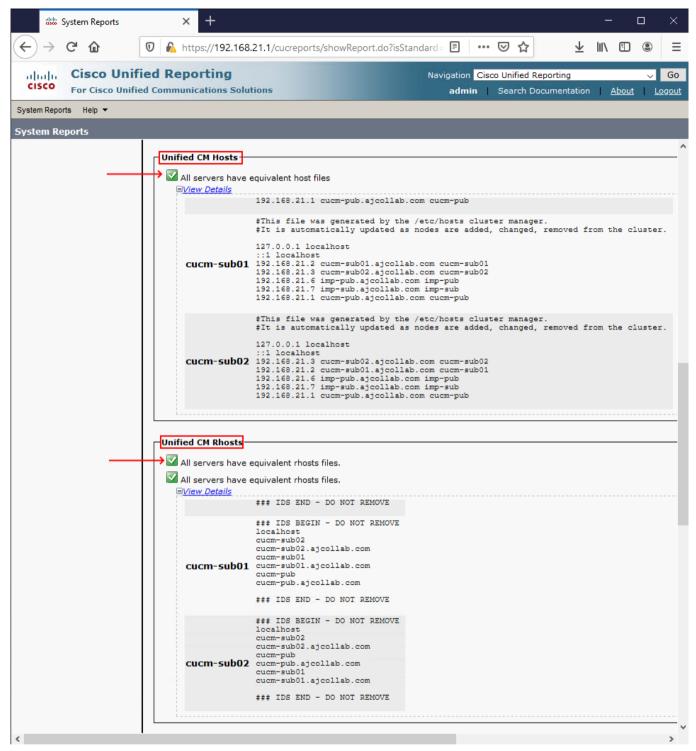
- If A Cisco DB and A Cisco DB Replicator are in stopped state, try to restart from CLI
- utils service restart A Cisco DB
- utils service restart A Cisco DB Replicator

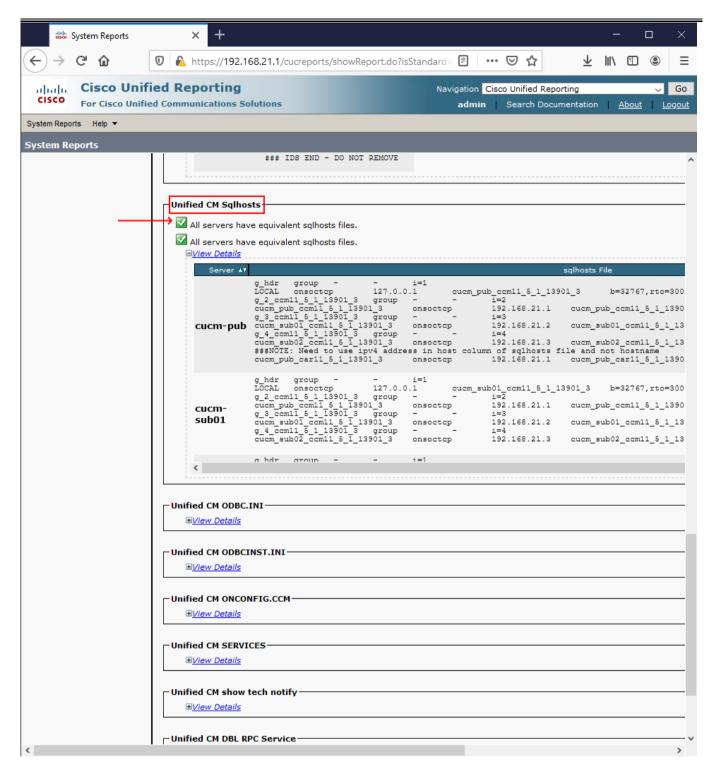


Step 6: Ensure Replication Server List (cdr list serv) is populated for all the nodes



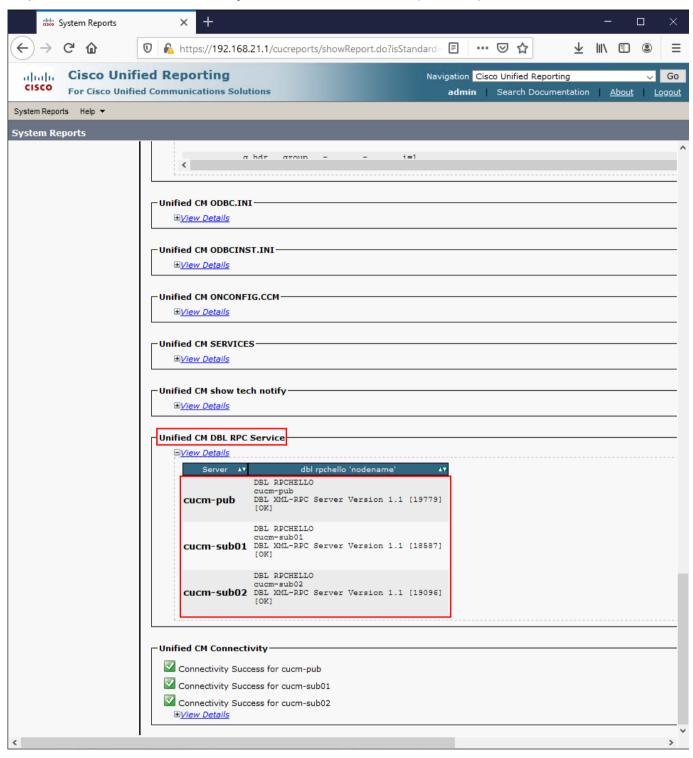
Step 7: Verify CM Hosts, CM Rhosts and CM Sqlhosts are in Sync





- If you find any mismatch, there is a possibility of an incorrect activity when an IP address changes or updates to the Hostname on the server
- Restart the following services from the CLI of the publisher server and check if the mismatch is cleared Generate a new report every time you make a change on the GUI/CLI to check if the changes are included
 - o utils service restart Cluster Manager
 - utils service restart A Cisco DB

Step 8: Ensure that the Database Layer Remote Procedural Call (DBL RPC) hello is successful



Step 9: Diagnose Test and NTP Status

```
21_cucm-pub_192.168.21.1 - SecureCRT
                                                                                                                   File Edit View Options Transfer Script Tools Window Help
D 🖺 🗥 🖨 🌣 🚎 🎖 🛭 🖀
✓ 21_cucm-pub_192.168.21.1 🛽 ✓ 22_cucm-sub01_192.168.21.2 🗸 23_cucm-sub02_192.168.21.3
                                                                                                                        4 b
admin:utils diagnose test
Log file: platform/log/diag1.log
Starting diagnostic test(s)
test - disk_space
                               Passed (available: 996 MB, used: 12685 MB)
     - disk_files
                               This module must be run directly and off hours
skip
test - service_manager

    Passed

test - tomcat
                              Passed
     - tomcat_deadlocks
test - tomcat_keystore
                            : Passed
test - tomcat_connectors
                            : Passed
test - tomcat_threads
     - tomcat_memory
                              Passed
     - tomcat sessions
skip - tomcat_heapdump
                            : This module must be run directly and off hours
: Error, intra-cluster communication is broken, unable to connect to [192.168.21.6]
test - validate_network
communication is broken, unable to connect to [192.168.21.7] Legys
test - raid
test - system_info
                            : Passed (Collected system information in diagnostic log)
    - ntp_reachability
                            : Warning
The NTP service is restarting, it can take about 5 minutes.
test - ntp_clock_drift
The local clock is not synchronised
None of the designated NTP servers are reachable/functioning or legitimate.
test - ntp_stratum
                            : Warning
The local clock is not synchronised
None of the designated NTP servers are reachable/functioning or legitimate.
skip - sdl_fragmentation : This module must be run directly and off hours
skip - sdi_fragmentation : This module must be run directly and off hours
Diagnostics Completed
 The final output will be in Log file: platform/log/diag1.log
 Please use 'file view activelog platform/log/diag1.log' command to see the output
 dmin:
```

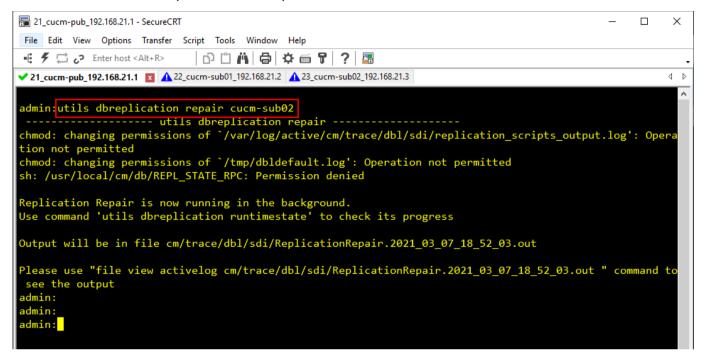
```
31_cucm-pub_192.168.21.1 - SecureCRT
File Edit View Options Transfer Script Tools Window Help
• € 🗲 🖾 💸 Enter host <Alt+R> 💮 🖺 🛗 🖨 🔯 📾 🖫 🔞 🔞

✓ 21_cucm-pub_192.168.21.1 
✓ 22_cucm-sub01_192.168.21.2 
✓ 23_cucm-sub02_192.168.21.3

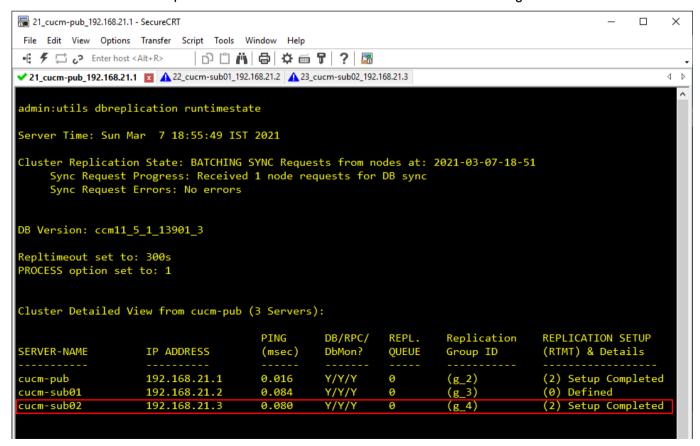
                                                                                                                           4
admin:utils ntp status 🤇
ntpd (pid 6475) is running...
     remote
                       refid
                                   st t when poll reach
                                                            delay
                                                                     offset jitter
 192.168.31.1
                  .INIT.
                                   16 u
                                          - 1024
                                                             0.000
                                                                      0.000
                                                                               9.999
unsynchronised ﴿
  polling server every 8 s
Current time in UTC is : Sun Mar 7 10:39:55 UTC 2021
Current time in Asia/Kolkata is : Sun Mar 7 16:09:55 IST 2021
admin:
```

Step 10: DB Repair Node

- If the utils dbreplication runtimestate command shows that there are error/mismatched tables, run the command
 - utils dbreplication repair NODE



Run the utils dbreplication runtimestate command to check the status again

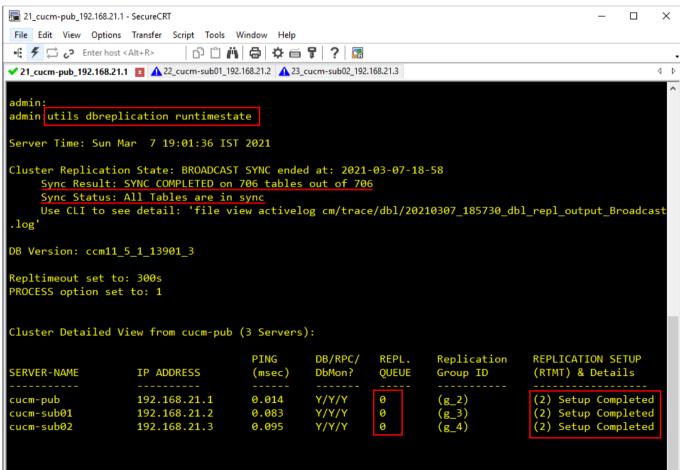


Step 11: Reset Database using Rebuild Command

- This will re-initiate the DB Replication on CUCM-SUB01 from scratch
- This is a time-consuming process
- DB of SUB01 will re-sync with CUCM PUB

```
21_cucm-pub_192,168,21,1 - SecureCRT
                                                                              ×
File Edit View Options Transfer Script Tools Window Help
4
                                                                                   Þ
admin utils dbreplication rebuild cucm-sub01
WARNING !!!
This command can only be run from the publisher.
This command will run a combination of the following CLI commands on specified nodes:
 utils dbreplication stop
 utils dbreplication dropadmindb or dropadmindbforce
 and utils dbreplication reset
This command may also affect other nodes including the publisher.
This command can take a considerable amount of time.
Do you want to continue? (y/n):y \leftarrow
Start rebuilding replication...
The process can be terminated by "delete process 19793" on a separate cli session
The detail status can be checked by "file view activelog cm/trace/dbl/sdi/rebuild_2021_03_07_18_48_24.o
Starting rebuild command...
Number of nodes to be rebuilt: 1
Stopping replication setup...
Removing server(s) from replication network...
Signaling subscriber(s) for replication setup...
Rebuild command completed successfully...
->cucm-sub01
Rebuild Completed: 1 \longleftarrow Failed: 0
Duration: 1.66 minute
admin:
```

```
21_cucm-pub_192,168,21.1 - SecureCRT
File Edit View Options Transfer Script Tools Window Help
■ # □ • Enter host < Alt+R>
                           - | P 🖺 A | 🖨 | 🌣 📾 🎖 | ? | 🔞
✓ 21 cucm-pub 192,168,21.1 🗵 🛕 22_cucm-sub01_192.168.21.2 🛕 23_cucm-sub02_192.168.21.3
                                                                                                             4 b
admin:utils dbreplication runtimestate
Server Time: Sun Mar 7 18:55:49 IST 2021
Cluster Replication State: BATCHING SYNC Requests from nodes at: 2021-03-07-18-51
     Sync Request Progress: Received 1 node requests for DB sync
     Sync Request Errors: No errors
DB Version: ccm11_5_1_13901_3
Repltimeout set to: 300s
 ROCESS option set to: 1
Cluster Detailed View from cucm-pub (3 Servers):
                                                   DB/RPC/
                                        PING
                                                                       Replication
                                                              REPL .
                                                                                        REPLICATION SETUP
SERVER-NAME
                     IP ADDRESS
                                        (msec)
                                                   DbMon?
                                                              QUEUE
                                                                       Group ID
                                                                                        (RTMT) & Details
cucm-pub
                     192.168.21.1
                                        0.016
                                                   Y/Y/Y
                                                              0
                                                                        (g_2)
                                                                                        (2) Setup Completed
cucm-sub01
                                                                       (g_3)
                                                                                       (0) Defined
                     192.168.21.2
                                        0.084
                                                   Y/Y/Y
                     192.168.21.3
                                                                                        (2) Setup Completed
cucm-sub02
                                                                        (g_4)
```



Step 12: DBreplication Setprocess

- If CUCM Nodes are Spread Across WAN, ensure that the nodes have network connectivity well under 80 ms (Round Trip Time)
- Changing this parameter improves the replication setup performance, but consumes additional system resources
 - o utils dbreplication setprocess <1-40>

Step 13: Replication Timeout

- The replication timeout (Default: 300 Seconds) is the time that the publisher waits for all the subscribers in order to send their defined messages
- Calculate the replication timeout based on the number of nodes in the cluster
 - Server 1-5 = 1 Minute Per Server Servers
 - Server 6-10 = 2 Minutes Per Server
 - Servers >10 = 3 Minutes Per Server

Example:

Example: 12 Servers in Cluster:

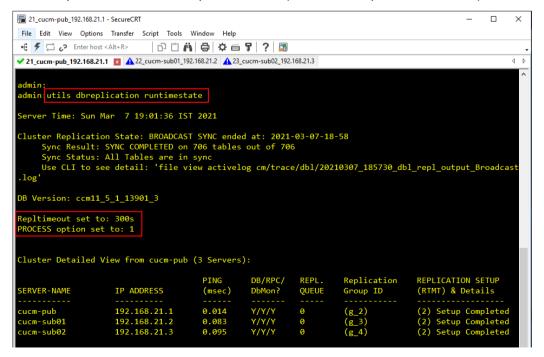
Server 1-5 x 1 min = 5x1 = 5 min

Server $6-10 \times 2 \text{ min} = 5 \times 2 = 10 \text{ min}$

Server $11-12 \times 3 \text{ min} = 2 \times 3 = 6 \text{ min}$

Repltimeout should be set to 21 Minutes

- show tech repltimeout (To check the current replication timeout value)
- utils dbreplication setrepltimeout (To set the replication timeout)



Step 14: Contact Cisco TAC

- If none of the above steps are not helping to fix the DB replication, please contact Cisco TAC
- While contacting TAC, make sure you collect the below details

You may also interested to refer Support Forum Document

Chapter 1 Module 3 - CUC Cisco On-Premise Collaboration Solution Cisco Unity Connection (CUC)



Cisco Unity Connection - CUC

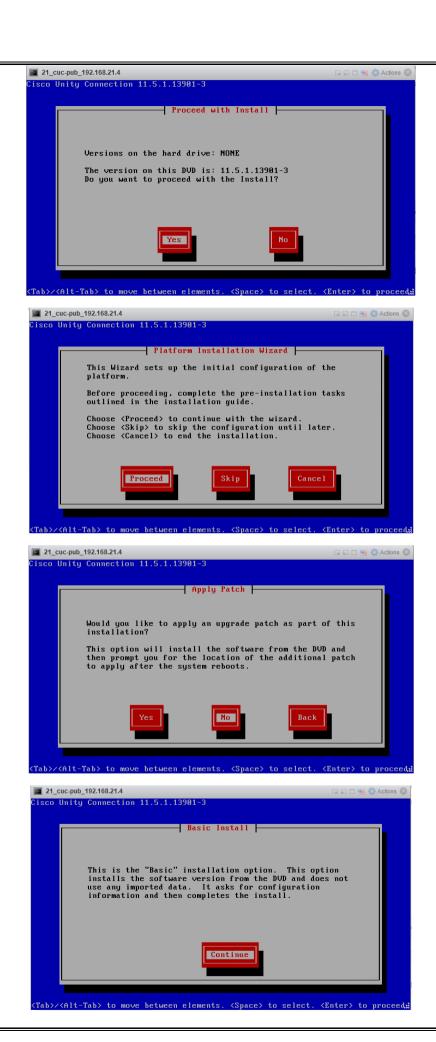


- Unified Voice Messaging and basic IVR solution system integrated to CUCM
- CUCM can't offer any voice mail solution natively, hence we integrate CUC with CUCM
- CUC can also be used for basic IVR auto attend functionality called Call Handlers. It provides greetings for external callers "Thank you for calling AJ Labs, Please Press 1 for enquiries, 2 for upcoming courses, 3 for other helps, etc."
- Cisco Unity Connection is deployed as one Publisher and one Subscriber Active-Active nodes. Only 2 nodes will be in the cluster

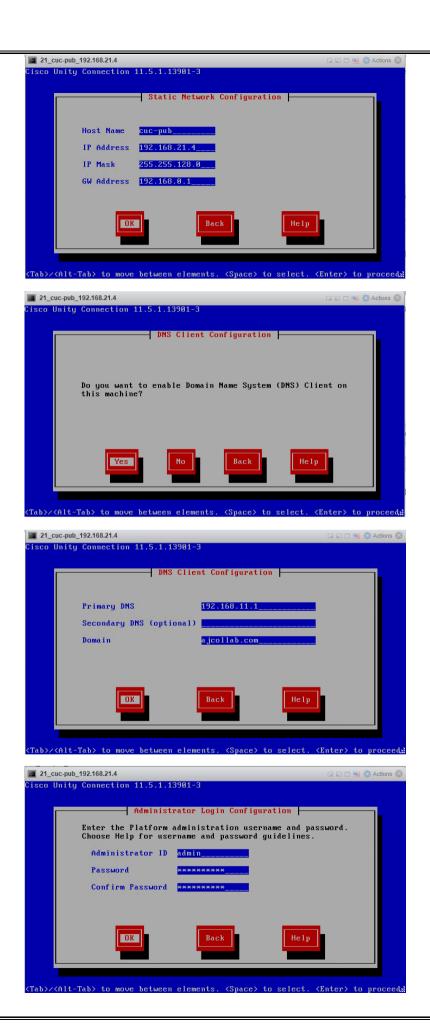
[Lab] Installation PUB and SUB

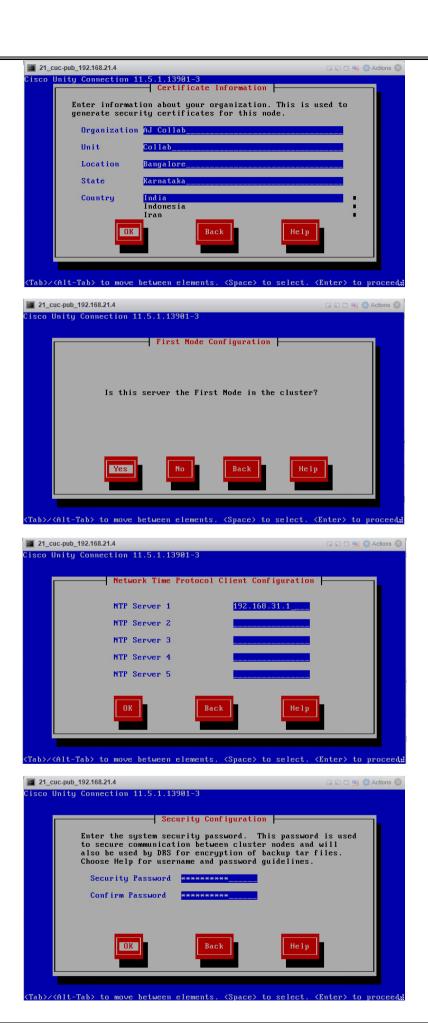
- The installation of Cisco Unity Connection is exactly like CUCM, in fact we use same ISO file to install CUCM and CUC
- The only difference is the virtual hardware requirement, the OVA of CUC is different, you can download the CUC 11.5 OVA from Cisco

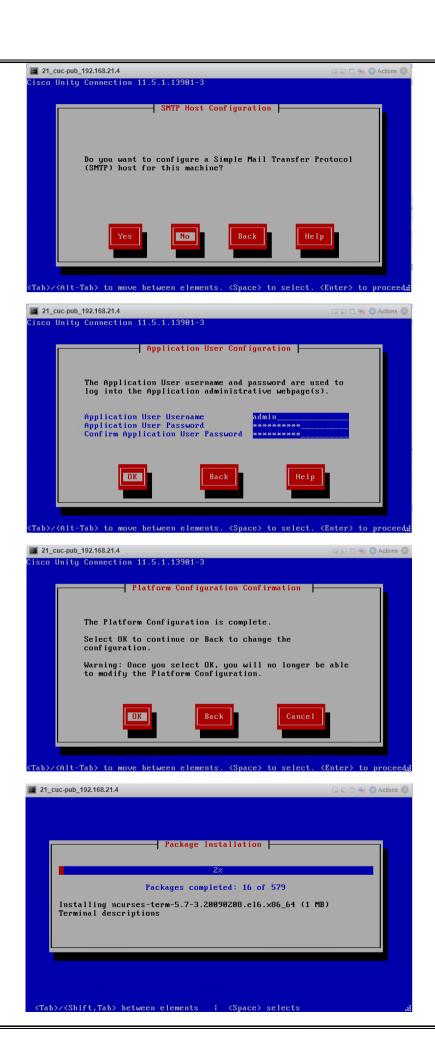




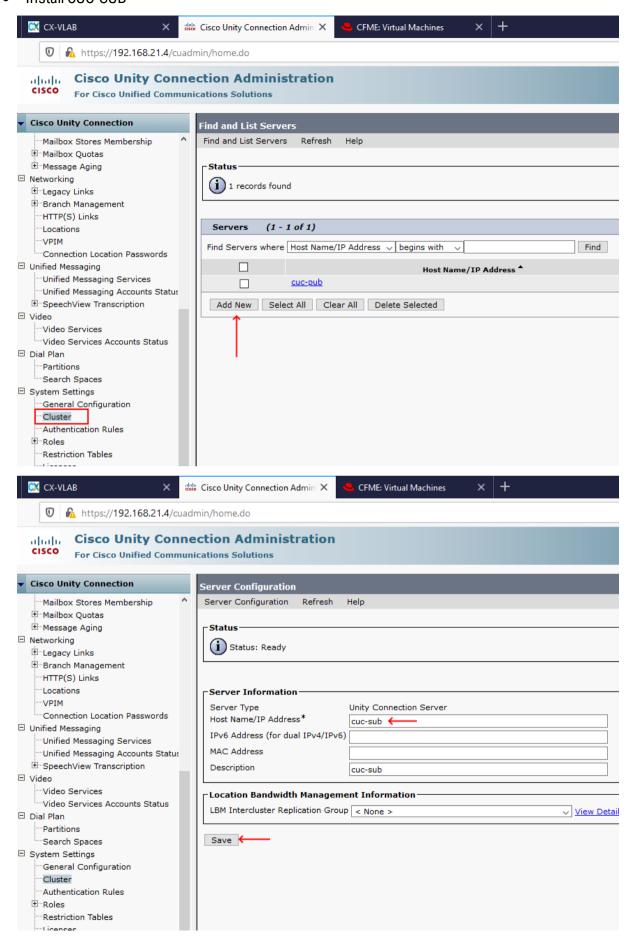


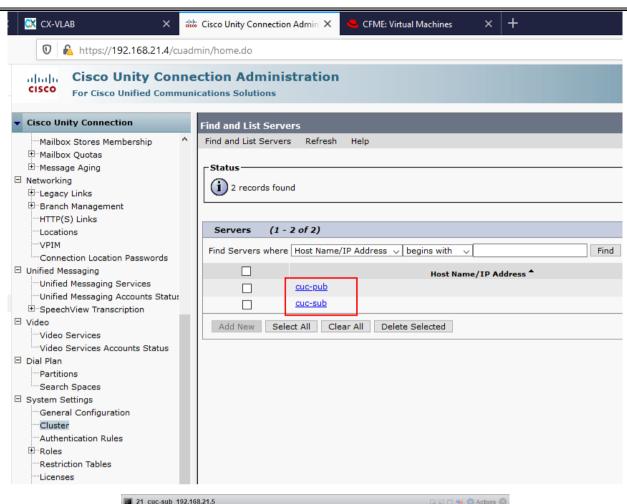


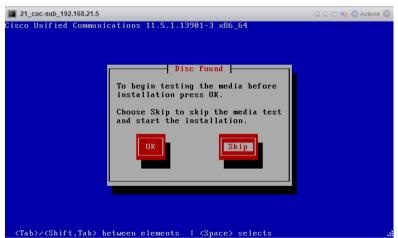


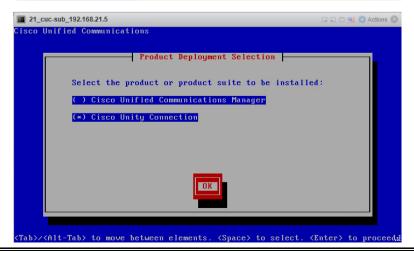


Install CUC SUB

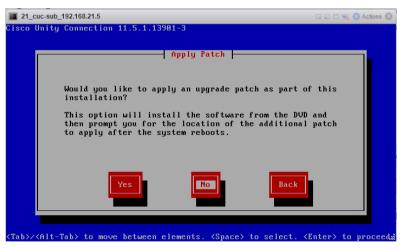


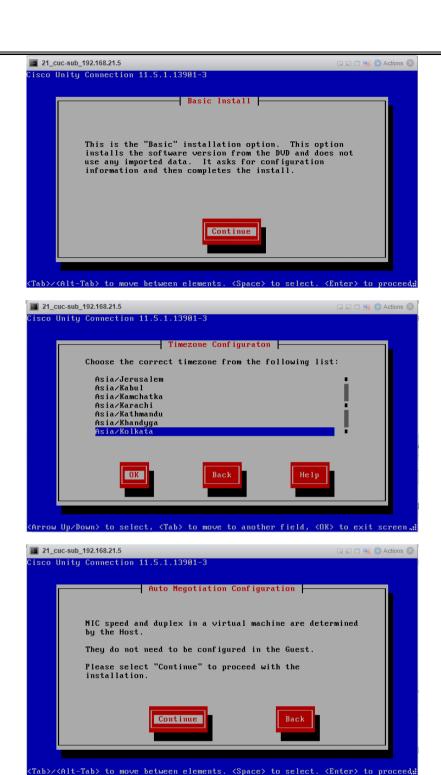


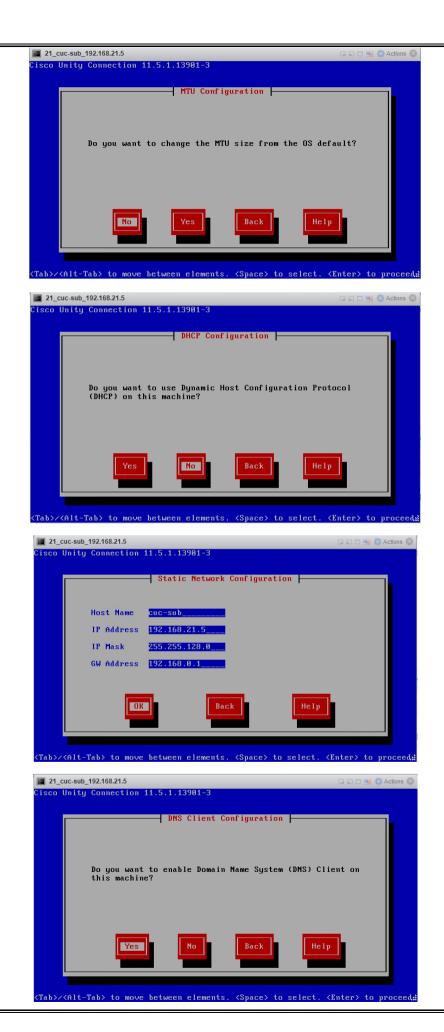


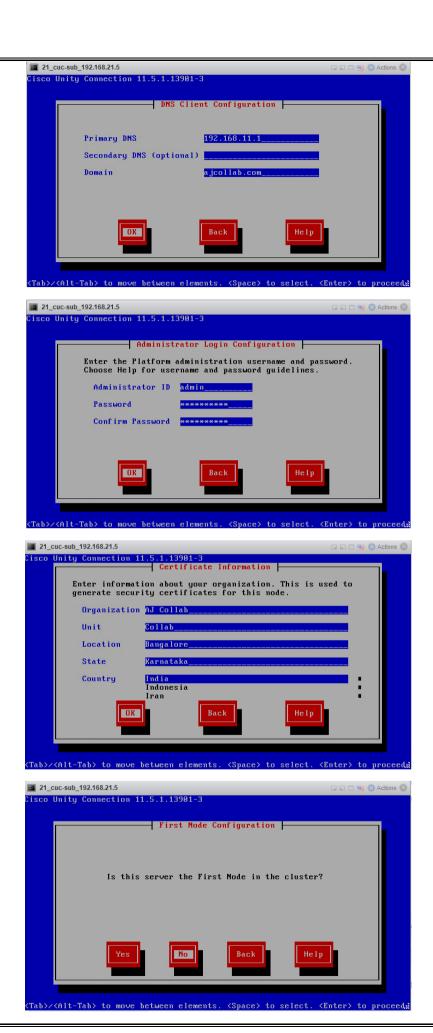


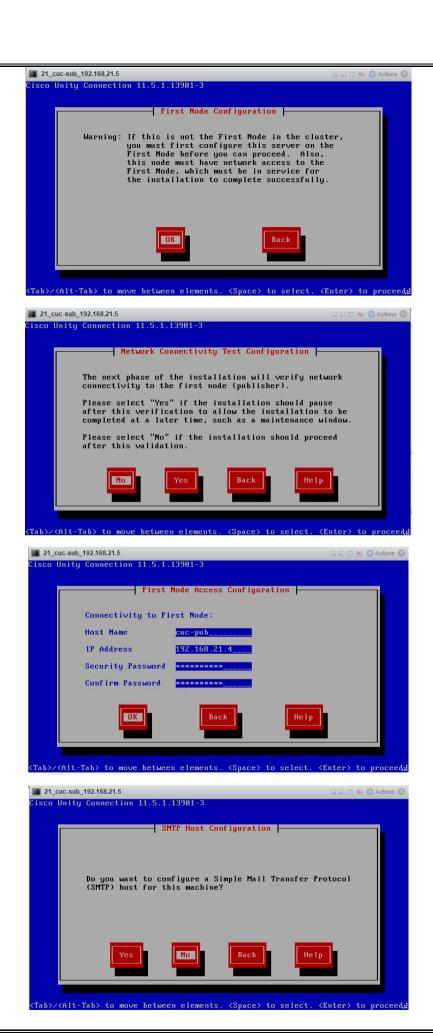


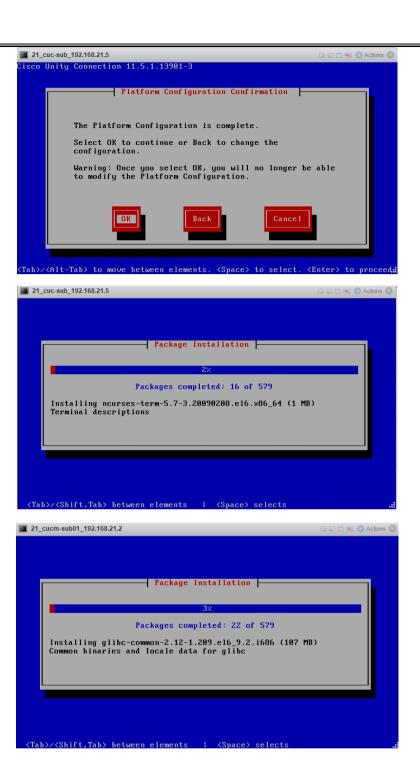






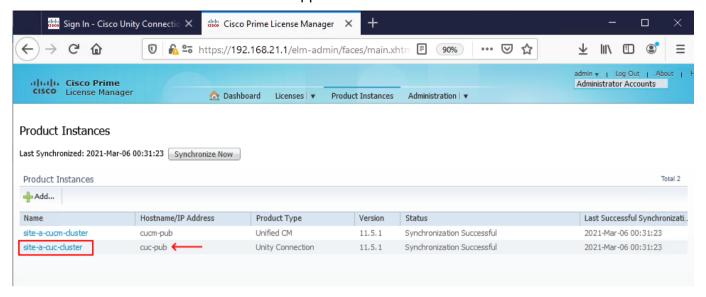






[Lab] CUC Licensing

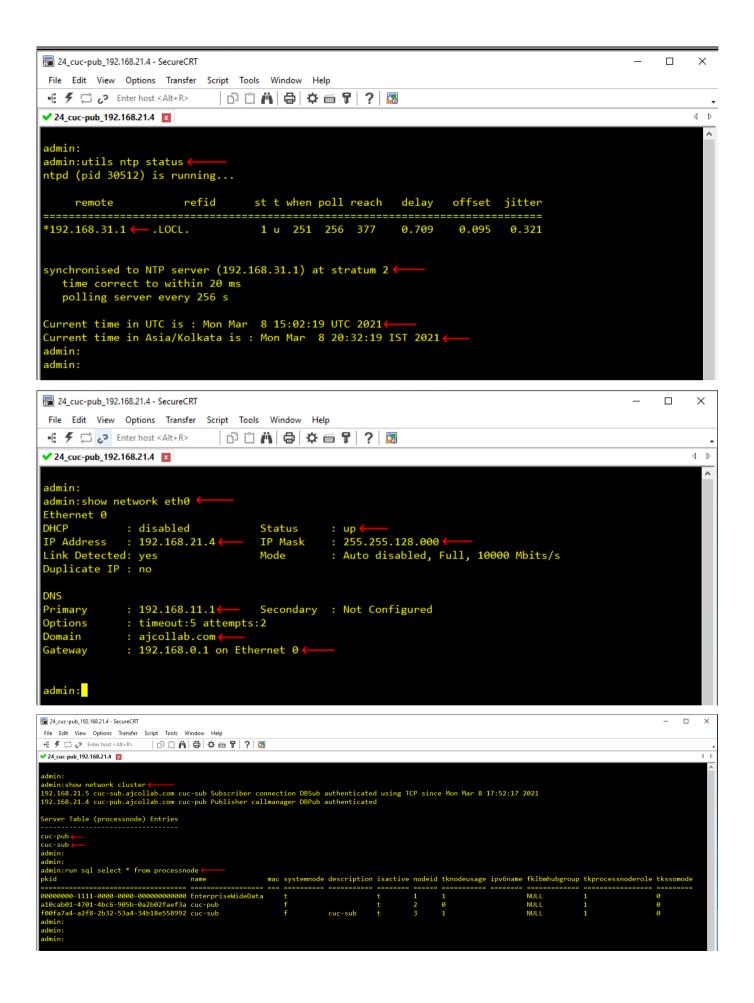
- Licensing of Cisco Unity Connection is exactly similar to CUCM
- You just need to add the product instance to Prime License Manager and Sync
- You must install user licenses that support voicemail



[Lab] Basic Health check in CUC

```
24_cuc-pub_192,168,21,4 - SecureCRT
File Edit View Options Transfer Script Tools Window Help
                             ■ F 🗀 🗘 Enter host <Alt+R>
✓ 24_cuc-pub_192.168.21.4 🗵
                                                                                                               4 b
 ommand Line Interface is starting up, please wait ...
   Welcome to the Platform Command Line Interface
VMware Installation:
        1 vCPU: Intel(R) Xeon(R) CPU E5-2643 v2 @ 3.50GHz
        Disk 1: 160GB, Partitions aligned
        4096 Mbytes RAM
admin:show myself
Machine Name :
account name
                 : admin 	
privilege level : 4 €
command count : disabled
logging setting : disabled admin:
admin:
admin:show status 

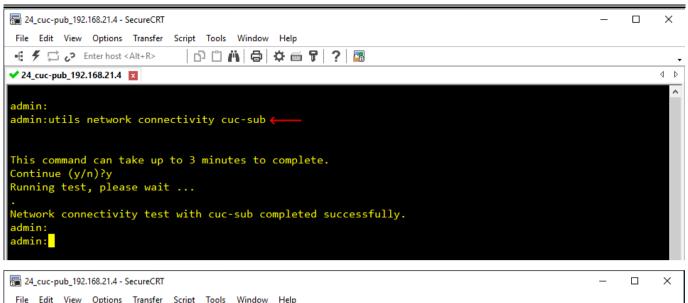
Host Name
                    : cuc-pub
                    : Mon Mar 8, 2021 20:31:15 
                    : India Standard Time (Asia/Kolkata) ←
Time Zone
Locale
                    : en US.UTF-8
                   : 11.5.1.13901-3
Product Ver
Unified OS Version : 6.0.0.0-2
Uptime: ←
 20:31:16 up 2:40, 1 user, load average: 0.37, 0.22, 0.14
            65.00% System: 00.00% ← IRQ:
CPU Idle:
                                07.00%
                                                    28.00%
                                           User:
                                                    00.00%
  IOWAIT:
Memory Total:
                       3925628K
        Free:
                       136848K
        Used:
                       3788780K
      Shared:
     Buffers:
                        50324K
                                            Free
                                                             Used
                         Total
                                                         9547732K (34%)
45028K (1%)
Disk/active
                                       19377956K
Disk/inactive
                    29223916K
102005088K
                                       27687740K
Disk/logging
                                       73843136K
                                                         22973704K (24%)
admin:
                                                                                                         CAP NUM ...
                                                         ssh2: AES-256-CTR 46, 7 47 Rows, 103 Cols Xterm
```

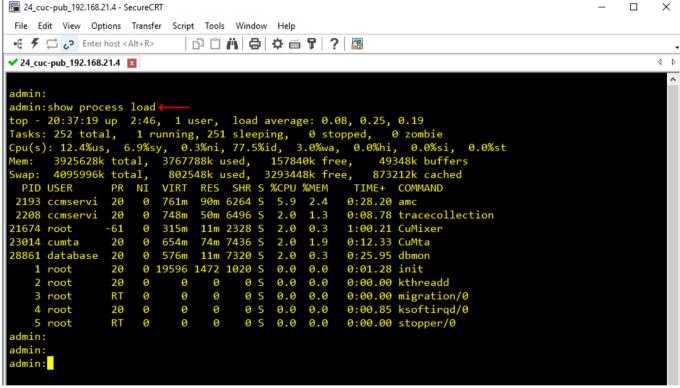


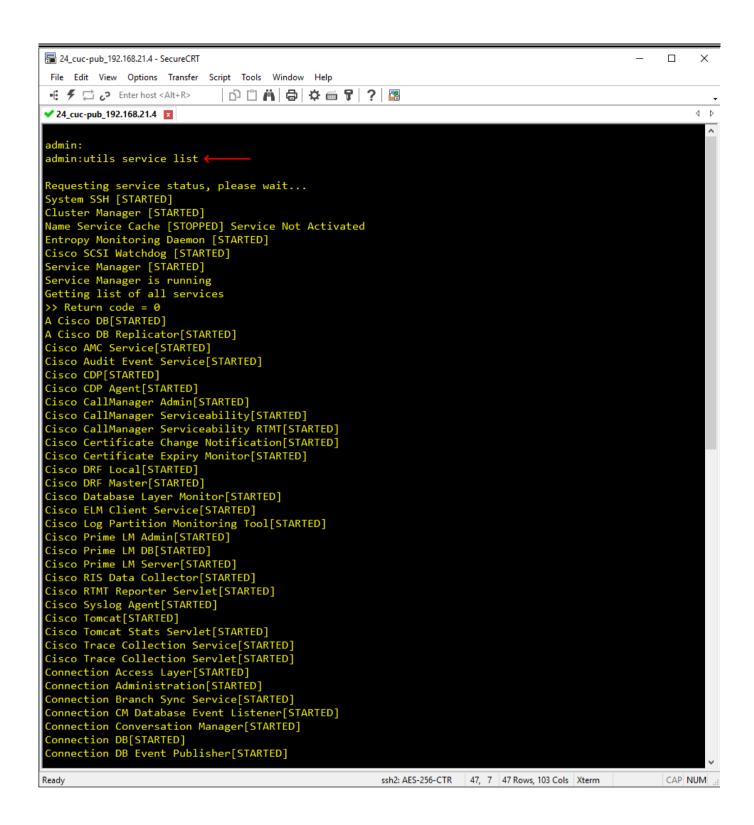
```
24_cuc-pub_192.168.21.4 - SecureCRT
                                                                                                                                                                                               File Edit View Options Transfer Script Tools Window Help
 ■ F 🗀 🕫 Enter host <Alt+R>
                                                     | D 🖺 A | 🖨 🕸 📾 🎖 | ? | 🜃
✓ 24_cuc-pub_192.168.21.4 🗵
                                                                                                                                                                                                        4
                                                                                                                                                                                                           Þ
admin:utils network ping cuc-sub ——
PING cuc-sub.ajcollab.com (192.168.21.5) 56(84) bytes of data.
64 bytes from cuc-sub.ajcollab.com (192.168.21.5): icmp_seq=1 ttl=64 time=0.145 ms
64 bytes from cuc-sub.ajcollab.com (192.168.21.5): icmp_seq=2 ttl=64 time=0.154 ms
64 bytes from cuc-sub.ajcollab.com (192.168.21.5): icmp_seq=3 ttl=64 time=0.166 ms
64 bytes from cuc-sub.ajcollab.com (192.168.21.5): icmp_seq=4 ttl=64 time=0.181 ms
--- cuc-sub.ajcollab.com ping statistics ---
4 packets transmitted, 4 received, 0% packet loss, time 3005ms rtt min/avg/max/mdev = 0.145/0.161/0.181/0.018 ms
admin:
admin:
admin:utils network traceroute cuc-sub traceroute to cuc-sub (192.168.21.5), 30 hops max, 60 byte packets 1 cuc-sub.ajcollab.com (192.168.21.5) 0.213 ms 0.138 ms 0.135 ms
admin:
admin:
admin:
```

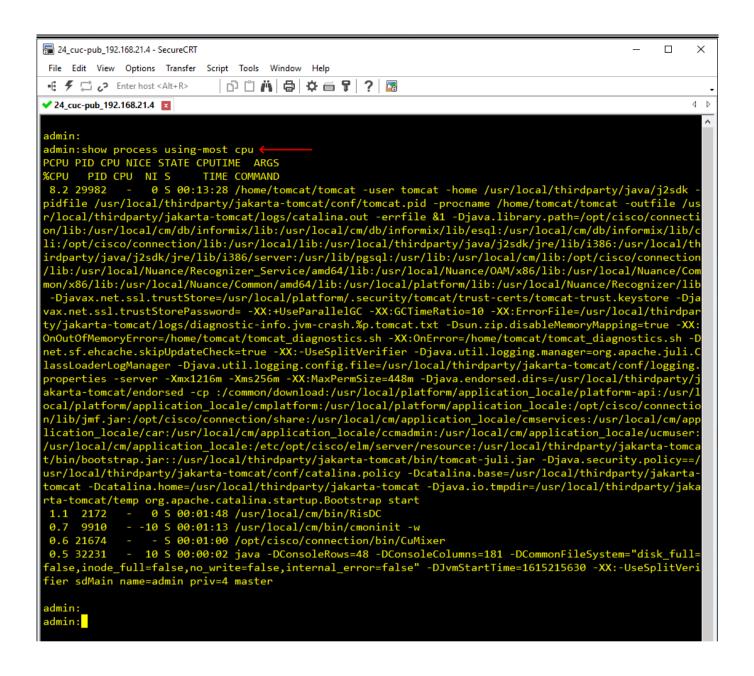
```
24_cuc-pub_192,168,21,4 - SecureCRT
 File Edit View Options Transfer Script Tools Window Help
                                 ■ F 🖾 🞝 Enter host < Alt+R>
✓ 24 cuc-pub 192.168.21.4 🗵
                                                                                                                           4
                                                                                                                             Þ
 admin:show web-security \leftarrow
   Version: V3
   Serial Number: 5BDBC7E3D110AD3FF02D7B168DCBA011
   SignatureAlgorithm: SHA256withRSA (1.2.840.113549.1.1.11)
  Issuer Name: L=Bangalore, ST=Karnataka, CN=cuc-pub.ajcollab.com, OU=Collab, O=AJ Collab, C=IN
   Validity From: Sun Mar 07 01:17:14 IST 2021
            To: Fri Mar 06 01:17:13 IST 2026
  Subject Name: L=Bangalore, ST=Karnataka, CN=cuc-pub.ajcollab.com, OU=Collab, O=AJ Collab, C=IN
 Key: RSA (1.2.840.113549.1.1.1)

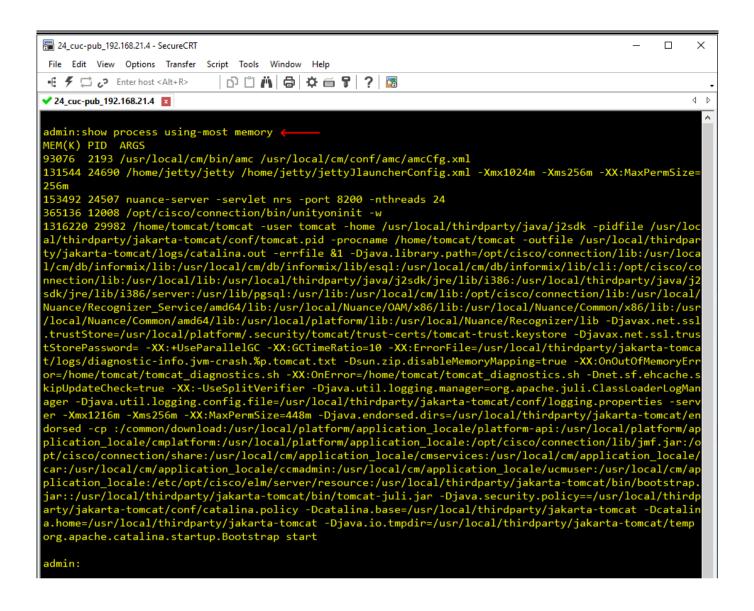
Key value: 3082010a0282010100ef5fdba65eeed6e53488b9128371163ca5359d65c5a37102ebddb0e643ab178893b232
064b9e0afd051c8949b37dba94201a05f9d6a5b3496b9ea63f0a02c74928d18efaabd856d32b025f1f5306ed3fba90478846d4f
 d130fa8dfb3c462be396f084ad78679ab95a48e479fe6286ba8dc3423215fc34fbd62fdfd49d30a21facc5bed73d74dbdd4e3b4
2e46786725b41934d803c6912799f091266d3de4e131757af6f46c7112f7a4ac54d5061297ad257faea93cf2ad3d78f6ad46170
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  28dc9a8d8d52104b8865cf246b39f0203010001
   Extensions: 4 present
      Extension: KeyUsage (OID.2.5.29.15)
      Critical: false
      Usages: digitalSignature, keyEncipherment, dataEncipherment, keyCertSign,
      Extension: ExtKeyUsageSyntax (OID.2.5.29.37)
      Critical: false
      Usage oids: 1.3.6.1.5.5.7.3.1, 1.3.6.1.5.5.7.3.2,
      Extension: SubjectKeyIdentifier (0ID.2.5.29.14)
      Critical: false
      keyID: 7f35e641e8e2e2c6ba1c6efab3338eae00251710
      Extension: BasicConstraints (OID.2.5.29.19)
      Critical: true
      pathLenConstraint: 0
   Signature:
       0: c4 14 40 0b cf 60 15 ef 85 43 cf cf a2 f9 3c da [..@..`...C....<.]
   0010: 3e bf ae 74 9a 8e d3 9d 80 ff 6a e9 6c 2f 50 56 [>..t....j.l/PV]
   0020: 5e c0 b2 8d b2 b7 c5 8e 0b 85 b0 52 2f a2 74 e9 [^...........R/.t.]
0030: 34 ab 39 42 3c 72 09 21 12 67 77 92 0b 40 f1 d7 [4.98<r.!.gw..@..]
   0040: 26 ec c1 4c 7f 26 ed b8 33 d9 88 93 49 9e 2e b4 [&..L.&..3...I...]
                                                               ssh2: AES-256-CTR 47, 63 47 Rows, 103 Cols Xterm CAP NUM
Ready
```

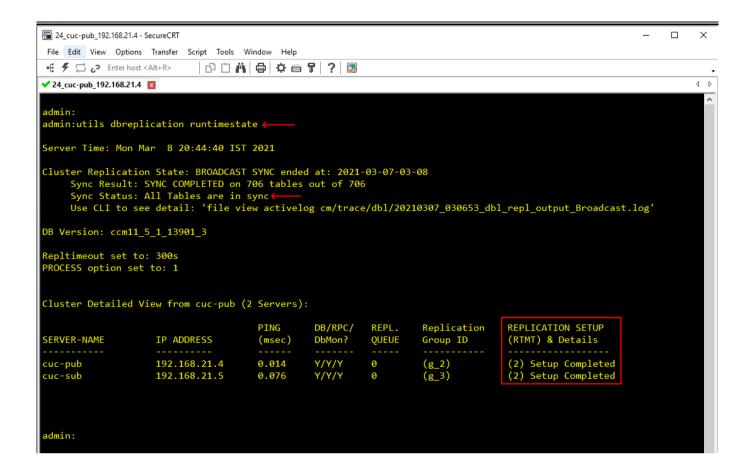


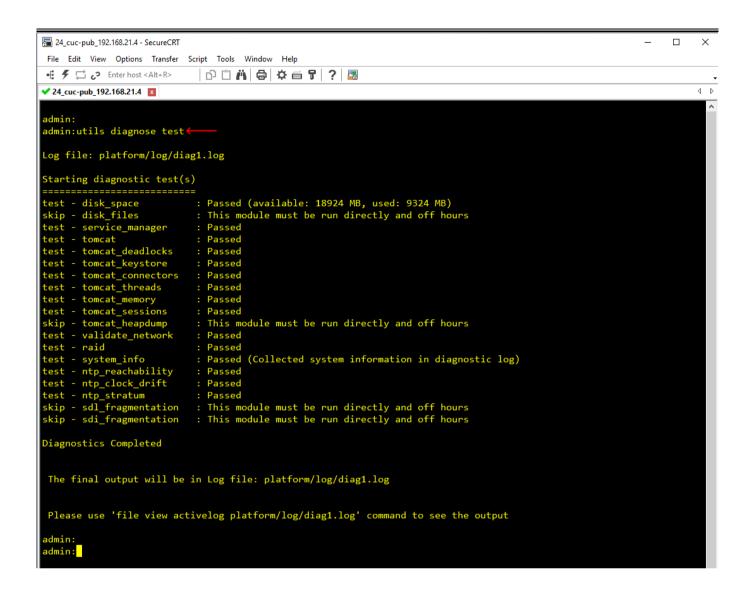


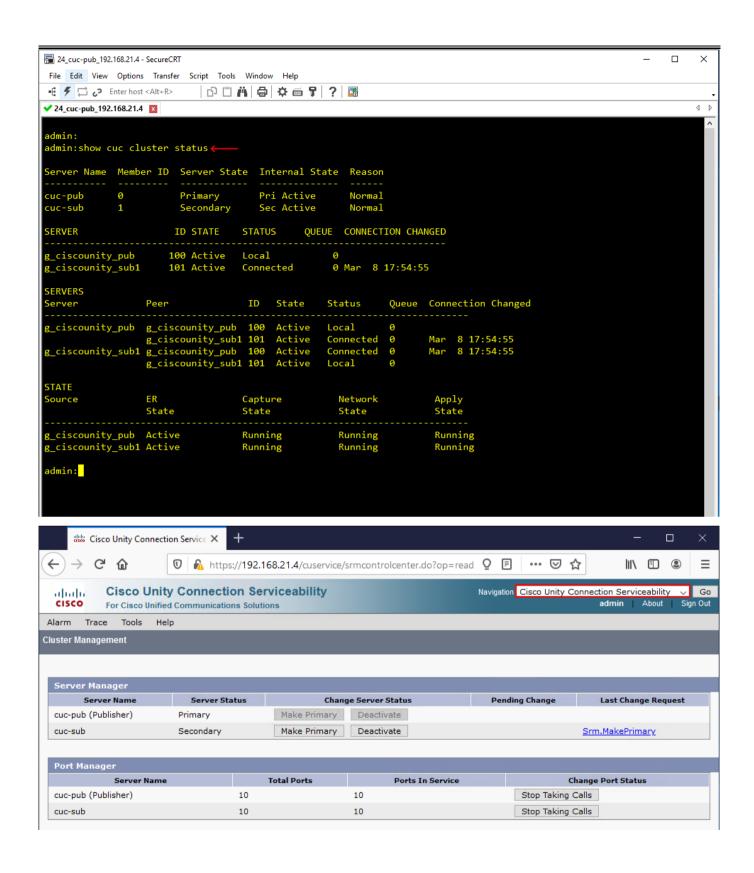




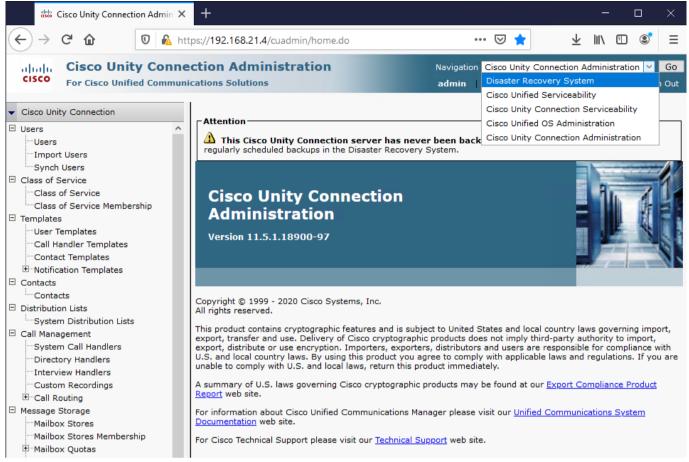






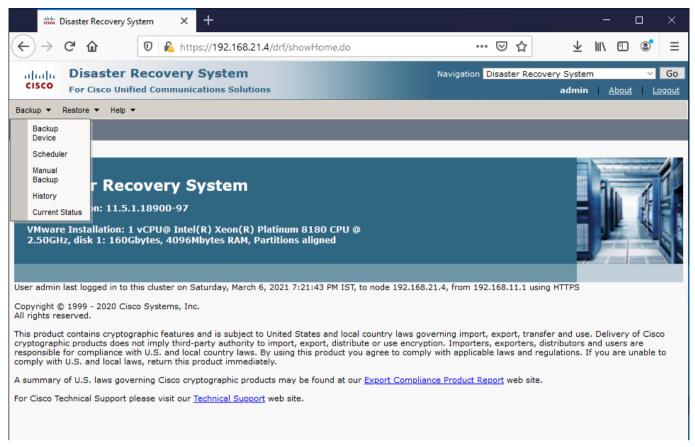


Web GUI and Five Consoles in CUC



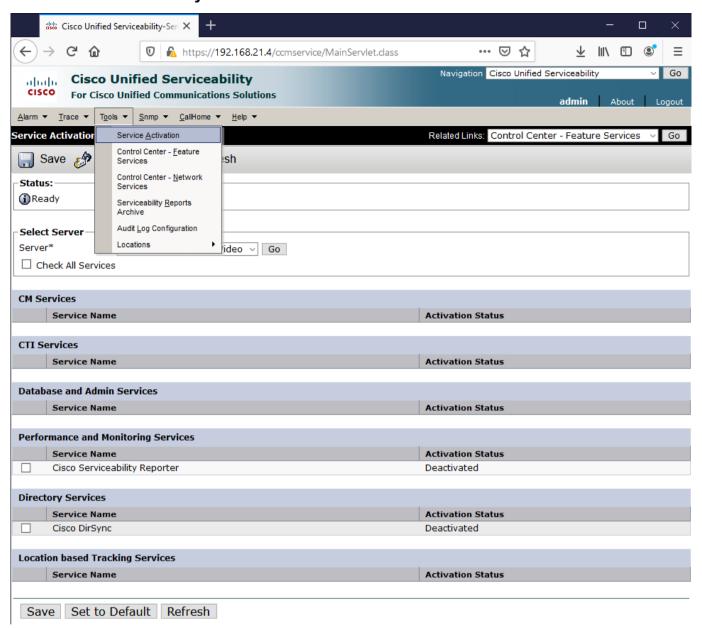
- There are 5 different web consoles available in CUC to administrate
- We usually browse the IP Address or FQDN of CUC PUB to get access to the web interface
- Cisco Tomcat Service is responsible for delivering the web interface over HTTPS
- Both the CUC nodes will have the web interface but all of them connects to the DB of CUC PUB if
 PUB is running
- The only time the CUC SUB talks to its own DB via HTTPS web GUI, when the CUC PUB is down
- During such scenario, we will not be able to change any configuration on the cluster, we just can see the things (read only mode) since CUC PUB is the read write copy of DB
- Unlike CUCM, here CUC works as Active-Active pair (Primary and Secondary). CUC-PUB is also participating in the voicemail / IVR functionality

Disaster Recovery System



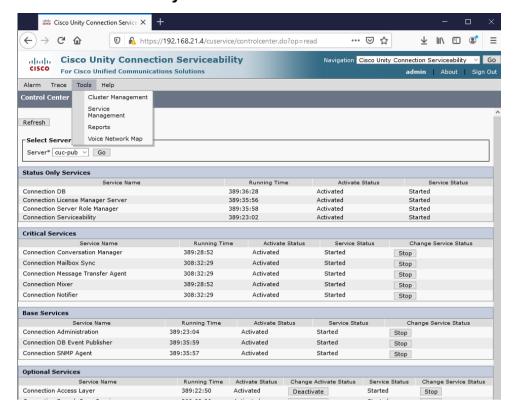
- Used to configure backup-device and perform manual and scheduled auto backup of the CUC cluster
- Also used to restore the DB in case of major database corruption or failures
- Cluster backup is stored to SFTP server as flat files
- It is not recommended to use the vmware native backup option like vmware Snapshot for CUC VM as gives bad performance and IO delay
- vmware image backup solutions like Veem is also not recommended to take CUC backup
- Also, it is not recommended to enable vmware vmotion and HA for CUC nodes
- The only recommended backup is via SFTP server
- We use Platform User Credentials (OS Admin credentials) to login to Cisco Unified Disaster
 Recovery System interface

Cisco Unified Serviceability

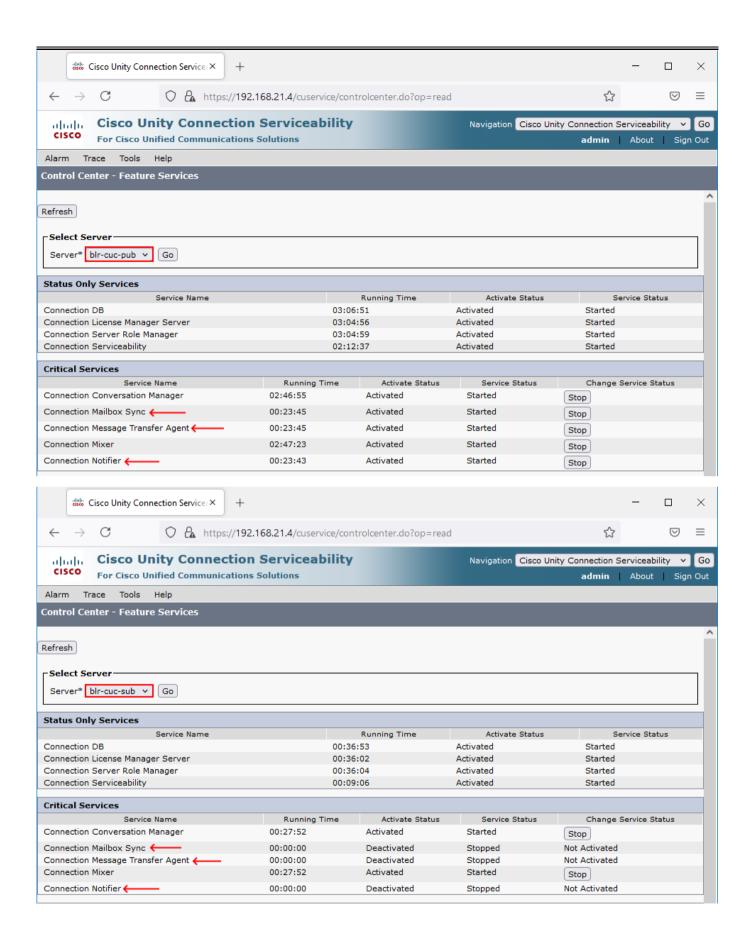


- There are only 2 feature services in CUC and that can be activated, started, stopped, and restarted from here. Most of the services are moved to different console called 'Cisco Unity Connection Serviceability'
- We use Application User Credentials to login to Cisco Unified Service Ability interface

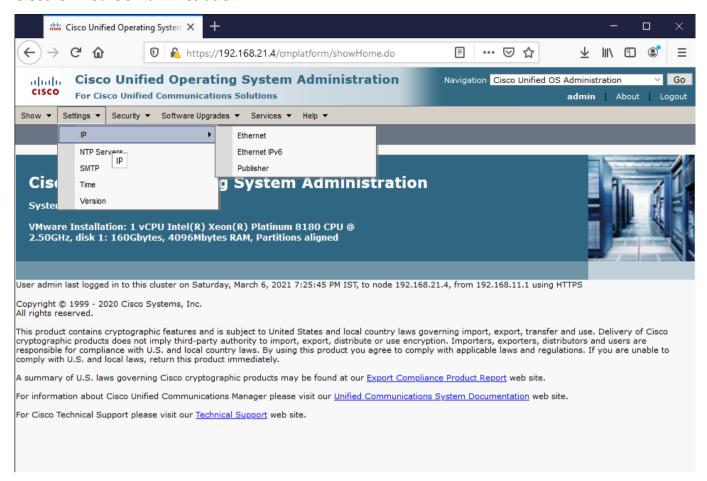
Cisco Unity Connection Serviceability



- Most of the services are moved to this console
- Connection Conversation Manager: Enables Unity Connection to handle calls. Disabling this service degrades the ability of Unity Connection to function. It runs in both primary and secondary CUC servers
- Connection Mailbox Sync: Synchronizes messages between Unity Connection and Exchange server. It runs only on CUC Primary
- Connection Message Transfer Agent: Enables the delivery of messages to the message store. It runs only on CUC Primary
- Connection Mixer: Enables the audio (media stream) for calls, recorded messages, and Text-to-Speech (TTS). It runs in both primary and secondary CUC servers
- Connection Notifier: This service enables notification of messages, such as turning message waiting indicators (MWIs) on and off. It runs only on CUC Primary
- Connection Mailbox Sync, Connection Message Transfer Agent and Connection Notifier runs on the CUC primary server, when it goes to secondary, these services switches between nodes
- We don't usually activate any services in CUC manually other than DirSync service for LDAP. Rest all services are by default activated
- We use Application User Credentials to login to Cisco Unified Service Ability interface

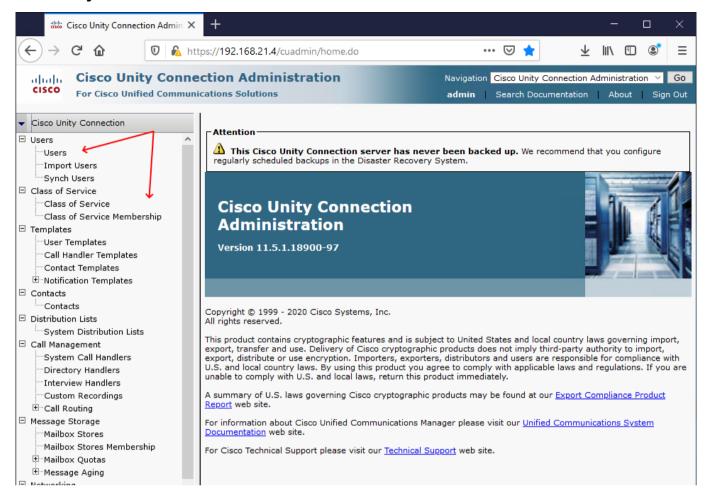


Cisco Unified OS Administration



- Here we can interact to the Linux OS of CUC like configuring IP Address, Changing DNS hostnames, Rebooting the node, Ping and much more
- CUC Node can be upgraded using this interface
- Some system status can be verified from here
- This is the same interface we access via SSH CLI
- We use Platform User Credentials (OS Admin credentials) to login to Cisco Unified OS Administration interface

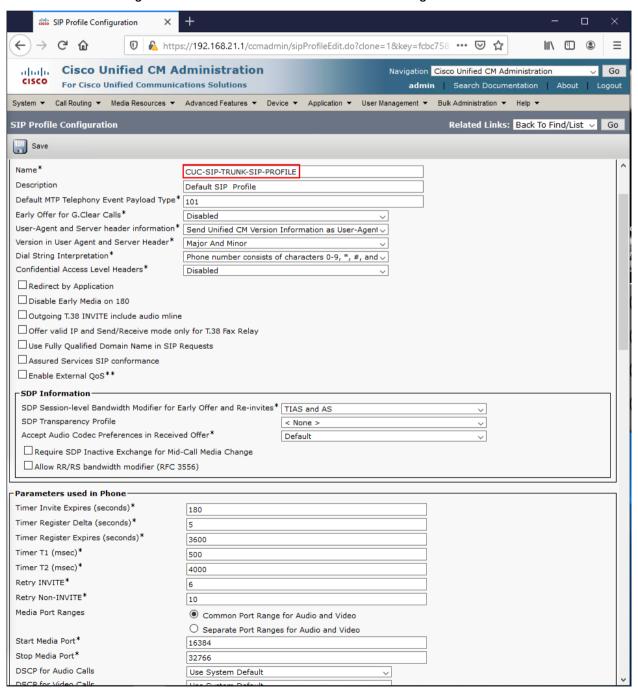
Cisco Unity Connection Administration

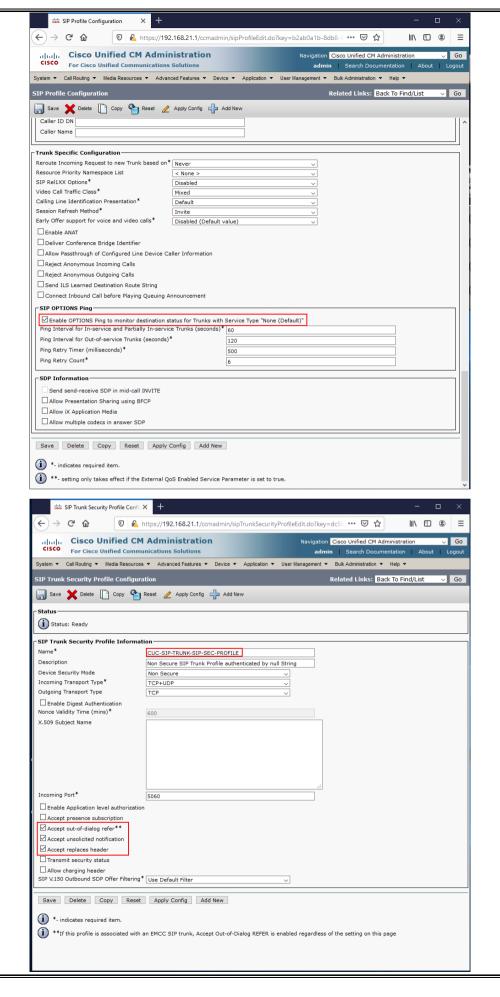


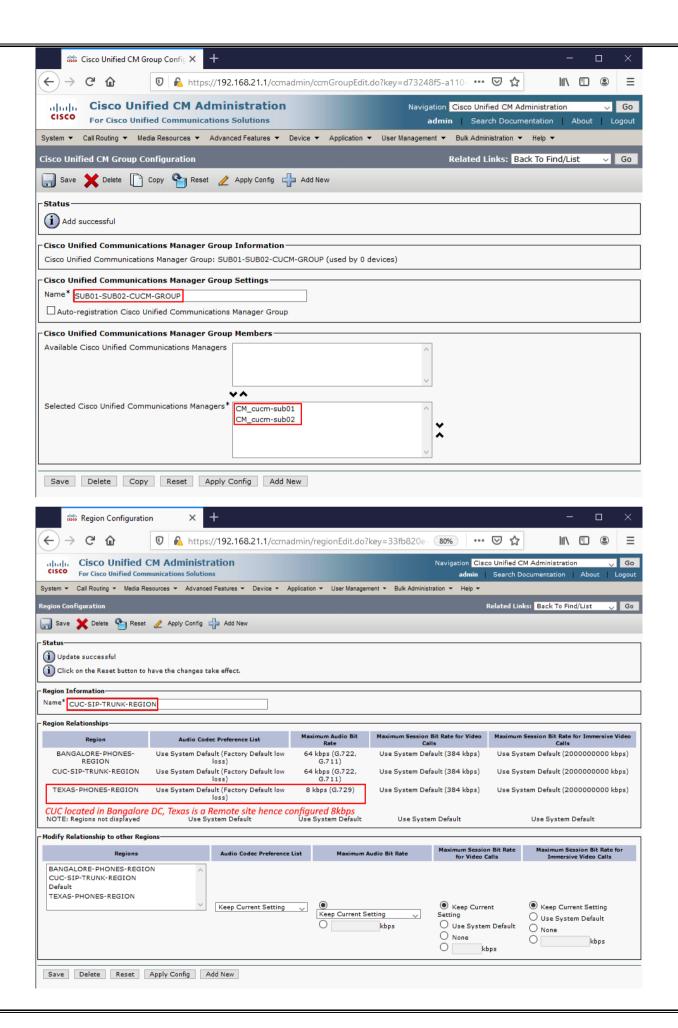
- This is the heart of CUC Administration
- Core day to day configurations like Adding Users, Deleting Users, Call Handler Configurations, etc.
 are done from Cisco Unity Connection Administration interface
- We will be here most of the time while dealing with CUC
- We use Application User Credentials to login to Cisco Unified CM Administration interface

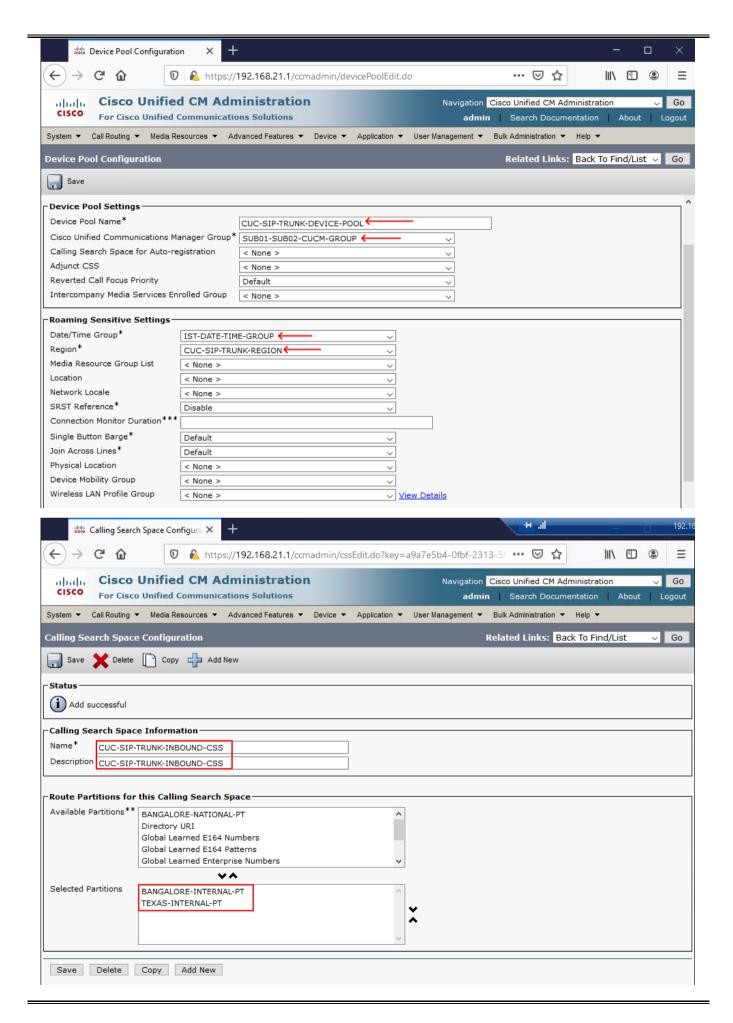
[Lab] Cisco Unity Connection SIP Integration with CUCM

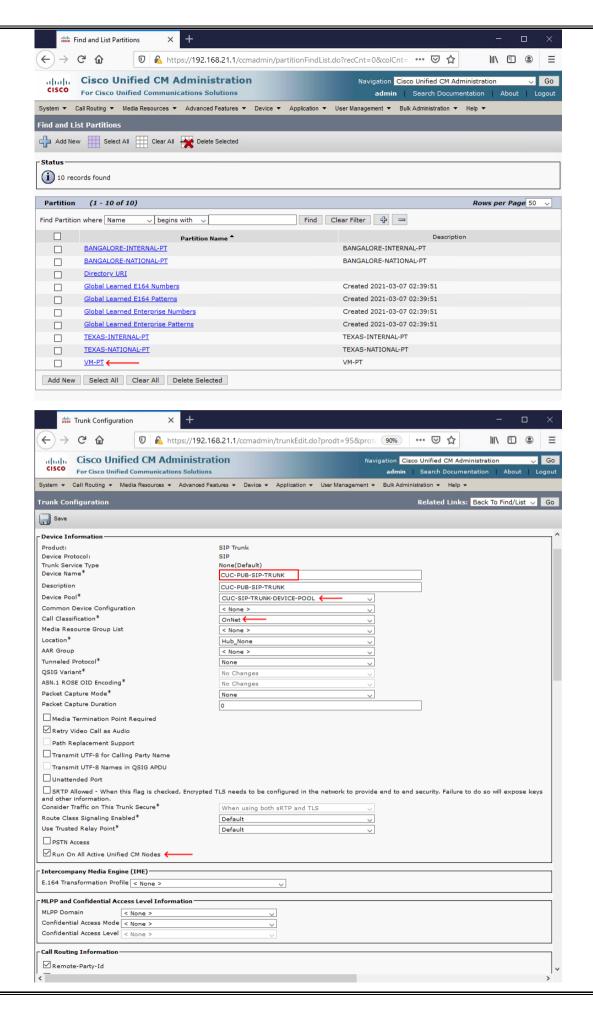
- CUC and CUCM can be integrated with each other either via SIP or SCCP protocol. SCCP is obsolete now hence we will be focusing on SIP integration
- In CUCM side we are creating a 2 SIP Trunks and 1 Route Pattern, hence we need to complete all the configurations for Trunks and Route Pattern
- When we have CUC PUB and SUB, it is recommended to have 2 SIP Trunks pointing to each CUC
 IPs
- I have configured CUC-SIP-PROFILE, CUC-SIP-SEC-PROFILE by copying the defaults, just like we
 did on CUBE integration section. I have not added those configurations here

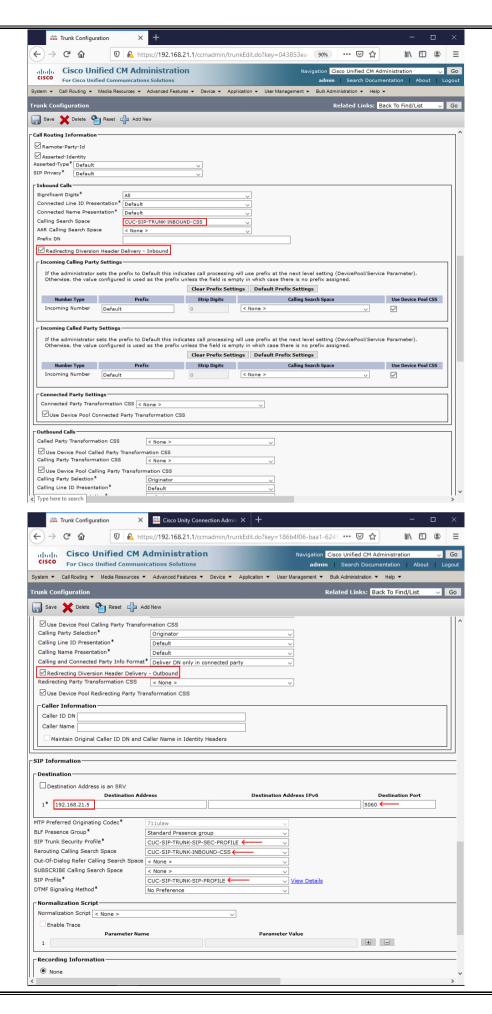


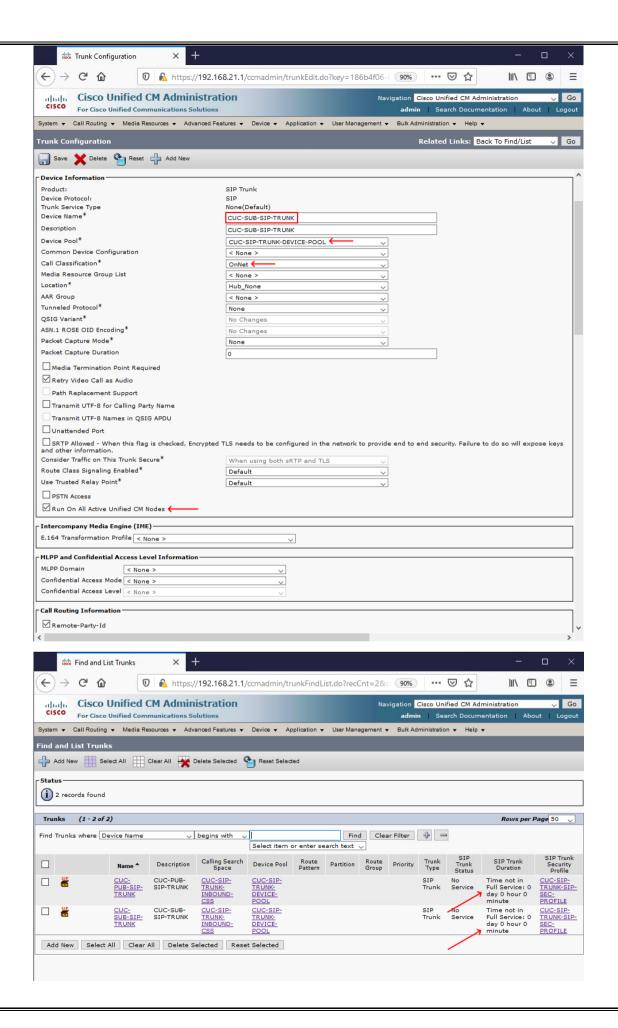


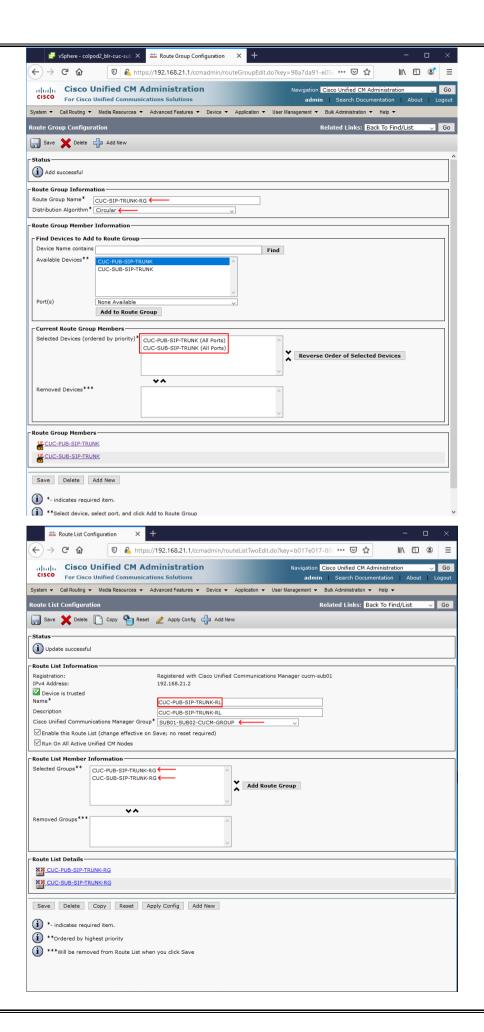


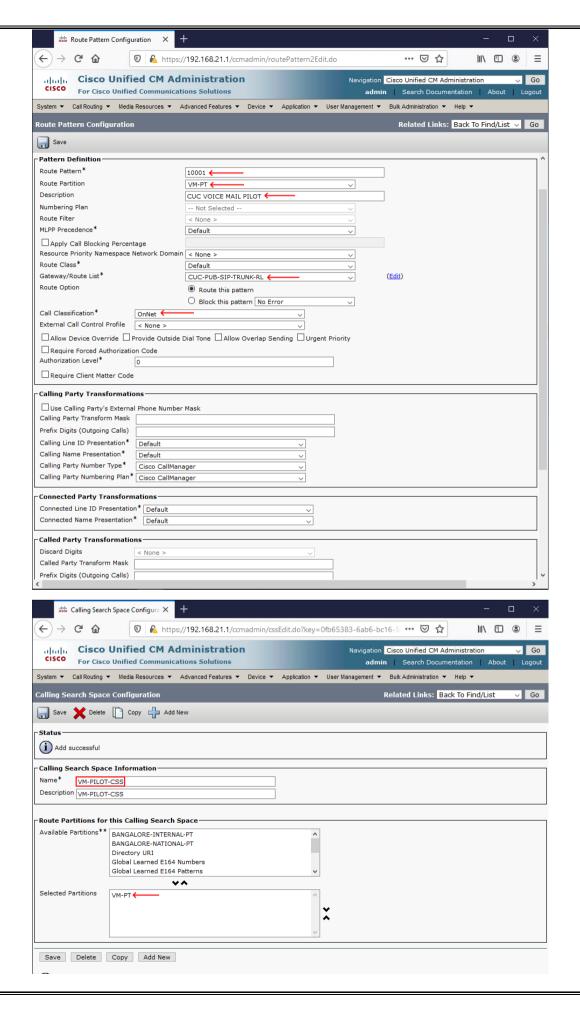


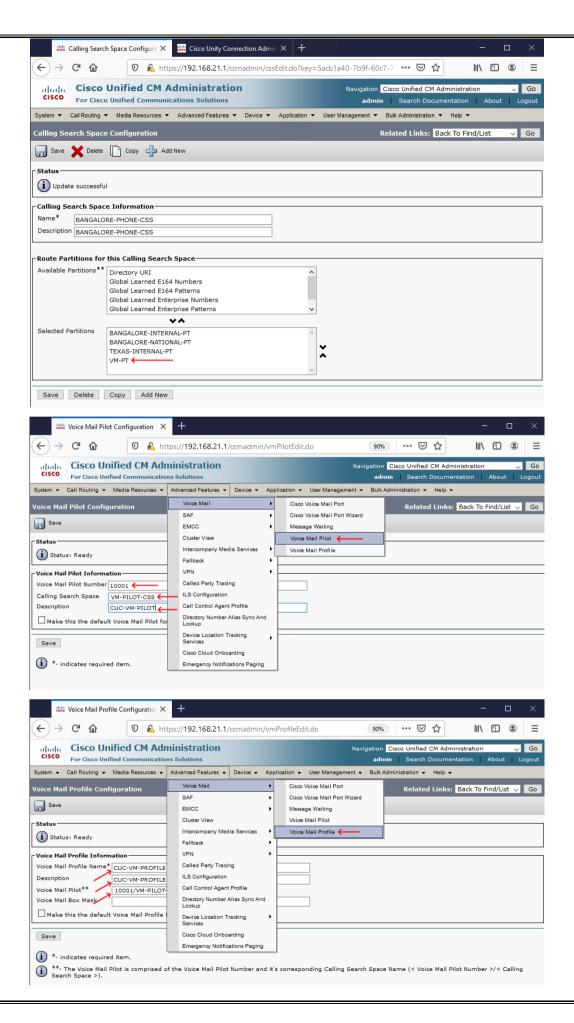


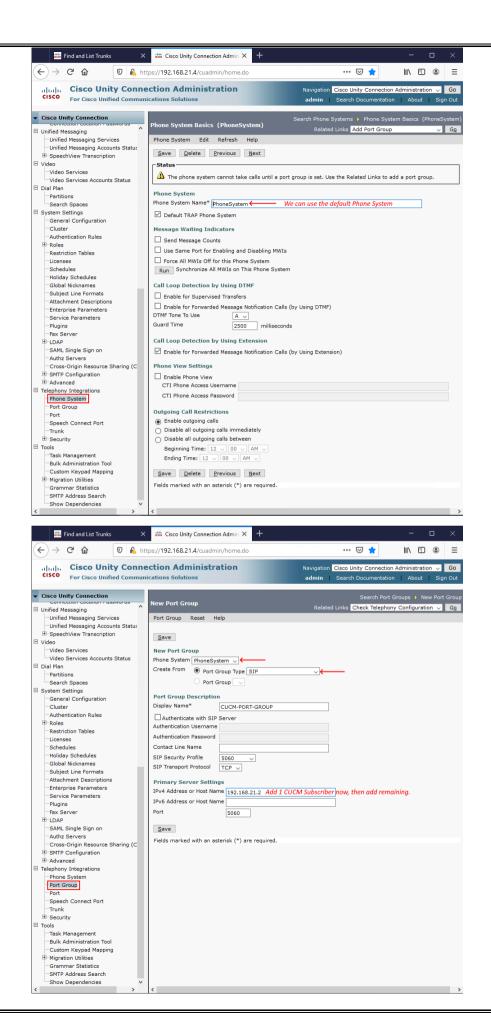


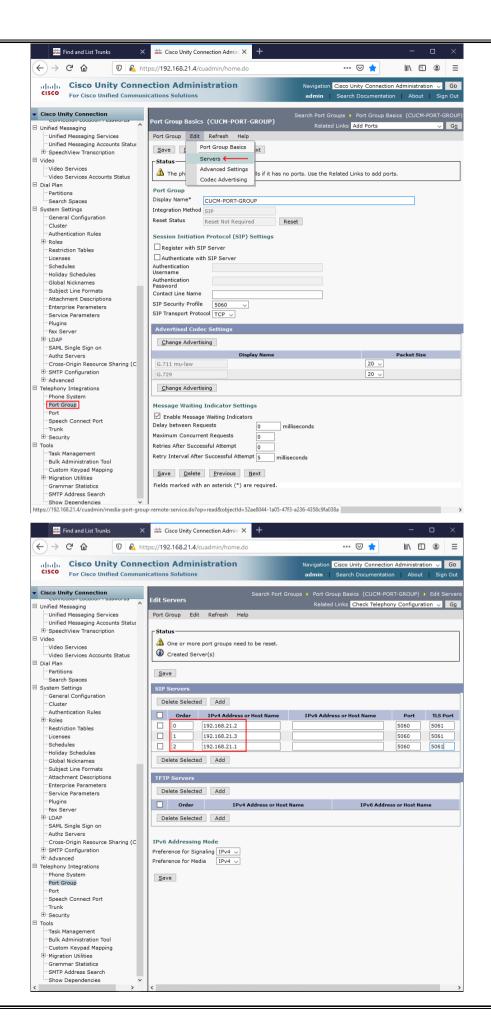


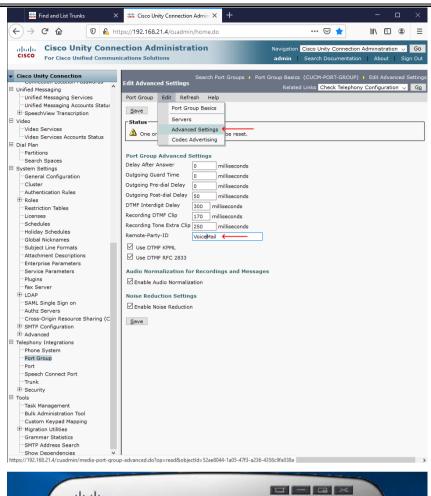




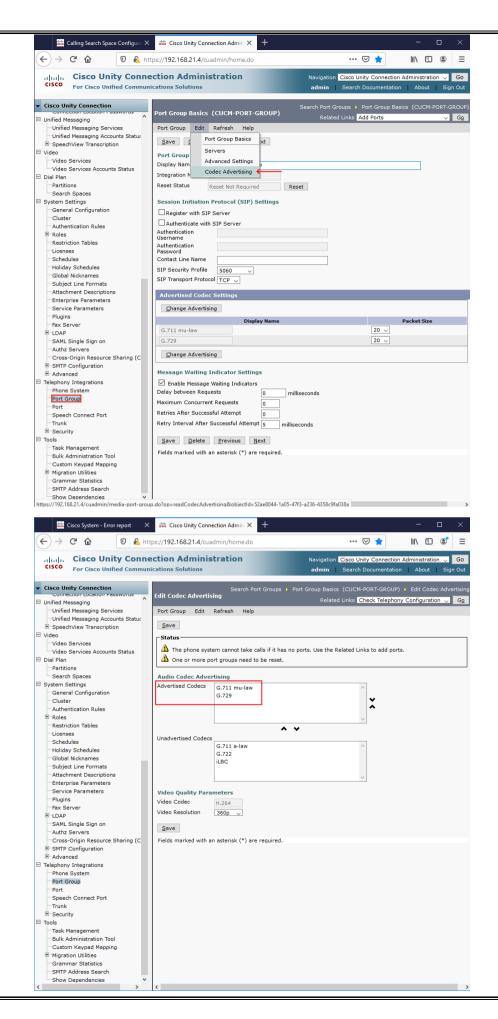


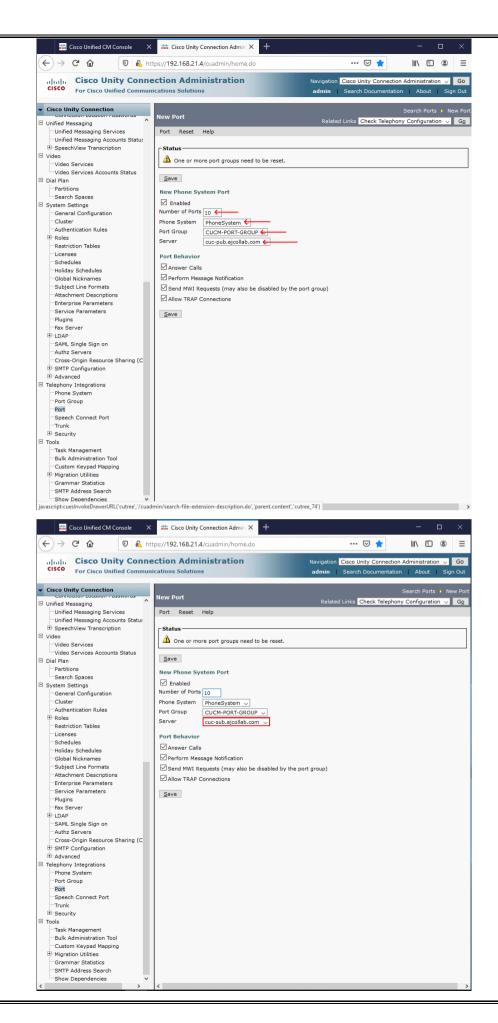


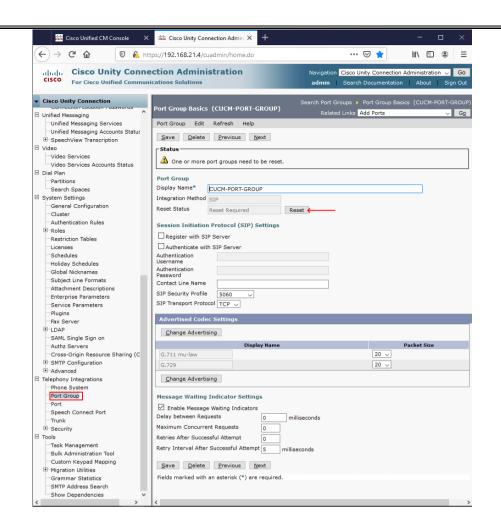


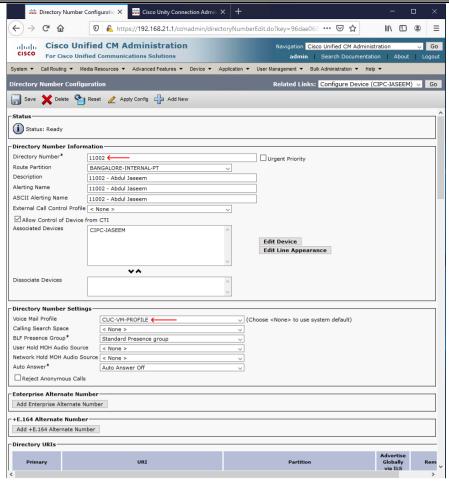




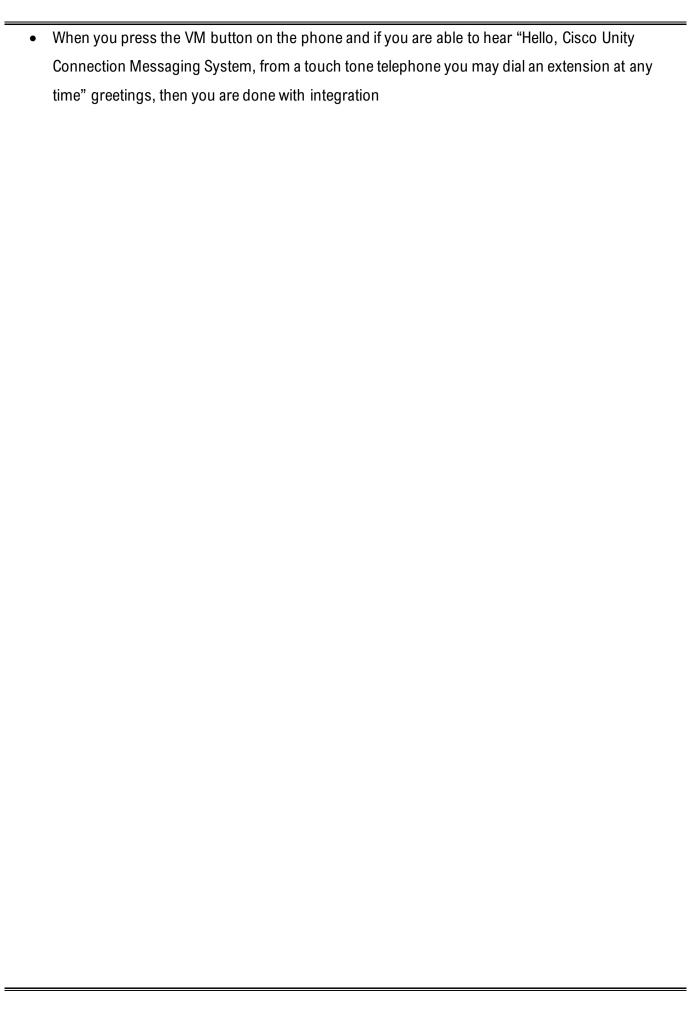










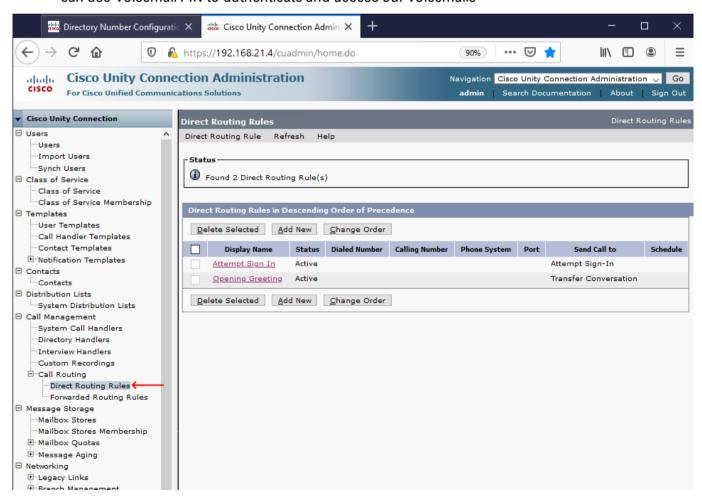


Understanding Call Routing Rules in Cisco Unity Connection

- When we make any call, there are 4 key parameters, Calling Number, Called Number, First Redirecting Number, Last Redirecting Number
- Once we route the call to CUC, Routing Rule decides how to treat the call
- CUC most of the time considers the Redirecting Number (only in forwarded call)

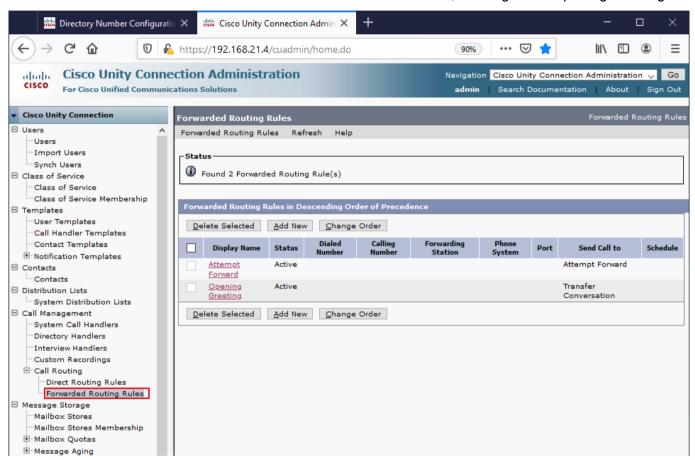
Direct Call

- When you hit voice message button on the phone or dial voicemail pilot directly
- Here we do not have any redirecting number since the user directly calling the pilot number
- This call is handled in 2 different ways in CUC
- If the Calling Number is not available in CUC Database, we hear the opening greetings "Hello, Cisco Unity Connection Messaging System, from a touch tone telephone you may dial an extension at any time". Now you might have understood why we got that greetings after the integration
- If the Calling Number is available in CUC database, it will ask for Attempt Sign In option, here we can use Voicemail PIN to authenticate and access our voicemails

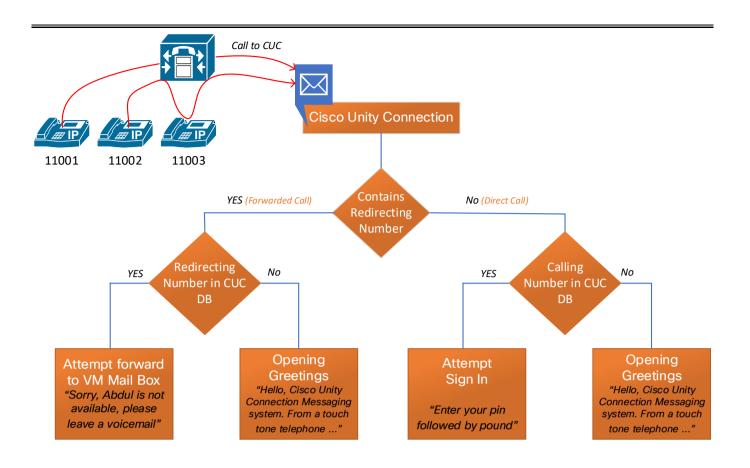


Forwarded Call

- When you forward your calls to Voicemail and someone calls you, this call will be a forwarded call
 to CUC and here we have the 'First Redirecting Number'
- This case CUC will always focus on the Redirecting Number and route the call
- If the redirecting number is available in the CUC database, then call will be forwarded to its voicemail box or call handler greetings
- If the Redirected Number is not available in the CUC database, it then goes to Opening Greetings

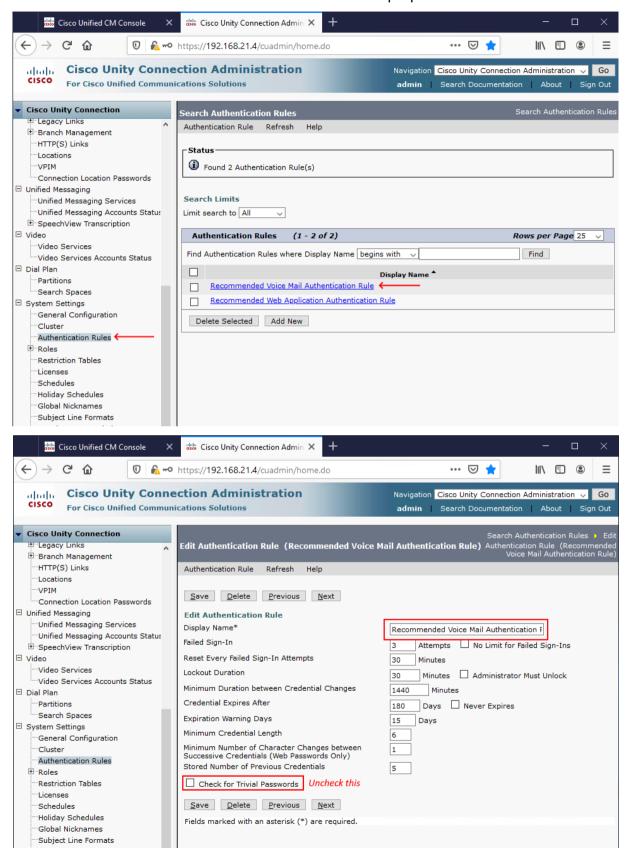


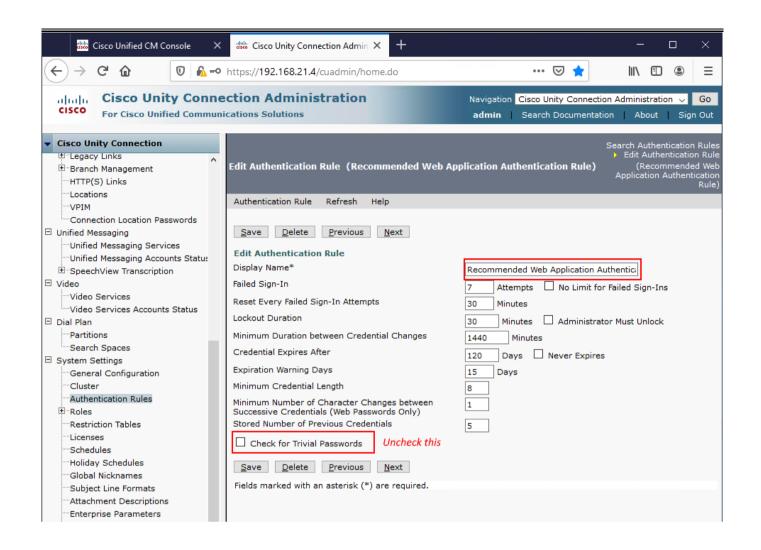
These are the default call routing treatment in CUC, we could add our own routing rules if we
wanted to modify some specific call rather than the default behavior



[Lab] Changing Authentication Rule

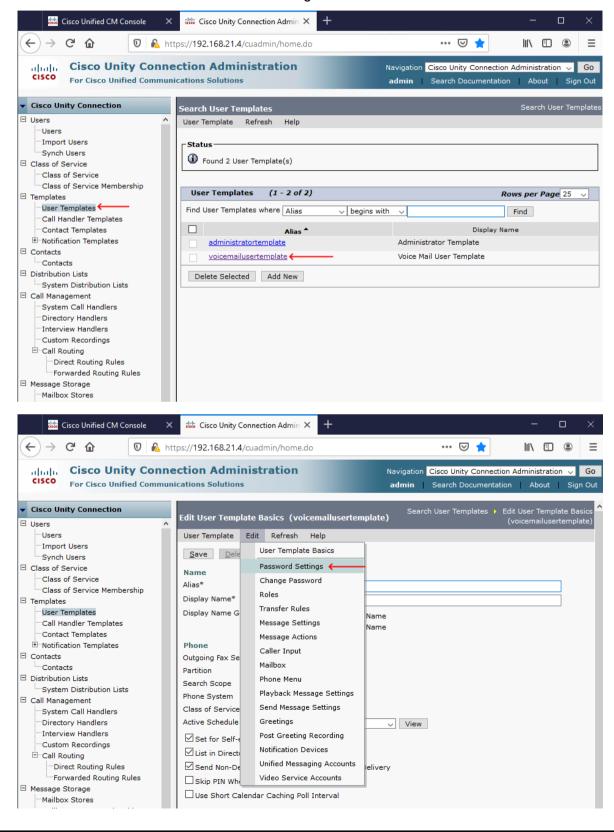
- Authentication rules are the password policy for voicemail user
- Let us disable 'Trivial Passwords' so that we can use simple password

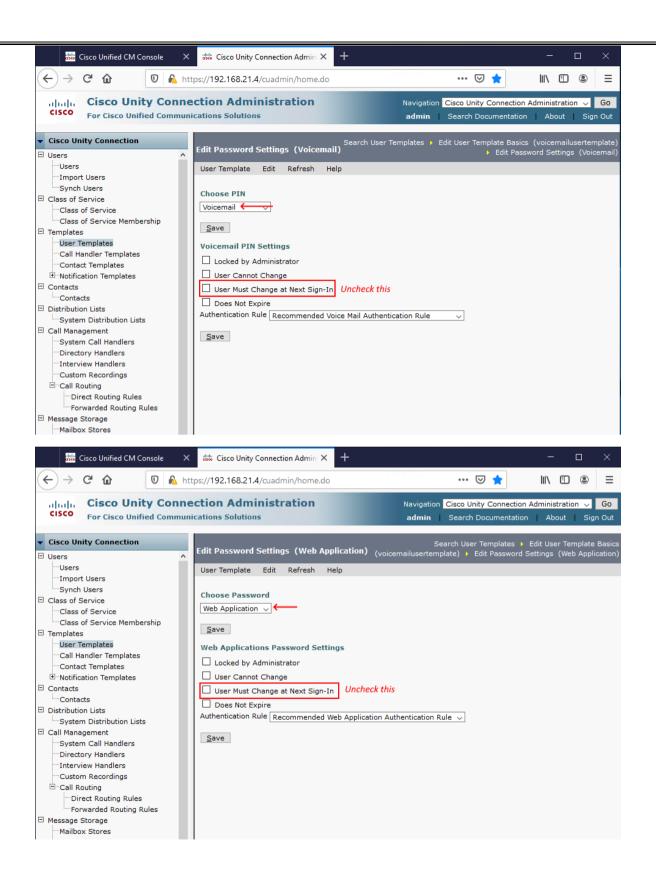


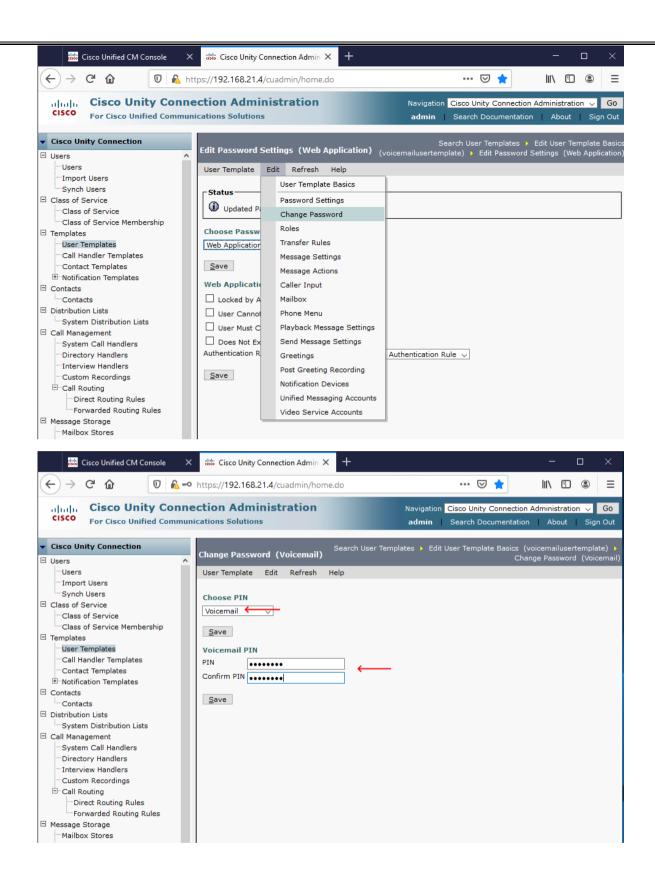


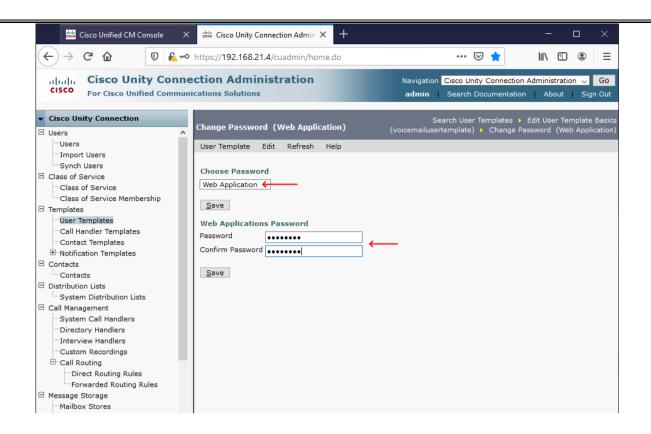
[Lab] Edit Voicemail Template

- Adding users in CUC is always based on voicemail template, hence edit the default 'voicemailusertemplate'
- We can set the default voicemail pin and web inbox password here
- Web inbox is a web GUI where we can manage the vociemails

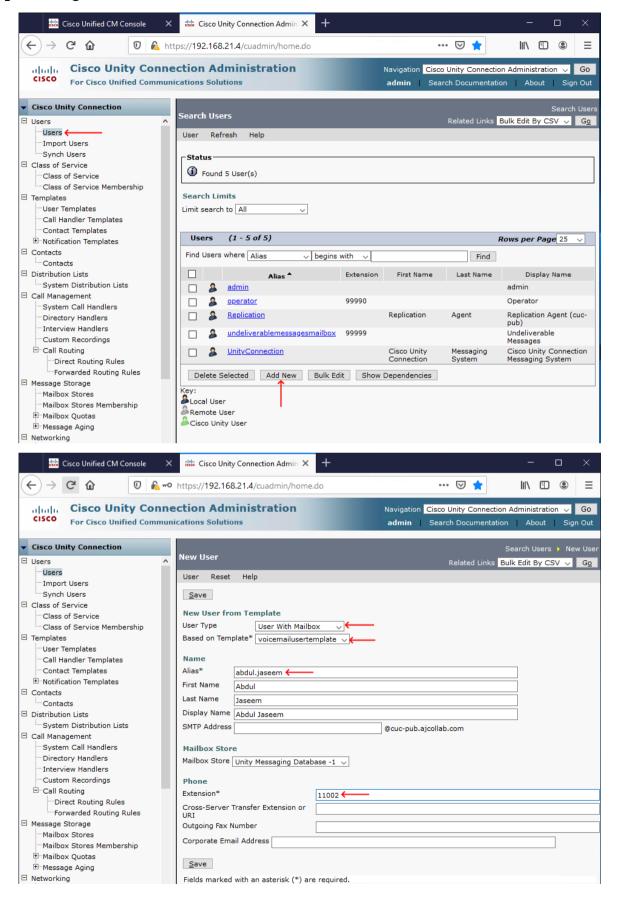


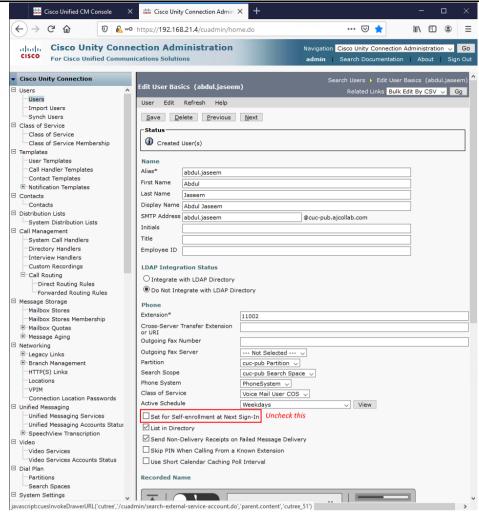






[Lab] Creating Voicemailbox User

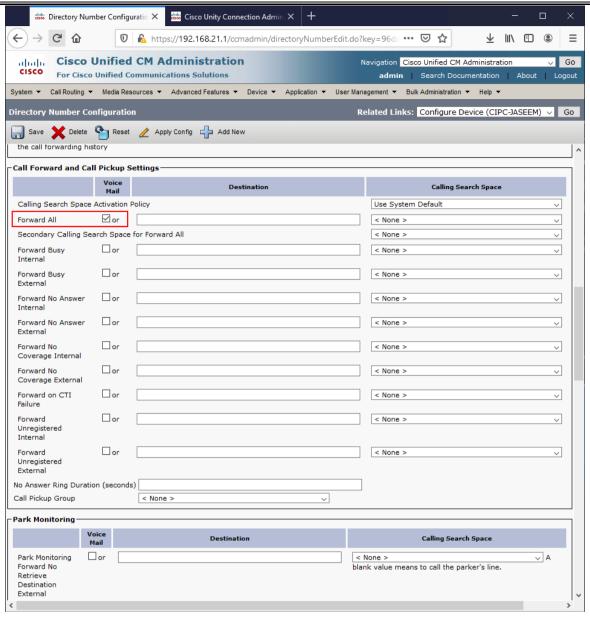






- Now you can hear "Enter your pin followed by pound" greetings since we created user and extension in CUC
- You can forward calls to voicemail and try to send a message from any other phone

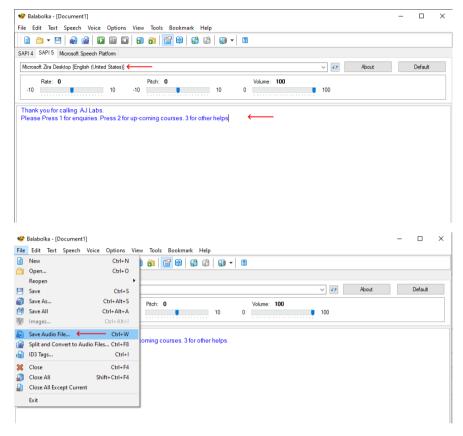
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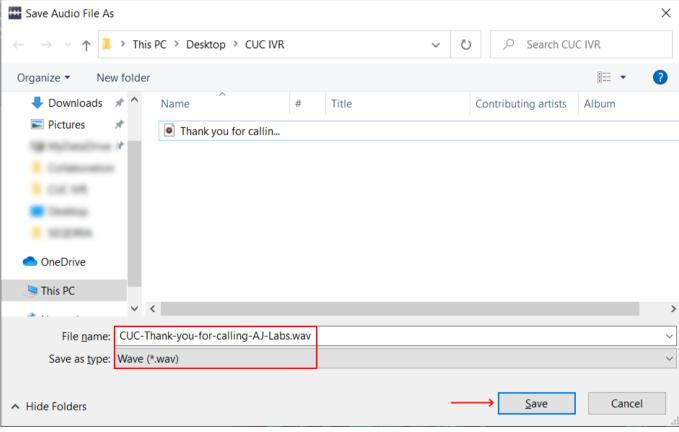
[Lab] Call Handler with Auto Attendant IVR

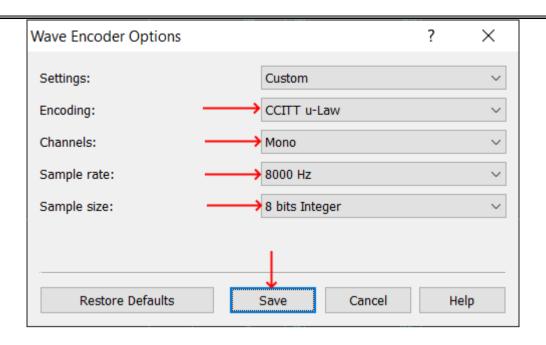
- Now let us design an IVR solution. First step we need to have a greeting file
- I have used 'Balabolka' software to create Text to Speech file, you may feel free to record your greetings file



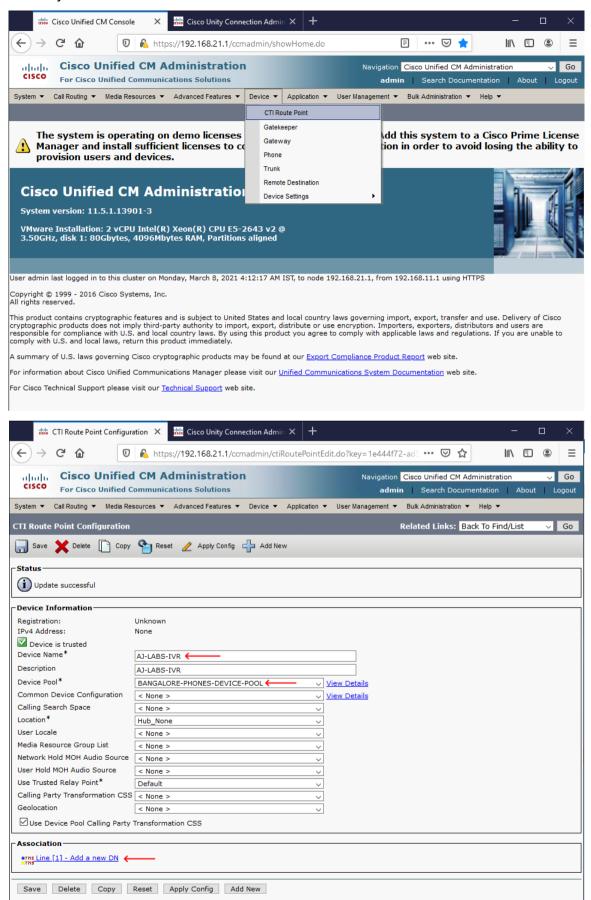
 Once you have the Audio file, whether recorded or TTS, you need to use 'Audacity' or 'WavePad Sound Editor' software to adjust the format and bitrate

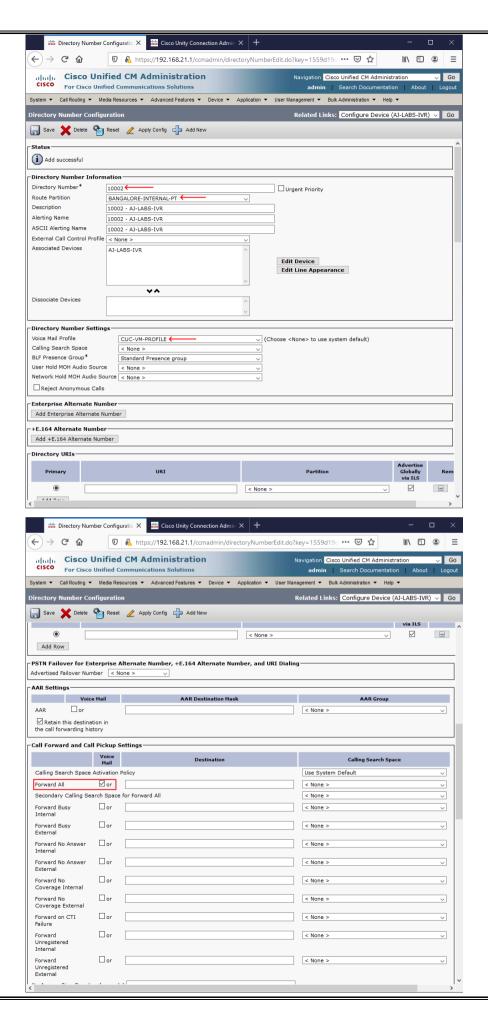


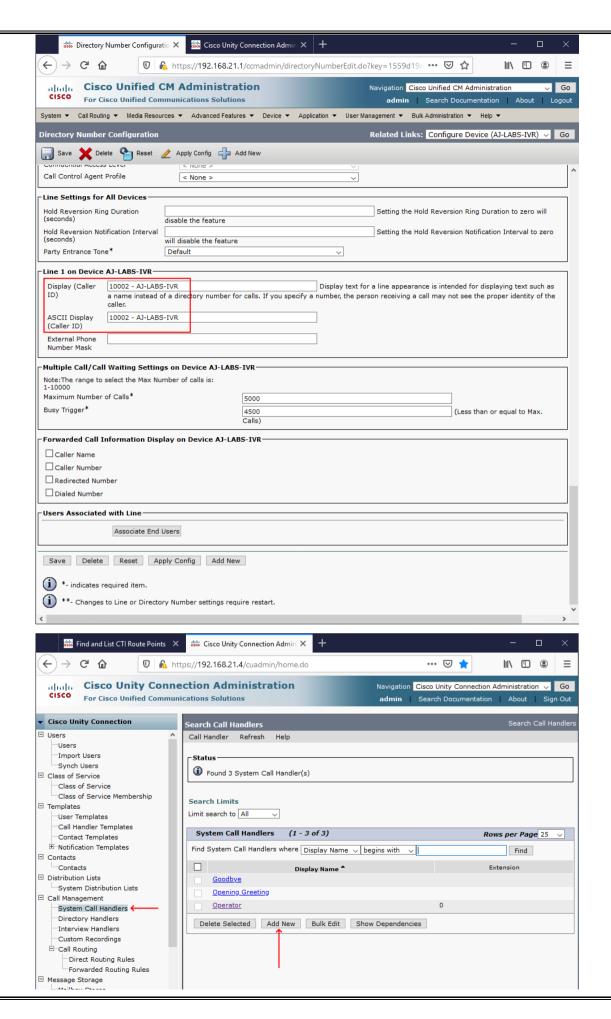


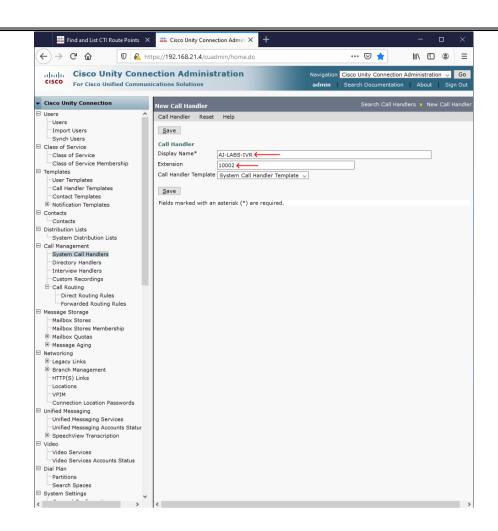


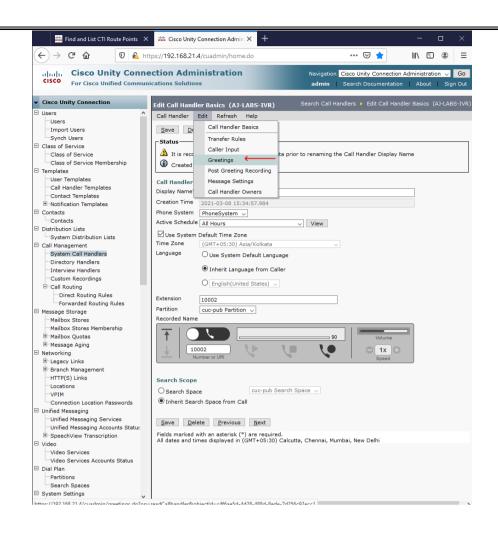
Create a Dummy CTI Route Point AJ-LABS-IVR

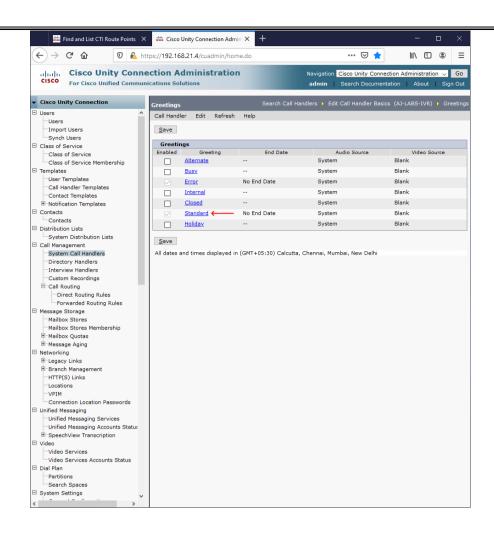


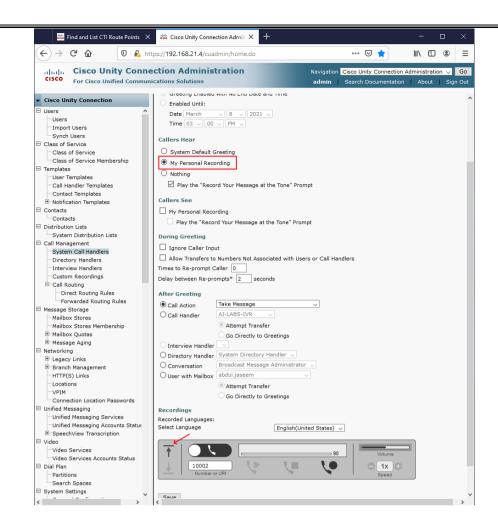


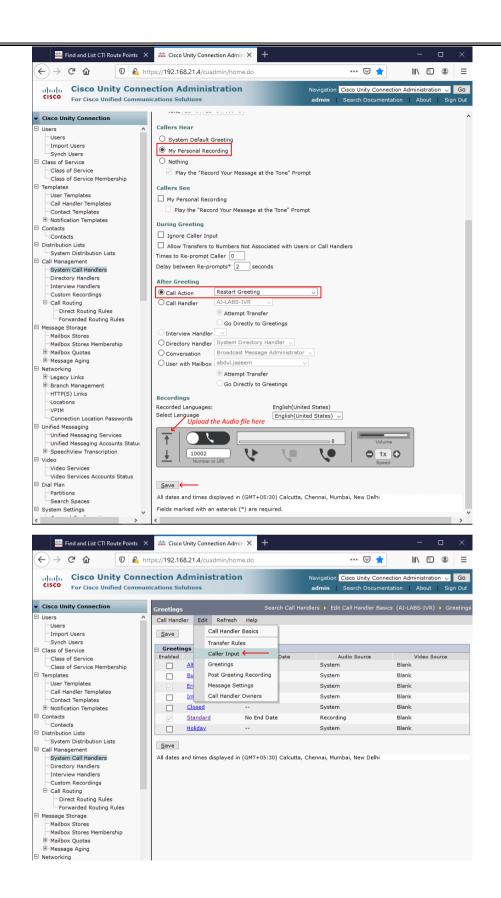


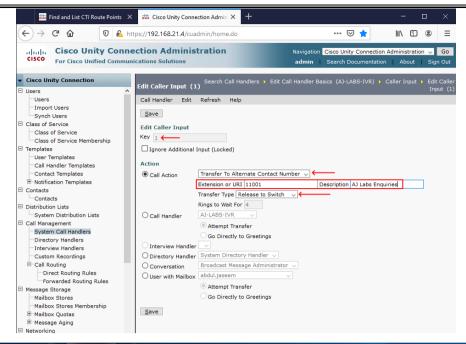






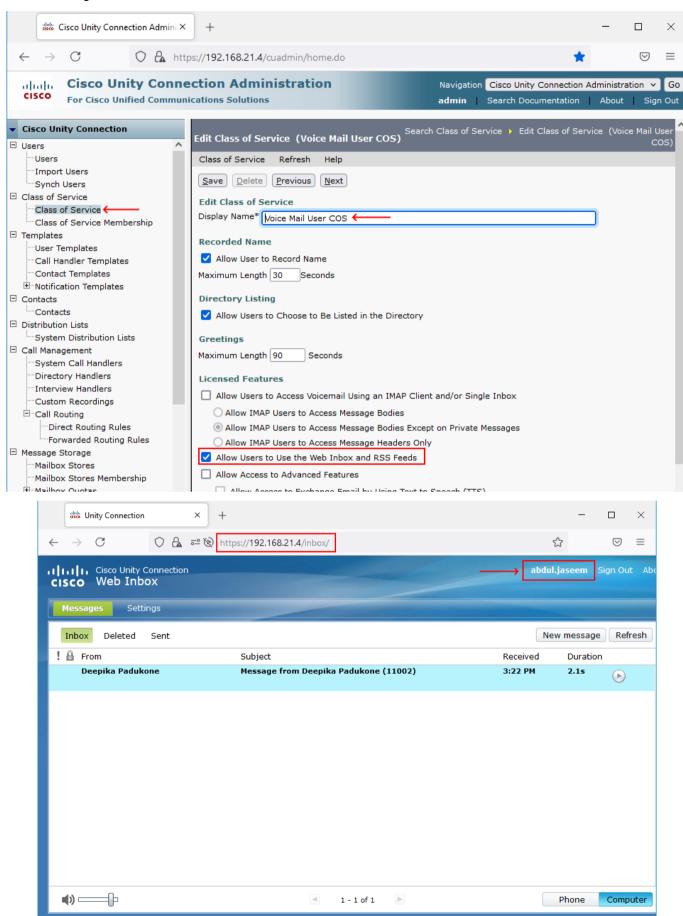




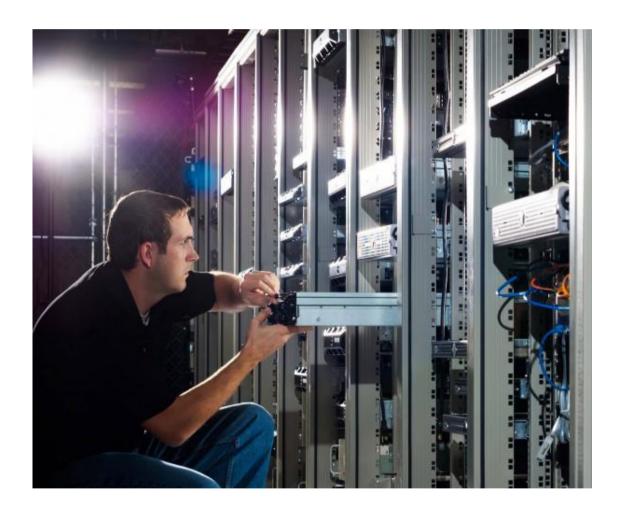




Cisco Unity Connection Web Inbox



Chapter 1 Module 4 - IMP Cisco On-Premise Collaboration Solution Cisco Unified IM and Presence (IMP)

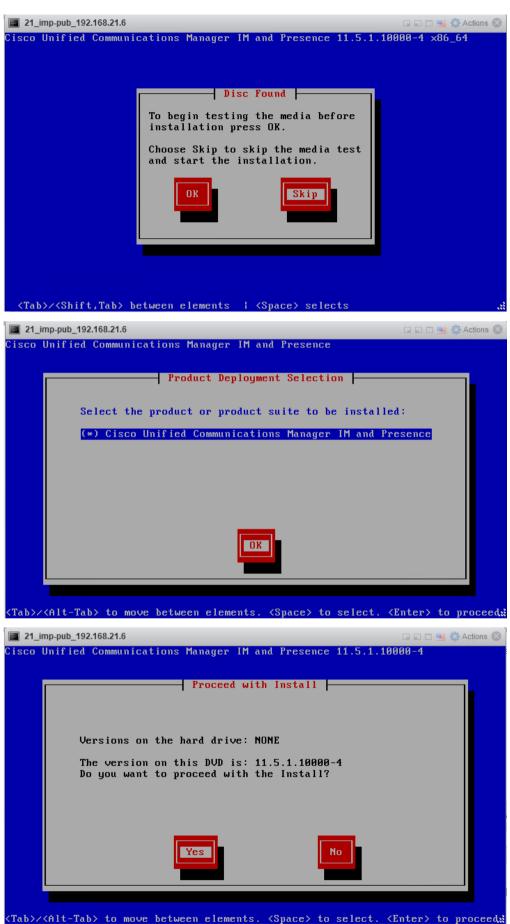


IM and Presence (IMP) Server

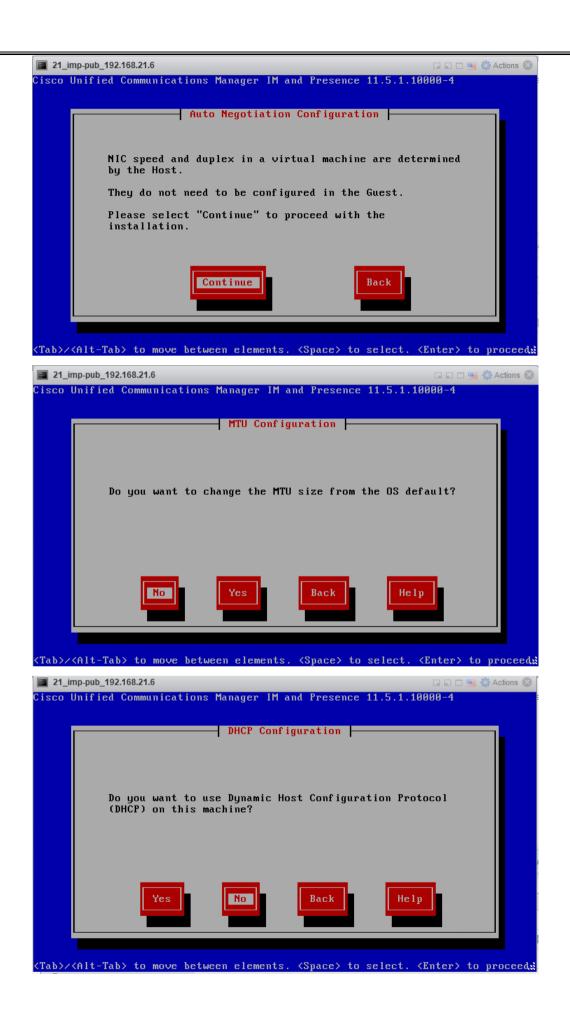


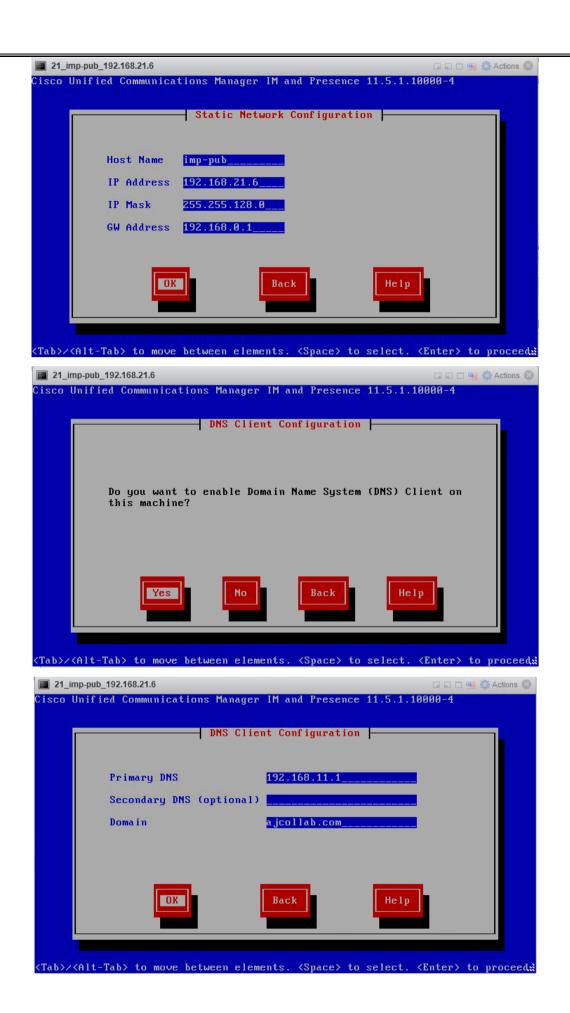
- The Instant Messaging & Presence (IM&P) service is an extensive messaging and presence collaboration service
- Provide Jaber Chat facility and presence (availability) service
- IM and Presence Servers are a part of the CUCM Cluster and installed as a Subscriber node to CUCM Publisher
- The first installed IM&P becomes IM and Presence Database Publisher and remaining are IM and Presence database subscribers
- IMP Publisher maintains the IMP database though IM&Ps are Subscriber to CUCM Publisher. That
 means the IMP Database replication is managed by IMP Publisher not CUCM Publisher

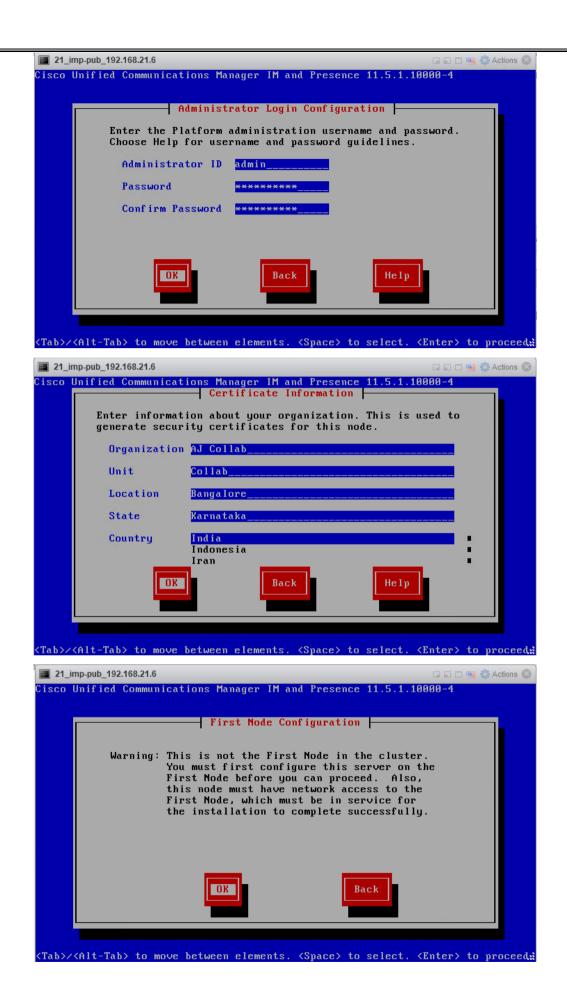
[Lab] IMP Installation

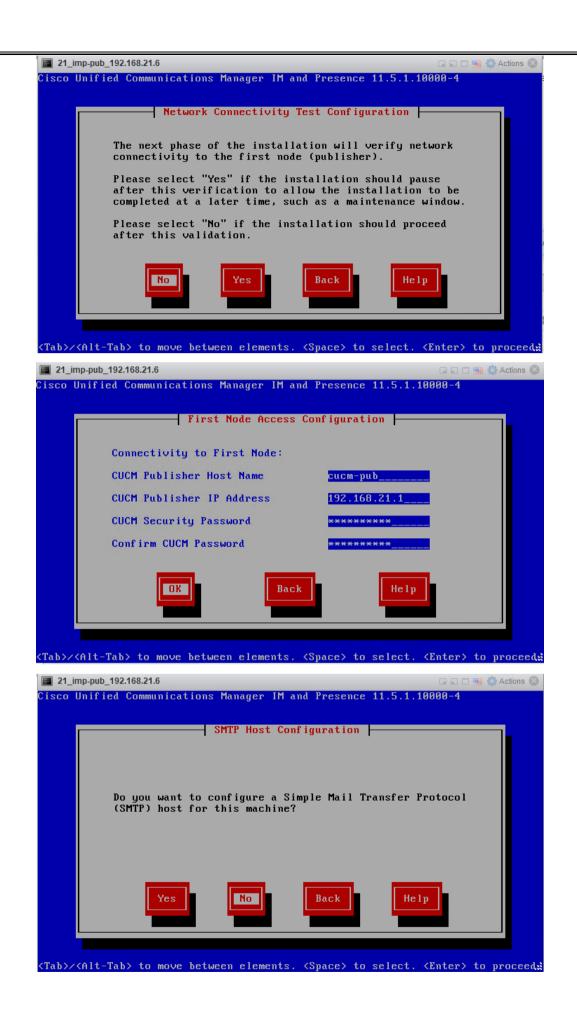








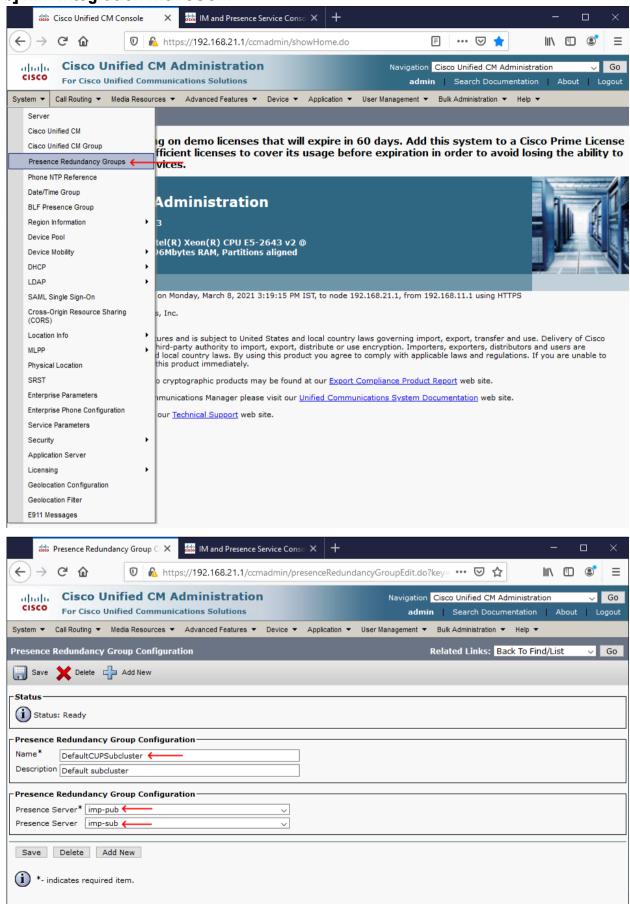


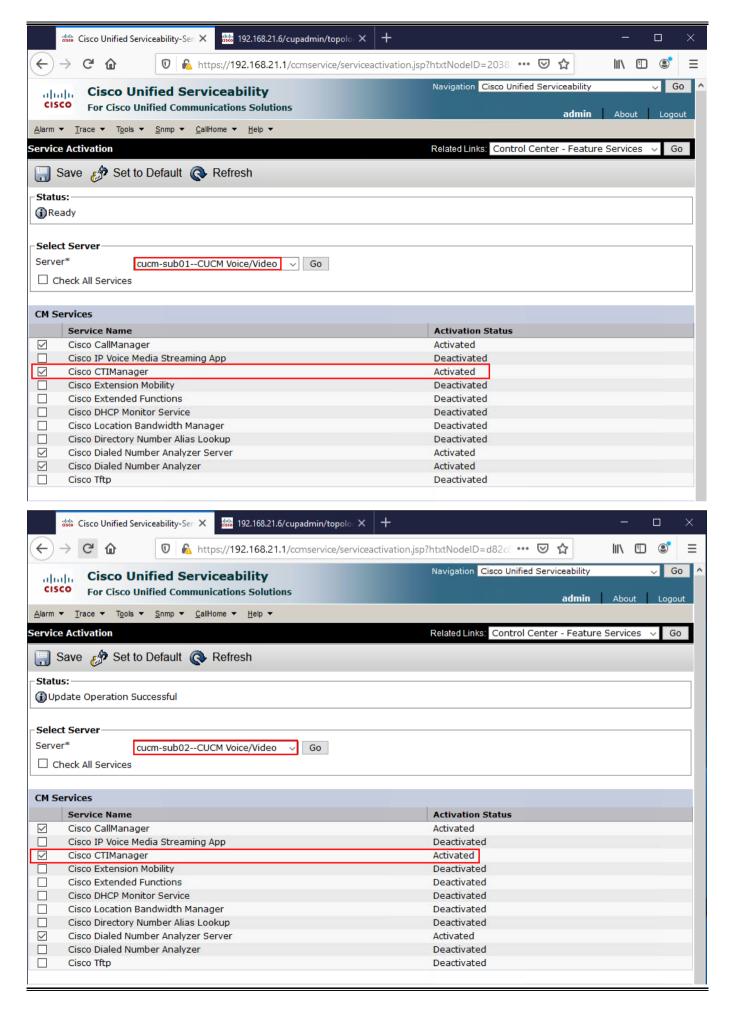


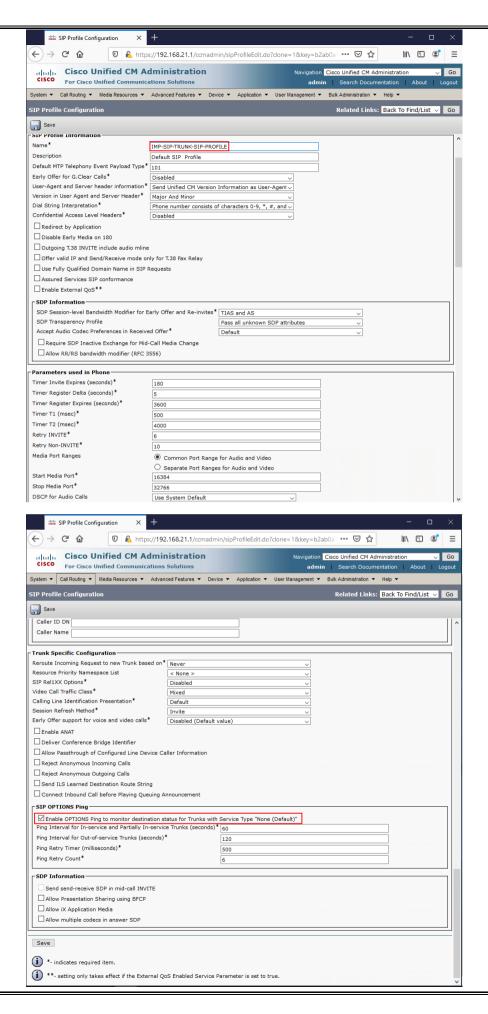


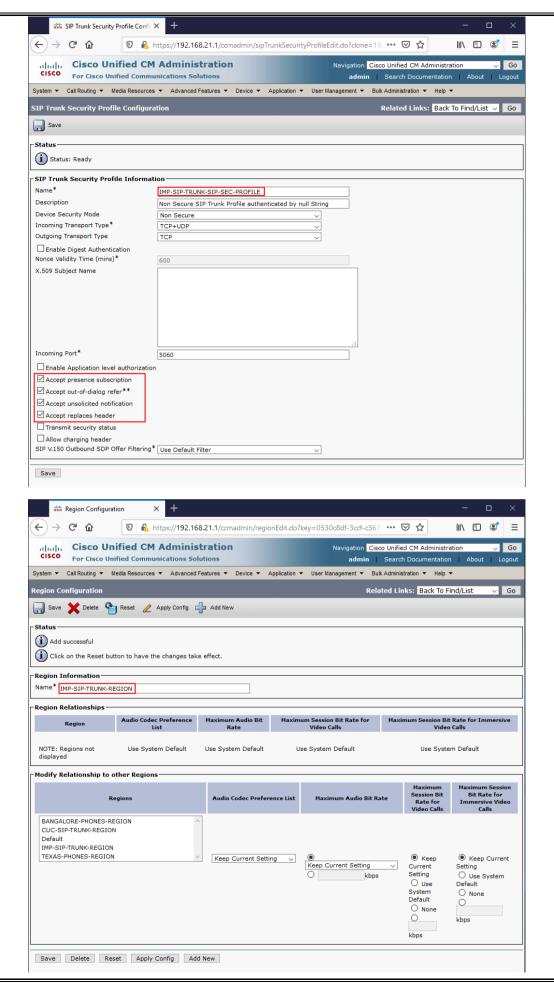
- There is no difference in IMP SUB Installation, it is exactly similar to IMP PUB, the whole cluster configuration is managed by CUCM PUB but the DB Replication is handled by IMP PUB
- The initially installed server becomes IMP PUB

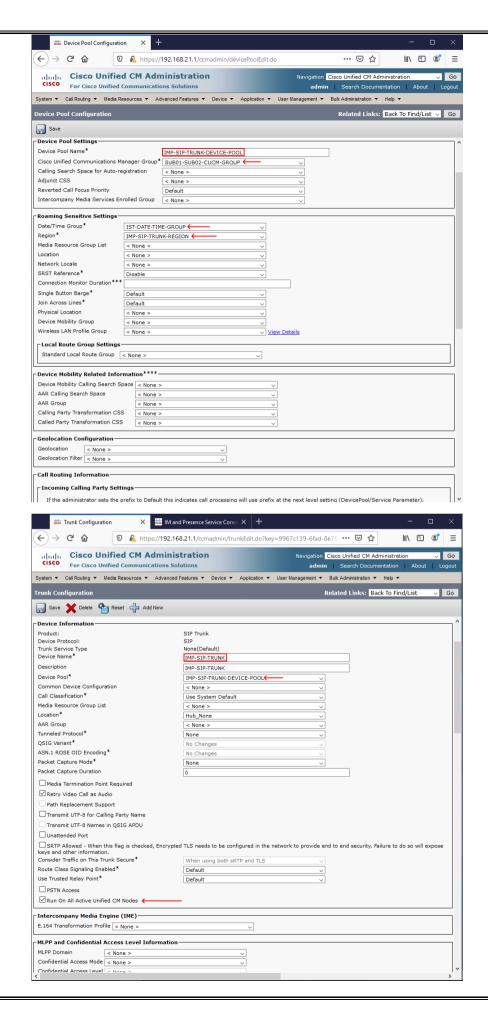
[Lab] IMP Integration with CUCM

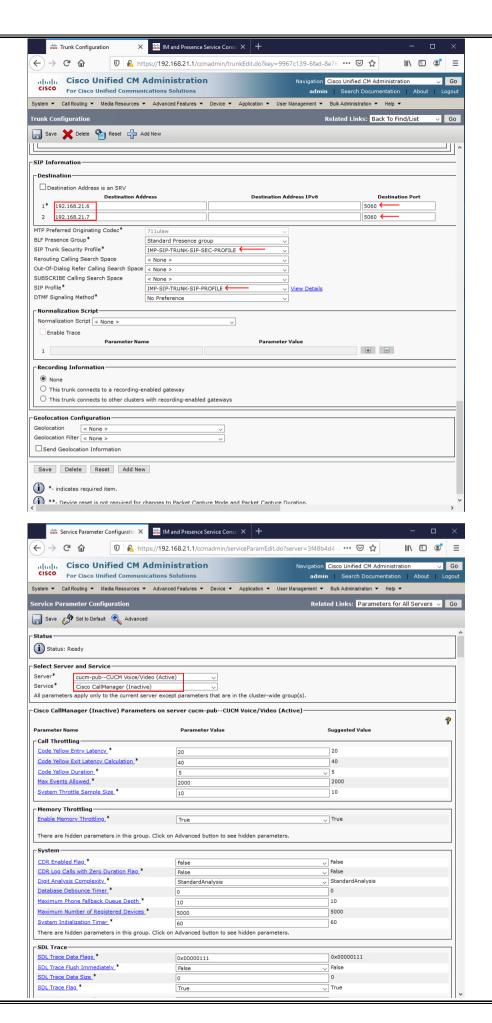


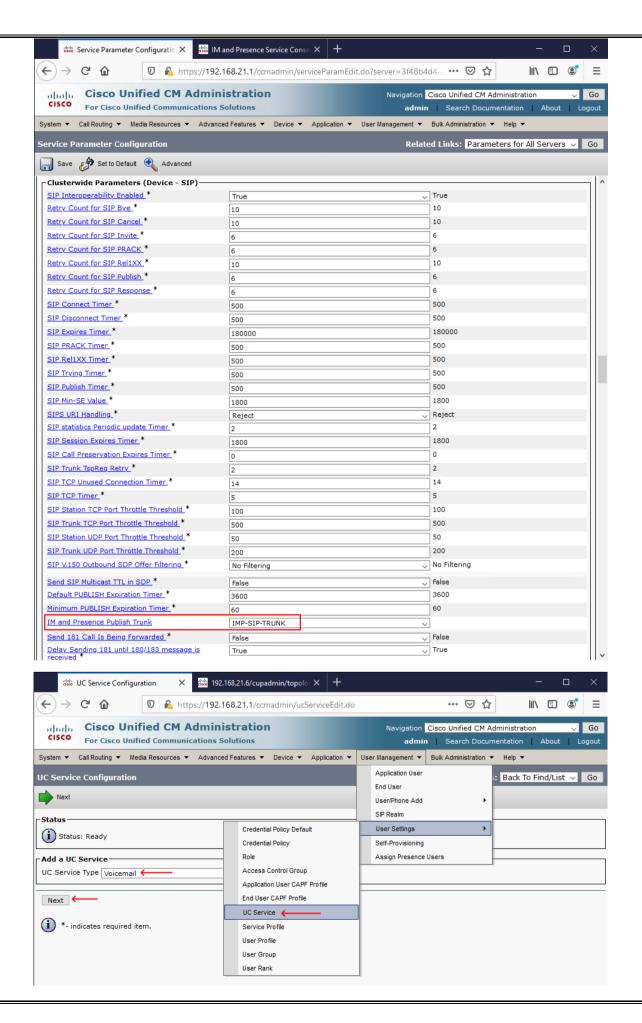


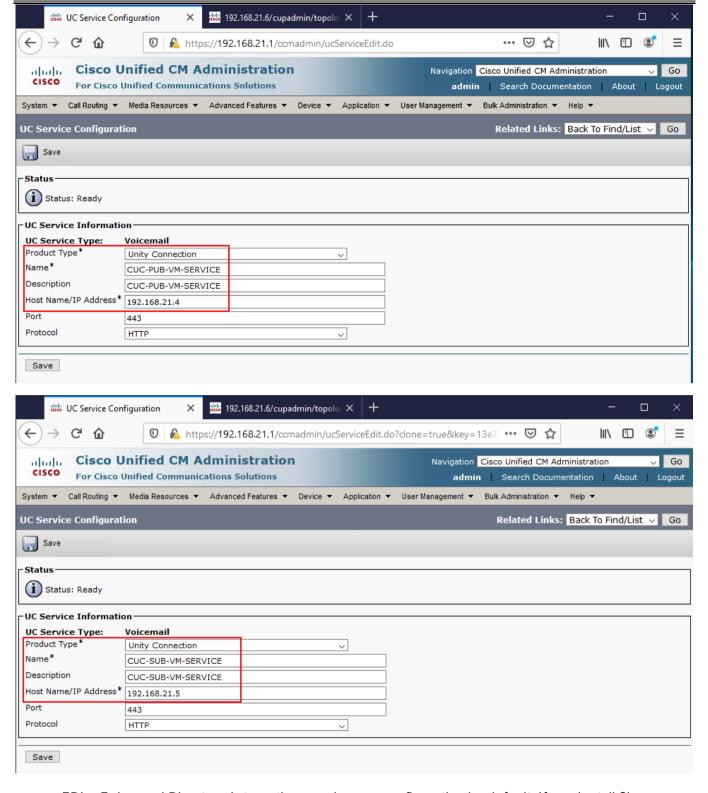




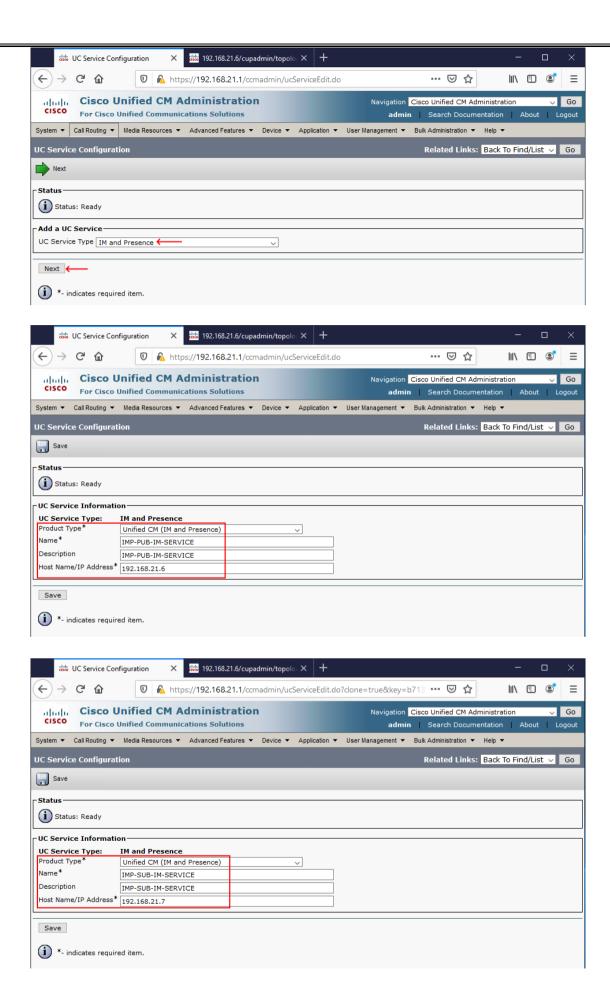


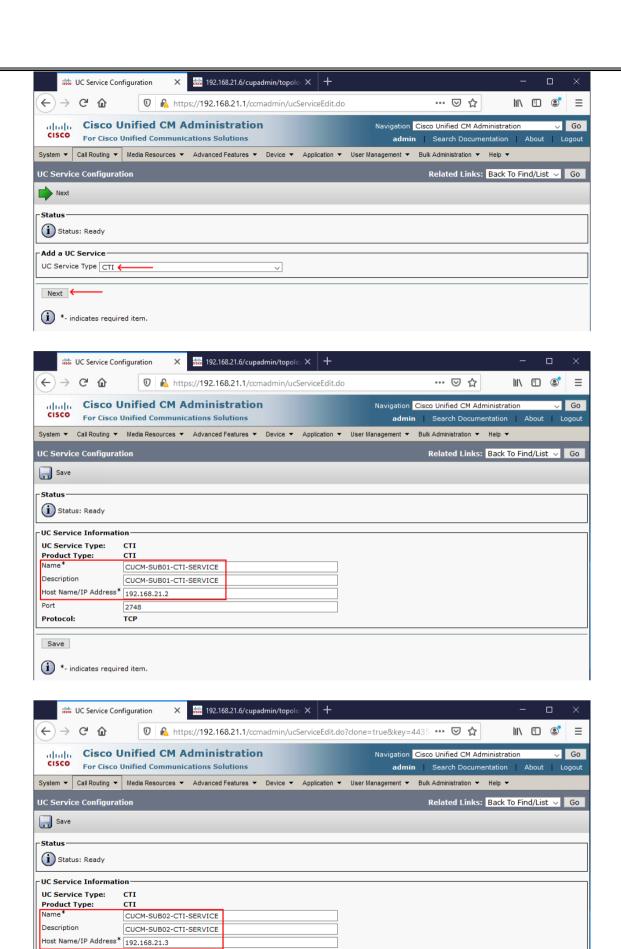






- EDI Enhanced Directory Integration requires no configuration by default. If you install Cisco
 Jabber for Windows on a workstation that is registered to an Active Directory domain, Cisco
 Jabber for Windows automatically discovers the directory service and connects to a Global Catalog
 in the domain.
- UDS User Data Service is an interface in Cisco Unified Communications Manager that makes contact information available to Cisco Jabber for Windows.





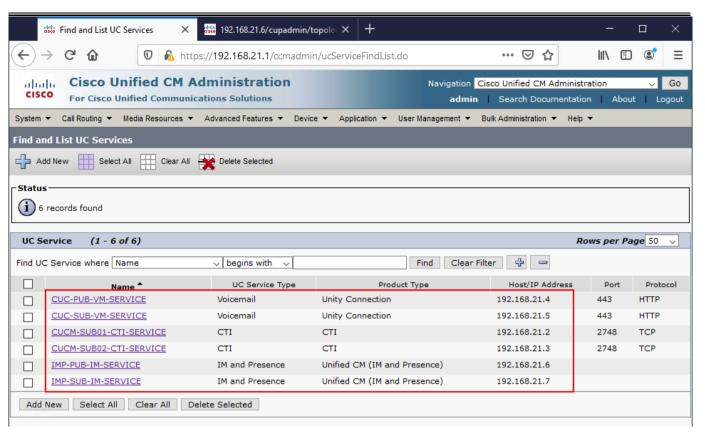
2748

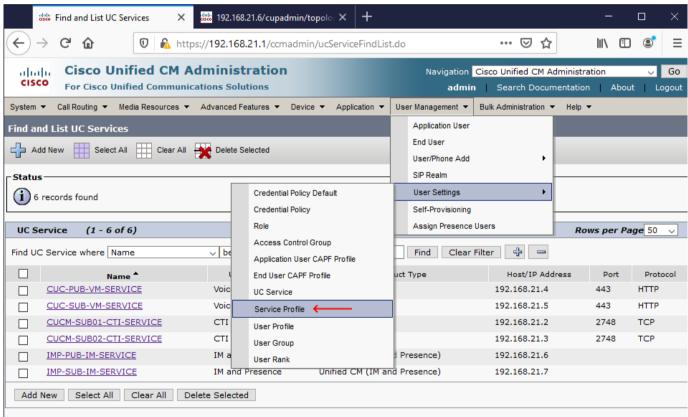
TCP

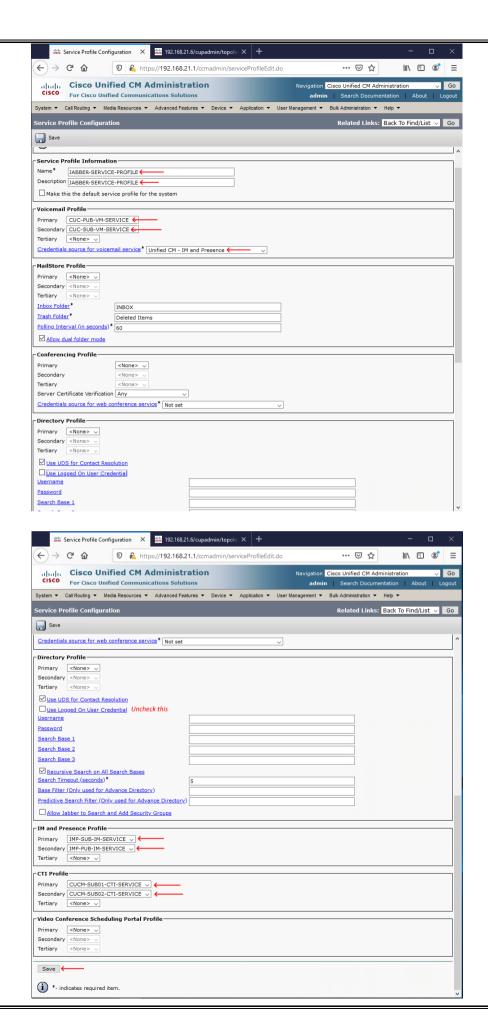
Protocol:

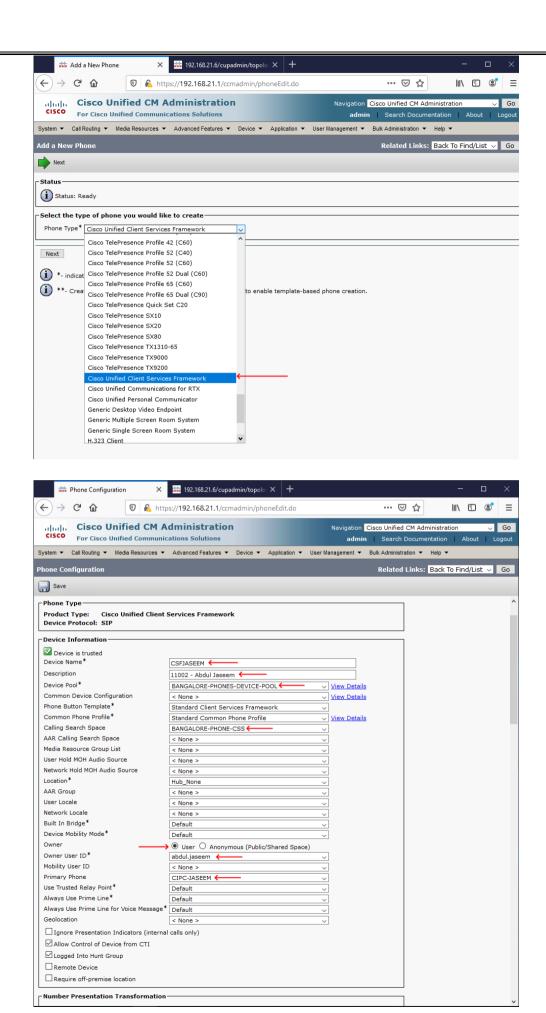
Save

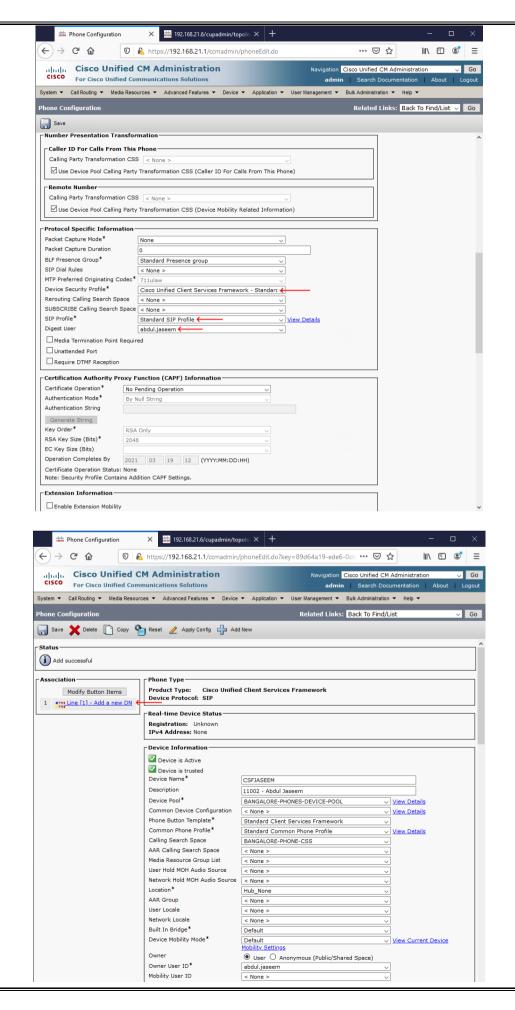
*- indicates required item.

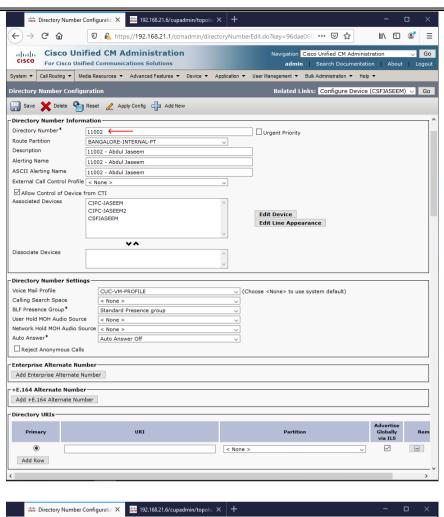


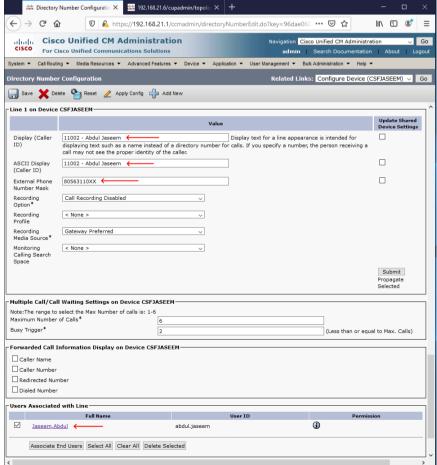


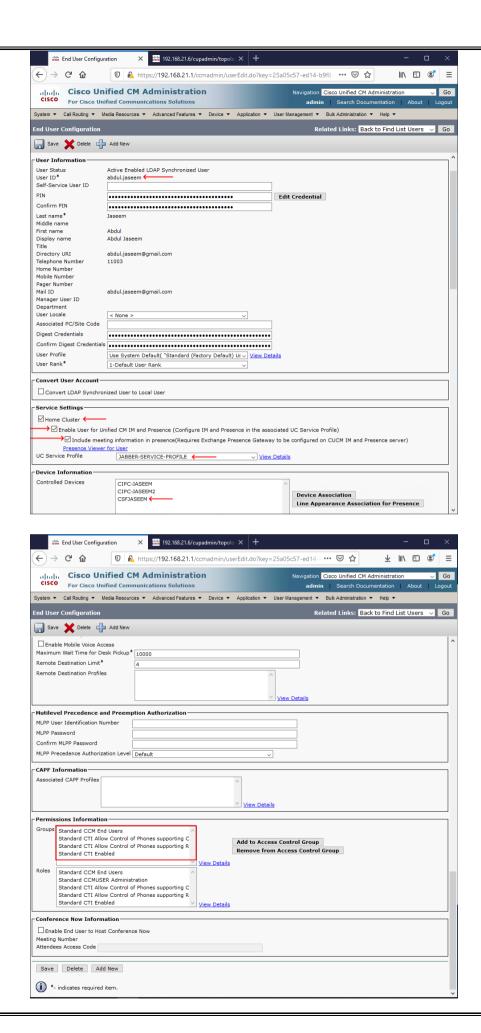


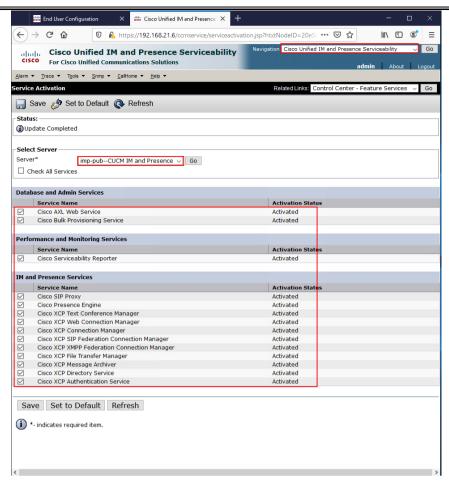


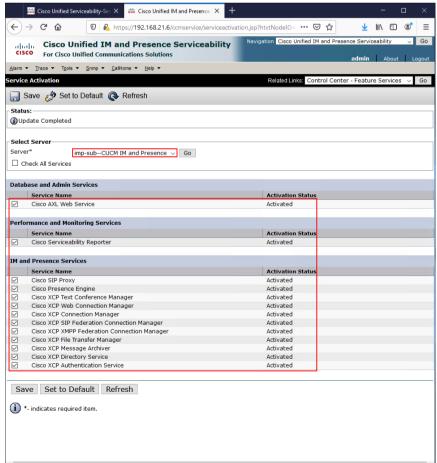


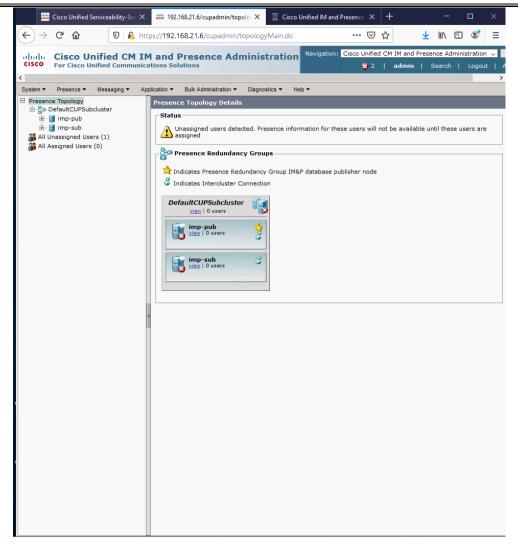


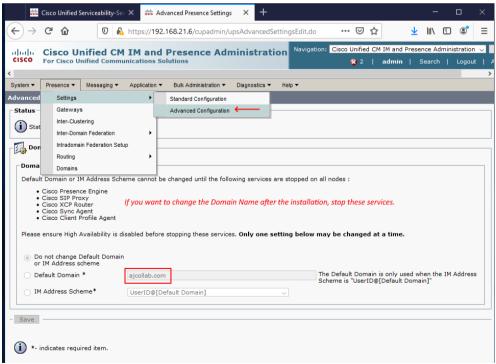


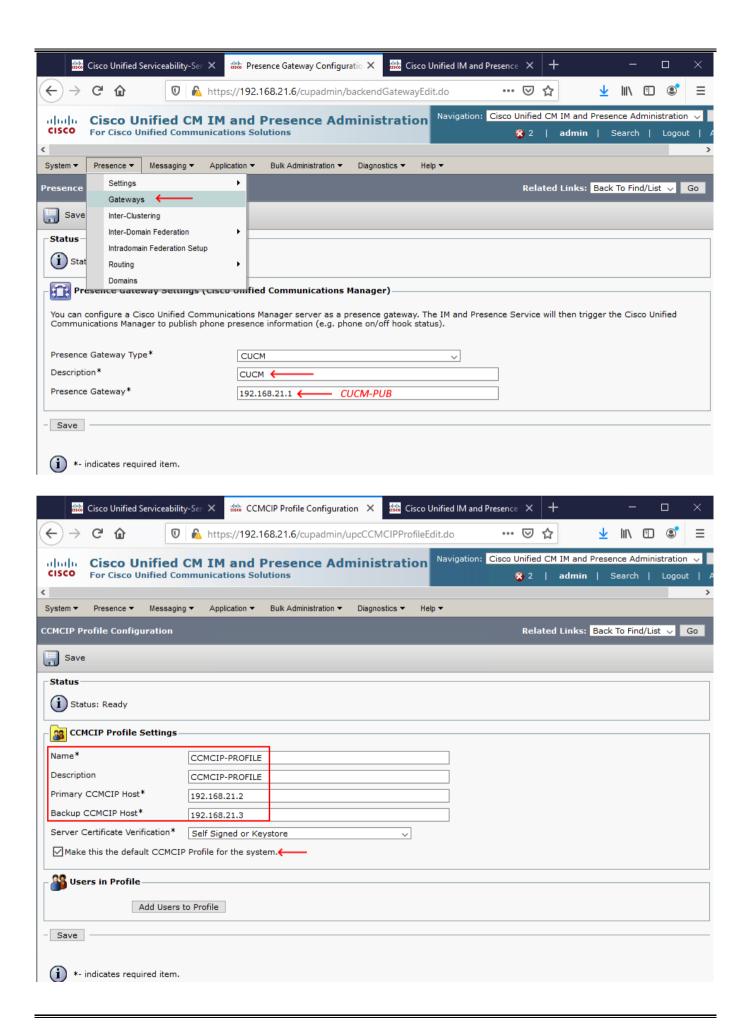


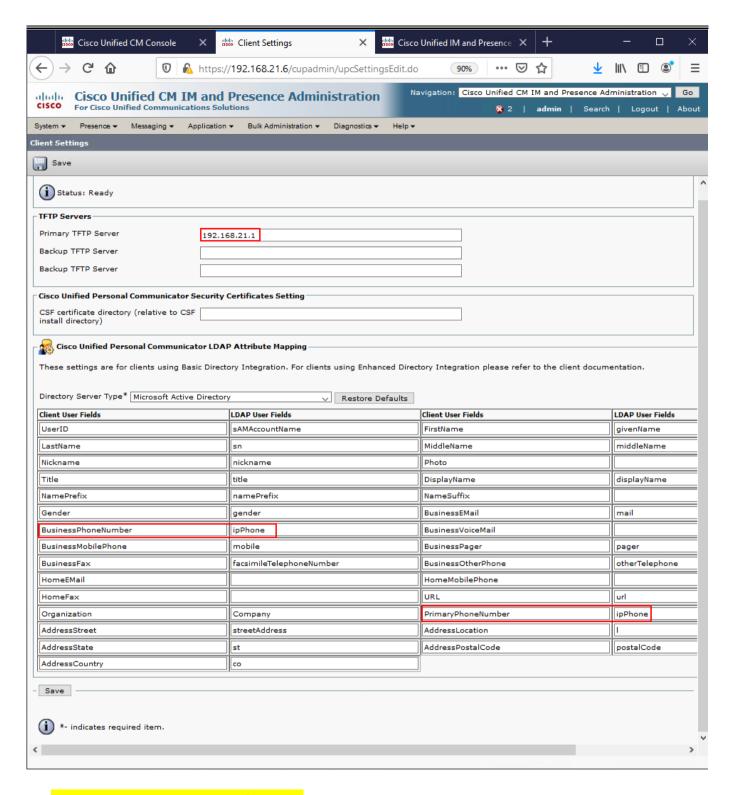




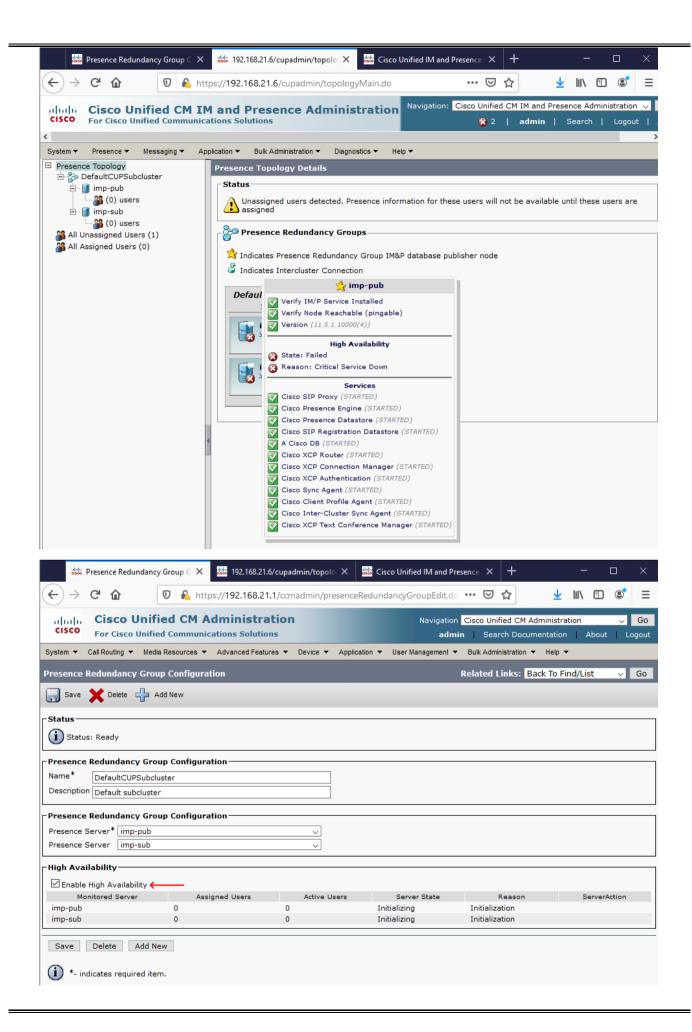


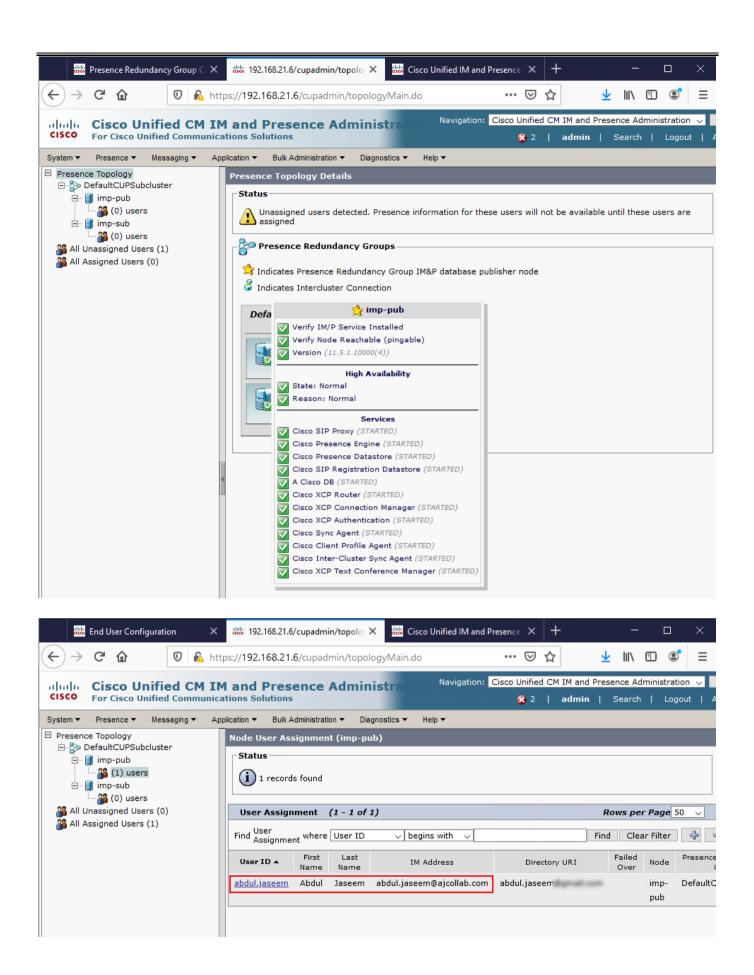


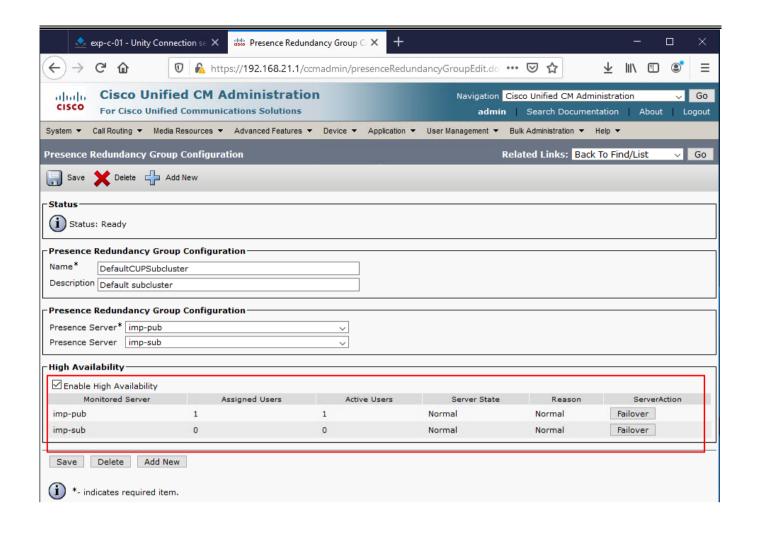


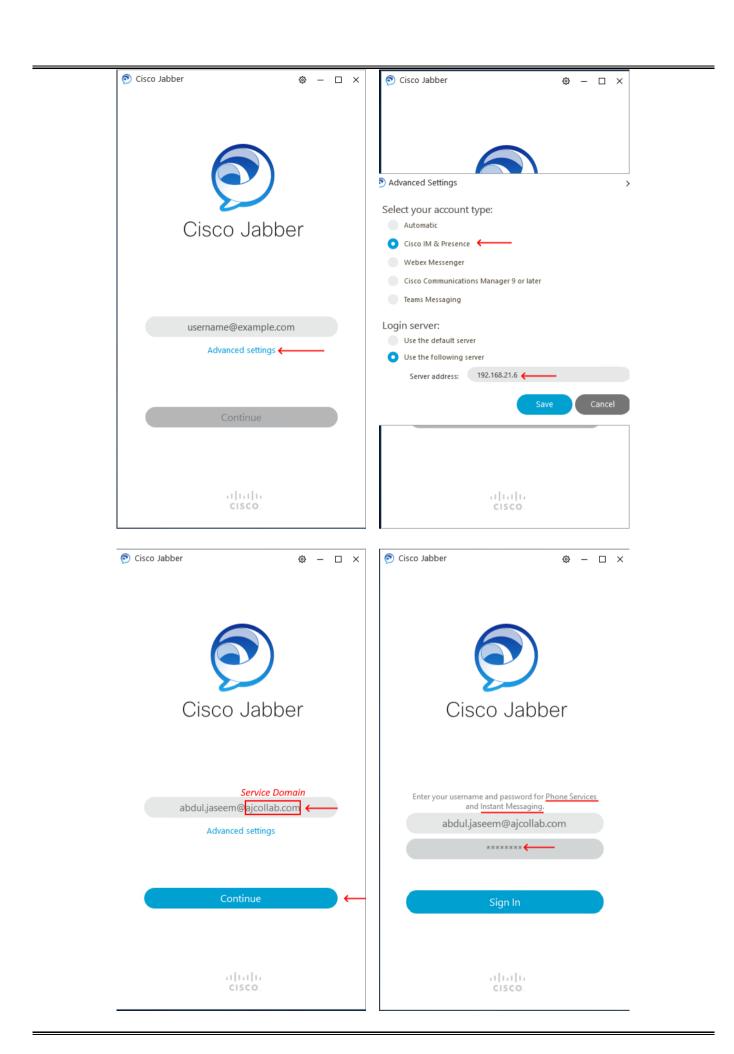


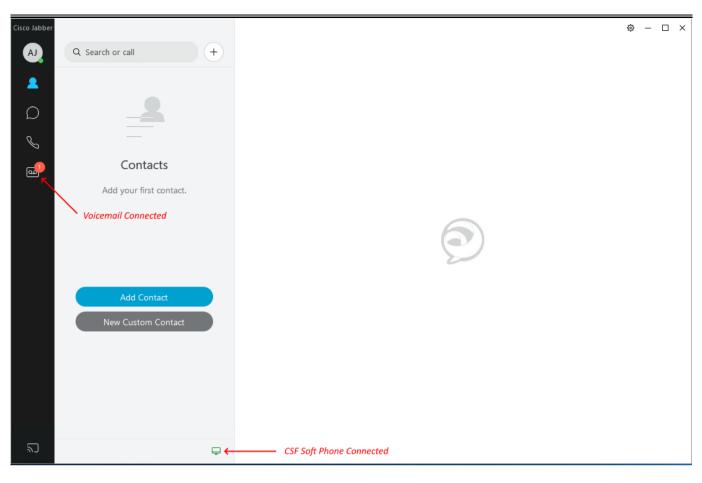
Restart IMP PUB and IMP SUB now

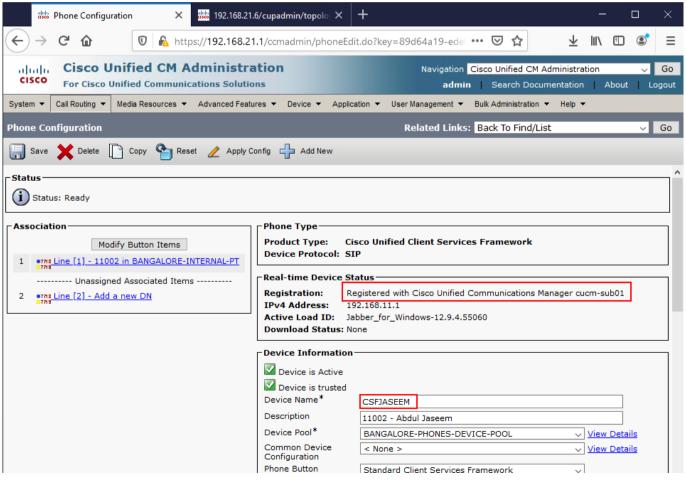






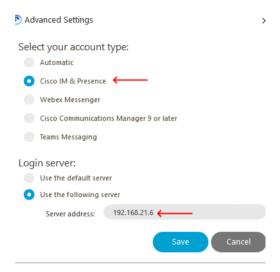




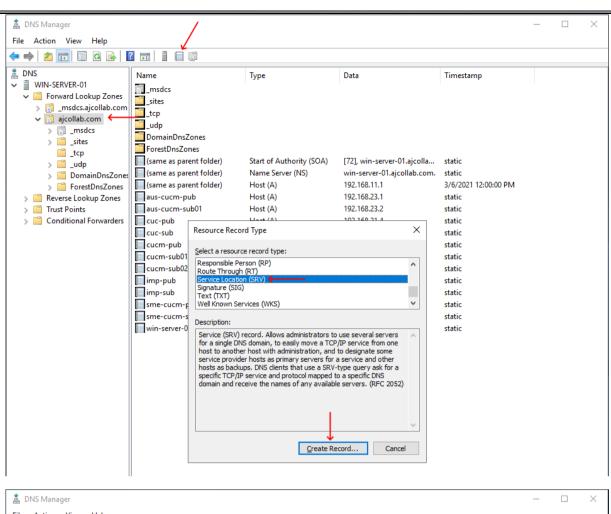


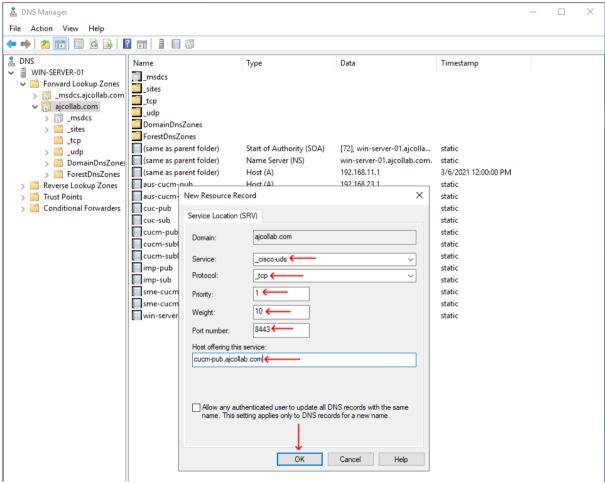
[Lab] DNS SRV Records for Jabber On-premise

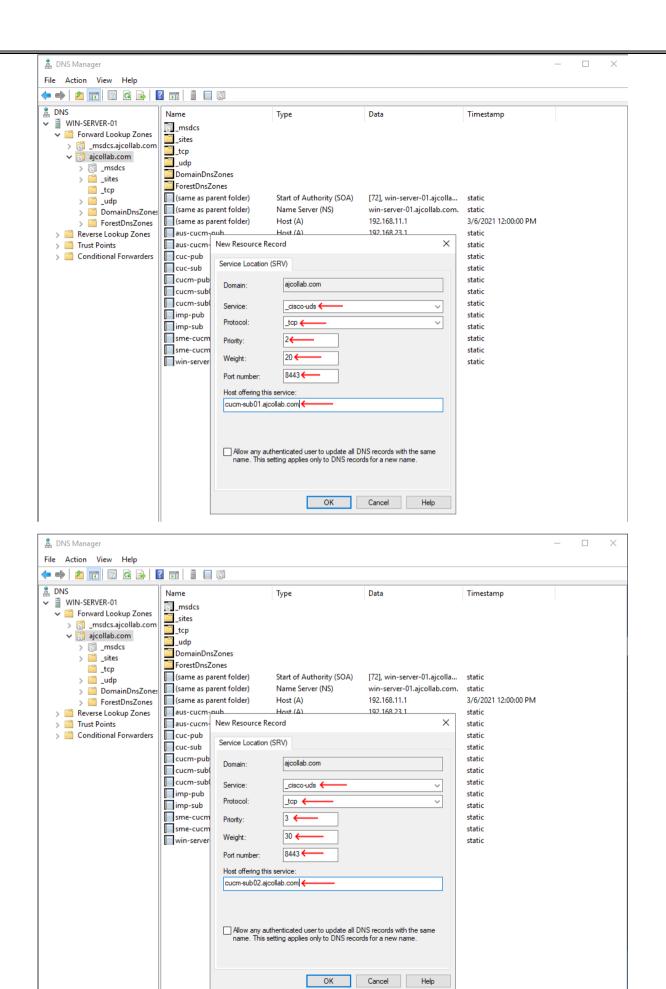
- As end user perspective, entering the IMP Server IP or FQDN on Jabber Advanced Settings not practical always
- We use DNS SRV (Service) Records to get rid of from the issue. SRVs are used for service discovery

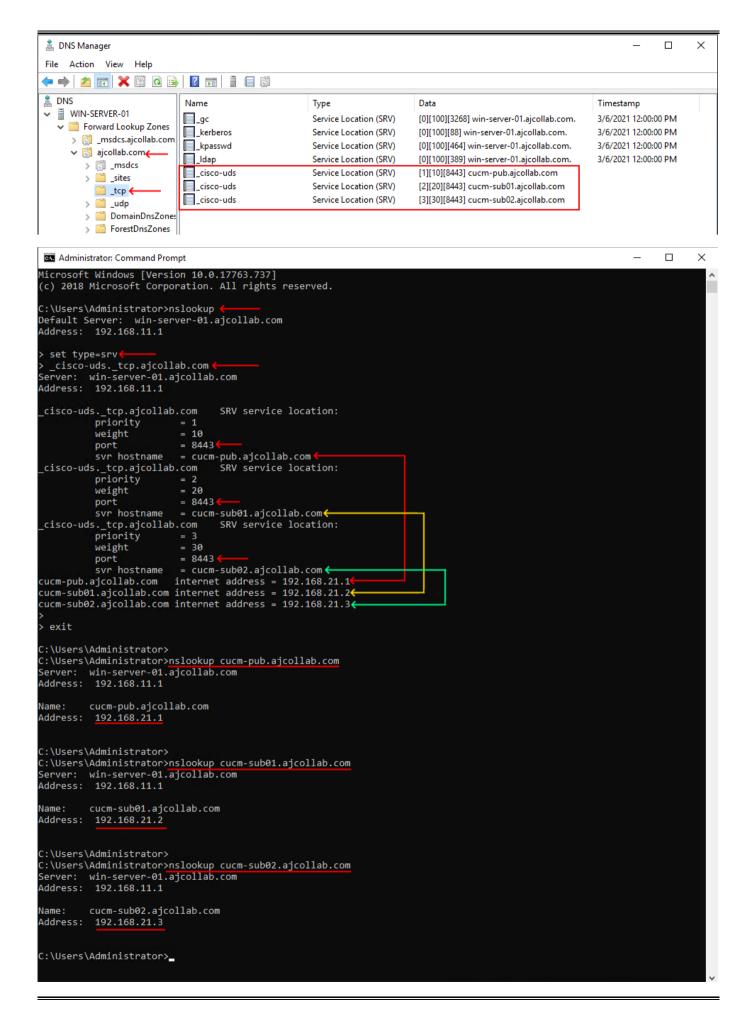


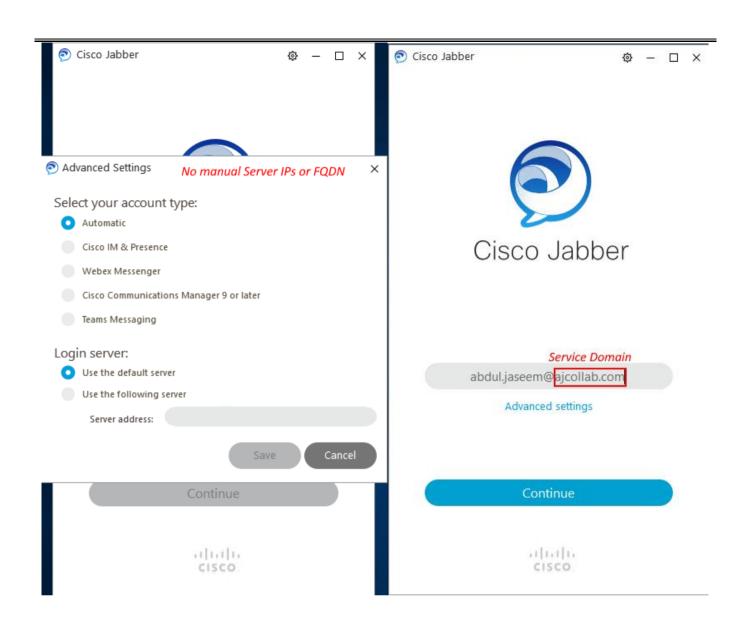
- When we enter the service domain (@ajcollab.com), Jabber can perform a DNS SRV check to identify the IP and CUCM server IP Address from its DNS Server
- If Jabber is inside corporate network, the DNS returns the private IP or IMP and CUCM servers
- If Jabber is outside the corporate network (on Internet), DNS returns the public IP of Expressway E server (This solution is called MRA, we will discuss that on the next chapter)
- Now let us configure DNS SRV records for the internal login so that users do not want to enter the
 IP manually while logging in

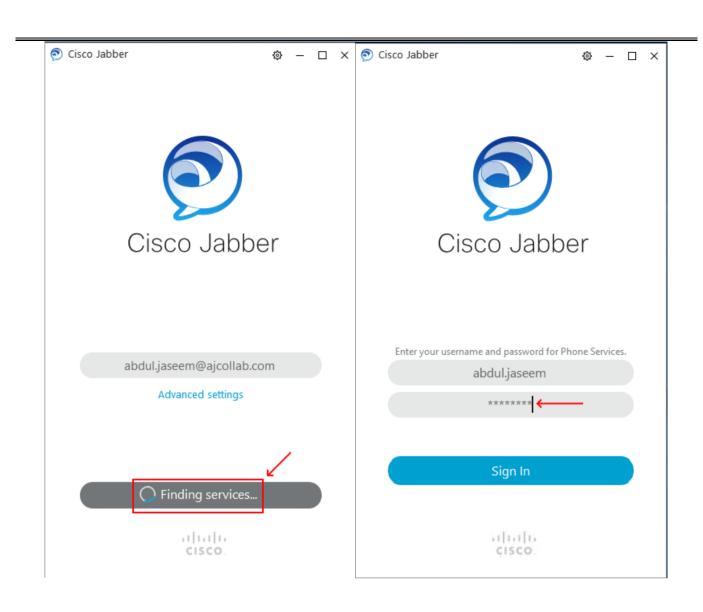


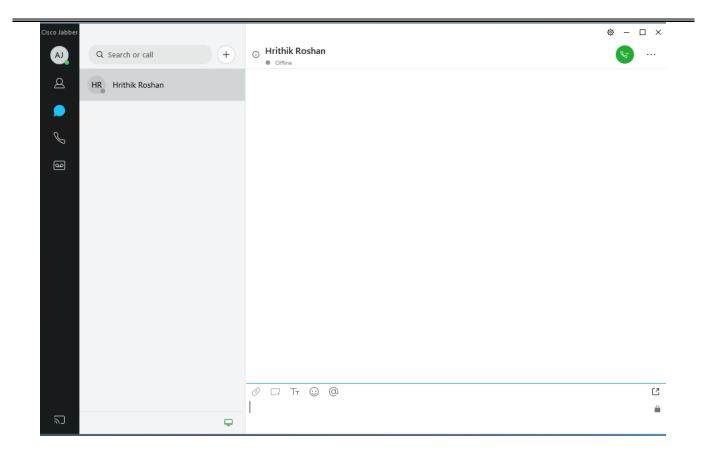










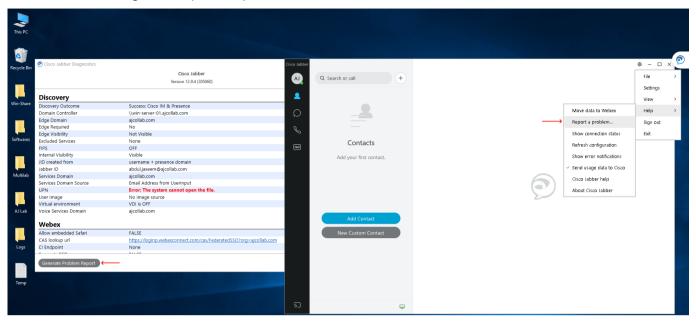


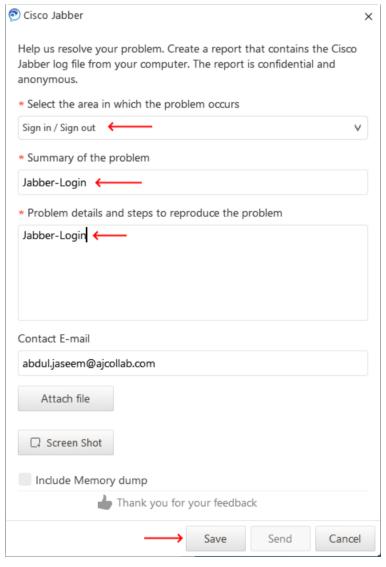
On-Prem Cisco Jabber Diagnostics & Problem Report (PRT)

- Hit "Ctrl + Shift + D" after launching Jabber
- This will give us the complete Service Discovery output

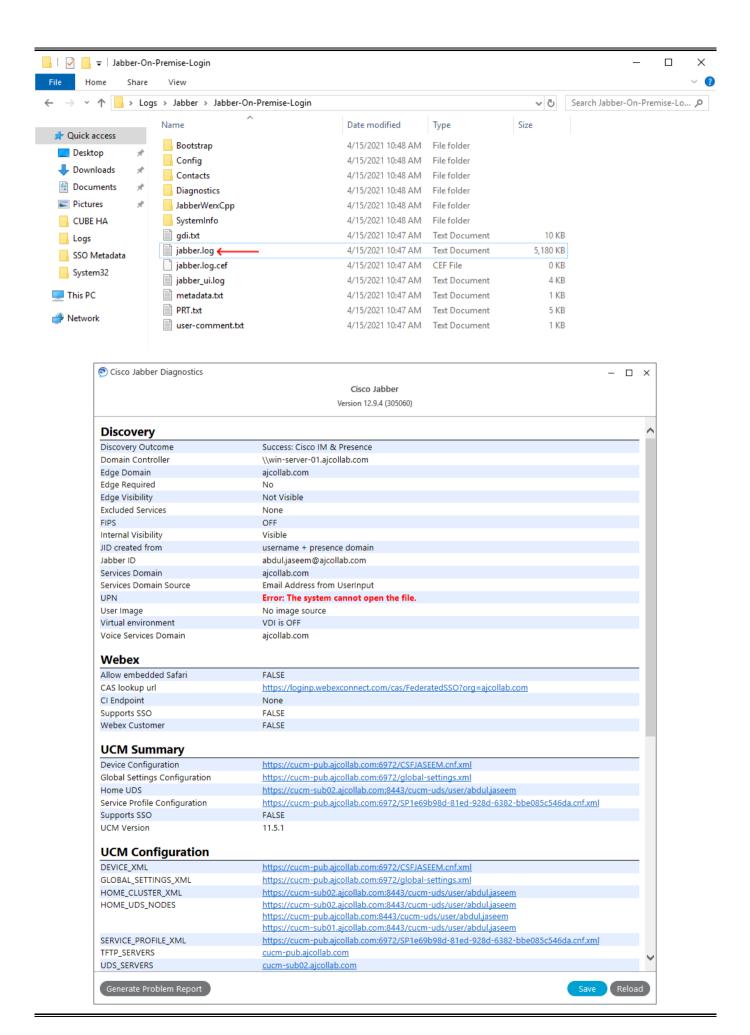
🔊 Cisco Jabber Diagnostics		- 🗆
	Cisco Jabber	
	Version 12.9.4 (305060)	
Discovery		
Discovery Outcome	Success: Cisco IM & Presence	
Domain Controller	\\win-server-01.ajcollab.com	
Edge Domain	ajcollab.com	
Edge Required	No	
Excluded Services	None	
FIPS	OFF	
Internal Visibility	Visible	
JID created from	username + presence domain	
Jabber ID	abdul.jaseem@ajcollab.com	
Services Domain	ajcollab.com	
Services Domain Source	Email Address from UserInput	
UPN	Error: The system cannot open the file.	
User Image	No image source	
Voice Services Domain	ajcollab.com	
UCM Summary		
Device Configuration	https://cucm-pub:6972/CSFJASEEM.cnf.xml	
Global Settings Configuration		
	nttps://cucm-pup:o9/z/giopai-settings.xmi	
	https://cucm-pub:6972/global-settings.xml https://cucm-sub01.aicollab.com:8443/cucm-uds/user/abdul.iaseem	
Home UDS	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration		
Home UDS	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml	
Home UDS Service Profile Configuration Supports SSO UCM Version	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML HOME_UDS_NODES	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration Supports SSO	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub02.ajcollab.com:8443/cucm-uds/user/abdul.jaseem	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML HOME_UDS_NODES SERVICE_PROFILE_XML TFTP_SERVERS	https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub02.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML HOME_UDS_NODES SERVICE_PROFILE_XML	https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml cucm-pub cucm-sub01.ajcollab.com cucm-pub.ajcollab.com	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML HOME_UDS_NODES SERVICE_PROFILE_XML TFTP_SERVERS	https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub02.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml cucm-pub cucm-sub01.ajcollab.com	
Home UDS Service Profile Configuration Supports SSO UCM Version UCM Configuration DEVICE_XML GLOBAL_SETTINGS_XML HOME_CLUSTER_XML HOME_UDS_NODES SERVICE_PROFILE_XML TFTP_SERVERS	https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml FALSE 11.5.1 https://cucm-pub:6972/CSFJASEEM.cnf.xml https://cucm-pub:6972/global-settings.xml https://cucm-pub:6972/global-settings.xml https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-sub01.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub.ajcollab.com:8443/cucm-uds/user/abdul.jaseem https://cucm-pub:6972/SP1e69b98d-81ed-928d-6382-bbe085c546da.cnf.xml cucm-pub cucm-sub01.ajcollab.com cucm-pub.ajcollab.com	

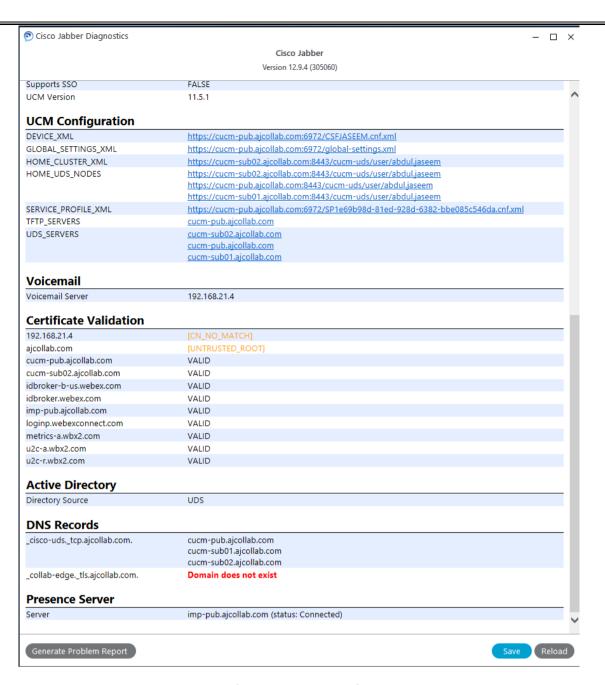
- Now let's capture a complete on-premise Jabber registration events in the form of PRT
- Go to Settings >> Help >> Report a Problem





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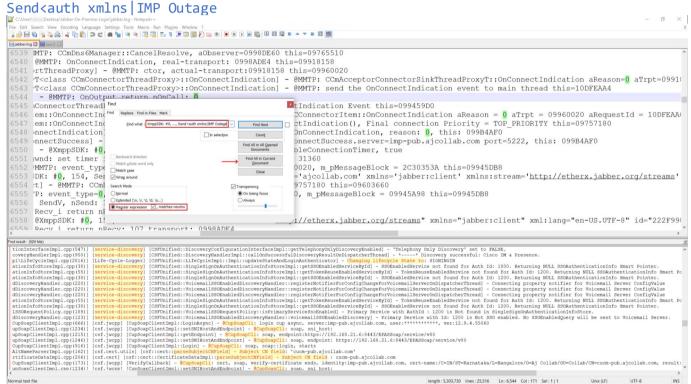


• You can download the sample PRT from here: Jabber On-Premise Login PRT

Jabber Login Flow - IM Service (XMPP)

- Service Discovery
- Configuration Request
- SOAP Login to IMP
- XMPP Login

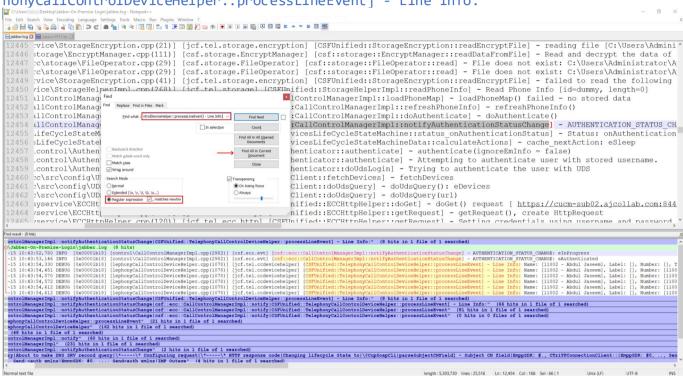
service-discovery|About to make DNS SRV record query|*----* Configuring
request|*----* HTTP response code|Changing lifecycle State
to|\@CupSoapCli|parseSubjectCNField] - Subject CN field|XmppSDK: #.,
CTriTPConnectionClient::|XmppSDK: #0, ..., Send:<auth xmlns|XmppSDK: #0, ...,</pre>



Jabber Login Flow - Phone Service

- Authenticate with CUCM
- Register the Soft Phone

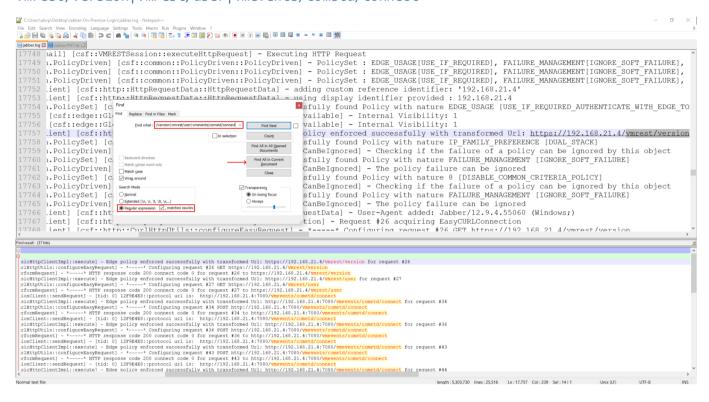
csf::ecc::CallControlManagerImpl::notifyAuthenticationStatusChange|CSFUnified::Telep honyCallControlDeviceHelper::processLineEvent] - Line Info:



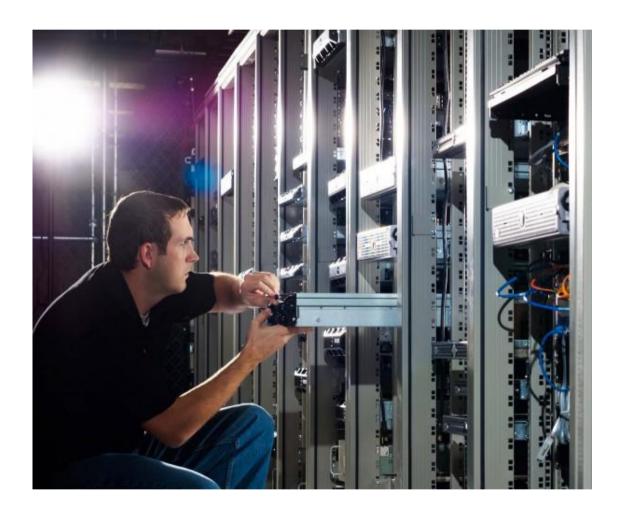
Jabber Login Flow - Voicemail (vmrest API)

Authenticate CUC via VMREST API

vmrest/version|vmrest/user|vmevents/cometd/connect



Chapter 1 Module 5 - Advanced Call Routing Cisco On-Premise Collaboration Solution CUBE, SME, Digit Manipulation & Media Resources

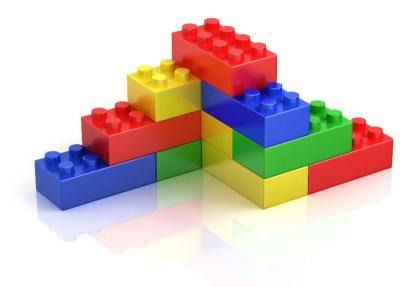


Session Initiation Protocol - SIP



- Open standard communications protocol for signaling and controlling multimedia communication sessions, it is an ideal VoIP protocol for interconnecting different VoIP system and networks
- Uses 5060 port number and Secure SIP (SIPS) uses 5061 by default. Goal of SIP is creating, modifying & terminating sessions
- Internet Engineering Talk Force (IETF) developed SIP as an alternative to H.323
- SIP is a peer-to-peer protocol where internet end points (User Agents) initiate sessions, similar to
 H.323
- SIP messages resemble with HTTP messages
- SIP uses SDP (Session Description Protocol) to exchange capabilities (Media IP, Port, Codec, DTMF Relay, etc.)
- Internet Telephony Service Provider (ITSP) uses SIP as their standard

Components of SIP



- User Agents (UA): The SIP message originator is termed as User Agent Client (UAC) and the server who respond to the SIP message termed as User Agent Server (UAS)
- Registrar Server: Receive SIP Registration request from SIP endpoints and register the endpoint to a database. CUCM acts as a Registrar Server during phone registration
- Proxy Server: Initial point of contact for user agents. It receives SIP request from UAC and forward
 it on behalf of the client to the next SIP server in the network. CUCM acts as a Proxy Server when
 routing the call from one SIP device to another SIP device or SIP Trunk
- Redirect Server: Finds a location of an endpoint. It provides next hope information to UAC. CUCM acts as a Redirect Server during Call Forward scenarios
- Location Server: Keeps the database of all User agents. It implements the mechanism to resolve the address. CUCM itself a Location Server since it stores all SIP endpoint details
- Presence Server: Provides availability (Presence information) of User Agents. IM & Presence
 Server is an example of this

SIP Request or Methods

- SIP Request or Methods are the SIP messages that takes an action
- Let's take a look at different SIP Requests that are common in CUCM
- REGISTER: Endpoint sends REGISTER request to the SIP Registrar Server. The SIP server provides
 a challenge to endpoint. User enters her/his valid user ID and password. The SIP server validates
 the user's credentials. It registers the user in its contact database and returns a response (200 OK)

```
REGISTER sip: cucm-sub01.ajcollab.com SIP/2.0 >> This is Request URI
Via: SIP/2.0/TCP 192.168.128.4:50136; branch=z9hG4bK687b7aa9 >> Indicates the path taken
From: <sip:11001@cucm-sub01.ajcollab.com>;tag=501cb00c71d500045362a3f9-3d42ea49 >> Contact
information of originator
To: sip:11001@cucm-sub01.ajcollab.com >> Indicates Recipient of the request
Call-ID: 501cb00c-71d50003-291a7902-22598ffe@192.168.128.4 >> unique identifier for SIP call
Max-Forwards: 70 >> Indicates how many proxies are supported
Date: Thu, 15 Apr 2021 21:00:30 GMT
CSeq: 101 REGISTER >> Number of Request type, response will have the same value
User-Agent: Cisco-CP8865/11.7.1 >> Details about the user agent who send the message
Contact: <sip:2f137c23-734b-444f-b8b3-
e5f7f3d17792@192.168.128.4:50136;transport=tcp>;+sip.instance="<urn:uuid:00000000-0000-0000-
0000-
501cb00c71d5>";+u.sip!devicename.ccm.cisco.com="SEP501CB00C71D5";+u.sip!model.ccm.cisco.com=
"36225"; video >> URI used to contact the sender. DN PKID
Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-cisco-
callinfo, X-cisco-serviceuri, X-cisco-escapecodes, X-cisco-service-control, X-cisco-srtp-
fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-8.5.1
Reason: SIP;cause=200;text="cisco-alarm:14 Name=SEP501CB00C71D5 ActiveLoad=sip8845_65.11-7-
1-17.loads InactiveLoad=sip8845_65.12-8-1-0001-455.loads Last=cm-closed-tcp" >> Used to
generate syslog alarm for last out of service
Expires: 3600 >> Lifespan of the registration, registration will timeout after 1 hour
Content-Type: multipart/mixed; boundary=uniqueBoundary >> identifies how the body is
Mime-Version: 1.0 >> Multipurpose Internet Mail Extension (MIME) version
Content-Length: 1337 >> Size of the message content in bytes
```

Note: **CSeq** or Command Sequence contains an integer and a method name. The CSeq number is incremented for each new request within a dialog

RSeq or Response Sequence Each provisional response is given a sequence number, carried in the RSeq header field in the response

INVITE: The INVITE method is used to establish media sessions between user agents. It is similar
to a Q.931 Setup message in ISDN. Responses to INVITEs are always acknowledged with the ACK

```
INVITE sip:x-cisco-serviceuri-abbrdial-1@192.168.21.2 SIP/2.0
Via: SIP/2.0/TCP 192.168.128.1:52552; branch=z9hG4bK3ac85a1b
From: "11002 - Abdul Jaseem" <sip: 11002@192.168.21.2>; tag=08cc6831d463009005a3afc3-1a06158a
To: <sip:x-cisco-serviceuri-abbrdial-1@192.168.21.2>
Call-ID: 08cc6831-d463007a-190deeb7-1977157c@192.168.128.1
Max-Forwards: 70
Date: Tue, 30 Mar 2021 18:43:19 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP9971/9.4.2
Contact: <sip:12534a53-2969-4d94-8a70-90f206a1f7da@192.168.128.1:52552;transport=tcp>;video
Expires: 180
Accept: application/sdp
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO
Remote-Party-ID: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;party=calling;id-type=subscriber;privacy=off;screen=yes >> Used to convey the calling telephone number and
source IP address. This gets displayed on the destination phone
Supported: replaces, join, sdp-anat, norefersub, resource-priority, extended-refer, X-cisco-
callinfo,X-cisco-serviceuri,X-cisco-escapecodes,X-cisco-service-control,X-cisco-srtp-
fallback, X-cisco-monrec, X-cisco-config, X-cisco-sis-7.0.0, X-cisco-xsi-8.0.1
Allow-Events: kpml,dialog >> Indicate which events or classes of events the notifier
supports. SIP KPML DTMF is supported here
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 785
Content-Type: <a href="mailto:application/sdp">application/sdp</a>
Content-Disposition: session; handling=optional
v=0 >> SDP Version 0
o=Cisco-SIPUA 16362 0 IN IP4 192.168.128.1 >> owner/creator and session identifier
s=SIP Call >> Session Name
m=audio 29938 RTP/AVP 102 9 124 0 8 116 18 101 >> Audio Port, Codec, DTMF Relay
c=IN IP4 192.168.128.1 >> Audio IP Address to establish
a=rtpmap:102 L16/16000 >> L16 Codec
a=rtpmap:9 G722/8000 >> G.722 Codec
a=rtpmap:124 ISAC/16000 >> ISAC Codec
a=rtpmap:0 PCMU/8000 >> G.711 ULaw Codec
a=rtpmap:8 PCMA/8000 >> G.711 alaw Codec
a=rtpmap:116 iLBC/8000 >> iLBC Codec
a=fmtp:116 mode=20 >> iLBC Sample size 20 msec
a=rtpmap:18 G729/8000 >> G.729 Codec
a=fmtp:18 annexb=no >> G.729 non annexb type
a=rtpmap:101 telephone-event/8000 >> In-band DTMF relay
a=fmtp:101 0-15
a=sendrecv >> Who receives the request, will send and receive audio
m=video 21814 RTP/AVP 126 97 >> Video Port, Codec, DTMF Relay
c=IN IP4 192.168.128.1 >> Video IP Address to establish
b=TIAS:1000000
a=rtpmap:126 H264/90000 >> H.264 Codec Mode 1
a=fmtp:126 profile-level-id=42801E;packetization-mode=1;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=rtpmap:97 H264/90000 >> H.264 Codec Mode 0
a=fmtp:97 profile-level-id=42801E;packetization-mode=0;level-asymmetry-allowed=1
a=imageattr:* recv [x=640,y=480,q=0.50]
a=recvonly >> Who receives the request, will receive video but don't send
```

ACK: The ACK method is used to acknowledge final responses to INVITE requests. Final responses
to all other requests are never acknowledged. Final responses are defined as 2XX, 3XX, 4XX, 5XX,
or 6XX class responses. The CSeq number is never incremented for an ACK

ACK sip:x-cisco-serviceuri-abbrdial-1@192.168.21.2:5060;transport=tcp SIP/2.0

Via: SIP/2.0/TCP 192.168.128.1:52552; branch=z9hG4bK1d3ef7e3

From: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;tag=08cc6831d463009005a3afc3-1a06158a

To: <sip:x-cisco-serviceuri-abbrdial-1@192.168.21.2>;tag=188~3f48b4d4-2f05-4b61-86ad-

47522208ed69-42836245

Call-ID: 08cc6831-d463007a-190deeb7-1977157c@192.168.128.1

Max-Forwards: 70

Date: Tue, 30 Mar 2021 18:43:28 GMT

CSeq: 101 ACK

User-Agent: Cisco-CP9971/9.4.2

Remote-Party-ID: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;party=calling;id-

type=subscriber;privacy=off;screen=yes

Content-Length: 0
Recv-Info: conference

Recv-Info: x-cisco-conference

CANCEL: The CANCEL method is used to terminate pending call attempts. When originator
disconnects the call before the call connects, we will see CANCEL instead of BYE message

CANCEL sip:2@192.168.21.2; user=phone SIP/2.0

Via: SIP/2.0/TCP 192.168.128.6:52570; branch=z9hG4bK2bbf3494

From: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;tag=84b517af4c4300296481aaad-69e78976

To: <sip:2@192.168.21.2>

Call-ID: 84b517af-4c430004-05451ecb-2eb41ce4@192.168.128.6

Max-Forwards: 70

Date: Sat, 17 Apr 2021 18:49:46 GMT

CSeq: 101 CANCEL

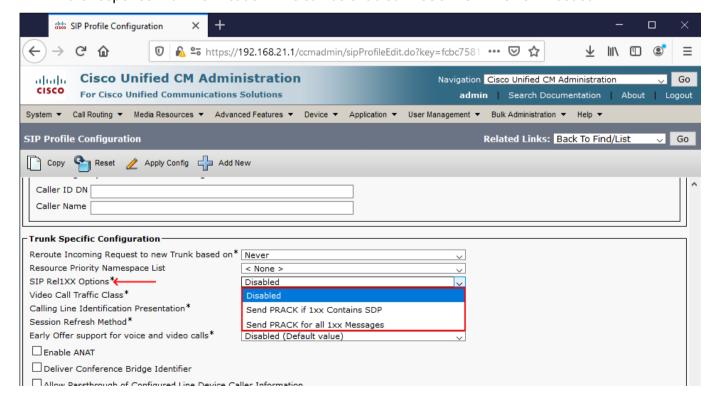
User-Agent: Cisco-CP9971/9.4.2

Content-Length: 0

• **BYE**: The BYE method is used to terminate an established media session. In ISDN, it is similar to a RELEASE message. A session is considered established if an INVITE has received a success class response (2XX) or an ACK has been sent. A BYE is sent only by user agents participating in the session. It is an end-to-end method, so responses are only generated by the other user agent

```
BYE sip:8044260389@192.168.31.3:5060; transport=tcp SIP/2.0
Via: SIP/2.0/TCP 192.168.21.2:5060; branch=z9hG4bK691ae67ef7
From: "11002 - Abdul Jaseem" <sip:8056311002@192.168.21.2>; tag=189~3f48b4d4-2f05-4b61-86ad-47522208ed69-42836246
To: <sip:8044260389@192.168.31.2>; tag=279F53-23D0
Date: Tue, 30 Mar 2021 18:43:24 GMT
Call-ID: cbbab000-6317149-67-215a8c0@192.168.21.2
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
P-Asserted-Identity: "11002 - Abdul Jaseem" <sip:8056311002@192.168.21.2> >> Verified identity of the user sending a SIP message
CSeq: 103 BYE
Reason: Q.850; cause=16 >> Disconnect reason (16 is normal call clearance)
Session-ID: 7c8b24ebbb0140d0ab8cdeba452aa188; remote=e60906e41ff12aa229378d9b697ab189
Content-Length: 0
```

PRACK: It is an ACK for 1XX series informational message. The informational messages are not a
final response for INVITE; hence they acknowledge by PRACK. 2XX, 3XX, 4XX, 5XX, 6XX are
acknowledged by actual ACK method. The PRACK echoes the number in the RSeq and the CSeq of
the response in a RAck header. This can be enabled in CUCM SIP Profile if needed



• **OPTIONS**: The OPTIONS method is used discover availability of user agent. The response to the request lists the capabilities of the user agent or server. It works as a Keepalive mechanism

OPTIONS sip:192.168.21.5:5060 SIP/2.0

Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK78c3275c37e

From: <sip:192.168.21.2>;tag=2074871202

To: <sip: 192.168.21.5>

Date: Sat, 17 Apr 2021 18:30:00 GMT

Call-ID: e9bb4980-7b12928-77a-215a8c0@192.168.21.2

User-Agent: Cisco-CUCM11.5

CSeq: 101 OPTIONS

Contact: <sip:192.168.21.2:5060;transport=tcp>

Max-Forwards: 0
Content-Length: 0

Session Refresh: If an active session is more than the value specified in the 'Session-Expires'
header, then re-Invite or Update used to refresh the session

SIP/2.0 200 OK

Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK673ca1eae4

From: "11002 - Abdul Jaseem" <sip:8056311002@192.168.21.2>;tag=189~3f48b4d4-2f05-4b61-86ad-

47522208ed69-42836246

To: <sip:8044260389@192.168.31.2>;tag=279F53-23D0

Date: Tue, 30 Mar 2021 20:52:44 GMT

Call-ID: cbbab000-6317149-67-215a8c0@192.168.21.2

CSeq: 102 INVITE

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,

REGISTER

Allow-Events: telephone-event

Remote-Party-ID: "Bangalore PSTN - 8044260389"

<sip:8044260389@192.168.31.3>;party=called;screen=yes;privacy=off

Contact: <sip:8044260389@192.168.31.3:5060;transport=tcp>

Supported: replaces Supported: sdp-anat

Server: Cisco-SIPGateway/IOS-15.5.2.S

Session-Expires: 1800; refresher=uas >> Refresh must come from UAS after 1800/2=900 Sec = 15

Min

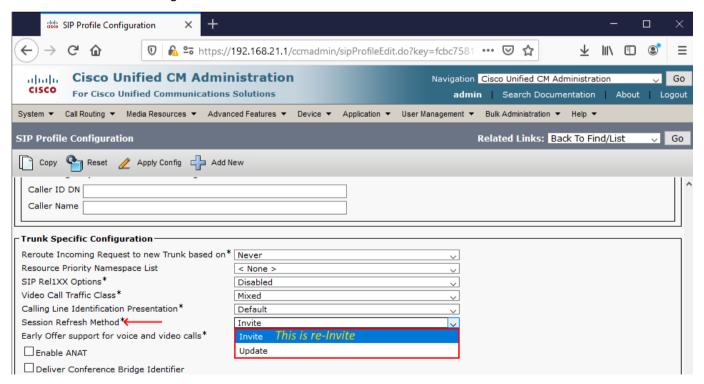
Require: timer Supported: timer

Content-Type: application/sdp

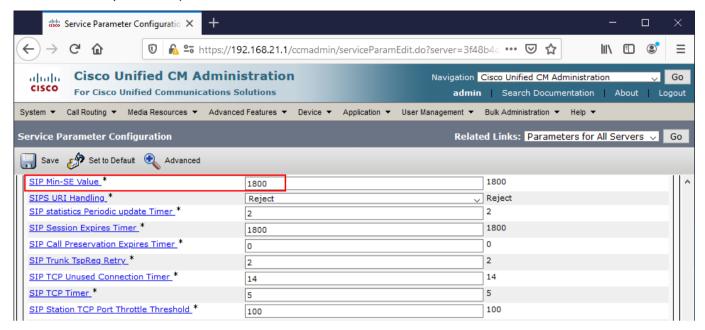
Content-Disposition: session; handling=required

Content-Length: 318

This can be configured in CUCM SIP Profile



- The session expire time can be customized in CUCM Service Parameter
- The request (whether re-Invite or Update is decided by SIP Profile and the expire timer decided by Service parameter)



When the call goes out, the session-expire value again depended on the next hop proxy (UAS) or
 ITSP

SIP Responses

- INFORMATIONAL (1XX): The informational class of responses 1XX are end-to-end responses and used to indicate call progress. Informational responses
 - 100 TRYING: This special case response is only a hop-by-hop request. It is never forwarded
 - 180 Ringing: This response is used to indicate that the INVITE has been received by the user agent and that alerting is taken place
 - 182 Call Queued: This response is used to indicate that the INVITE has been received and will be processed in a queue.
 - The 183 Session Progress: Indicates that information about the progress of the session. 183 is an end-to-end response. A typical use of this response is to allow a UAC to hear ring tone, busy tone, or a recorded announcement. 183 session progress can have SDP and PRACK used to acknowledge it. A one-way media connection is established from the calling party's telephone switch to the called party's telephone switch in the PSTN prior to the call being answered
- SUCCESS (2XX): Success class responses indicate that the request has succeeded or has been accepted
 - 200 OK: Used to accept a session invitation, sometimes it has a body with media properties
 (SDP) of the UAS (called party)
 - 202 Accepted: Response indicates that the UAS has received and understood the request, but that the request may not have been authorized or processed by the server
- **REDIRECT (3XX)**: Server has returned possible locations. The client should retry request at another server. Generally sent by a SIP server acting as a redirect server in response to an INVITE.
 - 300 Multiple Choices: Multiple Contact header fields, which indicate that the location service has returned multiple possible locations. They should be tried in the order in which they were listed in the response
 - 301 Moved Permanently: This redirection response contains a Contact header field with the new permanent URI of the called party. The address can be saved and used in future INVITE requests
 - 302 Moved Temporarily: This redirection response contains a URI that is currently valid but that is not permanent. During call forward situation, we will see 302
 - o 305 Use Proxy: This redirection response contains a URI that points to a proxy server
 - 380 Alternative Service: This response returns a URI that indicates the type of service that the called party would like. An example might be a redirect to a voicemail server

- CLIENT ERROR (4XX): The request has failed due to an error by the client. The client may retry the request
 - 400 Bad Request: This response indicates that the request was not understood by the server
 - 401 Unauthorized: This response indicates that the request requires the user to perform authentication & the authentication may fail.
 - 402 Payment Required: This response is a placeholder for future definition in the SIP protocol. It could be used to negotiate call completion charges
 - 403 Forbidden: This response is used to deny a request
 - 404 Not Found: This response indicates that the user identified by the Request-URI cannot be located by the server
 - 405 Method Not Allowed: This response indicates that the server or user agent has received and understood a request but is not willing to fulfill the request. An example might be a REGISTER request sent to a user agent.
 - 406 Not Acceptable: This response indicates that the request cannot be processed due to a requirement in the request message.
 - 407 Proxy Authentication Required: This request sent by a proxy indicates that the UAC must first authenticate itself with the proxy before the request can be processed
 - 408 Request Timeout: This response is sent when an Expires header field is present in an INVITE request, and the specified time period has passed
 - 415 Unsupported Media Type: This response sent by a user agent indicates that the media type contained in the INVITE request is not supported. For example, a request for a video conference to a PSTN gateway that only handles telephone calls will result in this response
 - 480 Temporarily Unavailable: This response indicates that the request has reached the correct destination, but the called party is not available for some reason. The reason phrase should be modified for this response to give the caller a better understanding of the situation. The response should contain a Retry-After header indicating when the request may be fulfilled
 - 483 Too Many Hops: This response indicates that the request has been forwarded the maximum number of times as set by the Max-Forwards header in the request
 - 486 Busy Here: This response is used to indicate that the user agent is busy. This response is equivalent to the busy tone in the PSTN
 - 487 Request Terminated: For pending Invites to terminate

- SERVER ERROR (5XX): The request has failed due to an error by the server. The request may be retried at another server
 - 501 Not Implemented: This response indicates that the server is unable to process the request because it is not supported. This response can be used to decline a request containing an unknown method
 - 502 Bad Gateway: This response is sent by a proxy that is acting as a gateway to another network, and indicates that some problem in the other network is preventing the request from being processed
 - 503 Service Unavailable: This response indicates that the requested service is temporarily unavailable. The request can be retried after a few seconds, or after the expiration of the Retry-After header field
 - 504 Gateway Timeout: This response indicates that the request failed due to a timeout encountered in the other network to which that the gateway connects
 - 505 Version Not Supported: This response indicates that the request has been refused by the server because of the SIP version number of the request. There is only one version of SIP (version 2.0) currently implemented
- GLOBAL FAILURE (6XX): The request has failed. The request should not be tried again at this or
 other servers
 - 600 Busy Everywhere: If there is a possibility that the Request could be answered in other locations, this response should not be sent
 - 604 Does Not Exist Anywhere: This response is similar to the 404 Not Found response but indicates that the user in the Request-URI cannot be found anywhere
 - 606 Not Acceptable: This response can be used to implement some session negotiation capability in SIP. This response indicates that some aspect of the desired session is not acceptable to the UAS, and as a result, the session cannot be established. The response may contain a Warning header field with a numerical code describing exactly what was not acceptable

Understanding LAB PSTN Setup

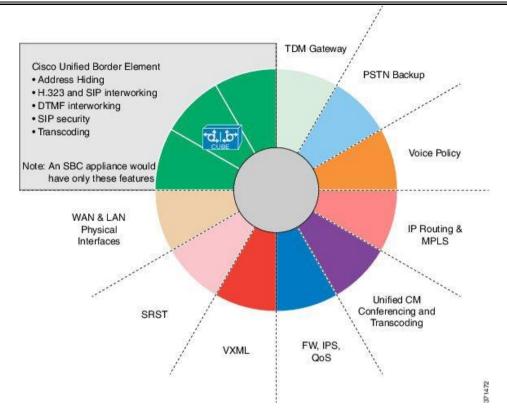
- To simulate external calls from CUCM, we have a simulated PSTN / ITSP running on 192.168.61.1
- In the backend, it is just another CUCM
- The simulated PSTN Phone is given below
- It has 3 PSTN numbers configured that corresponds to 3 Sites in the topology and 5 Speed Dials that simulates the incoming calls to the CUCM cluster from different locations



CUBE - Cisco Unified Border Element



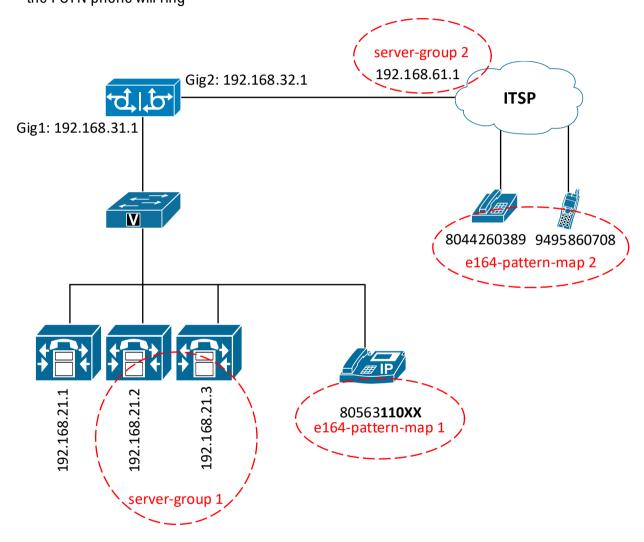
- To extend the phone calls to PSTN we use voice gateways or CUBE (Cisco Unified Border Element)
- Voice Gateways use H.323, MGCP or SIP protocol to talk to CUCM on the one leg and the other leg uses Circuit Switched Network (ISDN PRI - Primary Rate Interface).
- Some old deployments and locations use E1 R2 Signaling CAS (Channel Associated Signaling)
- These days PRIs are obsolete, and we have SIP providers are available, hence we deal with CUBE
- CUBE bridges voice and video connectivity between two separate VoIP networks. It is like a
 traditional voice gateway, except for the replacement of physical voice cards with an IP connection.
- CUBE is a voice enabled IOS router or Cisco CSR 1000v that runs some additional voice features
- You can still use the CUBE router to perform other tasks such as Routing, Access List, etc. It is
 just a router
- CUBE is the Session Border Controller solution from Cisco available in Integrated Services Router (ISR), Cisco Catalyst Edge Routers and Aggregation Services Router (ASR) and virtualized environments with the Cisco Cloud Services Router (CSR) and Catalyst Edge Software



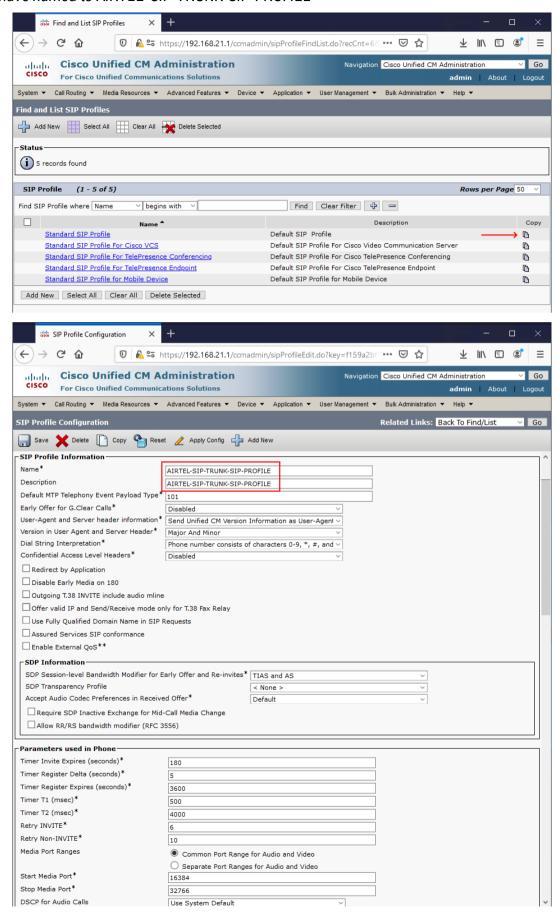
- CUBE talks to CUCM over SIP on one leg and the other leg connects to service provider (ITSP)
 also uses SIP
- allow connections sip to sip under voice service voip configurations makes an
 IOS router to perform CUBE functionality

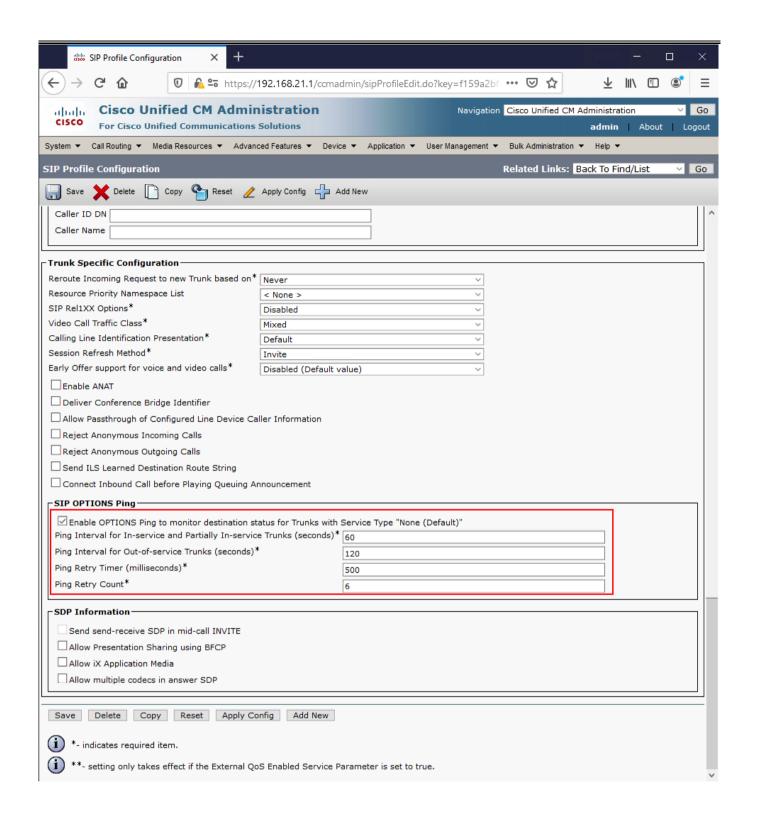
[Lab] CUBE Configuration

- Our aim is to make a call from our Lab IP Phone to '8044260389' which is a PSTN number in Bangalore
- Assume: Discussion happened between service provider that you have signed up for a specific SIP
 Account subscription from Airtel service provider
- In this lab, the service provider end is already configured. We must configure CUCM, CUBE Local side, CUBE Provider side
- You can see a PSTN phone with the same number configured. If all the configurations are in place,
 the PSTN phone will ring

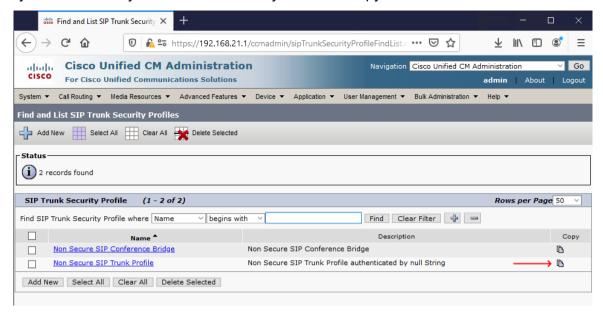


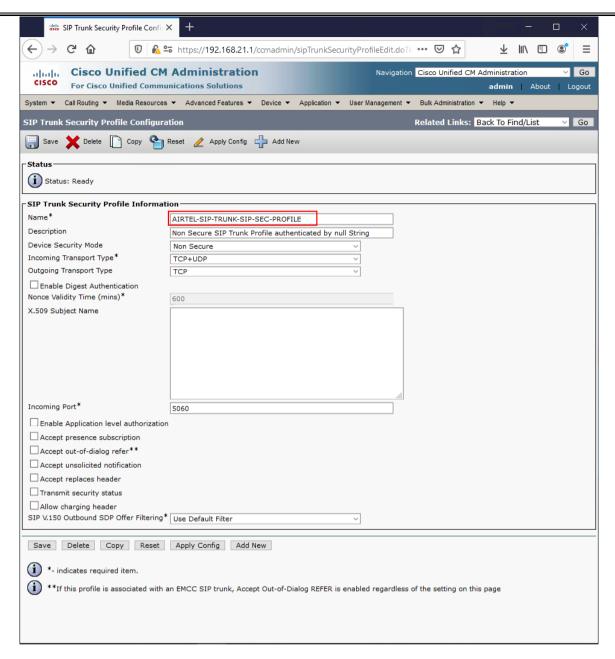
- Device >> Device Settings >> SIP Profile >> Copy Standard SIP Profile
- I have named to AIRTEL-SIP-TRUNK-SIP-PROFILE



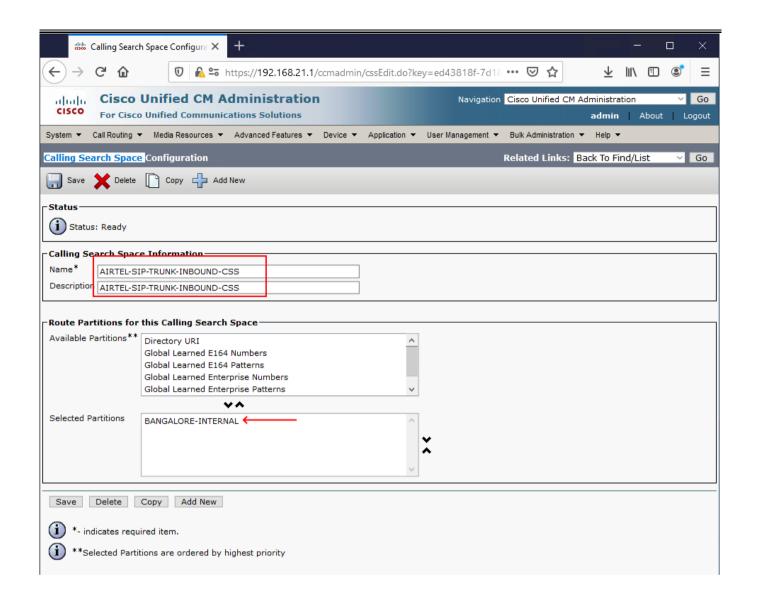


• System >> Security >> SIP Trunk Security Profile >> Copy Non Secure SIP Trunk Profile

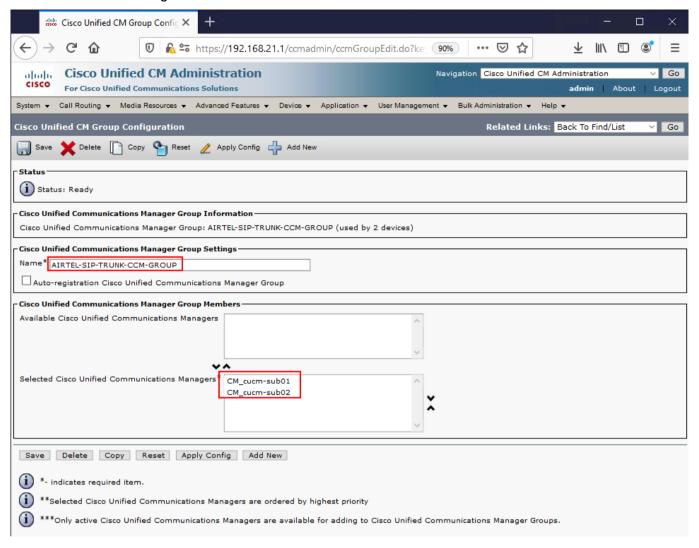




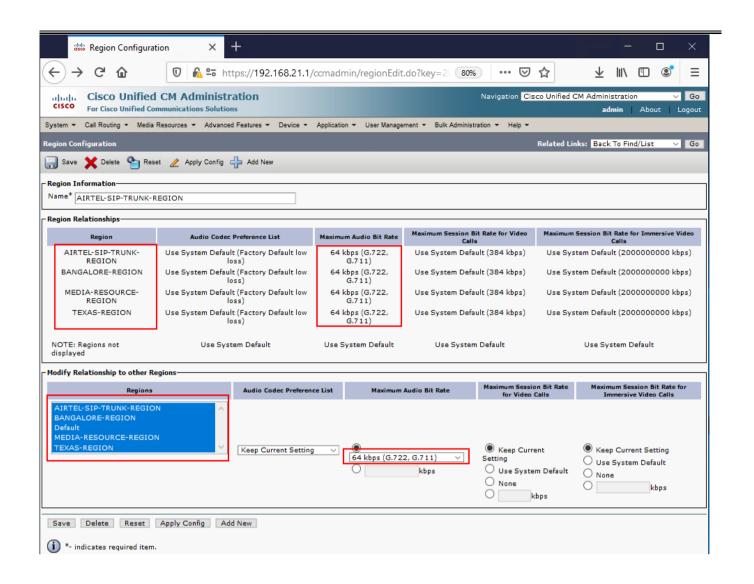
- Call Routing >> Class of Control >> Calling Search Space >> Add new CSS
- This is used when we receive calls via the SIP Trunk from ITSP



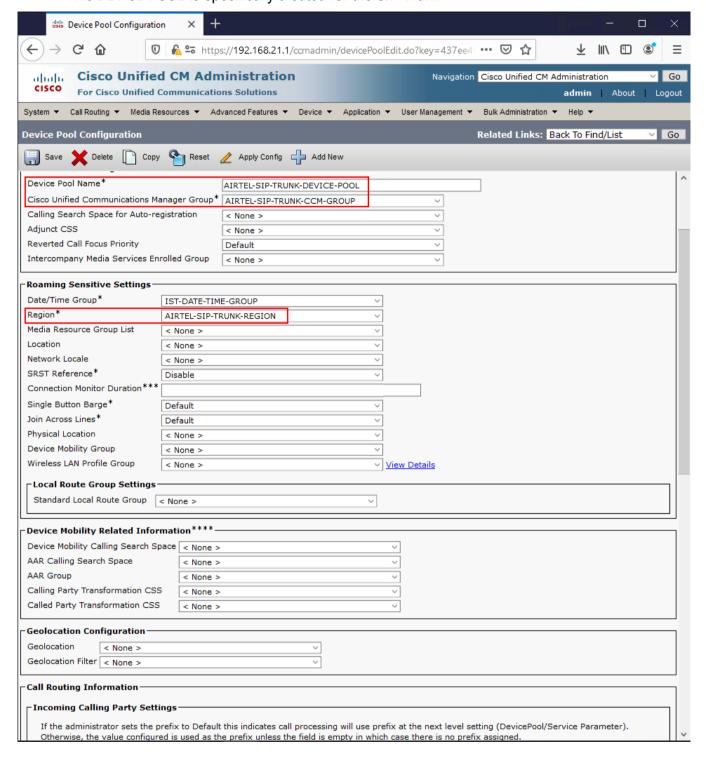
- Create a new CUCM Group
- This will be assigned to the SIP Trunk



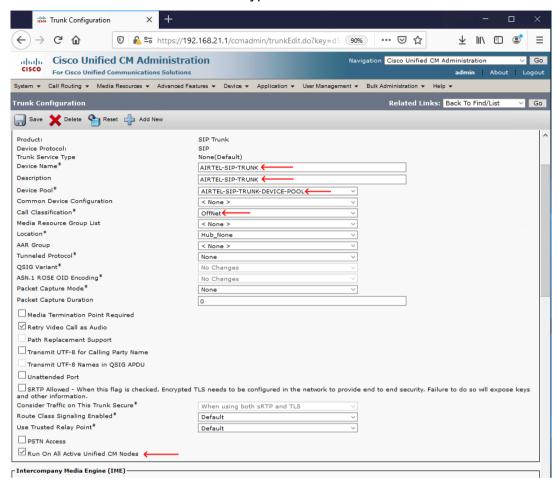
- Create a new REGION to control the Codec between devices and Sip Trunk
- I have set the codec to 64kbps

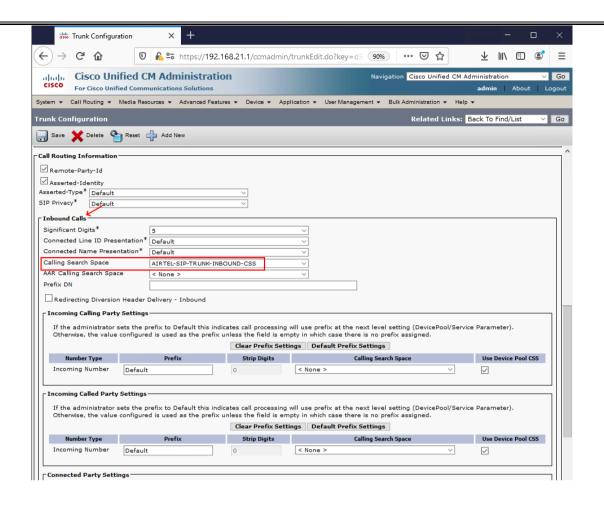


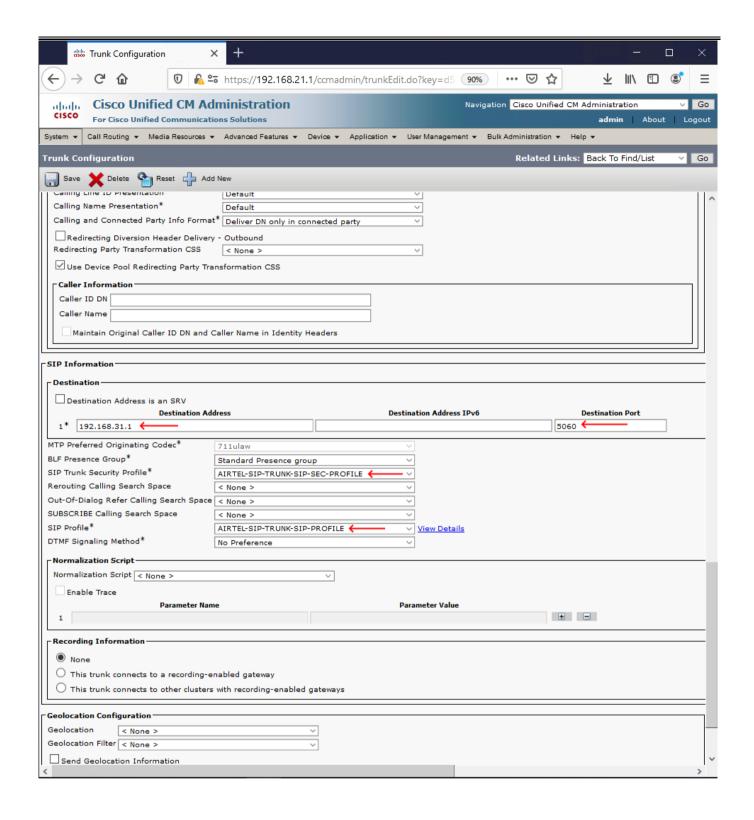
- Create a new DEVICE POOL and associate CUCM Group and REGION to it
- This DEVICE POOL is specifically created for the SIP Trunk



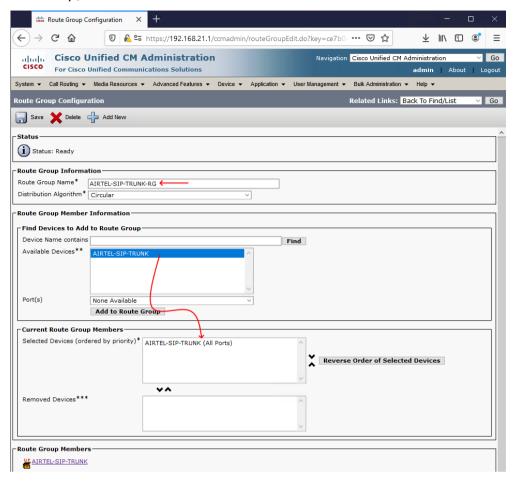
• Device >> Trunk >> Add New >> Trunk Type: SIP >> Next

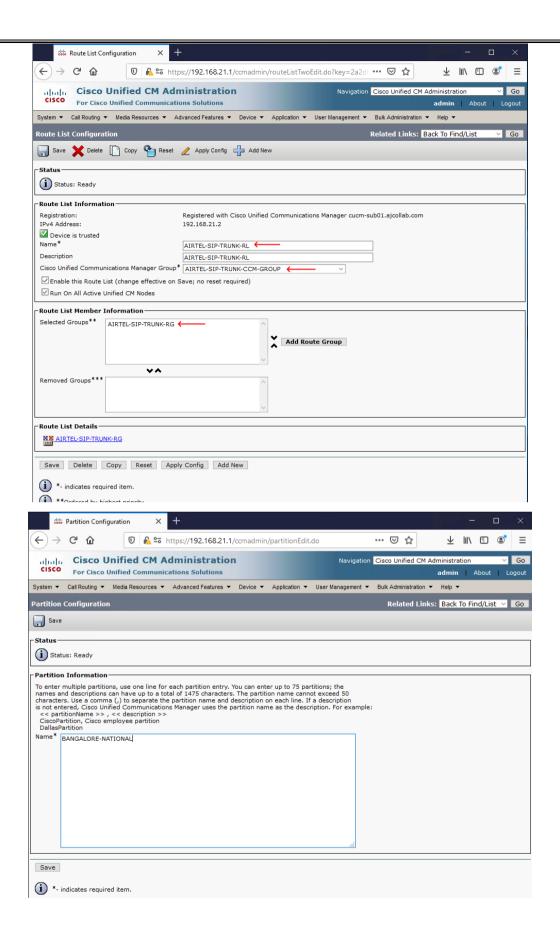


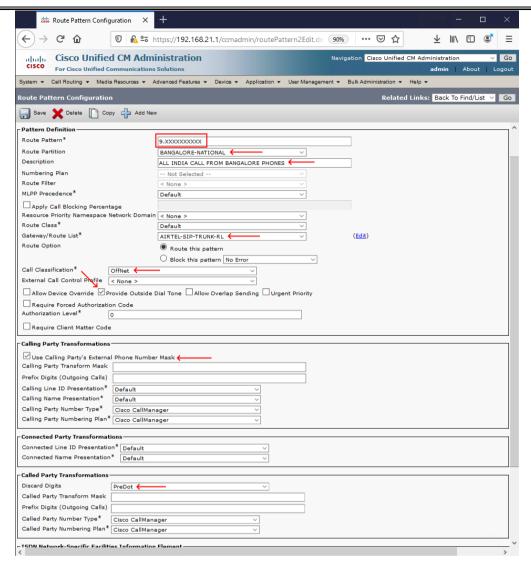




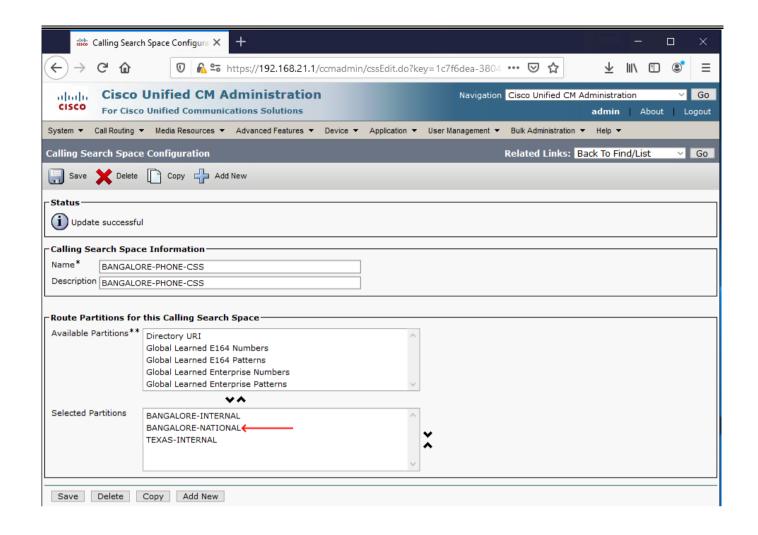
• Add Route Group, Route List and Route Pattern



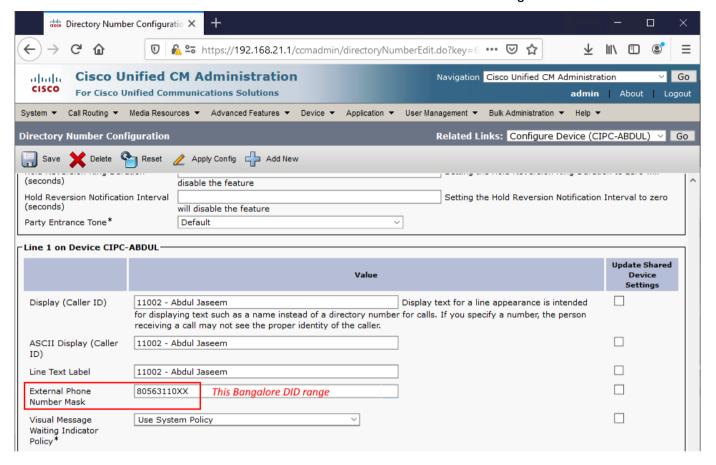




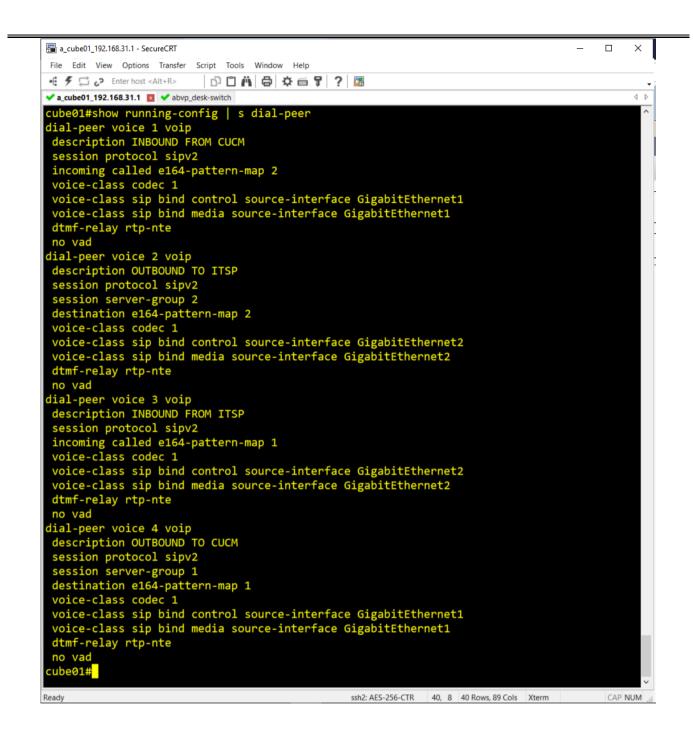
 In BANGALORE-PHONE-CSS, update the BANGALORE-NATIONAL partition so that Bangalore Phones can dial Indian National Number



- Update the Phone Line page External Phone Number Mask
- When the call hits the Route Pattern 9.XXXXXXXXXX, the calling number 11002 is transformed to 80563110XX = 8056311001
- Hence the PSTN Phone will see the actual DID number rather than 5-digit internal number



- Next you can use below configurations on the CUBE
- Referring the topology, we can understand the IP details and DID number details

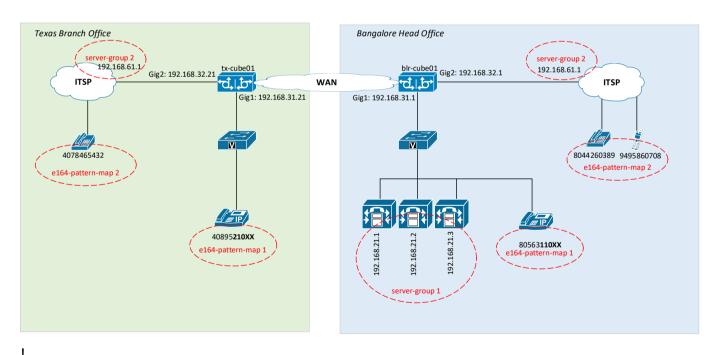


```
voice service voip
 ip address trusted list
  ipv4 192.168.21.1
  ipv4 192.168.21.2
  ipv4 192.168.21.3
  exit
 mode border-element
 allow-connections sip to sip
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
 codec preference 4 g729br8
voice class server-group 1
 description LOCAL CUCM SIDE
 ipv4 192.168.21.2
 ipv4 192.168.21.3
voice class server-group 2
 description ITSP SIDE
 ipv4 192.168.61.1
voice class e164-pattern-map 1
 description BANGALORE INTERNAL DID
  e164 ^80563110..$
  e164 ^8056310002$
voice class e164-pattern-map 2
 description BANGALORE ITSP NATIONAL NUMBERS
  e164 ^.....$
dial-peer voice 1 voip
 description INBOUND FROM CUCM
 incoming called e164-pattern-map 2
 session protocol sipv2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
no vad
dial-peer voice 2 voip
 description OUTBOUND TO ITSP
 session server-group 2
 destination e164-pattern-map 2
 session protocol sipv2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet2
```

```
voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
no vad
dial-peer voice 3 voip
description INBOUND FROM ITSP
 session protocol sipv2
 incoming called e164-pattern-map 1
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
no vad
dial-peer voice 4 voip
 description OUTBOUND TO CUCM
 session protocol sipv2
 session server-group 1
destination e164-pattern-map 1
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte
no vad
show voip rtp connections
show cube calls all
```



- We have one more CUBE in remote site TEXAS to make PSTN calls from Texas phones
- Verizon is the SIP Service Provider in Texas site follow the similar steps to configure SIP Trunk
- You have to create below configurations
 - VERIZON-SIP-TRUNK-SIP-PROFILE
 - VERIZON-SIP-TRUNK-SIP-SEC-PROFILE
 - VERIZON-SIP-TRUNK-INBOUND-CSS with TEXAS-INTERNAL
 - VERIZON-SIP-TRUNK-CCM-GROUP
 - VERIZON-SIP-REGION
 - VERIZON-SIP-TRUNK-DEVICE-POOL
 - VERIZON-SIP-TRUNK
 - VERIZON-SIP-TRUNK-RG
 - VERIZON-SIP-TRUNK-RL
 - Route Pattern 9.XXXXXXXXXXX on TEXAS-NATIONAL
 - Update TEXAS-PHONE-CSS
 - TX-CUBE Configurations



```
voice service voip
ip address trusted list
ipv4 192.168.21.1
ipv4 192.168.21.2
ipv4 192.168.21.3
mode border-element
allow-connections sip to sip
```

```
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
 codec preference 4 g729br8
voice class server-group 1
 description LOCAL CUCM SIDE
 ipv4 192.168.21.2
 ipv4 192.168.21.3
voice class server-group 2
 description ITSP SIDE
 ipv4 192.168.61.1
voice class e164-pattern-map 1
 description TEXAS INTERNAL DID
  e164 ^40895210..$
voice class e164-pattern-map 2
 description TEXAS ITSP NATIONAL NUMBERS
  e164 ^.....$
dial-peer voice 1 voip
 description INBOUND FROM CUCM
 session protocol sipv2
 incoming called e164-pattern-map 2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 no vad
dial-peer voice 2 voip
 description OUTBOUND TO ITSP
 session protocol sipv2
 session server-group 2
 destination e164-pattern-map 2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 no vad
dial-peer voice 3 voip
 description INBOUND FROM ITSP
 session protocol sipv2
 incoming called e164-pattern-map 1
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
```

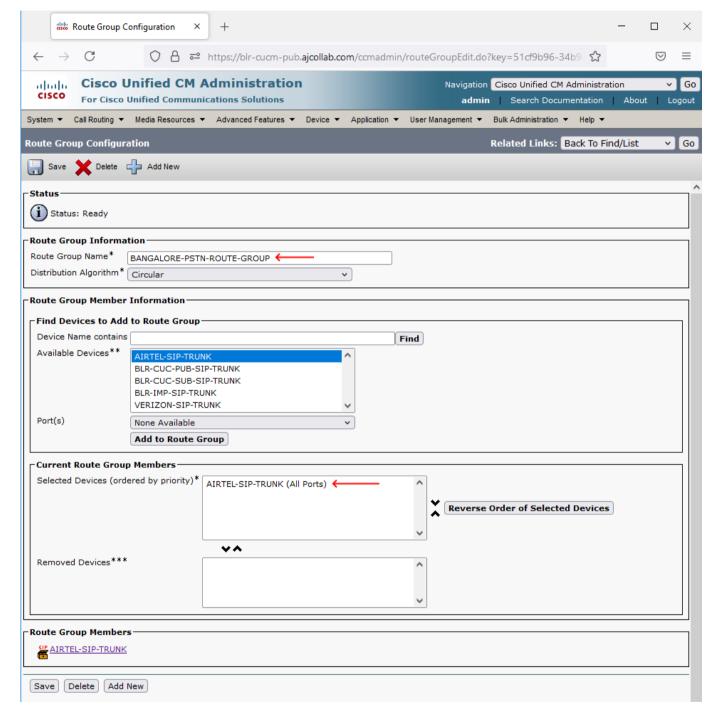
```
dtmf-relay rtp-nte
no vad
!
dial-peer voice 4 voip
description OUTBOUND TO CUCM
session protocol sipv2
session server-group 1
destination e164-pattern-map 1
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte
no vad
!
```

[Lab] Standard Local Route Group (SLRG)

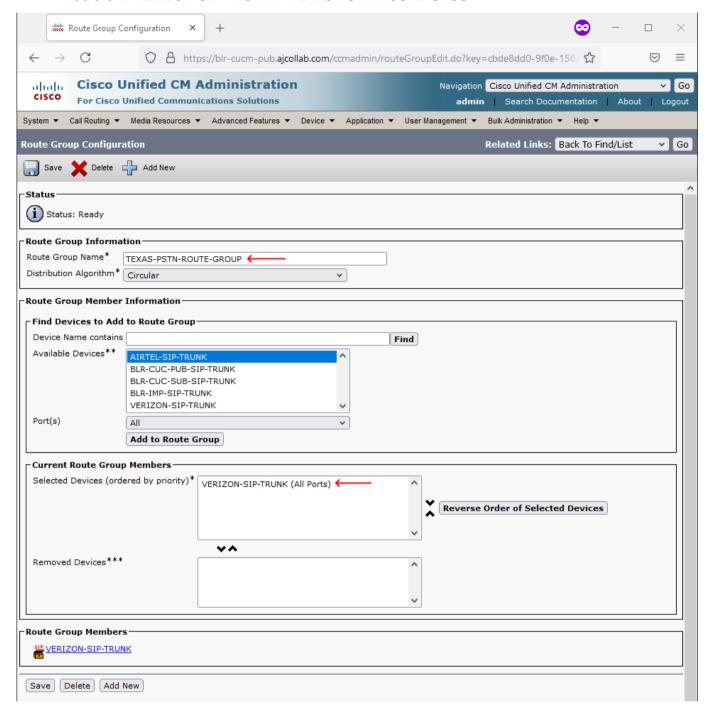
- SLRG reduces the complexity and size of Dial-plan in CUCM. LRG is a device pool parameter and introduced in CUCM 7
- It is a mechanism for device pool-based gateway / Trunk selection to route the calls
- Each device pools will have their own respective local gateway. Instead of multiple route pattern for multiple locations, we create generic pattern and point list and then to SLRG
- In our lab, instead of creating 9.XXXXXXXXXX in two different partition, we create just one route pattern 9.XXXXXXXXXX and one route list
- When Bangalore Phone initiates a call, it will go via Bangalore CUBE and if Texas Phone makes a call, it will go via Texas CUBE

Steps:

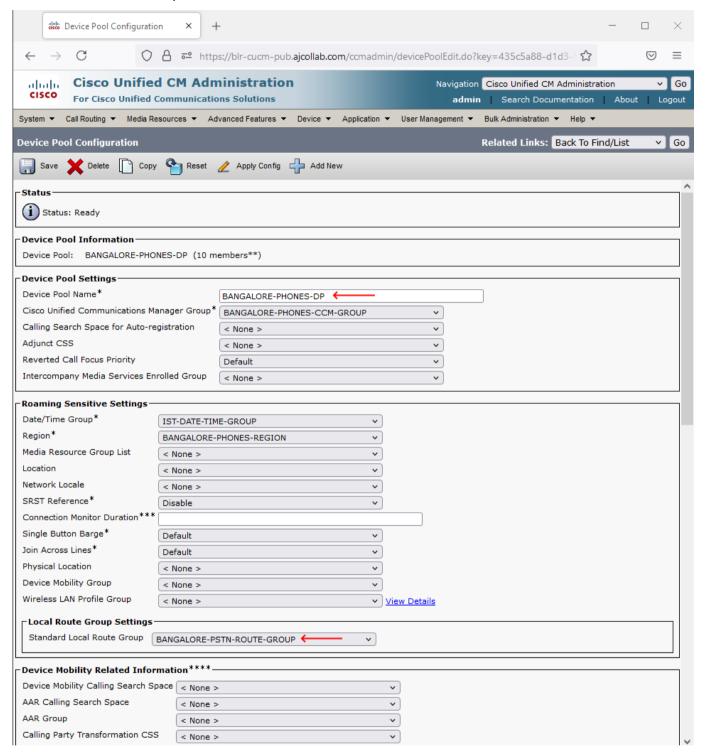
- Create 2 different Route Groups BANGALORE-PSTN-ROUTE-GROUP and TEXAS-PSTN-ROUTE-GROUP
- Add the in AIRTEL-SIP-TRUNK in BANGALORE-PSTN-ROUTE-GROUP



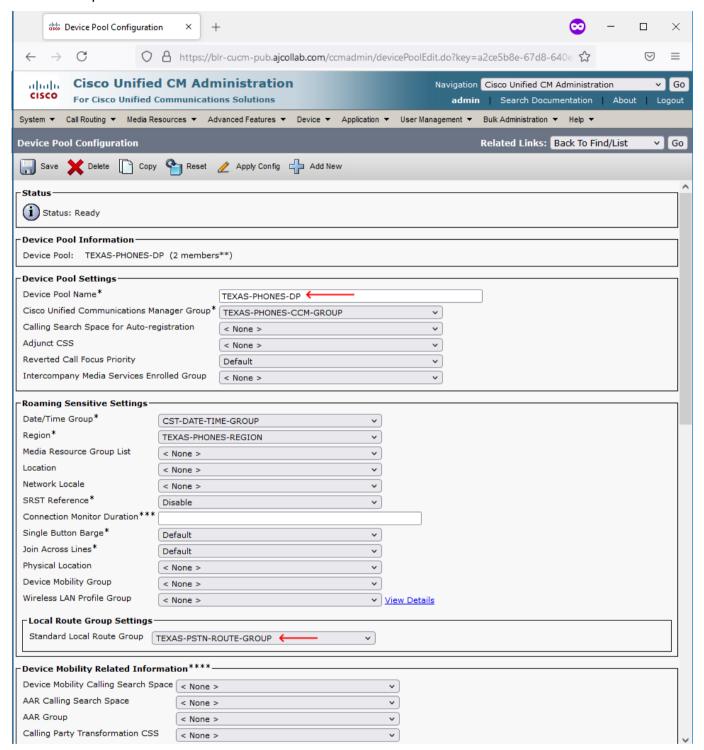
Add the in VERIZON-SIP-TRUNK in TEXAS-PSTN-ROUTE-GROUP



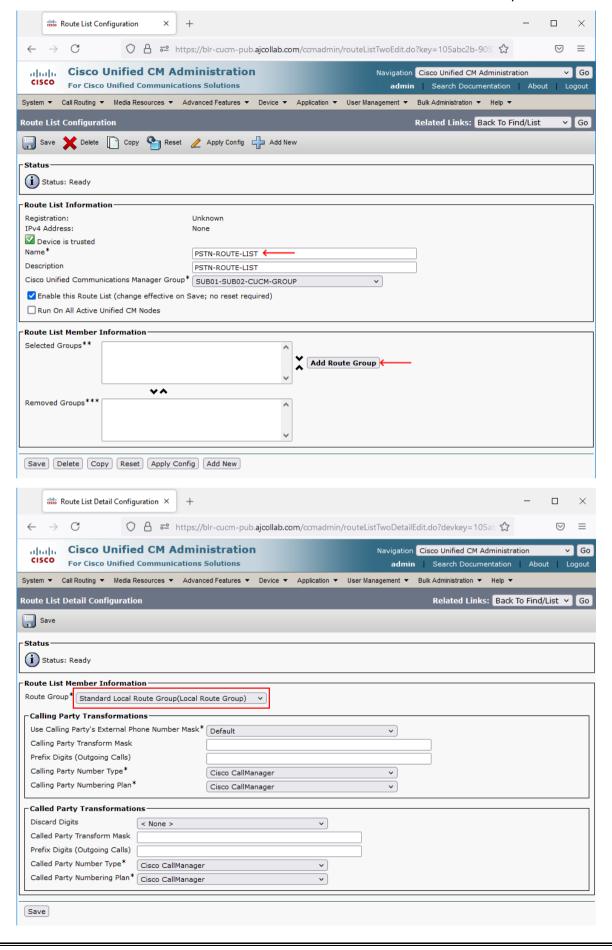
 On BANGALORE-PHONES-DEVICE-POOL add BANGALORE-PSTN-ROUTE-GROUP as the Standard Local Route Group



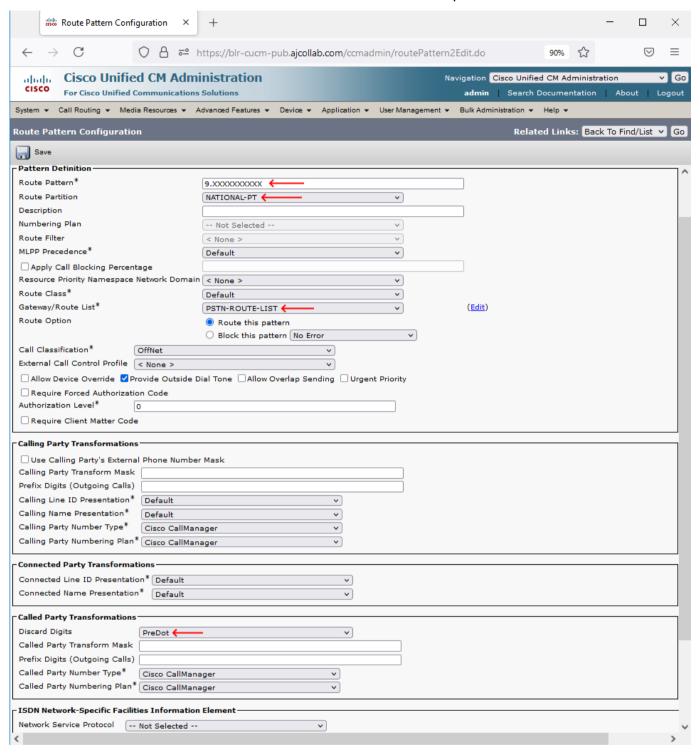
On TEXAS-PHONES-DEVICE-POOL add TEXAS-PSTN-ROUTE-GROUP as the Standard Local Route
 Group



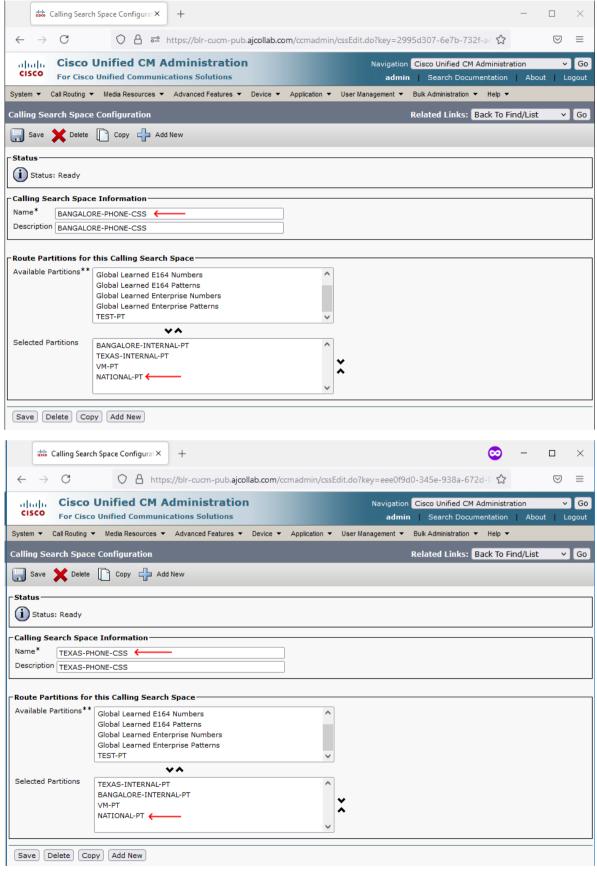
• Create a Route List: PSTN-ROUTE-LIST and select Standard Local Route Group



Create a Route Pattern: 9.XXXXXXXXXX in NATIONAL-PT and point to PSTN-CALL-ROUTE-LIST



Add the NATIONAL-PT to both Bangalore and Texas Phone CSS



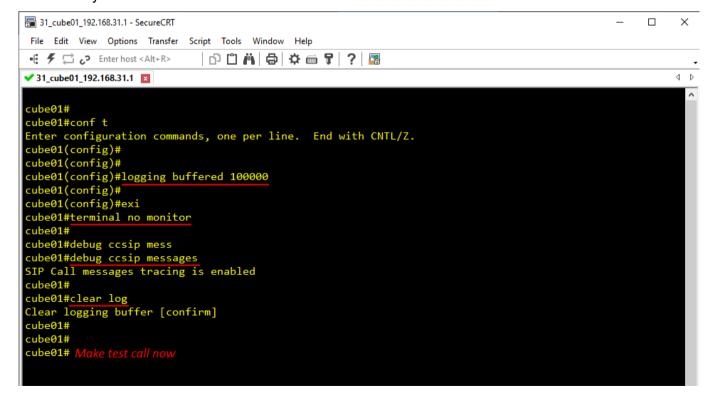
 Enable debugs on both Bangalore and Texas CUBEs then try calling the 10-digit PSTN number from Bangalore and Texas phones. See which gateway is getting hit.

CUBE Call Flow (with CUBE Debugs)

- **debug ccsip messages** debug command used to get the SIP logs from the CUBE router
- Since the CUBE interconnects CUCM and PSTN together, we can see the call leg between CUCM to CUBE and CUBE to ITSP in the CUBE debugs
- Use TranslatorX tool to analyses the SIP logs from CUBE
- While dealing with any CUBE debugs, it is always recommended to have below table filled

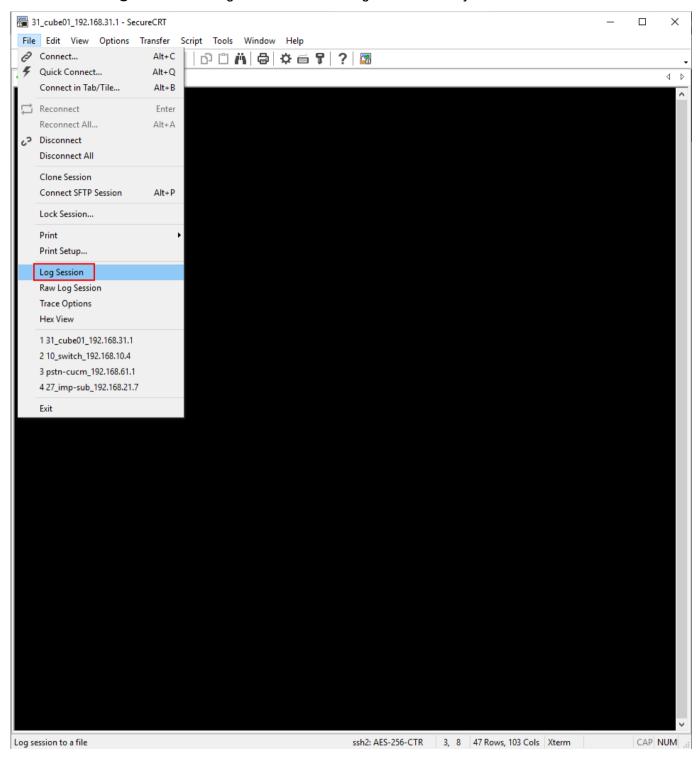
VALUES
80563 <mark>11002</mark>
8044260389
21:18 (Approx.)
14 Sec (Approx.)
192.168.128.4
192.168.21.1, 192.168.21.2, 192.168.21.3
192.168.31.1
192.168.32.1
192.168.61.1
192.168.130.1

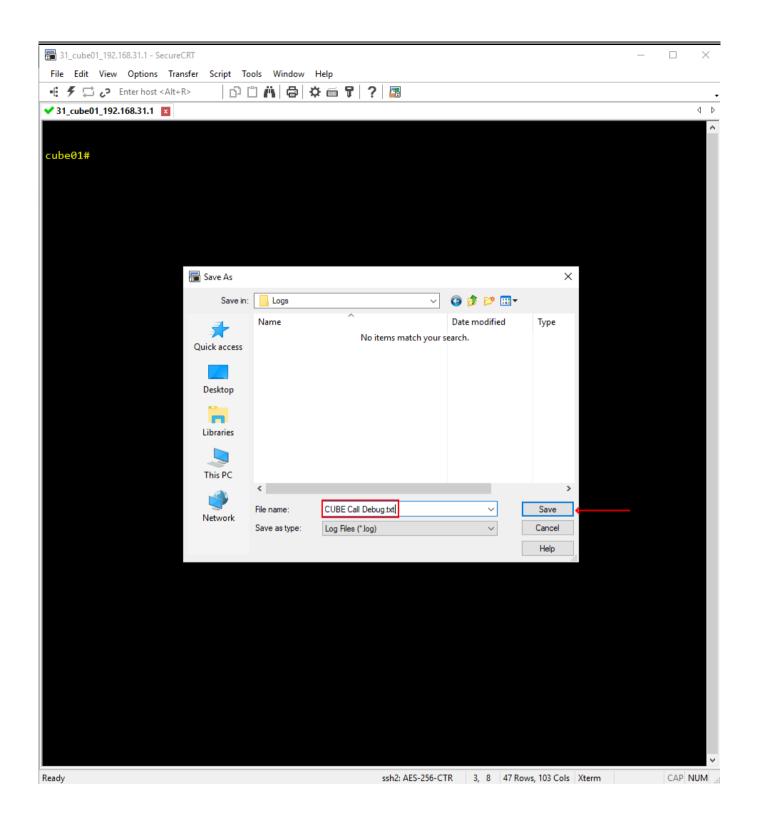
 If you have multiple calls on the CUBE, it is not recommended to have the logs collected via terminal monitor (directly on the SSH session), instead we can push the logs to a log buffer memory. If you have less calls, use terminal monitor and log the output to a Text file from Putty or Secure CRT.

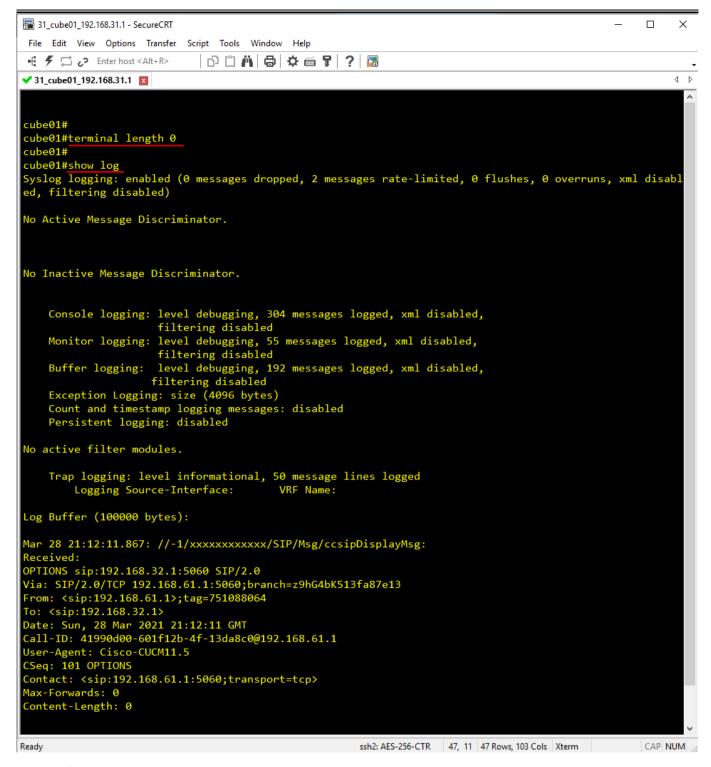


Right after the call, disable the debug using u all command

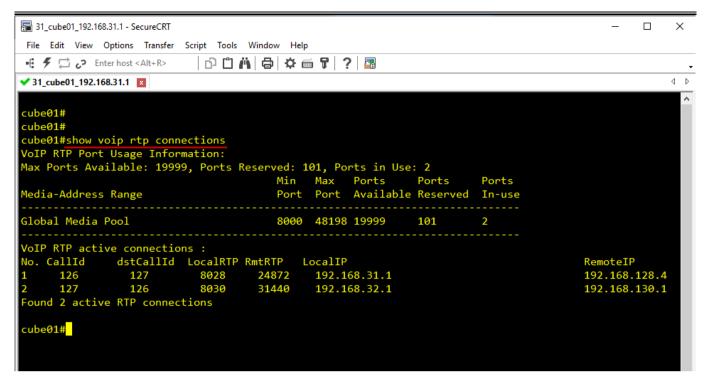
• show log command will give the content of log buffer memory



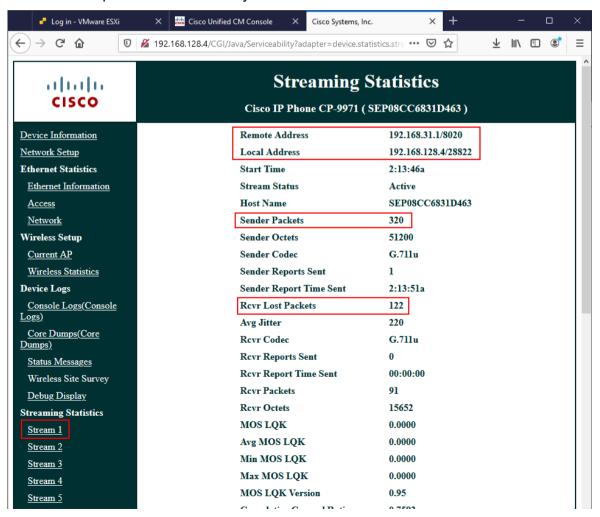




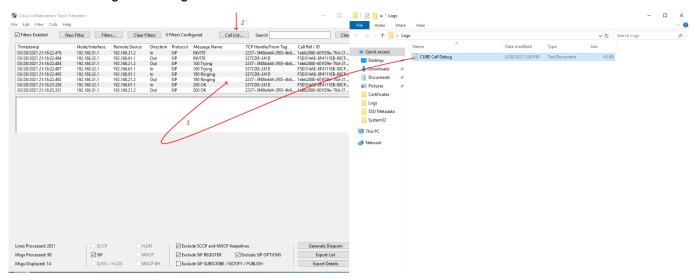
Optionally show voip rtp connections command will give the local and remote RTP IP
 Address and Port details, this will be available on the SDP as well. This has to be collected when the call is active.



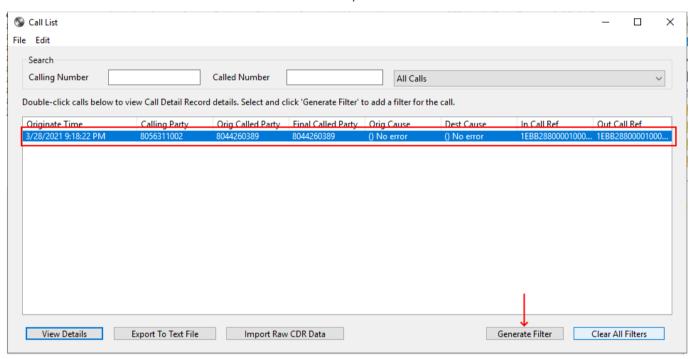
- Also Stream 1 from the IP Phone web page will give details about the Local (IP Phone) IP and Remote (CUBE Internal) IP along with RTP port numbers
- This will be helpful to isolate One-way Audio issue or No Audio issue as well



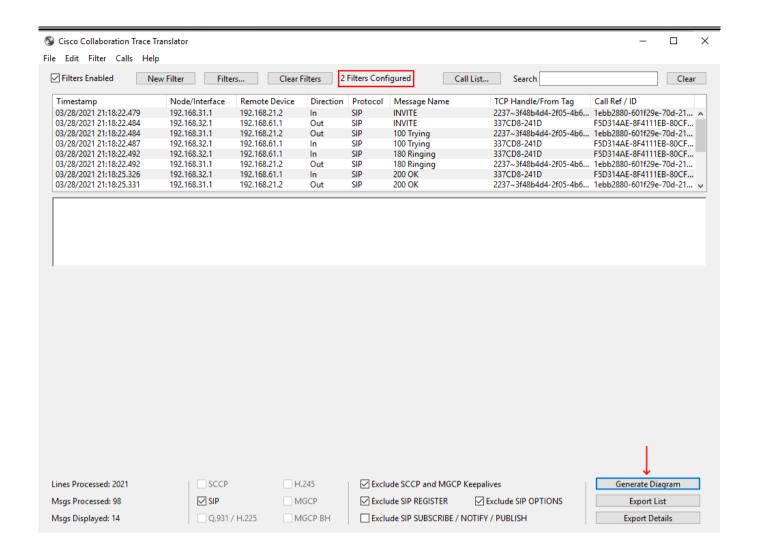
Now drag the debug text file to TranslatorX and click Call List button

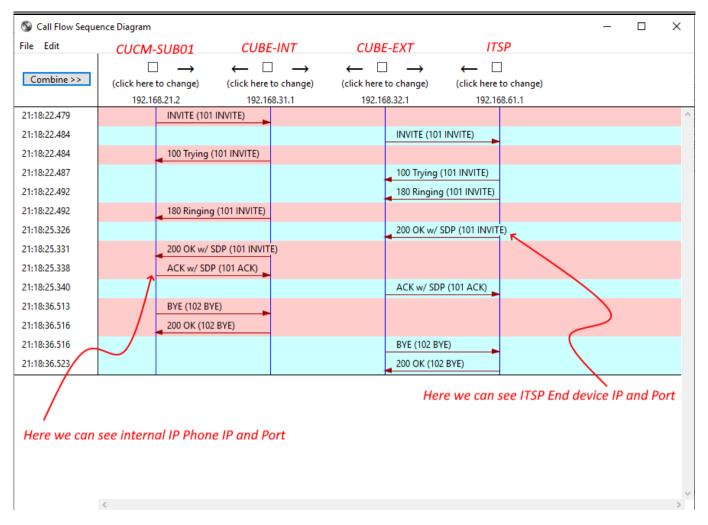


Click Generate Filter button on the TranslatorX, then close the Call List window



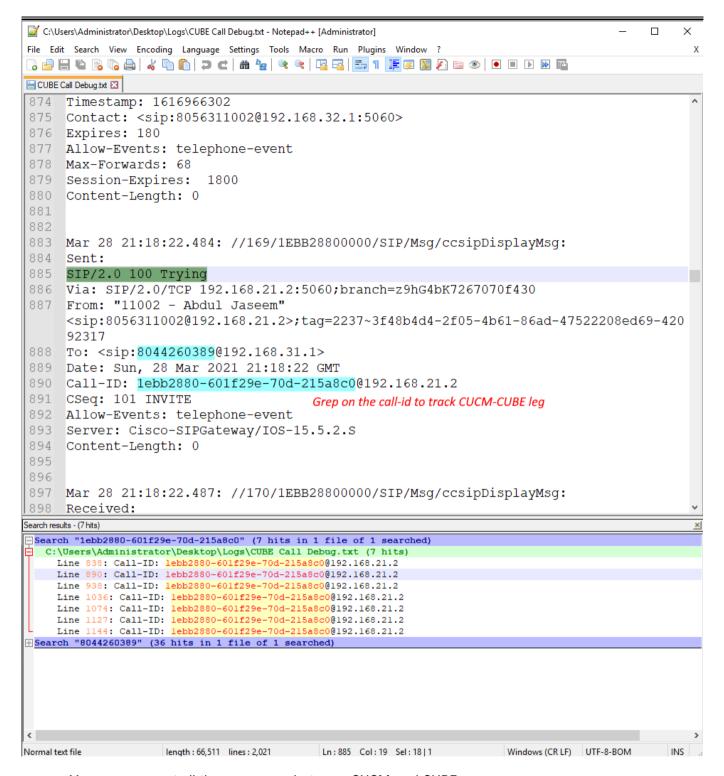
Click Generate Diagram





- Once you familiar with TranslatorX tool, you can alternatively analyze the logs from NotePad++
- Grep for the Calling or Called Number, then get the first Invite, there will be 2 Invites as you have seen in the above diagram

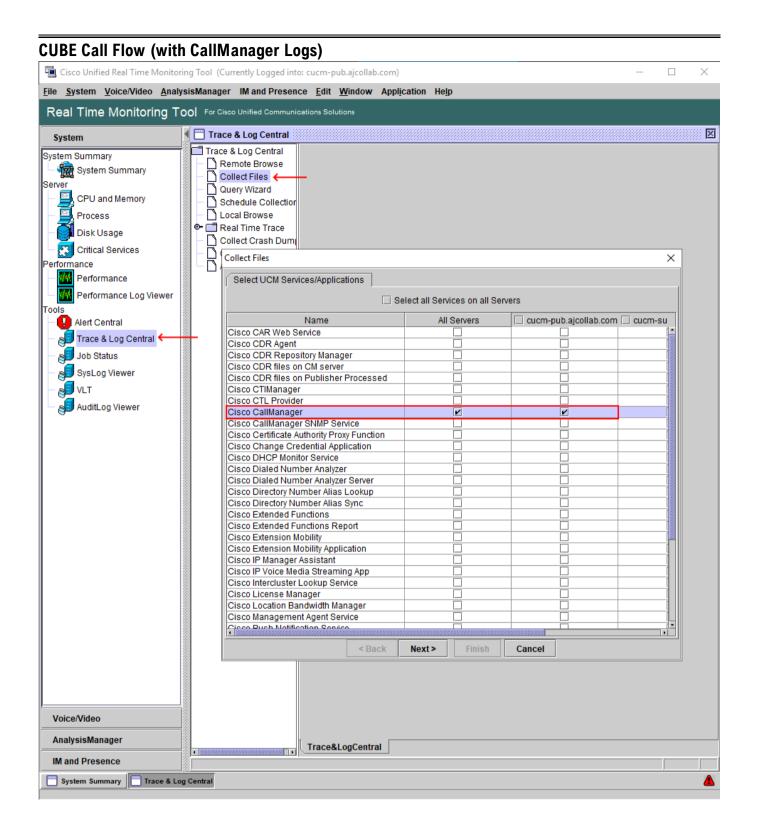
```
C:\Users\Administrator\Desktop\Logs\CUBE Call Debug.txt - Notepad++ [Administrator]
File Edit Search View Encoding Language Settings Tools Macro Run Plugins Window
 7 🖴 🕒 😘 😘 💫 | 사 🖍 🖍 🕳 😪 🔍 🤏 🔫 🖳 🚍 🖺 기 🎩 🐼 🔊 🖅 🗩 🗩 🗩 🗩
CUBE Call Debug.txt 🗵
830
831 Mar 28 21:18:22.479: //-1/xxxxxxxxxxx/SIP/Msq/ccsipDisplayMsq:
832 Received:
833 INVITE sip: 8044260389@192.168.31.1:5060 SIP/2.0
834 Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK7267070f430
835 From: "11002 - Abdul Jaseem"
       <sip:8056311002@192.168.21.2>;tag=2237~3f48b4d4-2f05-4b61-86ad-47522208ed69-420
      92317
836 To: <sip:8044260389@192.168.31.1>
837 Date: Sun, 28 Mar 2021 21:18:22 GMT
      Call-ID: 1ebb2880-601f29e-70d-215a8c0@192.168.21.2 → This will be unique for CUCM to CUBE leg
      Supported: timer, resource-priority, replaces
840 Min-SE:
                 1800
841
      User-Agent: Cisco-CUCM11.5 → Message came from CUCM
842 Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
      SUBSCRIBE, NOTIFY
843 CSeq: 101 INVITE
844 Expires: 180
845 Allow-Events: presence, kpml
846 Supported: X-cisco-srtp-fallback, X-cisco-original-called
847
      Call-Info:
       <sip:192.168.21.2:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
848
      Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
      Session-ID:
       Cisco-Guid: 0515582080-0000065536-0000000013-0034973888
850
851
      Session-Expires:
                            1800
Search results - (36 hits)
 Search "8044260389" (36 hits in 1 file of 1 searched)
  C:\Users\Administrator\Desktop\Logs\CUBE Call Debug.txt (36 hits)
     Line 833: INVITE sip:8044260389@192.168.31.1:5060 SIP/2.0 Invite from CUCM to CUBE
     Line 836: To: <sip:8044260389@192.168.31.1>
     Line 861: INVITE sip: 8044260389@192.168.61.1 SIP/2.0
     Line 865: To: <sip:8044260389@192.168.61.1>
     Line 888: To: <sip:8044260389@192.168.31.1>
     Line 902: To: <sip:8044260389@192.168.61.1>
     Line 915: To: <sip:8044260389@192.168.61.1>;tag=111~6d68c7b1-6dc3-4f9e-b3d4-c3249872f990-25383825
     Line 925: P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1> Line 925: P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>
     Line 926: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>;party=called;screen=yes; Line 926: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>;party=called;screen=yes;
     Line 927: Contact: <sip:8044260389@192.168.61.1:5060>
     Line 936: To: <sip:8044260389@192.168.31.1>;tag=337CE0-1651
     Line 942: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.31.1>;party=called;screen=yes;
     Line 942: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip: 8044260389 @192.168.31.1>; party=called; screen=yes;
Normal text file
                     length: 66,511 lines: 2,021 Ln: 834 Col: 20 Pos: 26,717
                                                                    Windows (CR LF) UTF-8-BOM
                                                                                                       INS
```

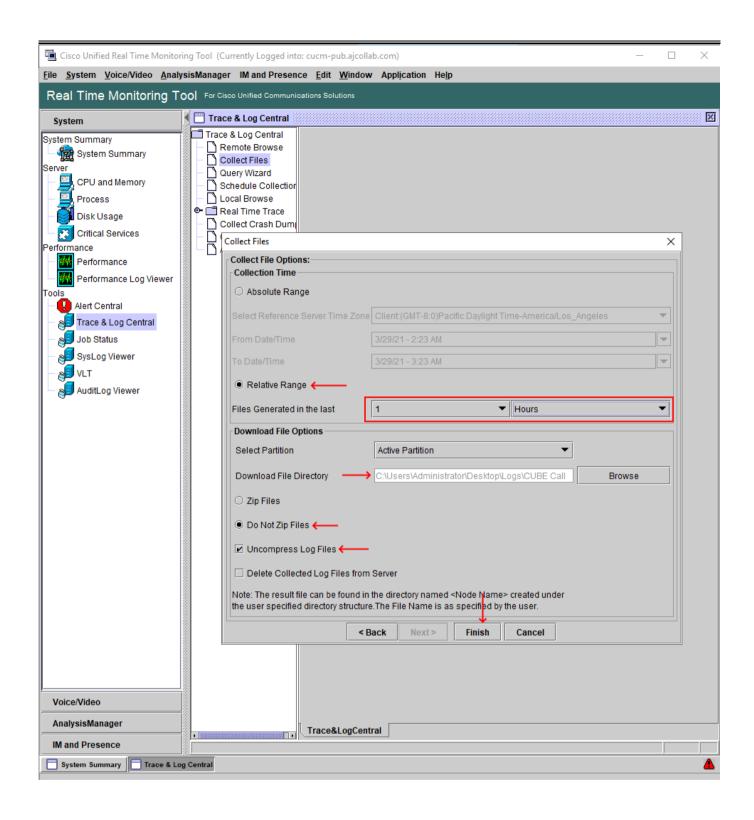


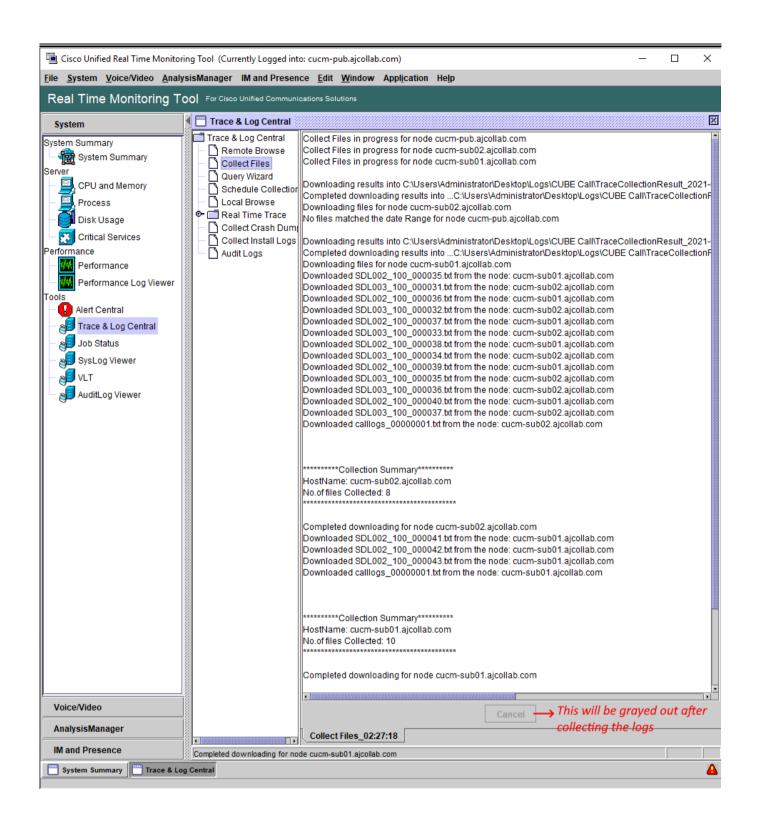
- You can now get all the messages between CUCM and CUBE
- To track the CUBE-ITSP leg, identify the call-id of second Invite and search for the call-id

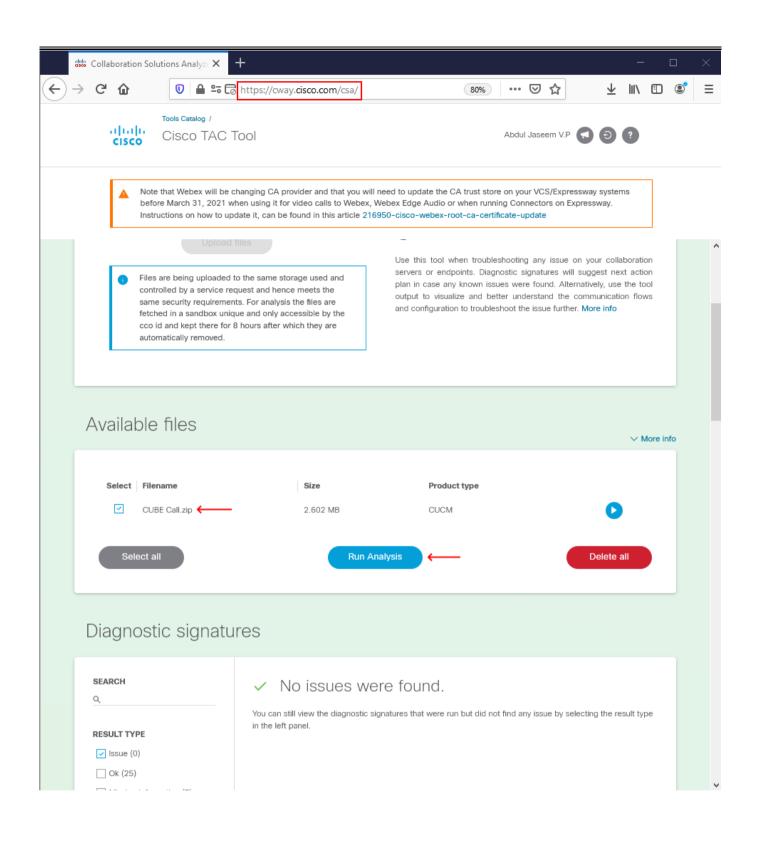
```
C:\Users\Administrator\Desktop\Logs\CUBE Call Debug.txt - Notepad++ [Administrator]
 File Edit Search View Encoding Language Settings Tools Macro Run Plugins Window
  3 🖴 🗎 🖺 🥦 🥱 😭 🔏 | & 🖍 🖍 🖒 | ⊃ C | ## 🛬 | 🤏 🥞 | 🖫 🖂 🔜 1 | 🎩 🐼 🔊 🖼 🐠 | • ● | • □ 🗈 🕪 🖼
 CUBE Call Debug.txt 🗵
 855 Max-Forwards: 69
           Content-Length: 0
 856
 857
 858
 859 Mar 28 21:18:22.484: //170/1EBB28800000/SIP/Msg/ccsipDisplayMsg:
 860 Sent:
 861 INVITE sip:8044260389@192.168.61.1 SIP/2.0
 862 Via: SIP/2.0/UDP 192.168.32.1:5060; branch=z9hG4bK186DE
 863 Remote-Party-ID: "11002 - Abdul Jaseem"
            <sip:8056311002@192.168.32.1>;party=calling;screen=yes;privacy=off
 864 From: "11002 - Abdul Jaseem" <sip:8056311002@192.168.32.1>;tag=337CD8-241D
 865 To: <sip:8044260389@192.168.61.1>
 866 Date: Sun, 28 Mar 2021 21:18:22 GMT
 867 Call-ID: F5D314AE-8F4111EB-80CFBC96-4640BB65@192.168.32.1
 868 Supported: timer, resource-priority, replaces, sdp-anat
 869 Min-SE: 1800
 870 Cisco-Guid: 0515582080-0000065536-0000000013-0034973888
 871 User-Agent: Cisco-SIPGateway/IOS-15.5.2.S Message originated from CUBE
 872 Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
            NOTIFY, INFO, REGISTER
 873 CSeq: 101 INVITE
 874 Timestamp: 1616966302
 875
            Contact: <sip:8056311002@192.168.32.1:5060>
 876 Expires: 180
 877 Allow-Events: telephone-event
 878 Max-Forwards: 68
 879 Session-Expires:
                                                 1800
Search results - (36 hits)
 Search "8044260389" (36 hits in 1 file of 1 searched)
    C:\Users\Administrator\Desktop\Logs\CUBE Call Debug.txt (36 hits)
          Line 833: INVITE sip: 8044260389@192.168.31.1:5060 SIP/2.0
          Line 836: To: <sip:8044260389@192.168.31.1>
         Line 861: INVITE sip:8044260389@192.168.61.1 SIP/2.0 Invite from CUBE to ITSP
          Line 865: To: <sip: 8044260389@192.168.61.1>
         Line 888: To: <sip:8044260389@192.168.31.1>
         Line 902: To: <sip:8044260389@192.168.61.1>
         Line 915: To: <sip: 8044260389@192.168.61.1>:tag=111~6d68c7b1-6dc3-4f9e-b3d4-c3249872f990-25383825
         Line 925: P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>
         Line 925: P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>
         Line 926: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>;party=called;screen=yes; Line 926: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>;party=called;screen=yes;
         Line 927: Contact: <sip:8044260389@192.168.61.1:5060>
         Line 936: To: <sip:8044260389@192.168.31.1>;tag=337CE0-1651
         Line 942: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.31.1>;party=called;screen=yes; Line 942: Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.31.1>;party=called;screen=yes; verified to the state of the
Normal text file
                                     length : 66,511 lines : 2,021 Ln : 871 Col : 42 Sel : 29 | 1 Windows (CR LF) UTF-8-BOM
```

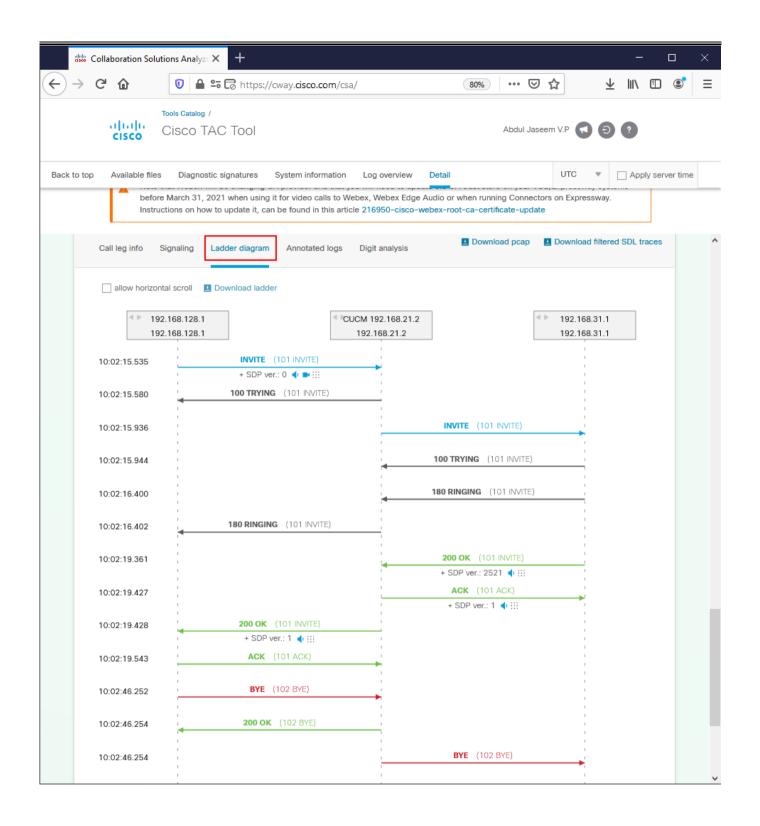
- The transaction between CUCM and CUBE is also available in Cisco CallManager logs that can be collected using RTMT Software
- From the CallManager logs, we could see the IP Phone to CUCM signaling as well







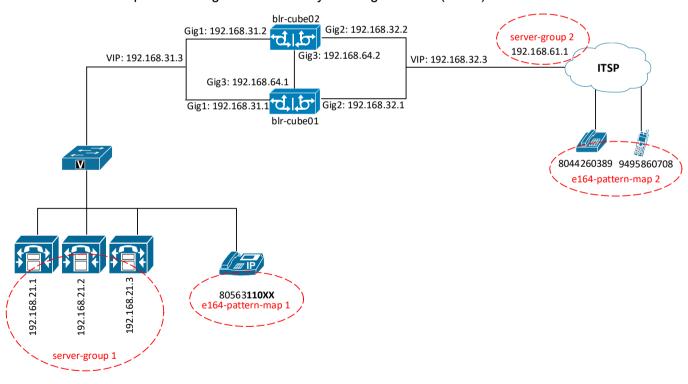




CUBE High Availability (HA)



- The High Availability (HA) feature provides failover capability of Cisco Unified Border Element
 (CUBE) on two routers, one active and one standby
- When the active router goes down for any reason, the standby router takes over seamlessly,
 preserving and processing your calls
- We configure a Virtual IP Address to make the two CUBEs to appear like one device
- HA is accomplished using the Hot Standby Routing Protocol (HSRP)



CUBE Base Configuration

- CUBE01 and CUBE02 should have similar call routing configurations (dial-peers, voice-class, etc.)
- Make sure your calls are routing via each CUBEs separately
- Below are the base configurations of CUBE

```
voice service voip
 ip address trusted list
  ipv4 192.168.21.1
  ipv4 192.168.21.2
  ipv4 192.168.21.3
 mode border-element
 allow-connections sip to sip
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 g729r8
 codec preference 4 g729br8
voice class server-group 1
 description LOCAL CUCM SIDE
 ipv4 192.168.21.2
 ipv4 192.168.21.3
voice class server-group 2
 description ITSP SIDE
 ipv4 192.168.61.1
voice class e164-pattern-map 1
 description BANGALORE INTERNAL DID
  e164 ^80563110..$
  e164 ^8056310002$
voice class e164-pattern-map 2
 description BANGALORE ITSP NATIONAL NUMBERS
  e164 ^.....$
dial-peer voice 1 voip
 description INBOUND FROM CUCM
 session protocol sipv2
 incoming called e164-pattern-map 2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
no vad
dial-peer voice 2 voip
```

```
description OUTBOUND TO ITSP
 session protocol sipv2
 session server-group 2
 destination e164-pattern-map 2
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 no vad
dial-peer voice 3 voip
 description INBOUND FROM ITSP
 session protocol sipv2
 incoming called e164-pattern-map 1
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 no vad
dial-peer voice 4 voip
 description OUTBOUND TO CUCM
 session protocol sipv2
 session server-group 1
 destination e164-pattern-map 1
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
no vad
```

- Now, monitor the status of the internal and external interfaces of CUBE- this is Gig1 and Gig2
- Monitoring the line-protocol status of the interfaces allows the redundancy application to determine when a resource has gone offline
- These configurations have to be applied on CUBE01 and CUBE02

CUBE01

```
!
track 1 interface GigabitEthernet1 line-protocol
track 2 interface GigabitEthernet2 line-protocol
!

CUBEO2
!
track 1 interface GigabitEthernet1 line-protocol
track 2 interface GigabitEthernet2 line-protocol
```

Configure the Redundancy Application that tracks Gig1 and Gig2 status

```
CUBE01
                                         CUBE02
!
redundancy
                                         redundancy
application redundancy
                                         application redundancy
group 1
                                         group 1
name CUBE-HA
                                         name CUBE-HA
priority 101
                                         priority 100
timers delay 30 reload 60
                                         timers delay 30 reload 60
control GigabitEthernet3 protocol 1
                                         control GigabitEthernet3 protocol 1
data GigabitEthernet3
                                         data GigabitEthernet3
track 1 shutdown
                                         track 1 shutdown
track 2 shutdown
                                         track 2 shutdown
ļ
```

Enable the redundancy in the voice service configuration of each CUBE

```
CUBE01

!

voice service voip

redundancy-group 1
!

!
```

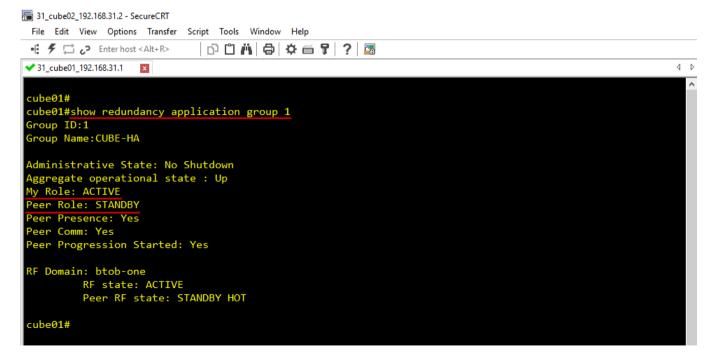
- Now let's configure the internal and external Virtual Interfaces
- The Command "redundancy rii" (Redundant Interface Identifier) is used for generating a Virtual IP (VIP) and Virtual MAC (VMAC)

CUBE01

```
!
interface GigabitEthernet1
redundancy rii 1
redundancy group 1 IP 192.168.31.3 exclusive
interface GigabitEthernet2
redundancy rii 2
redundancy group 1 IP 192.168.32.3 exclusive
!

CUBEO2
!
interface GigabitEthernet1
redundancy rii 1
redundancy group 1 IP 192.168.31.3 exclusive
interface GigabitEthernet2
redundancy rii 2
redundancy group 1 IP 192.168.32.3 exclusive
!
```

- show redundancy application group 1 OR (show redundancy application group) command will give the status of HA
- show redundancy application transport group 1





```
v 31_cube02_192.168.31.2 ▼

cube02#

cube02#show redundancy application group 1

Group ID:1

Group Name:CUBE-HA

Administrative State: No Shutdown

Aggregate operational state: Up

My Role: STANDBY

Peer Role: ACTIVE

Peer Presence: Yes

Peer Comm: Yes

Peer Progression Started: Yes

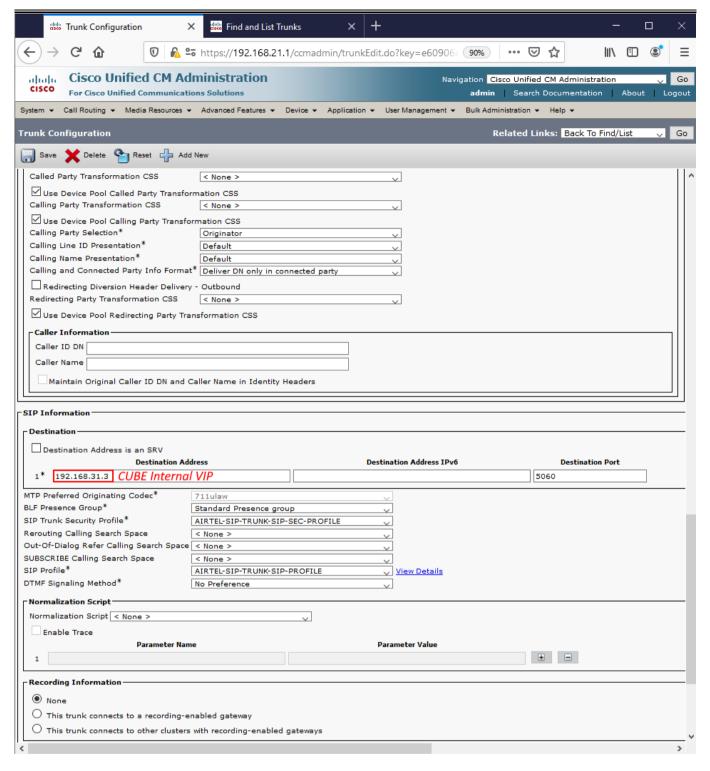
RF Domain: btob-one

    RF state: STANDBY HOT

    Peer RF state: ACTIVE

cube02#
```

- redundancy application reload group 1 self command used to flip between Active and Standby state
- Now configure CUCM SIP Trunk Destination to VIP: 192.168.31.3



Also communicate to the ITS that you will be sending SIP traffic from the external VIP:

192.168.32.3

When CUBE01 Active

DATA TO BE COLLECTED VALUES

 Calling Number
 8056311002

 Called Number
 8044260389

Time of the call 18:50:26(Approx.)

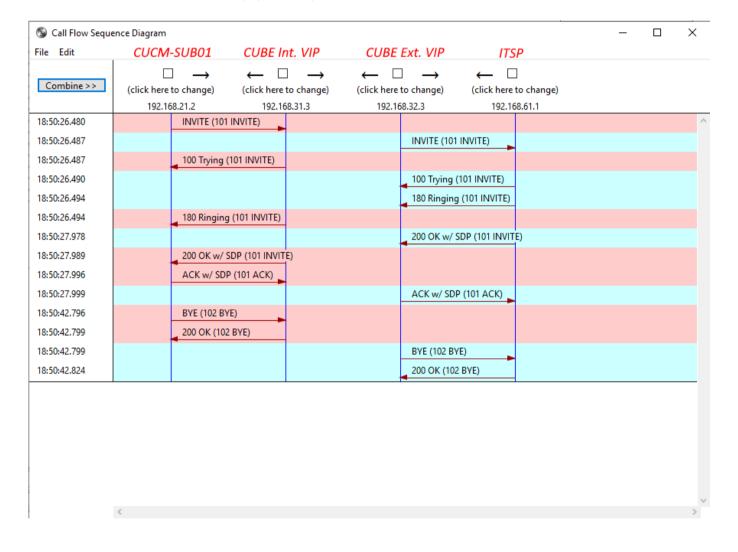
Duration of the call 16 Sec (Approx.)

Internal Phone IP 192.168.128.1

CUCM IPs 192.168.21.1, 192.168.21.2, 192.168.21.3

CUBE Internal IP 192.168.31.3 (VIP)
CUBE External IP 192.168.32.3 (VIP)

ITSP IP 192.168.61.1 ITSP End device (if possible) 192.168.130.2



When CUBE02 Active

DATA TO BE COLLECTED VALUES

 Calling Number
 8056311002

 Called Number
 8044260389

Time of the call 20:55:57 (Approx.)

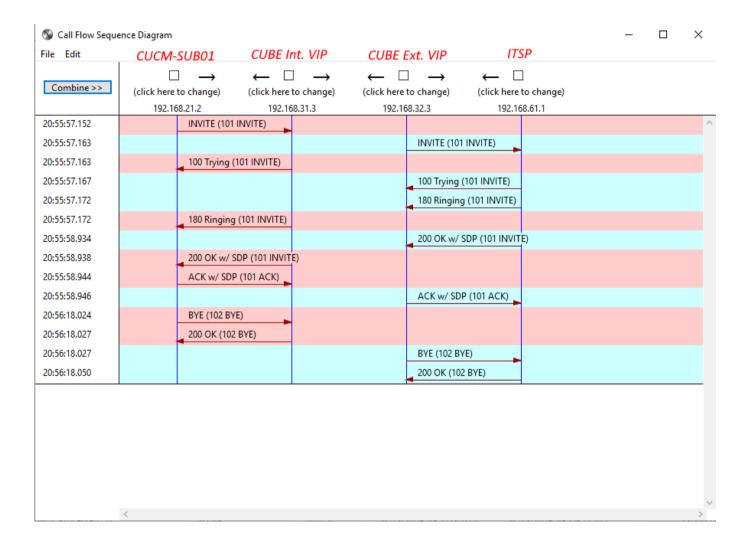
Duration of the call 21 Sec (Approx.)

Internal Phone IP 192.168.128.1

CUCM IPs 192.168.21.1, 192.168.21.2, 192.168.21.3

CUBE Internal IP 192.168.31.3 (VIP)
CUBE External IP 192.168.32.3 (VIP)

ITSP IP 192.168.61.1 ITSP End device (if possible) 192.168.130.2



CUCM Digit Manipulation



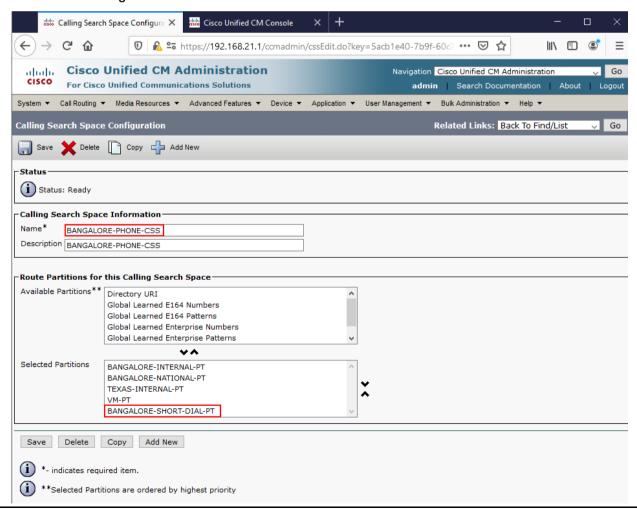
- Automatic Number Identifier ANI (Calling Number / Originating Number) and Dialed Number Identification Service - DNIS (Called Number / Destination Number / Dilled Number) can be modified at different levels in CUCM
- Complex CUCM Dial Plan contain different modification of digit at various degrees
- Digits can be modified from Translation Pattern, Transformation Pattern, Route Pattern, Route List-Route Group level, etc.

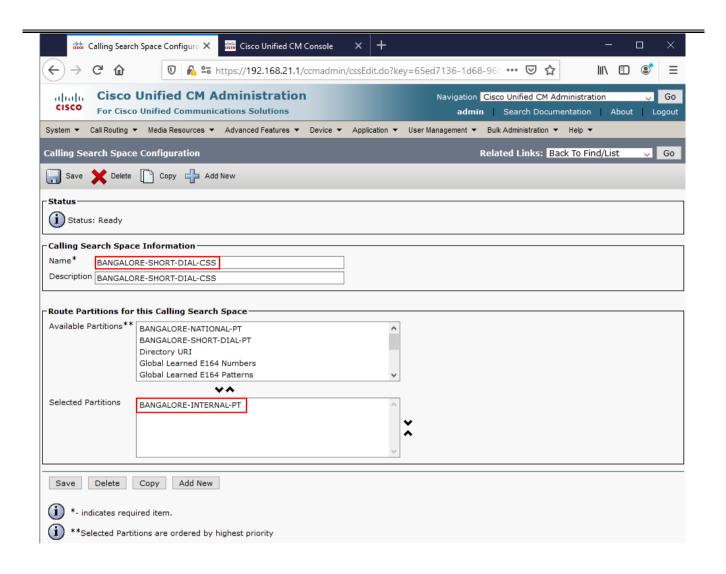
Translation Pattern

- Translation Pattern has the highest preference while matching a DNIS
- In our cluster, the Bangalore location uses 11XXX and Texas location uses 21XXX
- There is a requirement to implement short dial for intra location calls
- That means when Bangalore Phone calls another Bangalore Phone, user just has to dial 1XXX only (not 11XXX)
- Similarly, when Texas Phone calls another Texas Phone, use has to dial 1XXX (not 21XXX)
- For intra location calls, user has to dial the complete extension number
- This can be implemented by Translation pattern

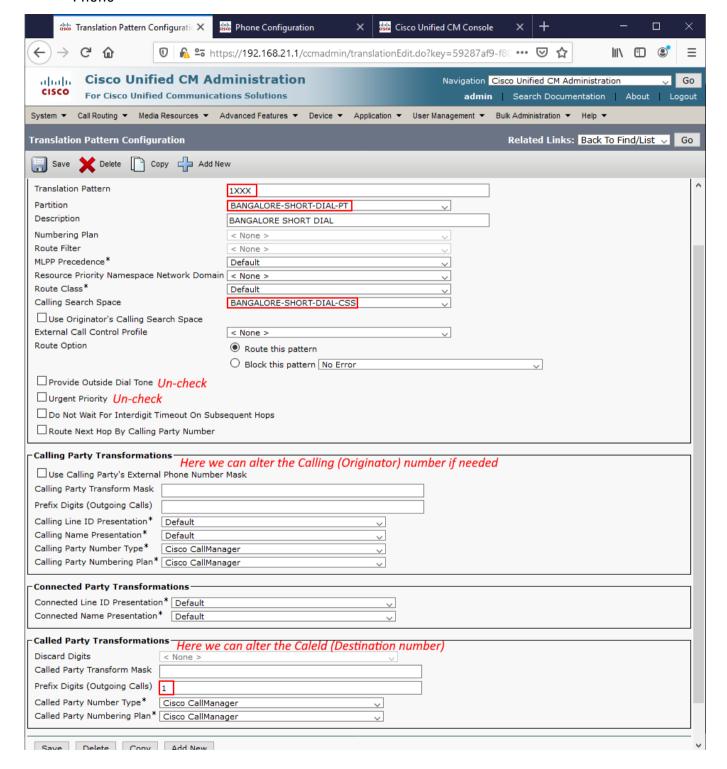
PATTERN	PARTITON	CSS	PREFIX	RESULT
1XXX	BANGALORE-SHORT-DIAL-PT	BANGALORE-SHORT-DIAL-CSS	1	11XXX
E.g. 1001				E.g. 11001

- Add BANGALORE-SHORT-DIAL-PT to BANGALORE-PHONE-CSS so that the Bangalore Phones can reach 1XXX translation pattern
- Add BANGALORE-INTERNAL-PT to BANGALORE-SHORT-DIAL-CSS so that the Translation Pattern can reach Bangalore Phones

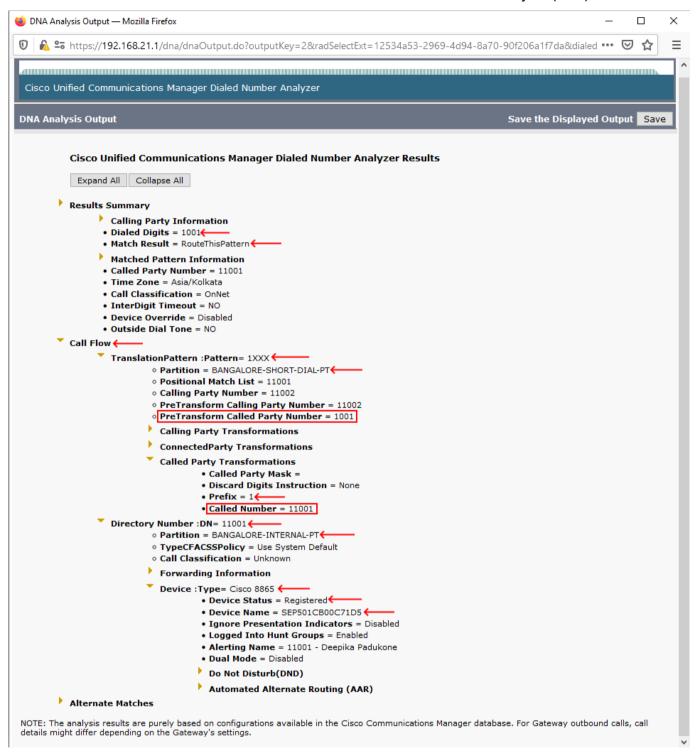




- This Translation Pattern 1XXX will add a prefix 1 to the Called (Dialed or Destination) number,
 hence the final called number becomes 11XXX
- Proper Partition will ensure that other location phones will not be able to access this pattern
- Proper CSS will help to further route the call after adding prefix to the appropriate Bangalore
 Phone

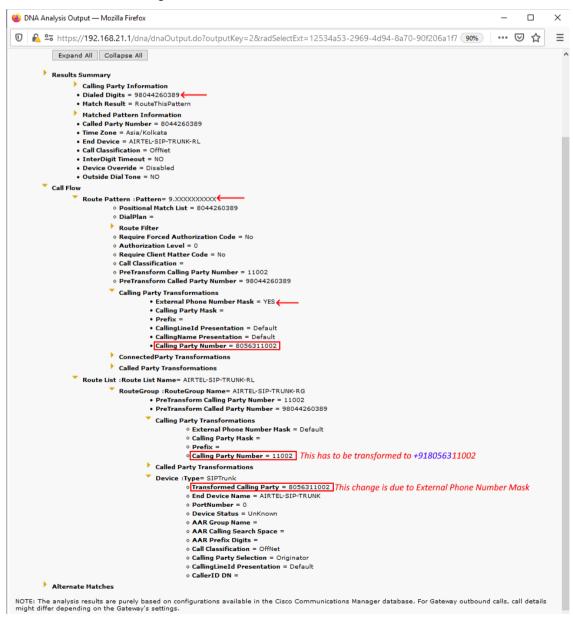


We could see the entire call flow and modification in Dialed Number Analyzer (DNA)



Calling and Called Party Transformations

- Calls through transformation patterns are not routable. It is just to modify the calling and called number presentation at the exit point
- This will match after the call routing completed and right before sending the call to next hope
- There is a requirement from the ITSP that when we route the calls to ITSP via CUBE, the calling and called numbers should be in e.164 format, i.e. 80563110XX has to be +9180563110XX,
 Otherwise ITSP will drop the call
- We have used External Phone Number Mask in the Route Pattern configuration, that will ensure the 80563110XX as the calling number instead of 110XX that goes to ITSP
- We can alter the External Phone Number mask to +9180563110XX to achieve the requirement, but
 we will end up editing External Phone number mask of all the phones
- Let's see the current calling behavior



Invite received at the CUBE with External Phone Number Mask

```
*Mar 31 11:15:15.988: //-1/xxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:8044260389@192.168.31.3:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK3f500da025
From: "11002 - Abdul Jaseem" <sip: 80563110020192.168.21.2>; tag=79~3f48b4d4-2f05-
4b61-86ad-47522208ed69-43078165
To: <sip:8044260389@192.168.31.3>
Date: Wed, 31 Mar 2021 11:15:16 GMT
Call-ID: 5d6eff80-64159c4-40-215a8c0@192.168.21.2
Supported: timer, resource-priority, replaces
Min-SE:
        1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback, X-cisco-original-called
Call-Info: <sip:192.168.21.2:5060>;method="NOTIFY;Event=telephone-
event; Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 1567555456-0000065536-0000000002-0034973888
Session-Expires: 1800
P-Asserted-Identity: "11002 - Abdul Jaseem" <sip:8056311002@192.168.21.2>
Remote-Party-ID: "11002 - Abdul Jaseem"
<sip:8056311002@192.168.21.2>;party=calling;screen=yes;privacy=off
Contact: <sip:8056311002@192.168.21.2:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0
```

Invite received at the CUBE with Calling Party Transformation Mask

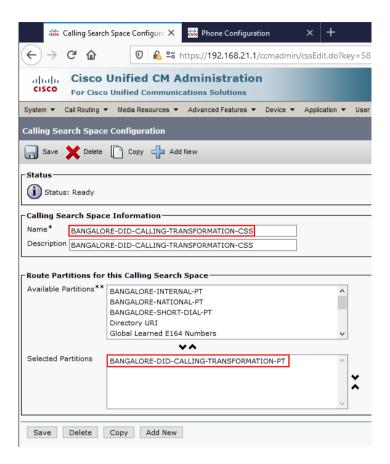
```
*Apr 1 17:18:18.032: //-1/xxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:8044260389@192.168.31.3:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK617da17d37
From: "11002 - Abdul Jaseem" \langle sip : +918056311002@192.168.21.2 \rangle; tag=111~3f48b4d4-2f05-
4b61-86ad-47522208ed69-43507474
To: <sip:8044260389@192.168.31.3>
Date: Thu, 01 Apr 2021 17:18:16 GMT
Call-ID: 3dbd0180-6610058-62-215a8c0@192.168.21.2
Supported: timer, resource-priority, replaces
Min-SE:
        1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback, X-cisco-original-called
Call-Info: <sip:192.168.21.2:5060>;method="NOTIFY;Event=telephone-
event; Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 1035796864-0000065536-0000000001-0034973888
Session-Expires: 1800
P-Asserted-Identity: "11002 - Abdul Jaseem" <sip:+918056311002@192.168.21.2>
Remote-Party-ID: "11002 - Abdul Jaseem"
<sip:+918056311002@192.168.21.2>;party=calling;screen=yes;privacy=off
Contact: <sip:+918056311002@192.168.21.2:5060;transport=tcp>;video;audio
Max-Forwards: 69
```

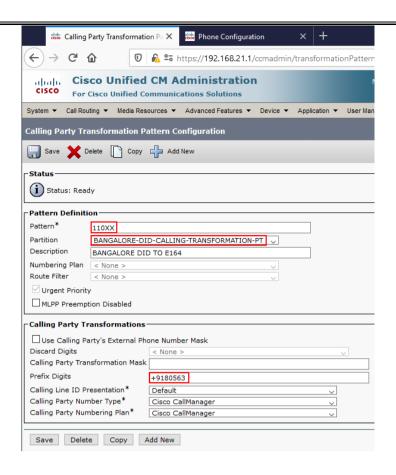
Max-Forwards: 69
Content-Length: 0

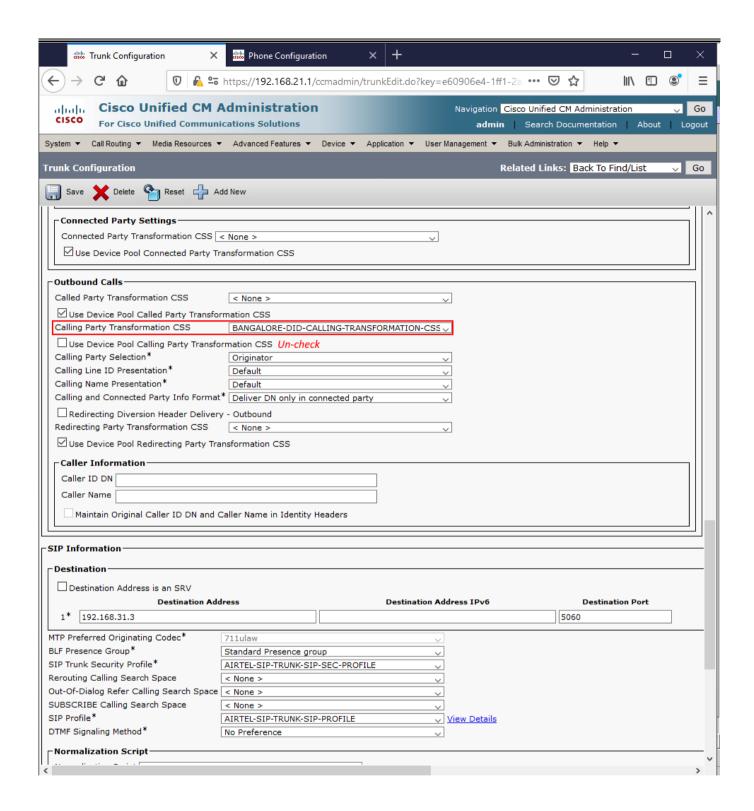
• Create a Partition and CSS for the Calling Party Transformation Mask

PATTERN PARTITON PREFIX RESULT

110XX BANGALORE-DID-CALLING-TRANSFORMATION-PT +9180563 +9180563110XX

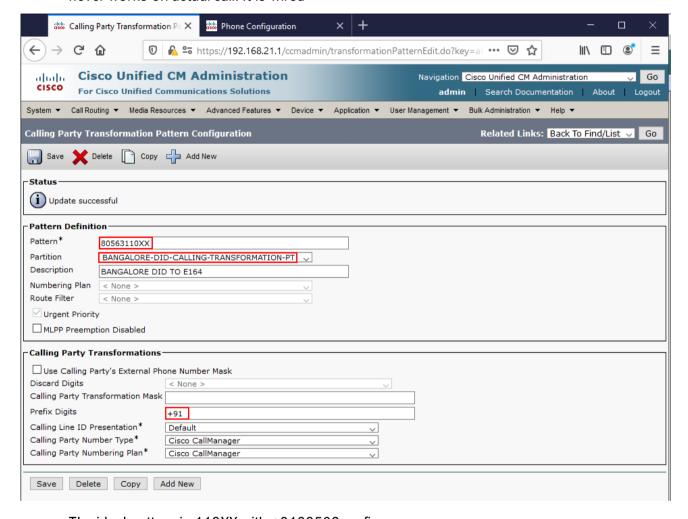






Note:

- Sometimes the modification works if we create the pattern for the External Phone Number mask format (shown below)
- DNA Result shows proper modification if we have pattern as 80563110XX with prefix 1, but it
 never works on actual call. It is wired



- The ideal pattern is 110XX with +9180563 prefix
- Always do real call testing while configuring Calling Party transformation Patterns
- Called Party Transformation will be used in the same way to modify the called (dialed or Destination number). I'm no showing the configuration as it affects my backend CUBE and ITSP routing if I modify the called number
- Other Digit Manipulation Technique such as Route Pattern Level, External Phone Number Mask,
 etc. are already discussed before

CUBE Digit Manipulation



- Apart from altering ANS and DNIS from CUCM level, we can modify numbers from CUBE level as well
- Voice Translation Profile are used to modify the digits whereas SIP Profile used to modify the SIP headers and SIP messages

Voice Translation Profile

- Voice Translation Rules are the method to modify calling (ANI) and called (DNIS) numbers in CUBE level
- Ideally, we keep CUBE dial plan minimal by doing all the translation and transformation at the
 CUCM level, but certain situation, we may have to implement digit manipulations at the CUBE level
- Scenario: We are not enabling External Phone Number mask and Calling Party transformation on CUCM; hence the calling party becomes 110XX that is being presented to ITSP
- ITSP is expecting calling party as the full DID 80563110XX
- We can apply CUBE level digit modification to archive this scenario

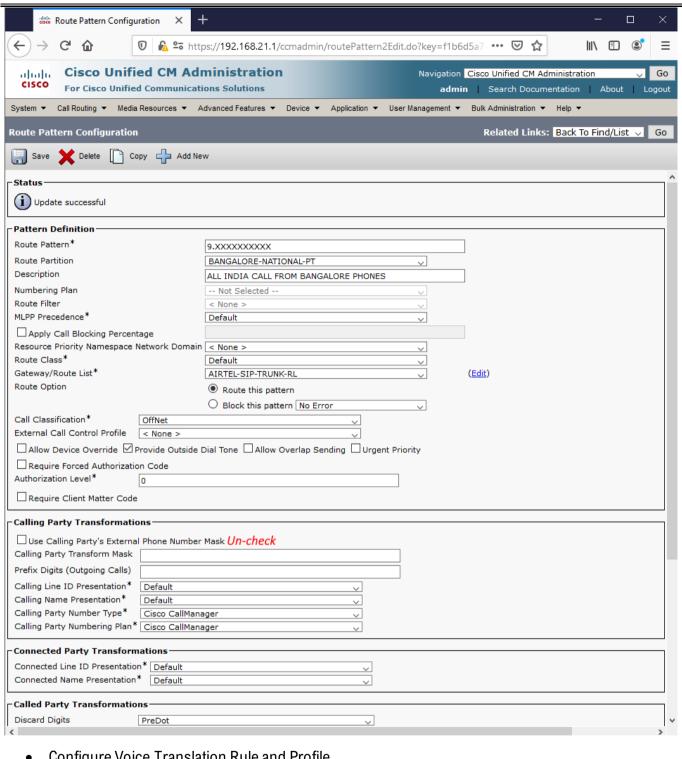
Wildcards	Use	Example	Result
	0 to 9 Match		
*	Match the character to the left 0 or	*	1., 2., . will repeat (Any
	more time		Match)
٨	Starting Match	/^14821/ /00123/	148219 = 001239
\$	Ending Match	/28148\$//00123/	728148 = 700123
^XXX\$	Exact Match	/^31148\$//00123/	3148 = 00123
[^X-Y]	Do not match a single digit from the range		
\(XXX\)	Creating set	\1	Calling set

Set Example:

Rule1 /^\(91\)35\(5551212\)480\(897\)\$/ /\1812\2465\3/

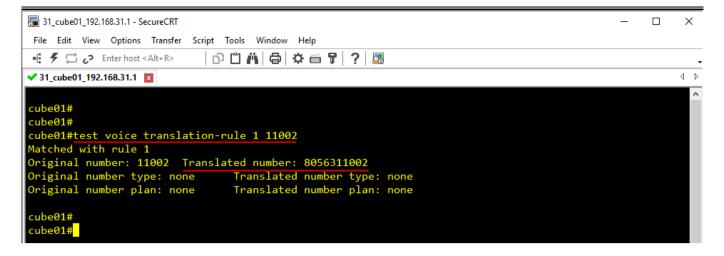
Result:

91 35 5551212 480 897 → 91 812 5551212 465 897



Configure Voice Translation Rule and Profile

```
voice translation-rule 1
 rule 1 /110\(..\)/ /80563110\1/
voice translation-profile ITSP-OUT
 translate calling 1
```



Apply the Voice Translation Profile under ITSP dial-peer

```
!
dial-peer voice 2 voip
description OUTBOUND TO ITSP
translation-profile outgoing ITSP-OUT
session protocol sipv2
session server-group 2
destination e164-pattern-map 2
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte
no vad
```

 Invite received at the CUBE after removing External Phone number mask and Calling Party transformation mask

```
*Apr 1 17:31:03.679: //-1/xxxxxxxxxxxX/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:8044260389@192.168.31.3:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK817b4b0b7a
From: "11002 - Abdul Jaseem" <sip: 11002@192.168.21.2>; tag=167~3f48b4d4-2f05-4b61-86ad-
47522208ed69-43507479
To: <sip:8044260389@192.168.31.3>
Date: Thu, 01 Apr 2021 17:31:03 GMT
Call-ID: 6e7eb00-6610357-82-215a8c0@192.168.21.2
Supported: timer, resource-priority, replaces
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback, X-cisco-original-called
Call-Info: <sip:192.168.21.2:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Cisco-Guid: 0115862272-0000065536-0000000003-0034973888
Session-Expires: 1800
P-Asserted-Identity: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>
Remote-Party-ID: "11002 - Abdul Jaseem"
<sip:11002@192.168.21.2>;party=calling;screen=yes;privacy=off
Contact: <sip:11002@192.168.21.2:5060;transport=tcp>;video;audio
Max-Forwards: 69
Content-Length: 0
```

 After applying the voice translation rule on the outbound dial-peer, we can see the calling number has been modified

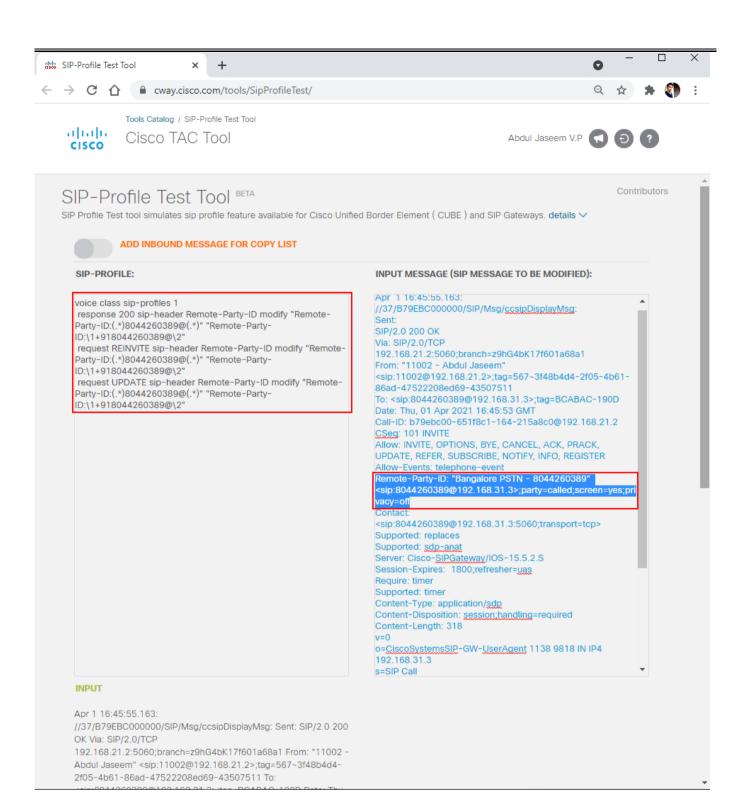
```
*Apr 1 17:31:03.684: //6/06E7EB000000/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:8044260389@192.168.61.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.32.3:5060;branch=z9hG4bK226A9
Remote-Party-ID: "11002 - Abdul Jaseem"
<sip:8056311002@192.168.32.3>;party=calling;screen=yes;privacy=off
From: "11002 - Abdul Jaseem" <sip: 8056311002@192.168.32.3>; tag=6FA932-1323
To: <sip:8044260389@192.168.61.1>
Date: Thu, 01 Apr 2021 17:31:03 GMT
Call-ID: DE1E5BC1-924611EB-80139BE5-FEC8FA29@192.168.32.3
Supported: timer, resource-priority, replaces, sdp-anat
Cisco-Guid: 0115862272-0000065536-0000000003-0034973888
User-Agent: Cisco-SIPGateway/IOS-15.5.2.S
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
CSeq: 101 INVITE
Timestamp: 1617298263
Contact: <sip:8056311002@192.168.32.3:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Session-Expires:
                  1800
Content-Length: 0
```

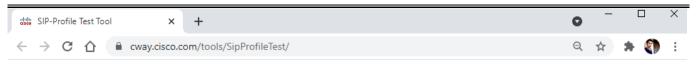
SIP Profile

- SIP Headers (From, To, Remote-Party-ID, Contact, etc.) can be modified using SIP profiles at
 CUBE level
- Previously we have seen voice translation rule to provide complete DID to the ITSP
- We have a requirement that when the call is connected to the ITSP end device, the Remote-Party-ID of ITS number has to be in E.164 format, that is +9198044260389 rather than 98044260389
- The +91 prefix must be added only after the call is connected to ITSP end device, while the phone ringing, there shouldn't be any change
- Usually Remote-Party-ID is presented in Invite, 180 Ringing, 2000K and Update. Since we need
 the modification after call connected, we modify the 2000K Remote-Party-ID header rather than
 Invite and 180 Ringing
- We must also consider REINVITE and UPDATE messages since those are used to refresh the session after session expire (ideally 15 minutes)

```
voice class sip-profiles 1
 response 200 sip-header Remote-Party-ID modify "Remote-Party-ID:(.*)8044260389@(.*)"
"Remote-Party-ID: \1+918044260389@\2"
 request REINVITE sip-header Remote-Party-ID modify "Remote-Party-ID:(.*)8044260389@(.*)"
"Remote-Party-ID: \1+918044260389@\2"
 request UPDATE sip-header Remote-Party-ID modify "Remote-Party-ID:(.*)8044260389@(.*)"
"Remote-Party-ID: \1+918044260389@\2"
dial-peer voice 1 voip
 description INBOUND FROM CUCM
 session protocol sipv2
 incoming called e164-pattern-map 2
 voice-class codec 1
 voice-class sip profiles 1
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 no vad
ı
```

- First "" Represents the match pattern and second "" represents the change pattern
- () represents a set in the match pattern and it can be called from change pattern using \X, where X is the number of set
- Above SIP Profile can be applied to the inbound dial peer from CUCM to CUBE
- You can use SIP Profile Test Tool https://cway.cisco.com/tools/SipProfileTest/ to verify the SIP Profile





Tools Catalog / SIP-Profile Test Tool

SIPGateway/IOS-15.5.2.S Session-Expires: 1800;refresher=uas Require: timer Supported: timer Content-Type: application/sdp Content-Disposition: session; handling=required Content-Length: 318 v=0 o=CiscoSystemsSIP-GW-UserAgent 1138 9818 IN IP4



Cisco TAC Tool







Abdul Jaseem V.P (1) (2) CISCO INPUT Apr 1 16:45:55 163: //37/B79EBC000000/SIP/Msg/ccsipDisplayMsg: Sent: SIP/2.0 200 OK Via: SIP/2 0/TCP 192.168.21.2:5060;branch=z9hG4bK17f601a68a1 From: "11002 -Abdul Jaseem" <sip:11002@192.168.21.2>;tag=567~3f48b4d4-2f05-4b61-86ad-47522208ed69-43507511 To: <sip:8044260389@192.168.31.3>;tag=BCABAC-190D Date: Thu, 01 Apr 2021 16:45:53 GMT Call-ID: b79ebc00-651f8c1-164-215a8c0@192.168.21.2 CSeq: 101 INVITE Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Allow-Events: telephone-event Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip_8044260389@192.168.31.3>;party=called;screen=yes;privacy=off Contact: <sip:8044260389@192.168.31.3:5060;transport=tcp> Supported: replaces Supported: sdp-anat Server: Cisco-SIPGateway/IOS-15.5.2.S Session-Expires: 1800;refresher=uas Require: timer Supported: timer Content-Type: application/sdp Content-Disposition: session; handling=required Content-Length: 318 v=0 o=CiscoSystemsSIP-GW-UserAgent 1138 9818 IN IP4 192.168.31.3 s=SIP Call c=IN IP4 192.168.31.3 t=0 0 m=audio 8064 RTP/AVP 0 8 18 101 c=IN IP4 192.168.31.3 a=rtpmap:0 PCMU/8000 a=rtpmap:8 PCMA/8000 a=rtpmap:18 G729/8000 a=fmtp:18 annexb=yes a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=ptime:20 OUTPUT Apr 1 16:45:55.163: //37/B79EBC000000/SIP/Msg/ccsipDisplayMsg: Sent: SIP/2.0 200 OK Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK17f601a68a1 From: "11002 -Abdul Jaseem" <sip:11002@192.168.21.2>;tag=567~3f48b4d4-2f05-4b61-86ad-47522208ed69-43507511 To: <sip:8044260389@192.168.31.3>;tag=BCABAC-190D Date: Thu, 01 Apr 2021 16:45:53 GMT Call-ID: b79ebc00-651f8c1-164-215a8c0@192.168.21.2 CSeq: 101 INVITE Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER Allow-Events: telephone-event Remote-Party-ID: "Bangalore PSTN - 8044260389" <sip:+918044260389@192.168.31.3>;party=called;screen=yes;privacy=off Contact: <sip:8044260389@192.168.31.3:5060;transport=tcp> Supported: replaces Supported: sdp-anat Server: Cisco-

2000K to CUCM before applying SIP Profile on dial-peer 1

```
Apr 1 16:45:55.163: //37/B79EBC000000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK17f601a68a1
From: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;tag=567~3f48b4d4-2f05-4b61-86ad-
47522208ed69-43507511
To: <sip:8044260389@192.168.31.3>;tag=BCABAC-190D
Date: Thu, 01 Apr 2021 16:45:53 GMT
Call-ID: b79ebc00-651f8c1-164-215a8c0@192.168.21.2
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
Allow-Events: telephone-event
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.31.3>;party=called;screen=yes;privacy=off
Contact: <sip:8044260389@192.168.31.3:5060;transport=tcp>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-15.5.2.S
Session-Expires: 1800; refresher=uas
Require: timer
Supported: timer
Content-Type: application/sdp
Content-Disposition: session; handling=required
Content-Length: 318
v=0
o=CiscoSystemsSIP-GW-UserAgent 1138 9818 IN IP4 192.168.31.3
s=SIP Call
c=IN IP4 192.168.31.3
t=0 0
m=audio 8064 RTP/AVP 0 8 18 101
c=IN IP4 192.168.31.3
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

2000K to CUCM after applying SIP Profile on dial-peer 1

Apr 1 16:50:12.493: //39/50CDD2800000/SIP/Msg/ccsipDisplayMsg:

Sent:

SIP/2.0 200 OK

Via: SIP/2.0/TCP 192.168.21.2:5060;branch=z9hG4bK18c7cad7293

From: "11002 - Abdul Jaseem" <sip:11002@192.168.21.2>;tag=587~3f48b4d4-2f05-4b61-86ad-

47522208ed69-43507513

To: <sip:8044260389@192.168.31.3>;tag=C0963A-16D6

Date: Thu, 01 Apr 2021 16:50:10 GMT

Call-ID: 50cdd280-651f9c2-16f-215a8c0@192.168.21.2

CSeq: 101 INVITE

Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,

REGISTER

Allow-Events: telephone-event

Remote-Party-ID: "Bangalore PSTN - 8044260389"

<sip:+918044260389@192.168.31.3>;party=called;screen=yes;privacy=off

Contact: <sip:8044260389@192.168.31.3:5060;transport=tcp>

Supported: replaces Supported: sdp-anat

Server: Cisco-SIPGateway/IOS-15.5.2.S
Session-Expires: 1800;refresher=uas

Require: timer Supported: timer

Content-Type: application/sdp

Content-Disposition: session; handling=required

Content-Length: 318

v=0

o=CiscoSystemsSIP-GW-UserAgent 5592 5791 IN IP4 192.168.31.3

s=SIP Call

c=IN IP4 192.168.31.3

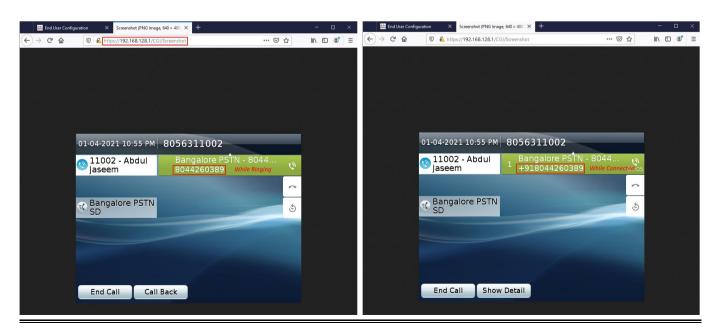
t=0 0

m=audio 8068 RTP/AVP 0 8 18 101

c=IN IP4 192.168.31.3
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15
a=ptime:20



SIP Normalization (Lua) Script



- SIP Normalization Script can be used to manipulate SIP Headers in CUCM, previously we have seen CUBE SIP Profile to manipulate SIP messages at the CUBE level
- We can learn SIP Normalization Script with an incident that I have faced in the past

Problem Description: External PSTN Call to CUC Mainline number (Auto Attendant) is getting Opening Greetings "Hello, Welcome to Cisco Unity Connection Messaging System" instead of Main Line Greetings

Data Collection:

- O When was the issues started?
- o Was the setup working before?
- o Is it a new implementation?
- What is the behavior when you dial the mainline from an internal extension (IP Phone)?
- Was there any recent UC Network Change?

DATA TO BE COLLECTED

- CallManager Logs, CUC Logs (Connection Conversation Manager, Connection Mixer) and CUBE (debug ccsip messages) Logs when external PSTN number calls the mainline number
- Cisco CallManager Logs and CUC Logs (Connection Conversation Manager) when an internal extension calls mainline number internally

VALUES

Call Sample 1 (Problematic Call)				
External Calling Number	8044260389			
Called Number (Main Line)	8056310002			
Time of the call	21:18 (Approx.)			
Duration of the call	14 Sec (Approx.)			
CUCM IPs	192.168.21.1, 192.168.21.2, 192.168.21.3			
CUBE Internal IP	192.168.31.1			
CUBE External IP	192.168.32.1			

ITSP IP	192.168.61.1
Call Sample 2 (Working Call)	
Internal Calling Number	11002
Called Number (Main Line)	10002
Time of the call	21:18 (Approx.)
Duration of the call	14 Sec (Approx.)
CUCM IPs	192.168.21.1, 192.168.21.2, 192.168.21.3
CUBE Internal IP	192.168.31.1
CUBE External IP	192.168.32.1

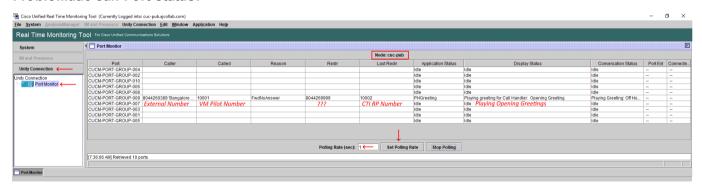
Initial Analysis and Debugging

ITSP IP

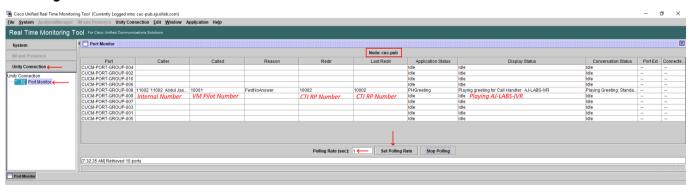
 Connect RTMT to CUC and go to Port Status Monitor, we will be able to see the CUC Calls in the Port Status Monitor Section

192.168.61.1

Problematic Call Port Status:

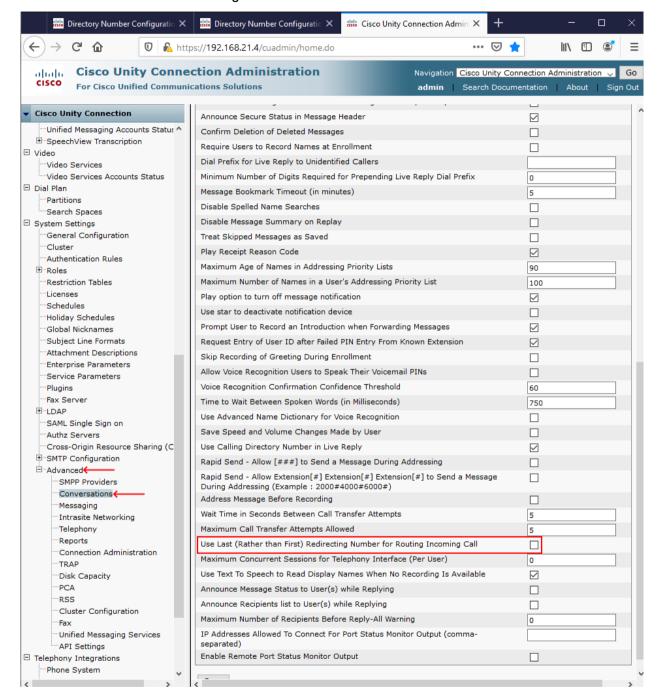


Working Call Port Status:



Initial Findings:

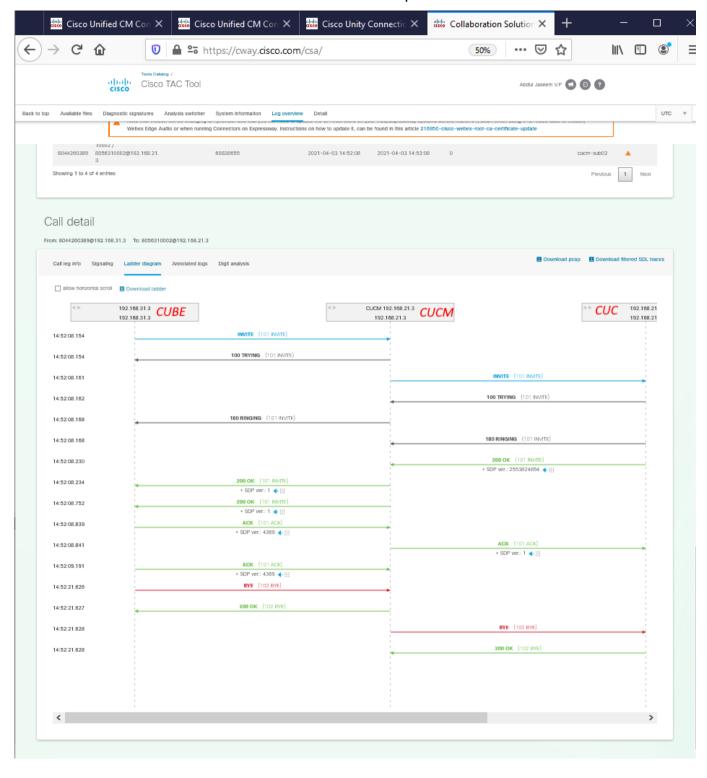
- We could see a bogus Redirecting Number 8044269999 in the call setup
- By default, CUC process the call based on the First Redirecting Number as specified on the Advanced Conversation Settings in CUC



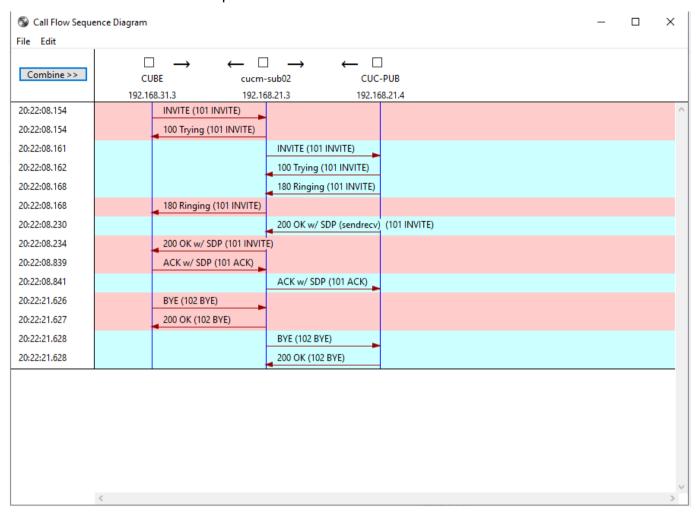
- CUC will look for the First Redirecting Number 8044269999 to route the call and there is no Voicemail Box or Call Handler with 8044269999
- Since First Redirecting Number is not 10002, CUC fails to deliver the call to 10002 Call Handler
- Now we need to identify how the bogus number is getting added in the external call

Cisco CallManager Logs from CUCM

- I have used Cisco Collaboration Solution Analyzer (CSA) Tool to track the call
- TranslatorX can also be used but more details will be processed in CSA Tool



• Here is the TranslatorX representation of the call



INVITE from CUBE to CUCM

```
00095448.002 | 20:22:08.154 | AppInfo | //SIP/SIPUdp/wait SdlDataInd: Incoming SIP UDP message
size 979 from 192.168.31.3:[51267]:
[6530, NET]
INVITE sip:8056310002@192.168.21.3 SIP/2.0
Via: SIP/2.0/UDP 192.168.31.3:5060;branch=z9hG4bK3912D1
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.31.3>;party=calling;screen=yes;privacy=off
From: "Bangalore PSTN - 8044260389" <sip: 8044260389@192.168.31.3>;tag=1D79C48-250B
To: <sip:80563100020192.168.21.3>
Date: Sat, 03 Apr 2021 14:52:08 GMT
Call-ID: FF51ED43-93C211EB-80AF8068-7CAB84EE@192.168.31.3
Supported: timer, resource-priority, replaces, sdp-anat
Min-SE: 1800
Cisco-Guid: 0678289792-0000065536-0000000029-0020818112
User-Agent: Cisco-SIPGateway/IOS-15.5.2.S
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
CSeq: 101 INVITE
Timestamp: 1617461528
Contact: <sip:8044260389@192.168.31.3:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Diversion: <sip:8044269999@192.168.61.1>; privacy=off; reason=unconditional; screen=yes
Session-Expires: 1800
Content-Length: 0
```

- Here we could see that Diversion Header is available when the call landed at CUCM from CUBE
- Also, the Domain Part of Diversion Header is 192.168.61.1 which doesn't belong to our network
- You can verify what is 192.168.61.1 because that is the entity added this diversion header
- Since the bogus Diversion is in the Invite from CUBE, we need to check CUBE debugs to further isolate the issue

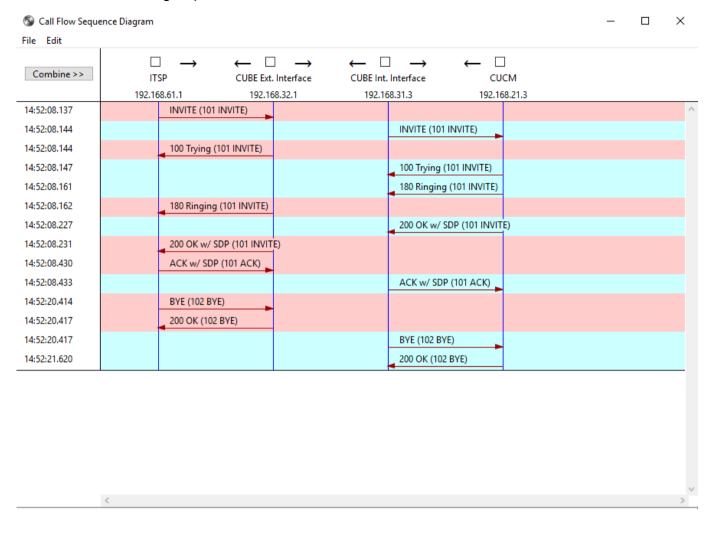
INVITE from CUCM to CUC

```
00095532.001 | 20:22:08.161 | AppInfo | SIPTcp - wait SdlSPISignal: Outgoing SIP TCP message
to 192.168.21.4 on port 5060 index 591
[6532, NET]
INVITE sip:10001@192.168.21.4:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.21.3:5060;branch=z9hG4bK8bd216e3bb5
From: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.21.3>;tag=3736~3f48b4d4-2f05-
4b61-86ad-47522208ed69-60828658
To: <sip:10001@192.168.21.4>
Date: Sat, 03 Apr 2021 14:52:08 GMT
Call-ID: 286de180-6818118-89e-315a8c0@192.168.21.3
Supported: timer, resource-priority, replaces
Min-SE: 1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSea: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=MIXED
Cisco-Guid: 0678289792-0000065536-0000000010-0051751104
Session-Expires: 1800
Diversion: "10002 - AJ-LABS-IVR"
<sip:10002@192.168.21.3>;reason=unconditional;privacy=off;screen=yes
Diversion: <sip:8044269999@192.168.61.1>; reason=unconditional; privacy=off; screen=yes
P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.21.3>
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.21.3>;party=calling;screen=yes;privacy=off
Contact: <sip:8044260389@192.168.21.3:5060;transport=tcp>
Max-Forwards: 68
Content-Length: 0
```

- Here we could see that First Redirecting Party is 8044269999 and Last Redirecting Party is 10002
 (CTI RP Number)
- Since the bogus number 8044269999 present in the INVITE, CUC unable to route the call to 10002
 Call Handler

CUBE Debugs

• For CUBE debugs, I prefer TranslatorX



INVITE received from ITSP

```
Apr 3 14:52:08.137: //-1/xxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:8056310002@192.168.32.1:5060 SIP/2.0
Via: SIP/2.0/TCP 192.168.61.1:5060; branch=z9hG4bK41448f6be7
From: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>;tag=49~6d68c7b1-6dc3-4f9e-
b3d4-c3249872f990-27361085
To: <sip:8056310002@192.168.32.1>
Date: Sat, 03 Apr 2021 14:52:08 GMT
Call-ID: 286de180-6818118-1d-13da8c0@192.168.61.1
Supported: timer, resource-priority, replaces
Min-SE:
       1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:192.168.61.1:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=VIDEO_UNSPECIFIED
Cisco-Guid: 0678289792-0000065536-0000000029-0020818112
Session-Expires: 1800
Diversion: <sip:8044269999@192.168.61.1>; reason=unconditional; privacy=off; screen=yes
P-Asserted-Identity: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.61.1>
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.61.1>:partv=calling:screen=ves:privacv=off
Contact: <sip:8044260389@192.168.61.1:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

- From this, it is clear that the Diversion Header is getting added from the ITSP provider
- We need to talk to ITSP to stop sending Diversion in the INVITE

What is Next?

- Do you think ITSP will resolve this immediately when you raise a case with them? They will take
 their own time to get this sorted. Well, will you be able to wait that long since this is affecting your
 business's main line number?
- We have to do some technique that can remove the bogus Diversion Header, you can either apply
 a SIP Profile at CUBE to remove it or SIP Normalization Script at CUCM at Trunk Level

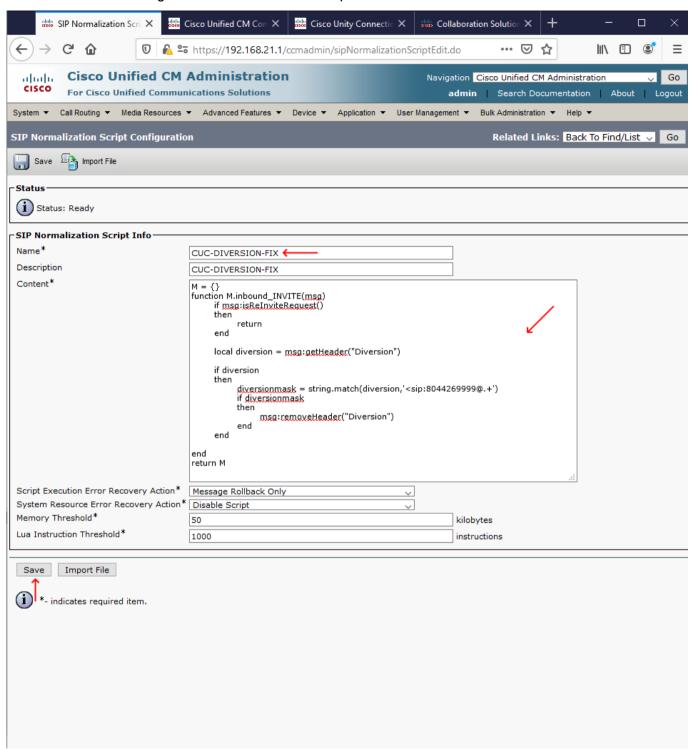
Resolution - SIP Normalization Script to Remove Diversion Header

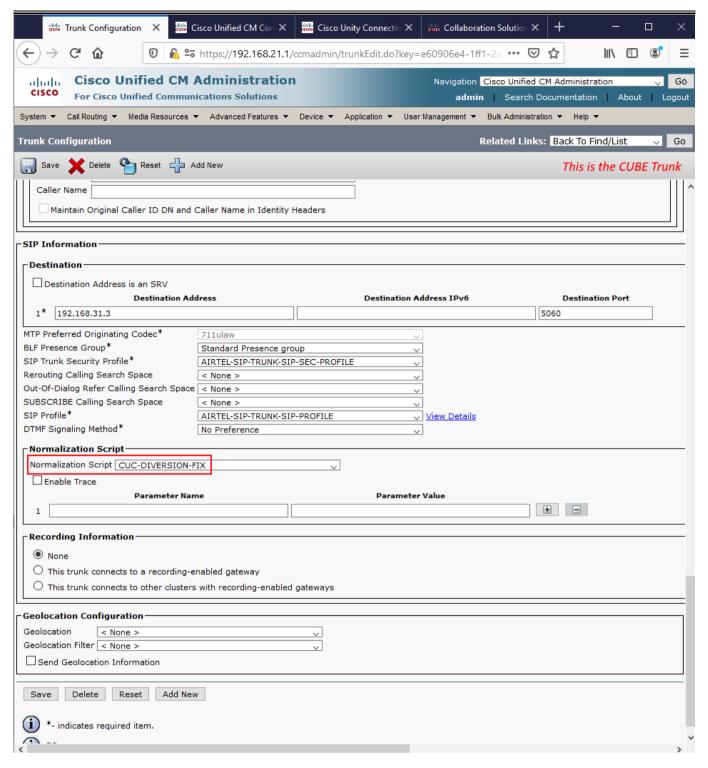
- SIP Normalization Scripts can be applied on SIP Trunks or SIP Profiles in CUCM
- In our case, Trunk level, we can either apply it on the CUCM-CUBE Trunk (AIRTEL-SIP-TRUNK)
 inbound direction or on CUCM-CUC Trunk (CUC-PUB-SIP-TRUNK, CUC-SUB-SIP-TRUNK)
- SIP Profile level can be done at AIRTEL-SIP-TRUNK-SIP-PROFILE or CUC-SIP-TRUNK-SIP-PROFILE
- Since CUC is always playing with Diversion Header, I don't prefer to apply anything that affects
 CUCM-CUC SIP Trunk
- I have decided to apply the Normalization Script on the CUCM-CUBE Trunk (AIRTEL-SIP-TRUNK)
 inbound direction
- Below Lua Script will remove the Diversion Header from Inbound INVITE SIP message

```
M = \{\}
function M.inbound INVITE(msg)
       if msg:isReInviteRequest()
       then
              return
      end
      local diversion = msg:getHeader("Diversion")
       if diversion
       then
             diversionmask = string.match(diversion, '<sip: 8044269999@.+')</pre>
              if diversionmask
              then
                    msg:removeHeader("Diversion")
              end
       end
end
return M
```

 All kind of fine tuning of SIP Messages are possible via Lua Script, here is the complete document to explore more on Lua Scripting: Developer Guide of SIP Normalization

Device >> Device Settings >> SIP Normalization Script >> Add New





A reset of SIP Trunk is required and that will terminate all the active call served by the SIPTrunk,
 hence recommended to apply this on off-business hours

Result

INVITE from CUBE to CUCM (before CUCM process the Script)

```
00111125.002 |21:49:31.668 | AppInfo | //SIP/SIPUdp/wait SdlDataInd: Incoming SIP UDP message
size 977 from 192.168.31.3:[52757]:
[7839, NET]
INVITE sip:8056310002@192.168.21.3 SIP/2.0
Via: SIP/2.0/UDP 192.168.31.3:5060;branch=z9hG4bK3C3B1
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.31.3>;party=calling;screen=yes;privacy=off
From: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.31.3>;tag=2279EC3-43B
To: <sip:8056310002@192.168.21.3>
Date: Sat, 03 Apr 2021 16:19:31 GMT
Call-ID: 34B2D0E2-93CF11EB-80B58068-7CAB84EE@192.168.31.3
Supported: timer, resource-priority, replaces, sdp-anat
Min-SE: 1800
Cisco-Guid: 1568682240-0000065536-0000000030-0020818112
User-Agent: Cisco-SIPGateway/IOS-15.5.2.S
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
CSea: 101 INVITE
Timestamp: 1617466771
Contact: <sip:8044260389@192.168.31.3:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Diversion: <sip:8044269999@192.168.61.1>; privacy=off; reason=unconditional; screen=yes
Session-Expires:
Content-Length: 0
     CUCM process the Script and Removes the Diversion
00111126.000 |21:49:31.676 |SdlSig
                                      |SIPNormalizeRea
                                                                              |wait
|SIPNormalization(3,100,81,1)
                                   |SIPHandler(3,100,82,1)
3,100,10,1.44^192.168.31.3^*
                                           |*TraceFlagOverrode
00111126.001 |21:49:31.676 |AppInfo |//SIP/SIPNormalization/trace_sip_message: After
inbound SIP Normalization msg is:
[7839, INT]
INVITE sip:8056310002@192.168.21.3 SIP/2.0
Date: Sat, 03 Apr 2021 16:19:31 GMT
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
From: "Bangalore PSTN - 8044260389" <sip:8044260389@192.168.31.3>;tag=2279EC3-43B
Allow-Events: telephone-event
Supported: timer, resource-priority, replaces, sdp-anat
Remote-Party-ID: "Bangalore PSTN - 8044260389"
<sip:8044260389@192.168.31.3>;party=calling;screen=yes;privacy=off
Cisco-Guid: 1568682240-0000065536-0000000030-0020818112
Content-Length: 0
User-Agent: Cisco-SIPGateway/IOS-15.5.2.S
To: <sip:8056310002@192.168.21.3>
Contact: <sip:8044260389@192.168.31.3:5060>
Expires: 180
Call-ID: 34B2D0E2-93CF11EB-80B58068-7CAB84EE@192.168.31.3
Via: SIP/2.0/UDP 192.168.31.3:5060;branch=z9hG4bK3C3B1
CSeq: 101 INVITE
Session-Expires:
                  1800
Max-Forwards: 69
```

Alternate Fix using CUBE SIP Profiles:

- This issue can also be addressed using SIP Profiles at the CUBE
- We can completely remove the Diversion Header by matching to the bogus number 8044269999
- Below are the configurations

```
!
voice class sip-profiles 2
  request INVITE sip-header Diversion modify "Diversion:(.*)8044269999@(.*)" ""
!
dial-peer voice 4 voip
  description OUTBOUND TO CUCM
  session protocol sipv2
  session server-group 1
  destination e164-pattern-map 1
  voice-class codec 1
  voice-class sip profiles 2
  voice-class sip bind control source-interface GigabitEthernet1
  voice-class sip bind media source-interface GigabitEthernet1
  dtmf-relay rtp-nte
  no vad
!
```

Other Alternative Fixes:

Apart from the Lua Script at CUCM and SIP Profile at CUBE, there are other possible solutions for this issue, but all of them affects the normal call routing in one or the other way,

- Disable 'Redirecting Diversion Header Delivery Inbound' on the CUBE Trunk
 This will remove the Diversion Header whatever calls that comes via the Trunk
- 2. Disable 'Redirecting Diversion Header Delivery Outbound' on CUC Trunk
 This parameter is required for Normal Call Routing to CUC, disabling this will entirely break CUC Voicemail
 and Call Handler integration
- 3. Enable 'Use Last (Rather than First) Redirecting Number for Routing Incoming Call'
 This will consider the last redirecting number while CUC routing the calls, this will change the entire call routing behavior of CUC including voicemail routing. Even though, it never breaks anything, but user experience will be changed

Inbound Dial Peer Match



- When a CUBE receives a call setup request, the gateway begins searching for an applicable incoming dial-peer for this call
- A dial-peer only needs to satisfy one of the 10 conditions for matching. It is not necessary for all
 the attributes to be configured in the dial-peer or that every attribute match the call setup
 information
- All dial-peers are searched based on the first match criteria. The gateway moves on to the next criteria only if no match is found
- After matching the inbound dial-peer, the properties of the specific dial-peer applied to the call behavior (like Codec, DTMF Relay, etc.)
- If none of the dial-peer matches, then the default dial-peer get applied (dial-peer 0)

Situation: A user having internal extension 1001 calls a PSTN number 9495860708

ANI: 1001

DNIS: 9495860708

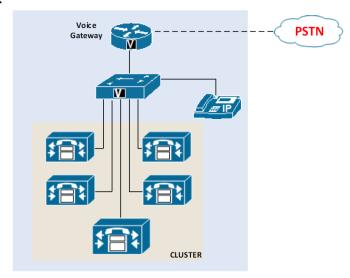
Preference	Match Criteria	Dial-peer Commands
1	URI	incoming uri via <uri-tag></uri-tag>
2	URI	incoming uri request <uri-tag></uri-tag>
3	URI	incoming uri to <uri-tag></uri-tag>
4	URI	incoming uri from <uri-tag></uri-tag>
5	Called Number	incoming called-number <dnis> incoming called e164-pattern-map <pattern-map-dnis-number></pattern-map-dnis-number></dnis>
6	Calling Number	answer-address <ani> incoming calling e164-pattern-map <pattern-map-ani-number></pattern-map-ani-number></ani>
7	Destination-pattern (ANI)	destination-pattern <ani></ani>
8	Port	Port through which call came in, this applies only for POTS dial-peer (PRI, FXO, FXS, etc.)
9	Carrier-ID	carrier-id source <string></string>
10	Default	dial-peer 0 (peer_tag=0, pid:0) This is a hidden dial-peer

Deployment Models



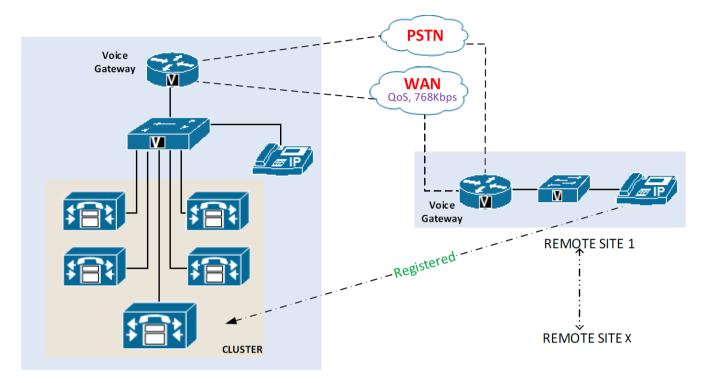
- CUCM Cluster can be deployed in many ways
- Since VoIP technology is network based; we can deploy the cluster that spread across the world

Single Site Deployment



- Call processing agent, DSP resources are located on single site, no telephony service provided over IP WAN. VoIP Traffic occupy in single site
- Number of device support is based on the OVA deployment and number of Subscriber nodes available
- Voice Gateway (with T1, E1, PRI, FXO, FXS) or CUBE used to connect PSTN for handling all external calls
- High bandwidth audio & Video communication between devices within the site, No transcoding required because only one codec used

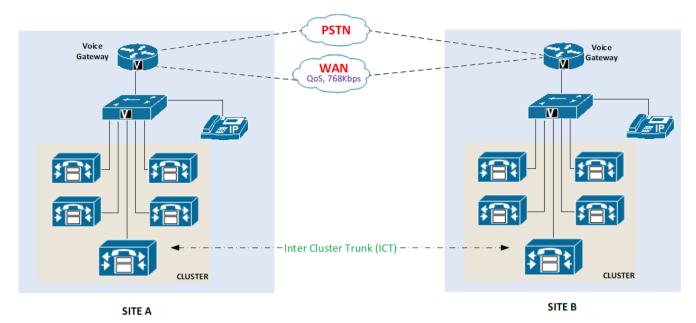
Multisite with Centralized Call Processing over WAN



CENTRAL SITE

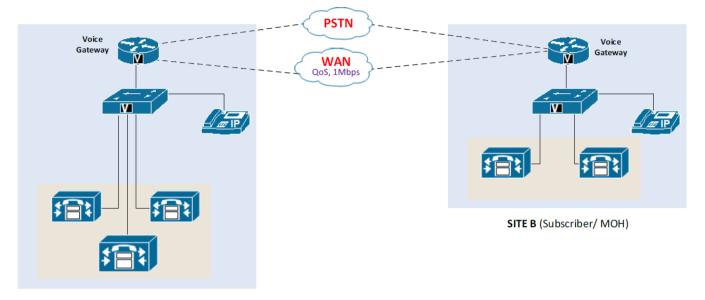
- Single call processing agent cluster provides telephony services to multiple remote sites over IP
 WAN
- Number of device support is based on the OVA deployment and number of Subscriber nodes available
- The remote site Voice Gateway have SRST capability to provide telephony service in the event if WAN failures
- External PSTN Calls are handled by site specific local Voice Gateway or CUBE
- We must have QoS enabled WAN to connect remote sites (768Kbps band width minimum).
- Call Admission control (use Location Mechanism) & AAR must be enabled for effective management of available bandwidth
- WAN connectivity options are Leased Lines, Frame Relay, ATM, MPLS VPN, IPSec VPN, etc.
- DSP resources are located at central site and remote sites to support local requirement
- High bandwidth audio between devices within the site & low bandwidth audio between devices in different site
- Free call between Central site and Remote sites over IP WAN
- TEHO (tail-end hop-off) can be implemented to bypass long distance PSTN calls via IP WAN
- Central dial-plan and Easy to administrate

Multisite with Distributed Call Processing over WAN



- Multiple independent sites each having its own call processing CUCM cluster
- Different sites are connected via SIP Trunks that uses IP WAN for network connectivity
- Each site can have again other remote sites
- External PSTN Calls are handled by site specific local Voice Gateway or CUBE
- WAN connectivity options are Leased Lines, Frame Relay, ATM, MPLS VPN, IPSec VPN, etc.
- High bandwidth audio between devices within the site, low bandwidth audio between devices in different sites
- Call Admission Control, AAR implemented to save bandwidth during inter cluster calls
- TEHO (tail-end hop-off) can be implemented to bypass long distance PSTN calls via IP WAN
- Session Management Edition (SME) used to interconnect if we have multiple clusters since it can reduce the dial-plan complexity and provide a kind of central dial-plan administration

Clustering over IP WAN



SITE A (Publisher/TFTP, Subsriber)

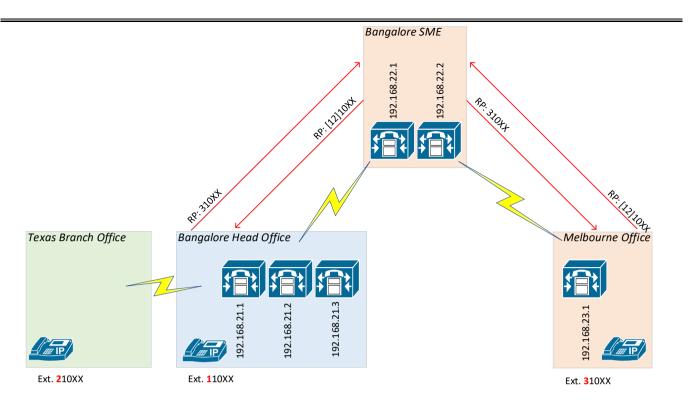
- Servers in same CUCM cluster located at different sites via IP WAN (QoS enabled 1Mbps minimum bandwidth)
- Round trip delay less than 80 msec for Database Replication over WAN
- Single point of administration, Unified Dial Plan

Note: I have seen many customers are using mixed deployment models. Multiple clusters with each cluster are distributed to different data centers, one cluster will have many remote sites with local gateways. SME interconnects all clusters for inter cluster call routing. Hence the deployment is not specific to only the above-described models. Consult Cisco Professional Service Team or Cisco Partners to get guidelines while implementing your infrastructure.

Advanced Inter Cluster Call Routing via Session Management Edition (SME)



- Cisco Unified Communications Manager Session Management Edition (SME) helps enterprises
 create a centralized architecture for Collaboration network
- It is just another CUCM Cluster that runs only Cisco CallManager feature service. Its primary function to interconnect other clusters together via SIP Trunk
- Imagine that we are expanding our AJCollab UC Network to some geographical areas and we
 planned to implement different CUCM Cluster apart from creating remote location
- We have installed a CUCM Cluster in Australia, now we have Bangalore CUCM Cluster supporting Bangalore and Texas Location and Australia CUCM Cluster supporting Melbourne location, we may extend our footprint in future to other countries
- Instead of creating multiple Inter Cluster SIP trunk between all these clusters (full mesh), we
 decided to have a central CUCM Cluster that acts as the bridge between remote clusters



- The CUCM Cluster that is connected to SME called Leaf Cluster, here we have Bangalore Leaf and Australia Leaf Clusters
- We create SIP Trunk between Bangalore CUCM and SME also Australia CUCM and SME
- This way, even if you expand the locations in future, the call routing configuration will be much easier, you just create Sip Trunk between SME and new CUCM cluster
- Ideally no phones will be registered in SME, but for testing purpose, you can register phones there. SME is just another CUCM

 Configuration Summary: I'm not adding detailed screenshots here since we already know how to create SIP Trunks. Make sure you create Region and Device Pool for respective trunks and set G.729 codec between Phone and SIP Trunk

1. Bangalore Cluster to SME

310XX in SME-PT

BLR-SME-RL

BLR-SME-RG

BLR-SME-SIP-TRUNK (192.168.22.1, 192.168.22.2) in BLR-SME-SIP-TRUNK-DP BLR-SME-SIP-TRUNK-INBOUND-CSS with BANGALORE-INTERNAL-PT, TEXAS-INTERNAL-PT) Update BANGALORE-PHONE-CSS and TEXAS-PHONE-CSS with SME-PT

2. SME to Bangalore Cluster

[12]10XX in BANGALORE-PT

SME-BLR-RL

SME-BLR-RG

SME-BLR-SIP-TRUNK (192.168.21.1, 192.168.21.2) in SME-BLR-SIP-TRUNK-DP SME-BLR-SIP-TRUNK-INBOUND-CSS with MELBOURNE-PT

3.SME to Australia Cluster

310XX in MELBOURNE -PT

SME-MEL-RL

SME-MEL-RG

SME-MEL-SIP-TRUNK (192.168.23.1) in SME-MEL-SIP-TRUNK-DP

SME-MEL-SIP-TRUNK-INBOUND-CSS with BANGALORE-PT

4. Australia Cluster to SME

[12]10XX in SME-PT

MEL-SME-RL

MEL-SME-RG

MEL-SME-SIP-TRUNK (192.168.22.1, 192.168.22.2) in MEL-SME-SIP-TRUNK-DP

MEL-SME-SIP-TRUNK-INBOUND-CSS with MELBOURNE-INTERNAL-PT)

Update MELBOURNE -PHONE-CSS with SME-PT

Inter Cluster Lookup Service (ILS)



- The ILS Intercluster Lookup Service feature enables different CUCM Cluster to exchange directory URI with other clusters in an ILS network, URI Replication provides support for Intercluster URI dialing
- ILS runs as a service on the Publisher node of a cluster
- We can call ILS Intercluster Lookup Service as Voice Dynamic Routing Protocol that advertise dial plan between clusters
- Each CUCM cluster node advertises its URIs and "SIP route string" to its neighbor's (or Hubs).
 CUCM cluster later creates a table with URIs and associated SIP route string
- Finally, SIP route strings are routed through SIP route patterns
- If you have just 2 clusters, we can make one as the ILS HUB and other as ILS SPOKE cluster, but ideally in with SME infrastructure, we make SME as the HUB Cluster
- All the CUCM Leaf Clusters will advertise the local URIs to SME HUB and SME HUB will advertise whatever it received to other clusters

Hub Cluster

- Each ILS network must have at least one hub cluster and they are the backbone of an ILS network
- Hub clusters exchange ILS updates with the other hub clusters in the ILS network and then relay that information to and from their spoke clusters
- You can connect a hub cluster to multiple other hub clusters, or you might configure a hub cluster
 as the only hub cluster in the network
- In addition, you can connect a hub cluster to multiple spoke clusters, or you might configure the hub cluster with no spoke clusters

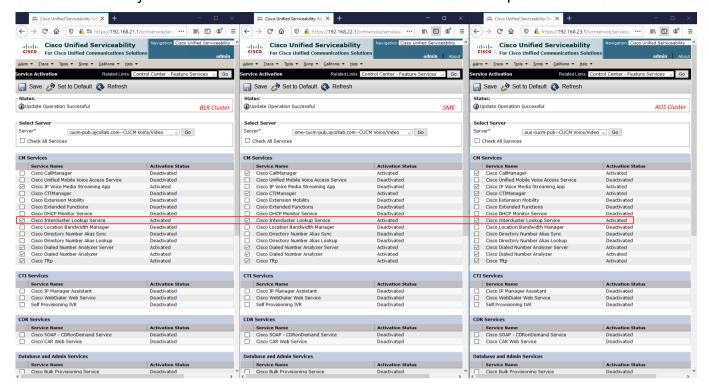
Spoke Cluster

- A spoke cluster in an ILS network relies on the hub cluster that it is connected to in order to relay
 ILS updates to and from the rest of the ILS network
- A spoke cluster can have only one hub cluster

[Lab] Inter Cluster Lookup Service (ILS) Configuration URI Dialing

Step 1: Activate Intercluster Lookup Service on all the CUCM Publisher

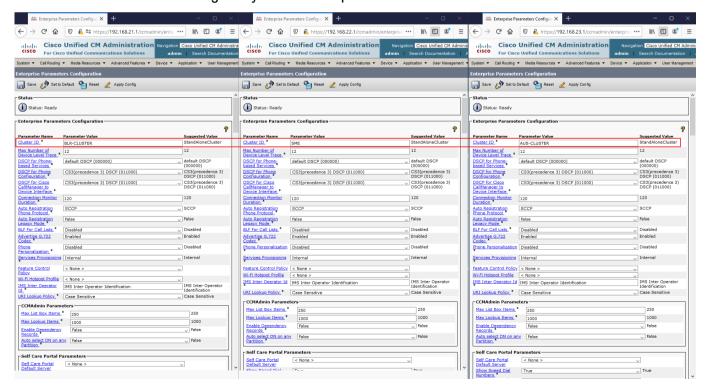
Go to Service Ability > Tools > Service Activation > Select 'Intercluster Lookup Service' and save



Step 2: Set Cluster ID for the CUCM Clusters

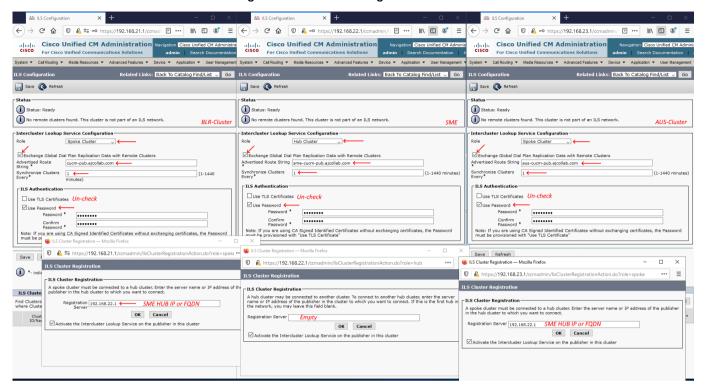
Cluster ID is a unique identifier for the cluster.

Go to CUCM Administration Page > System > Enterprise Parameter > Cluster ID

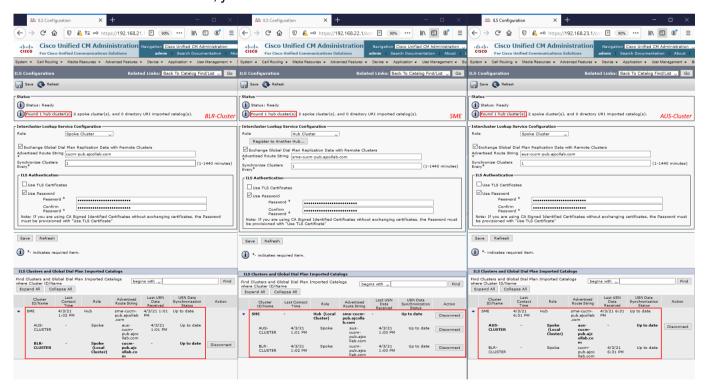


Step 3: ILS Configuration

Go to Advanced Features > ILS Configuration > and configure as follows



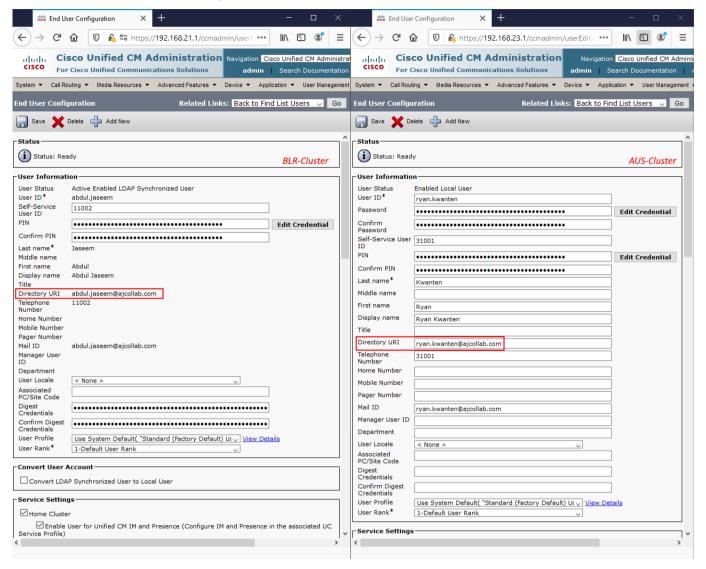
Refresh after 2 minutes, you will see the ILS Network has been established



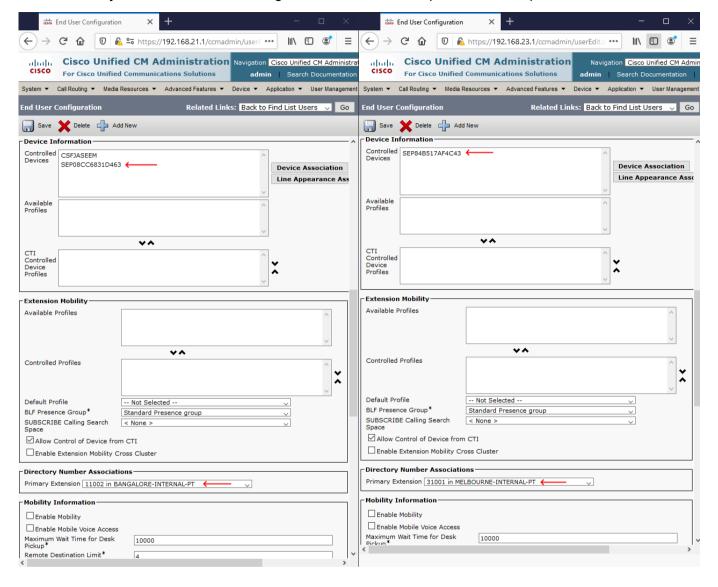
Step 4: Verify and Configure URIs

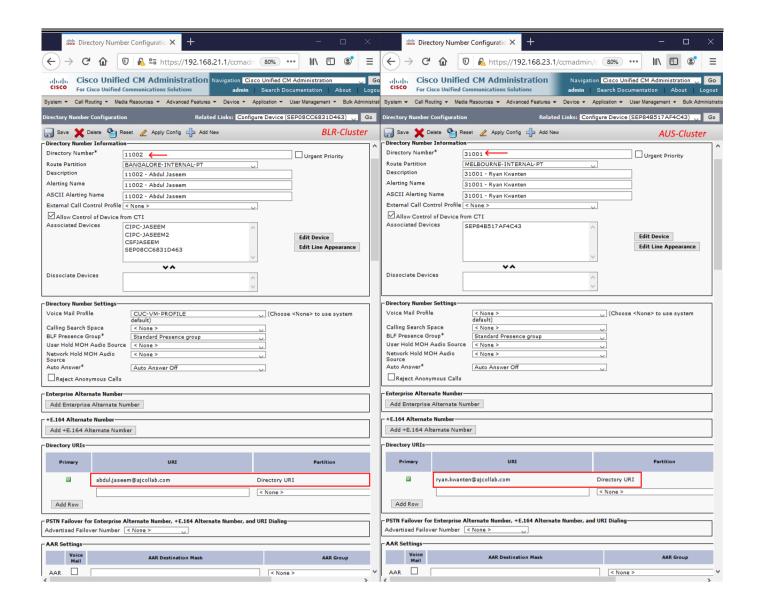
For Bangalore Clusters, the users are LDAP Sync, I do have an LDAP attribute that maps E-Mail address as Directory URI

In Australia Cluster, I have configured a local user with URI



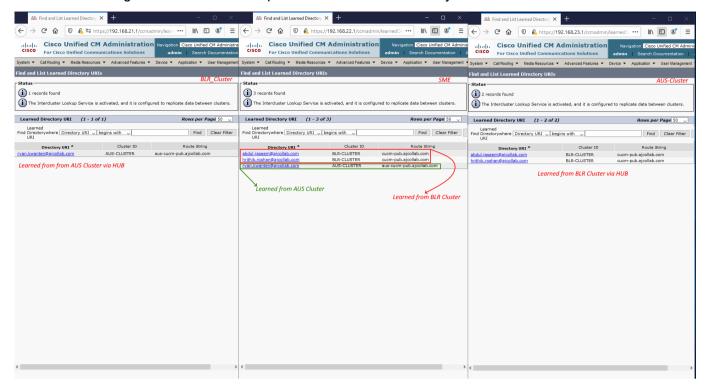
- Each directory URI will be attached to a DN in CUCM, this mapping is done on the Primary
 Extension field of End User configuration
- Primary extension must be configured, since this will map URI to the respective DN





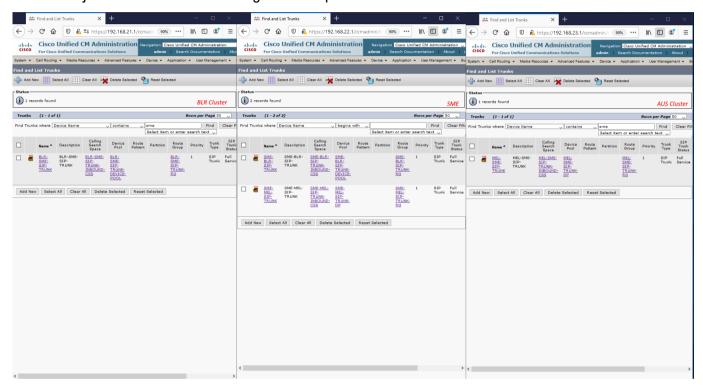
Step 5: Verify the Learned Directory URIs

Go to Call Routing > Global Dial Plan Replication > Learned Directory URIs

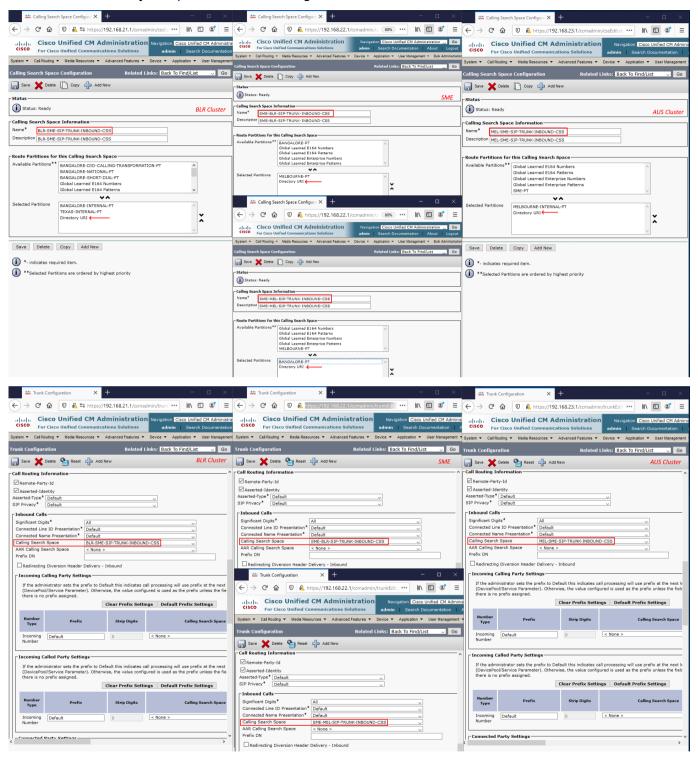


Step 6: SIP Trunk between Clusters

- Any of the URI learned via ILS will be having 2 unique values. One is the Route String and another
 one is Cluster ID. Route String is used to route the call back to the respective cluster via a
 separate SIP trunk
- ILS will only take care of advertising the URIs between clusters. They do not participate in call
 routing. For dialing the URI from one cluster to another, we need to have a separate SIP trunk
- We have already configured the SIP Trunk between BLR, SME and AUS Clusters for regular dialing,
 we just need to add URI dialing to the respective SIP trunks



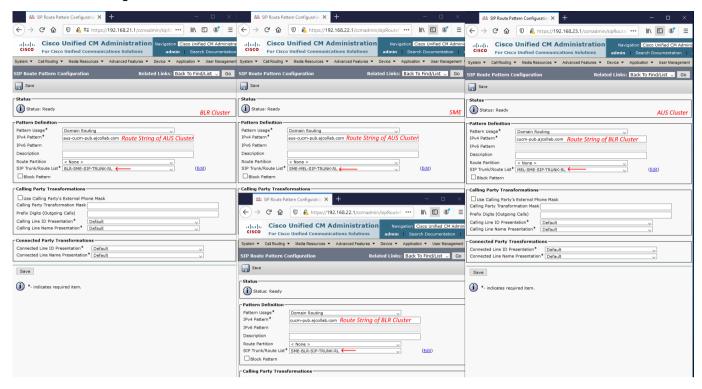
• By default, all the URIs will be assigned with 'Directory URI' partition. Hence you must have a CSS with Directory URI partition that is assigned as Inbound CSS at the SIP Trunks



Step 7: Configure SIP Route Pattern

The Advertise String of other ILS node will be the SIP Route Pattern.

Go to Call Routing > SIP Route Pattern > Add New and Point the SIP Trunk.



Step 8: Add Speed Dials to Test URI Dialing Phone Configuration Phone Configuration ← → C û l l https://192.168.23.1/ccmadmin/phoneEdi ··· II I ← → C 🔐 🕡 🖟 % https://192.168.21.1/ccmadmin/phon ••• 🕪 🗈 🖹 որոր. Cisco Unified CM Administration IIIIIII Cisco Unified CM Administration Navigation Cisco Unified CM Administration Navigation Cisco Unified CM Admi For Cisco Unified Communications Solutions For Cisco Unified Communications Solutions admin | Search Documentation System ▼ Call Routing ▼ Media Resources ▼ Advanced Features ▼ Device ▼ Application ▼ User Management System ▼ Call Routing ▼ Media Resources ▼ Advanced Features ▼ Device ▼ Application ▼ User Management Phone Configuration Related Links: Back To Find/List Phone Configuration Related Links: Back To Find/List Save X Delete Copy Reset Apply Config Add New Save X Delete Copy Reset / Apply Config Add New (i) Status: Ready i Status: Ready BLR Cluster **AUS Cluster** Association Association Phone Type Phone Type Modify Button Items Product Type: Cisco 9971 Modify Button Items Product Type: Cisco 9971 Device Protocol: SIP Device Protocol: SIP 1 Ine [1] - 11002 in BANGALORE-INTERNAL-PT 1 •ms Line [1] - 31001 in MELBOURNE-INTERNAL-PT Real-time Device Status-Real-time Device Status 2 •7712 Line [2] - Add a new DN Line [2] - Add a new DN Registered with Cisco U Registration: Registered with Cisco Registration: 3 @ 98044260389 abdul.jaseem@ajcollab.com 192.168.128.1 192.168.131.1 Active Load ID: sip9971.9-4-2SR2-2 Active Load ID: sip9971.9-4-2SR2-2 4 @ ryan.kwanten@ajcollab.com Add a new SD Inactive Load ID: sip9971.9-4-2SR4-1 Inactive Load ID: sip9971.9-4-2SR4-1 5 Add a new SD 5 Can Add a new SD Download Status: None Download Status: None 6 Can Add a new SD Add a new SD Device is Active Device is Active ----- Unassigned Associated Items --------- Unassigned Associated Items ----7 Add a new SD Device is trusted Add a new SD Device is trusted MAC Address* 08CC6831D46 MAC Address 84B517AF4C43 All Calls All Calls Description 11002 - Abdul 31001 - Rvan K Add a new BLF Directed Call Park 9 • Add a new BLF Directed Call Park BANGALORE-F MELBOURNE-PH 10 Call Park 10 Call Park Common Device Configuration < None > Common Device Configuration < None > 11 Call Pickup 11 Call Pickup Phone Button Template* Phone Button Template* Standard 997: Standard 9971 12 CallBack 12 CallBack Softkey Template Standard User Softkey Template 13 Group Call Pickup 13 Group Call Pickup Common Phone Profile* Standard Con Common Phone Profile* Standard Com 14 Hunt Group Logout 14 Hunt Group Logout Calling Search Space Calling Search Space BANGALORE-F MELBOURNE-PH 15 Intercom [1] - Add a new Intercom 15 Intercom [1] - Add a new Intercom AAR Calling Search Space AAR Calling Search Space < None > < None > Media Resource Group List 16 Malicious Call Identification 16 Malicious Call Identification Media Resource Group List User Hold MOH Audio User Hold MOH Audio Source < None > 17 Meet Me Conference 17 Meet Me Conference < None > 18 Mobility 18 Mobility Network Hold MOH Audio < None > Source 19 Other Pickup 19 Other Pickup Hub_None Hub_None 20 Quality Reporting Tool 20 Quality Reporting Tool AAR Group AAR Group 21 Redial 21 Redial User Locale < None > User Locale < None > Add a new SURL 22 Add a new SURL 22 Network Locale Network Locale < None > < None > Built In Bridge* 23 Add a new BLF SD Built In Bridge* 23 Add a new BLF SD Privacy* Default Privacy* Default 24 Answer Oldest 24 Answer Oldest Device Mobility Mode* Device Mobility Mode³ Default Device Mobility: Default 25 Do Not Disturb 25 Do Not Disturb evice Mobility Phone Configuration X Screenshot (PNG Image, 640 X Phone Configuration X Screenshot (PNG Image, 640 X ← → C û 0 % ~ 192.168.128.1/CGI/Screenshot ... ||\ □ ◎ ≡ ← → C û 0 № 192.168.131.1/CGI/Screenshot 03-04-2021 3:43 PM 8056311002 04-03-2021 9:13 PM 396531001 11002 - Abdul Kwanten Jaseem 5 5 Bangalore PSTN SD abdul.jaseem@
ajcollab.com

→ ryan.kwanten@ ajcollab.com

New Call

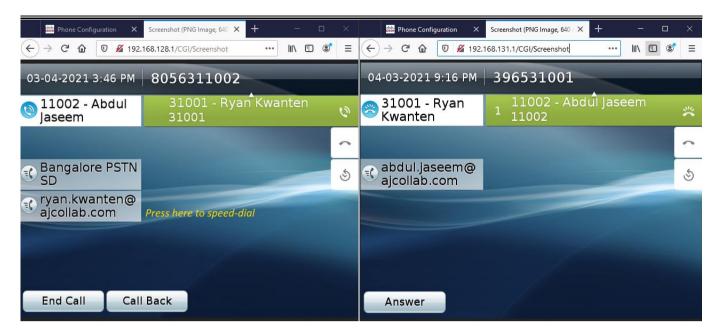
Forward All

Redial

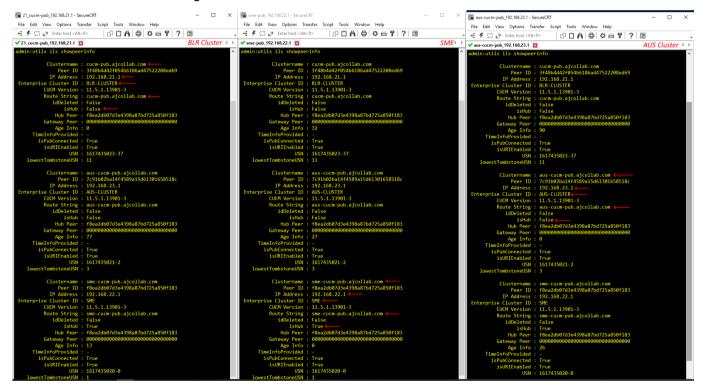
Call Back

Call Back

New Call Forward All Redial



 utils ils showpeerinfo CLI command will give a summary of ILS Network. This can be handy while troubleshooting ILS

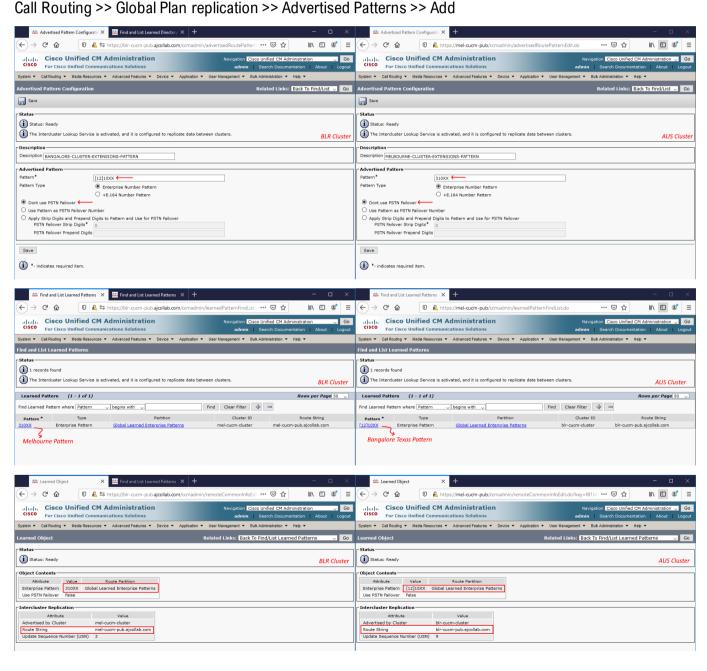


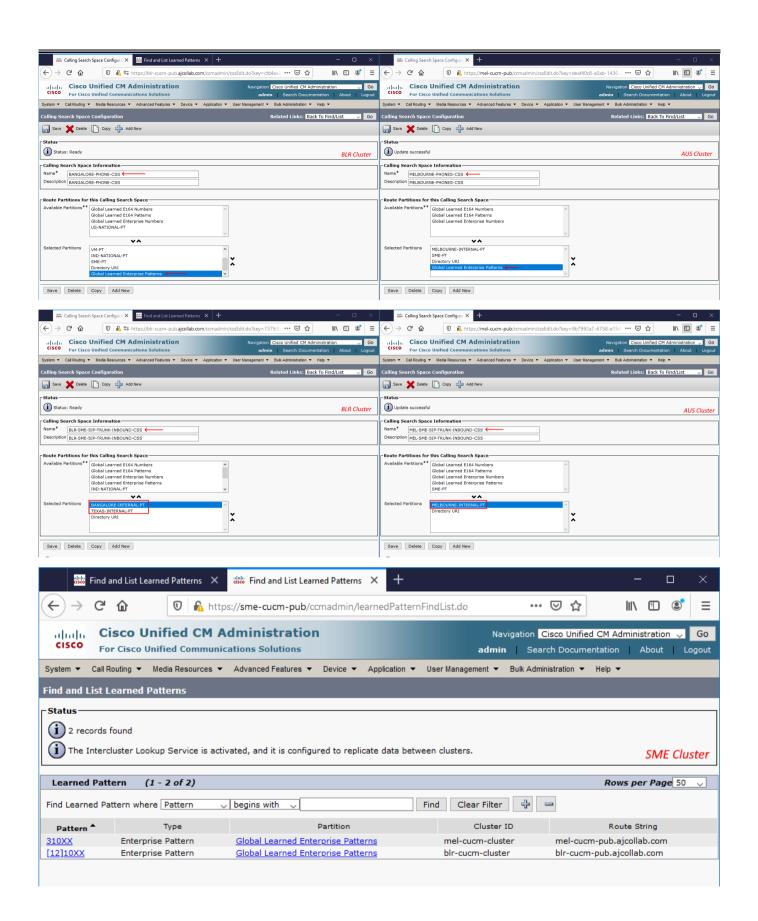
- ILS can be configured with TLS Authentication as well, for that we need to configure proper Tomcat Certificates in all the clusters, recommended to have CA Signed Multi SAN Certificates. While signing the certificate, please make sure the Common Name must match CUCM PUB FQDN (while generating Multi SAN CSR, there will be an '-m' added to the common name, this has to be removed manually before generating CSR)
- Detailed documentation of Certificate based ILS is explained here CUCM Inter Cluster Lookup
 Service

[Lab] Global Pattern Replication

- In our SME implementation, we have manually created route patterns 310XX and [21]10XX to route the call to SME and eventually from SME to respective leaf clusters
- It requires log of configuration overhead and effort, we can achieve this using Advertise Pattern in Global Dial plan Replication
- on Bangalore cluster we create [12]10XX pattern and advertise over ILS and in Melbourne cluster
 we create 310XX pattern and advertise over ILS
- Respective patterns will be advertised over SME HUB cluster along with its own Route String

 Call Pouting >> Clabel Plan replication >> Advertised Potterns >> Add





Media Resources

- Media Resources are Software or Hardware that support the media processing functions and features of VoIP network, CUCM itself plays most of these roles
- Voice Termination: Converting an audio into IP packet and vice versa. TDM legs must be terminated by hardware that performs coding/decoding and Packetizing of audio. This is performed by DSPs in the Gateway router. This is applicable to H.323 or MGCP Gateways
- Software based Media Resources are offered by CUCM Nodes where we Enable IP Voice Media
 Streaming App IPVMS service. Annunciator, MoH, Software CFB and Software MTP are activated
 with the IPVMS Service. After Enabling IPVMS, it recommended to put them in a dedicated region
 and device pool
- Hardware Media Resources are configured as SCCP endpoints in CUCM. The endpoint will be the DSP chips inside CUBE or Voice Gateway Router. Hardware CFB, Hardware MTP and XCODE are comes under this category

Software Only Media Resources

- Annunciator (ANN): Provides spoken messages ("The call cannot be completed as dialed...") and
 various tones (Ring Back, Fast Busy, etc.). Uses SCCP Signaling to get register to CUCM and alays
 establish one-way RTP stream
- Music On Hold (MOH): Provide music to the caller when their call is placed on Hold, Parked or during consultant transfer

Software or Hardware Media Resources

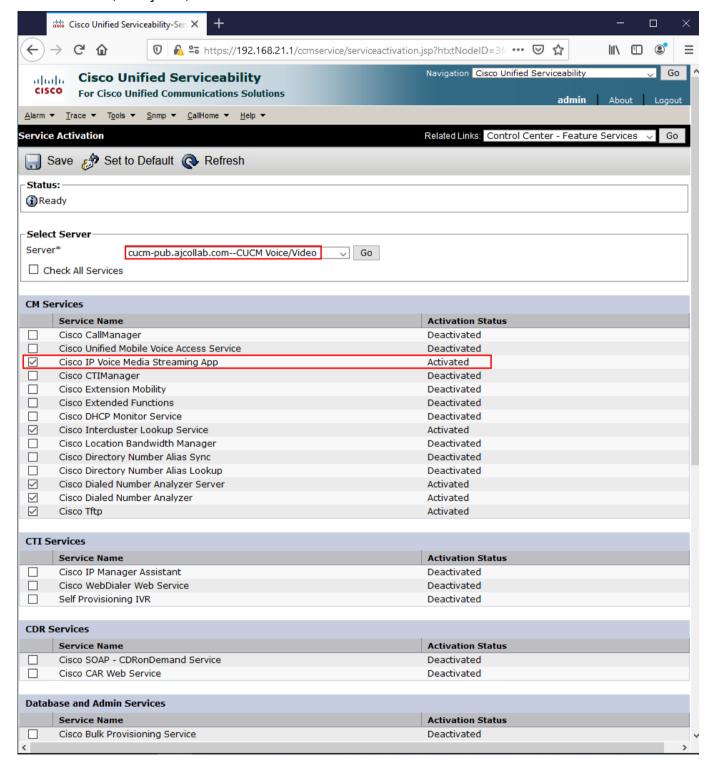
- Conference Bridge (CFB): Mixing of multiple audio stream together and provide a single stream for conference. Software conference bridge supports only G.711 call legs whereas Hardware Conference bridge supports any codec that is configured
- Media Termination Point (MTP): DTMF Internetworking between call legs (RFC 2833 to OOB).
 Software MTP supports only G.711 call leg DTMF internetworking whereas Hardware MTP supports any codec that is configured

Hardware Only Media Resources

 Transcoder (XCODE): Convert one Codec to another Codec. There are two types, Regular and Universal. Regular XCODE switch between G.711 to any other codec (G.711 to G.729), one leg has to be G.711 whereas Universal XCODE switch any codec to any codec (G.729 to iLBC)

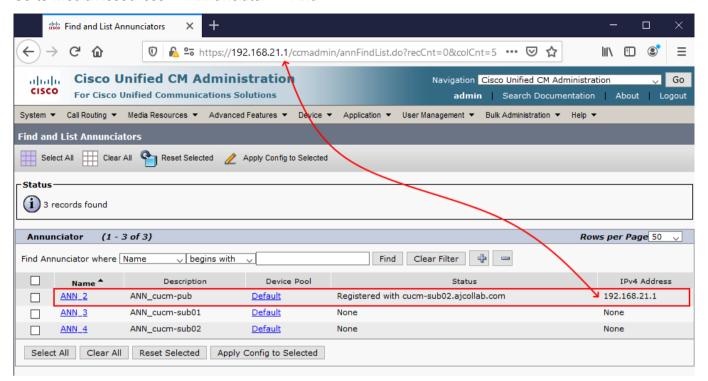
[Lab] Software Media Resource Activation

- Software media resources are provided by CUCM Nodes where we activate IPVMS (IP Voice Media Streaming App) Service
- In large enterprise production deployments, we can dedicate one CUCM Node just for IPVMS
 Service, in my Lab, I have activated on CUCM-PUB

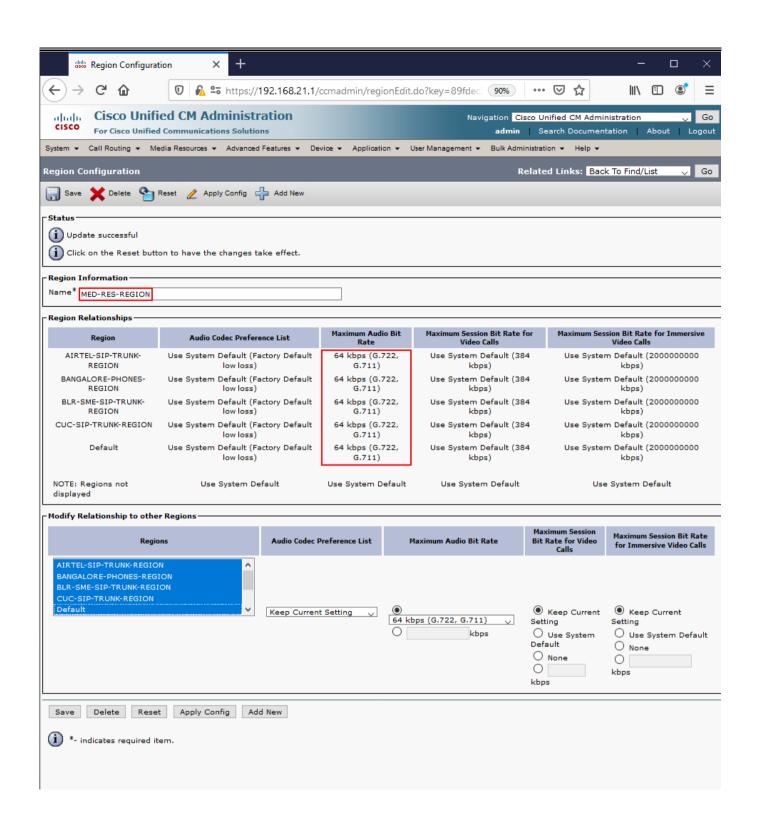


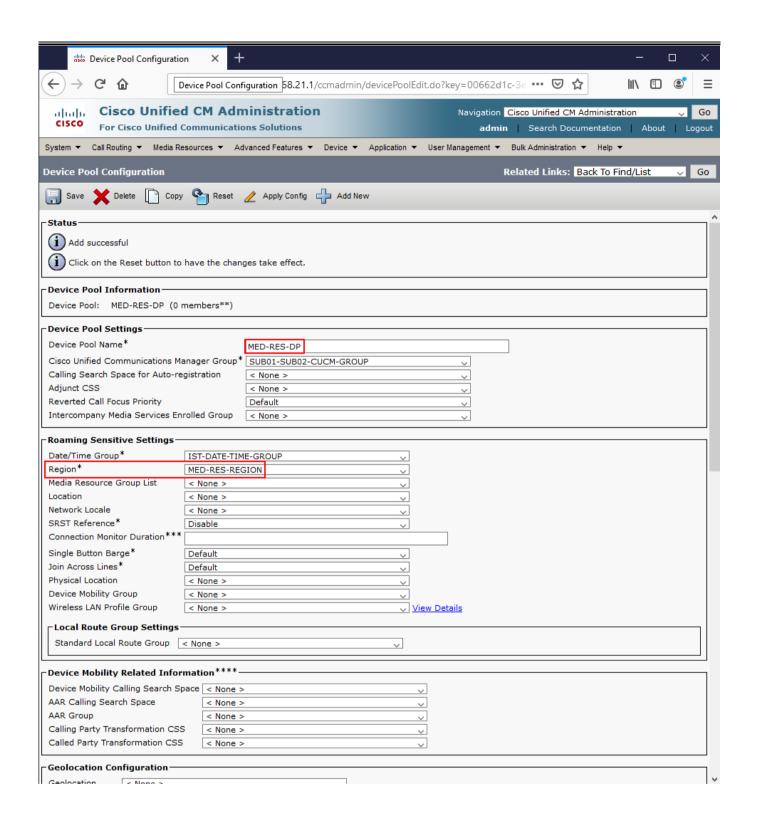
[Lab] Annunciator Configuration

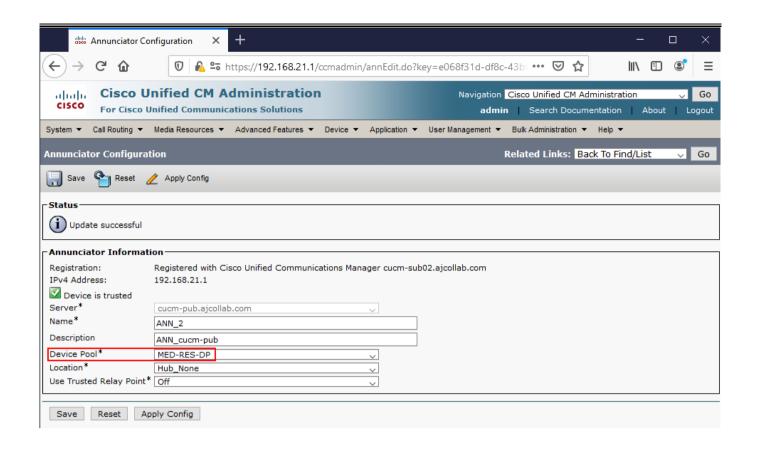
Go to Media Resources >> Annunciator >> Find



- We could see 3 Annunciators there and one is registered, and it has IP Address of 192.168.21.1
 (CUCM-PUB)
- Since we have activated IPVM only on CUCM-PUB, that is registered, and others are 'None' state
- Even after enabling IPVMS if you see the status of Annunciator is 'None', make sure in the Device
 Pool of Annunciator has a valid CUCM Subscriber with Cisco CallManager Service running
- In my case, the default device pool has default CUCM Group and it has one subscriber and hence it shows registered
- It is recommended to keep the media resources in separate Device Pool (and Region) so that we can control the codec relations
- I have created MEDIA-RES-REGION and MED-RES-DP to accomplish this
- Also, set the MED-RES-REGION to all other region as G.711 Codec. This is important as otherwise you may face media resource allocation failures



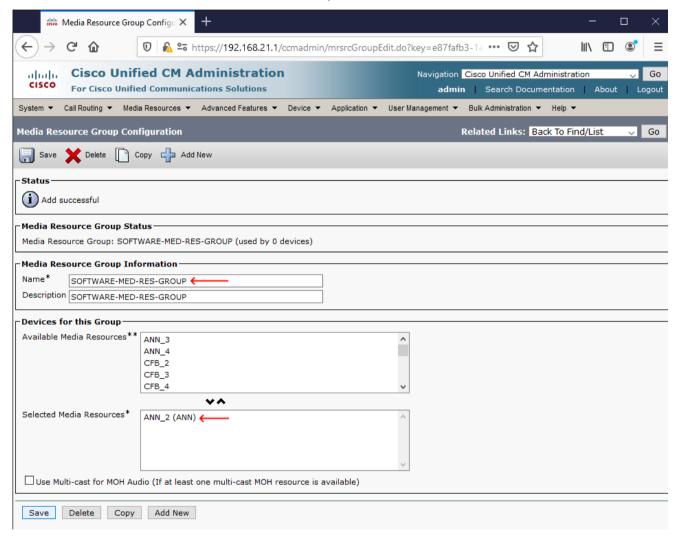




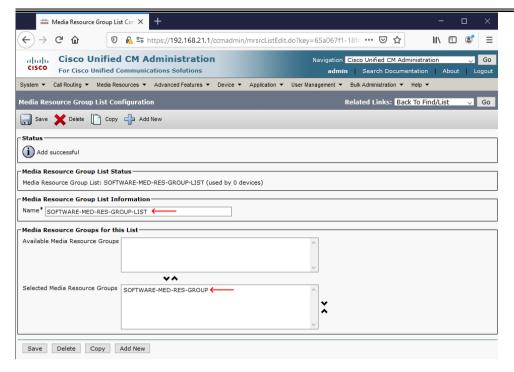
[Lab] Media Resource Access Control

- Media Resources are distributed to Phones, Trunks, and other devices via Media Resource Group (MRG) and Media Resource Group List (MRGL)
- All media resources are in a NULL media resource group by default and every device will be able to access it
- When you place the media resource inside and MRG, it can be access only for those devices who
 is having access to specific MRGL where the MRG and Resource is a part of

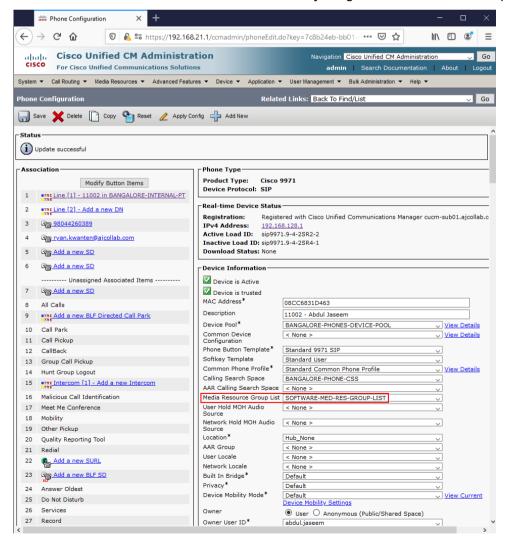
Go to Media Resources >> Media Resource Group >> Add New

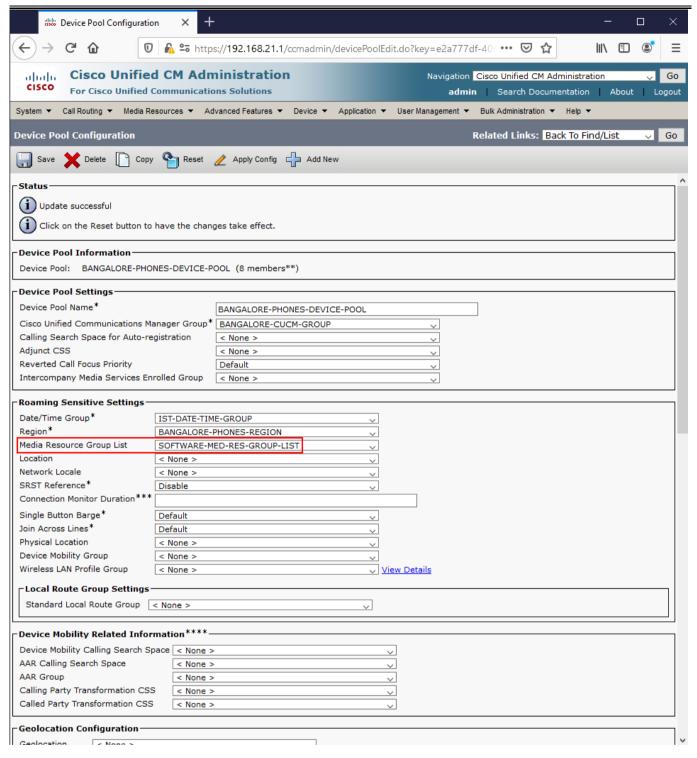


Go to Media Resources >> Media Resource Group List >> Add New



- Media Resource Group List applied to the Device Level or Device Pool Level
- MRGL in Device Pool Level serves everything that falls in the device pool





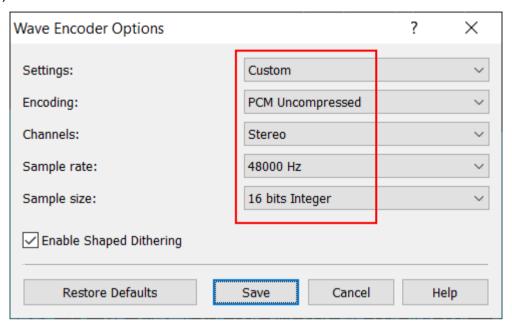
 To test Annunciator feature, dial any wrong directory number (that is no in the CUCM DB) from the phone, you will be able to hear "The call cannot be completed as dialed. Please confirm your directory and call again" instead of fast busy tone

[Lab] Music on Hold (MoH) Configuration

- The MoH sources makes music available when the call is on Hold
- Audio codecs supported by MoH server are G.711 alaw, G.711 ulaw, G.722, G.729
- If the held party doesn't support any of the above codecs, the MoH Server or the Held device invoke a Transcoder to perform audio codec conversion
- The CUCM integrated MoH server supports Unicast & Multicast streaming of music to the held party. Multicast method reduces the load of MoH server but requires multicast touting enabled at the network level
- Multicast also reduces the bandwidth because it enables multiple users to participate in a single audio stream. The recommended IP range of multicast IP is 239.1.1.1 to 239.255.255.255
- The MoH stream that an end point receives is determined by a combination two things. One is the source file configured in the Holding Party phone page and other one is the MoH resource on the held party's MRGL

Audio File Preparation

- The format of MoH file is 16-bit PCM .WAV, Stereo or Mono. 8, 16, 32, 48 kHz sampling rate
- Using any Audio converter application, you can generate this file. (I have used WavePad Sound Editor)



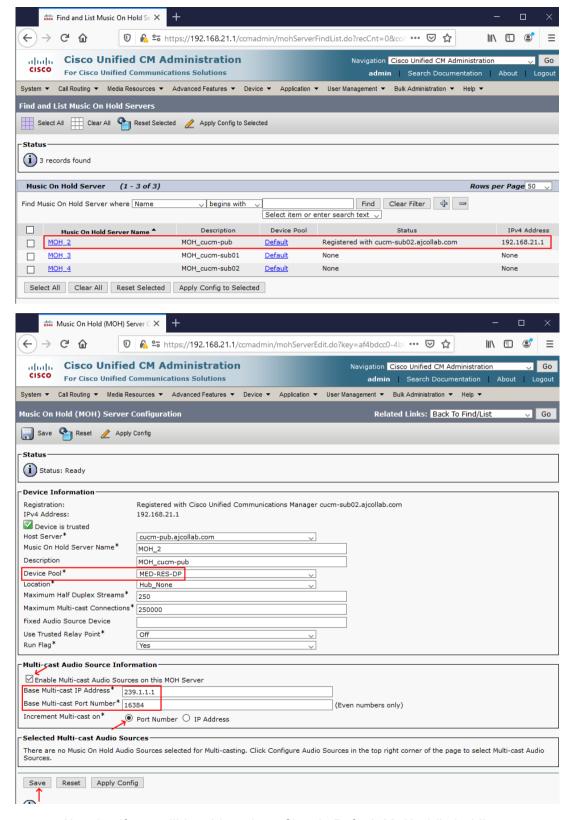
File >> Export >> Export as WAV >>

Note: You can download this MoH File from here for lab testing

Configure Music On Hold Server

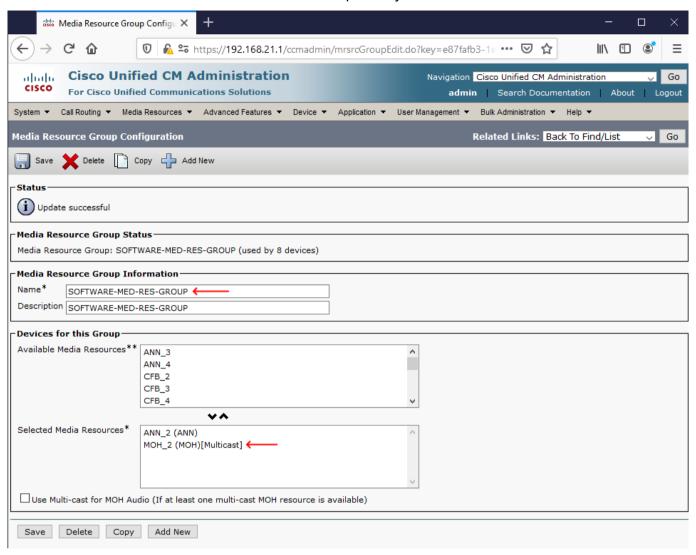
The IPVMS Service enable MoH server as well, we just need to change the Device Pool

Media Resources >> Music On Hold Servers >> Find



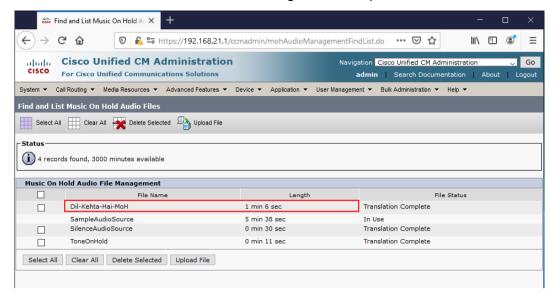
Now itself you will be able to hear Cisco's Default MoH while holding

• Place this MoH Server in MRG and MRGL respectively

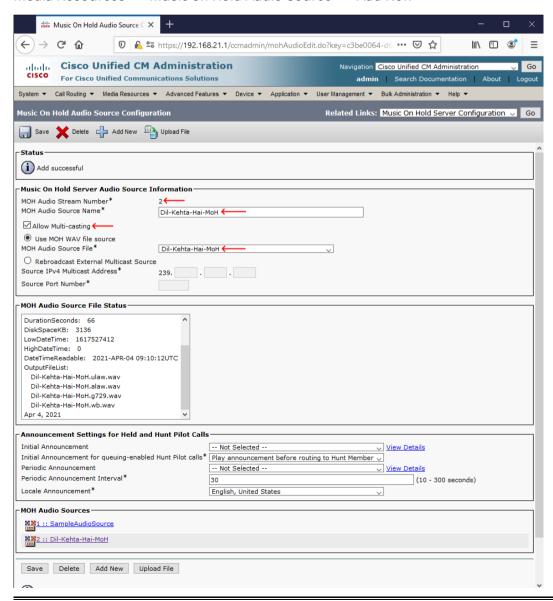


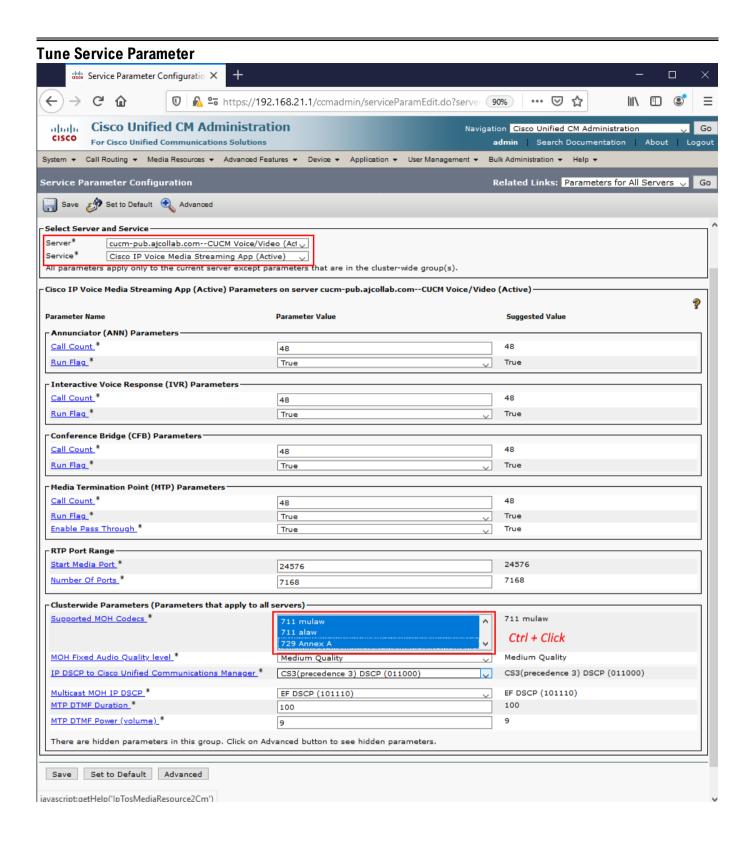
Upload MoH File

Media Resources >> MoH Audio File Management >> Upload File >> Select the file and upload

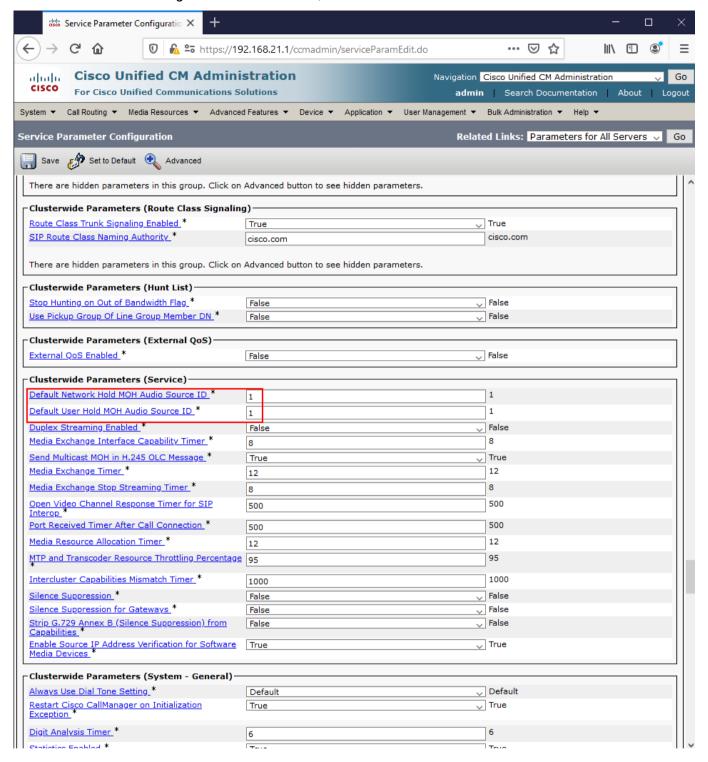


Media Resources >> Music on Hold Audio Source >> Add New

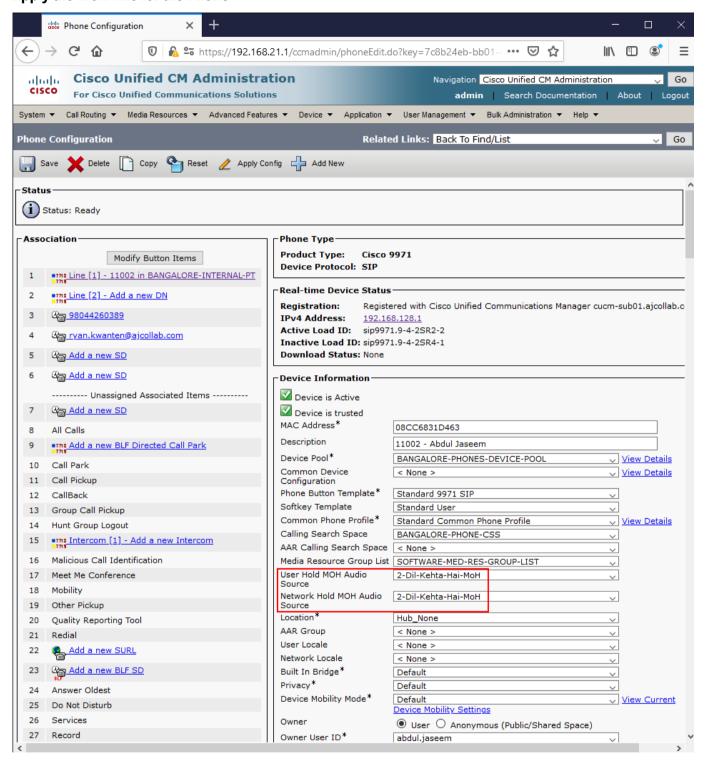




• In Cisco CallManager Service Parameter, we can alter the default MoH Audio Source



Apply the MoH File for the Phone



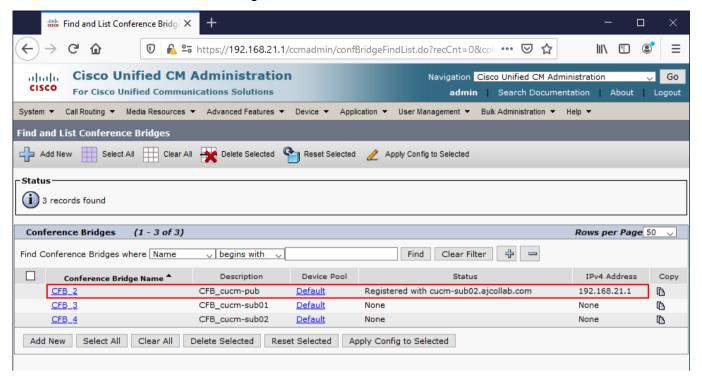
- Make a test from this phone to any other phone and hit Hold Button on this phone
- Now the other party will hear Music (Held Party should have MRGL Access with MoH Server)

[Lab] Software Conference Bridge (CFB) Configuration

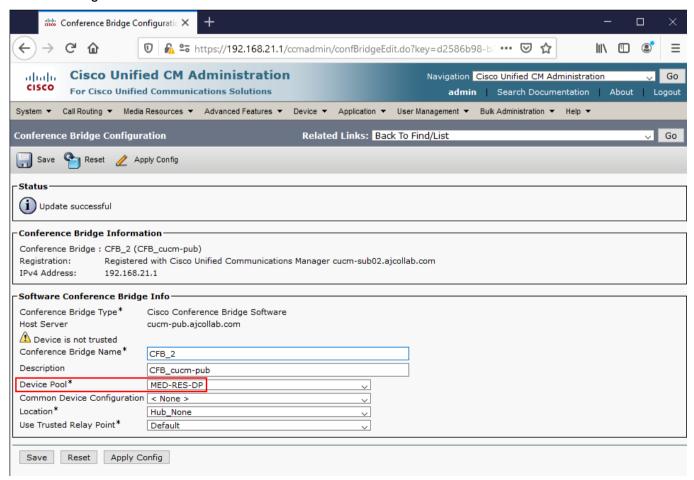
- Conference bridge handles conference calls (Ad Hoc or MeetMe)
- Enabling IPVMS will activate software Conference Bridge, since it is software based, it can support
 only G.711 call legs. Meaning whoever joins the conference must use G.711 as the codec
- Basic Ad Hoc Conference: Allows the conference controller (person who hit Conference soft key)
 to add specific participants to the conference. Here, a connected user hit conference call and dial
 another number then consult with him and press the Conference button again to start
 conferencing. Basic Ad Hoc is the default conference
- Advanced Ad Hoc: Any participant can add or remove other participants. It allows linking multiple
 Ad Hoc conferences together

Configure Software CFB

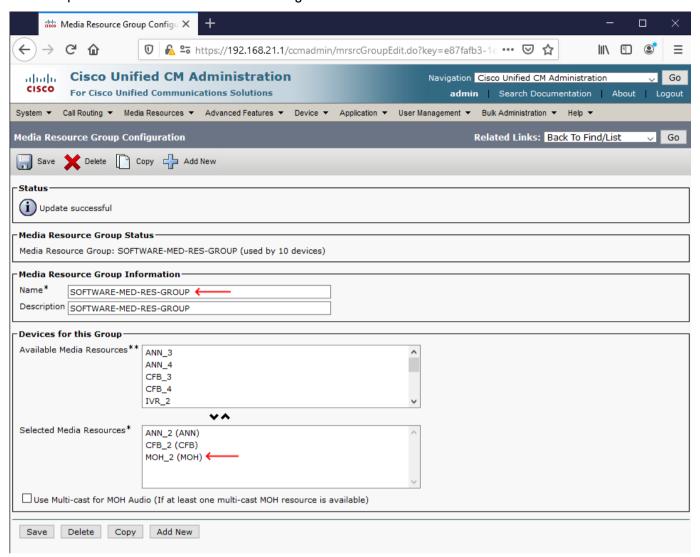
Media Resources >> Conference Bridge >> Find



Change the Device Pool

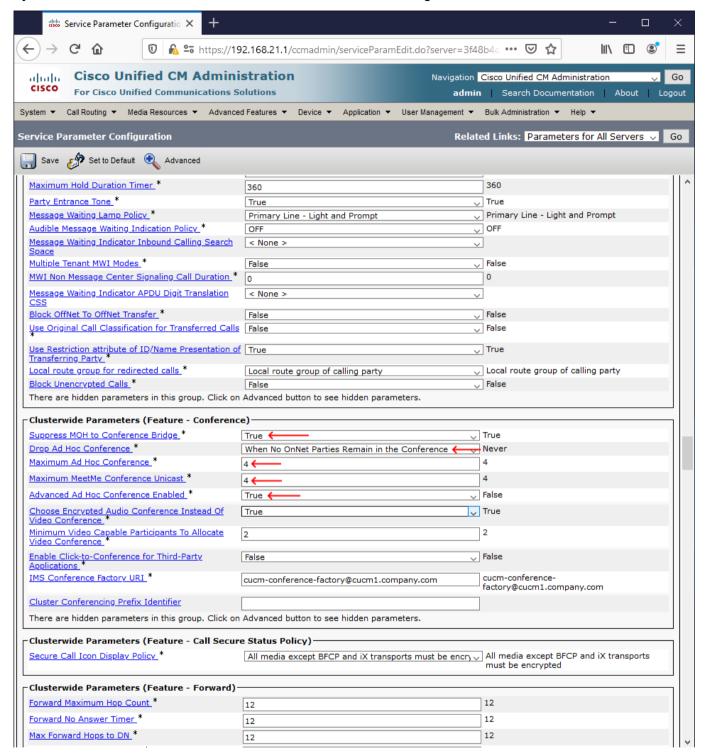


• Update the Software Conference bridge in the MRG and MRGL



Tune Service Parameters

System >> Service Parameter >> CUCM-PUB >> Cisco CallManager



Test Conference

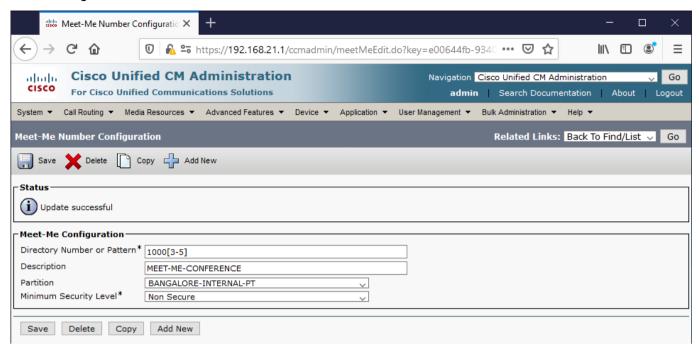
Make call to one number, hit 'Confrn' softkey, dial another number, hit 'Confrn' again



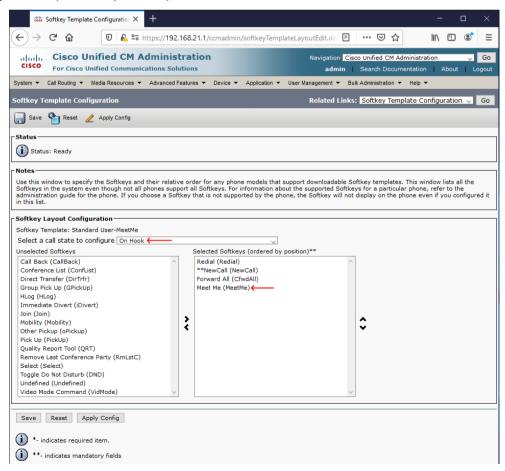
Note: Cisco IP Phone 88XX, 99XX Series phones has dedicated button for Conference, hence you don't find the softkeys sometimes

MeetMe Conference

Call Routing >> MeetMe Number/Pattern >> Add New



- Here 10003, 10004, 1005 are dedicated to MeetMe Conference
- To start the conference, anyone has to initiate the conference by pressing 'MeetMe' softkey and dialing 10003 (or other patterns)

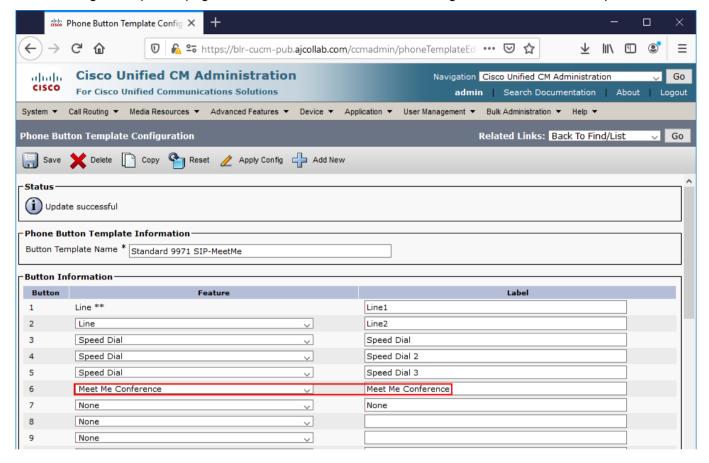


 Now others can simply dial 10003 and join the conference. Only the initiator has to use MeetMe softkey

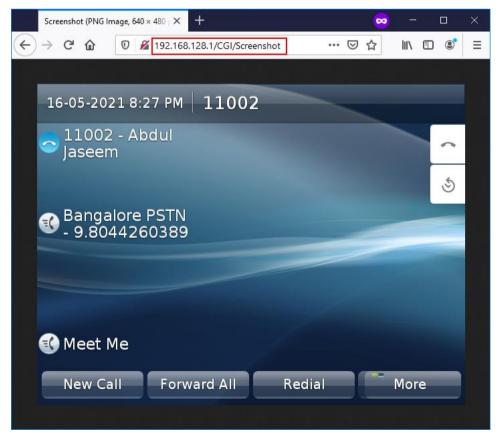




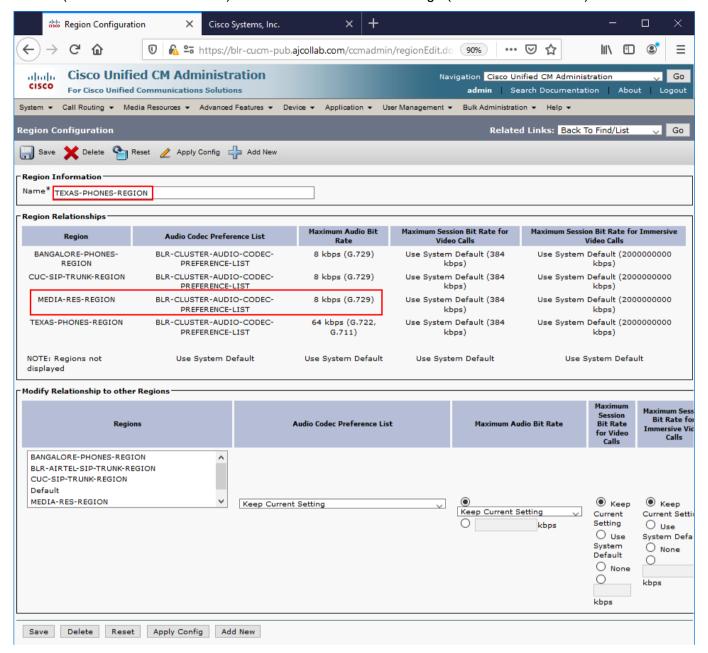
 IP Phone model Cisco 99XX and 88XX uses completely different softly rather than the one assigned in phone page. Hence the MeetMe button ins configured on the button template



http://192.168.128.1/CGI/Screenshot



Meet-Me conference from Texas Phones will fail since the Region relation between Texas Phones
 (TEXAS-PHONES-REGION) and Software Conference bridge (MEDIA-RES-REGION) set to G.729



To address this issue, we must use Hardware Conference bridge that can support any codecs

[Lab] Hardware Conference Bridge (CFB) Configuration

- IPVMS Service doesn't have any control on Hardware Media Resources
- Hardware CFB is configured on the IOS device (CUBE or Voice Gateways) and registered on CUCM as SCCP endpoint
- Hardware CFB support codecs that are configured on the IOS
- While designing the capacity, you can make use of DSP Calculator Tool to plan the resources





media-res# show version

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.5(3)M, RELEASE SOFTWARE (fc1)

Technical Support: http://www.cisco.com/techsupport

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Compiled Thu 23-Jul-15 00:28 by prod rel team

ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1

media-res# show inventory

NAME: "CISCO2901/K9", DESCR: "CISCO2901/K9 chassis, Hw Serial#: FGL200220A6, Hw

Revision: 1.0"

PID: CISCO2901/K9 , VID: V06 , SN: FGL200220A6

NAME: "PVDM3 DSP DIMM with 32 Channels on Slot 0 SubSlot 4", DESCR: "PVDM3 DSP DIMM

with 32 Channels"

PID: PVDM3-32 , VID: V01 , SN: FOC162318B5

NAME: "C1941/C2901 AC Power Supply", DESCR: "C1941/C2901 AC Power Supply"

PID: PWR-1941-2901-AC , VID: , SN:

media-res# show voice dsp group slot 0

dsp 1:

State: UP, firmware: 40.2.0
Max signal/voice channel: 32/32

Max credits: 480, Voice credits: 480, Video credits: 0

num_of_sig_chnls_allocated: 0
Transcoding channels allocated: 0

Transcouring channels allocated: 0

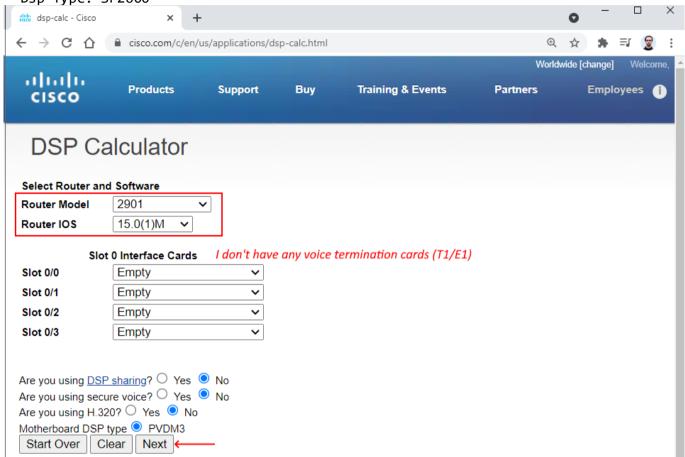
Group: FLEX_GROUP_VOICE, complexity: FLEX

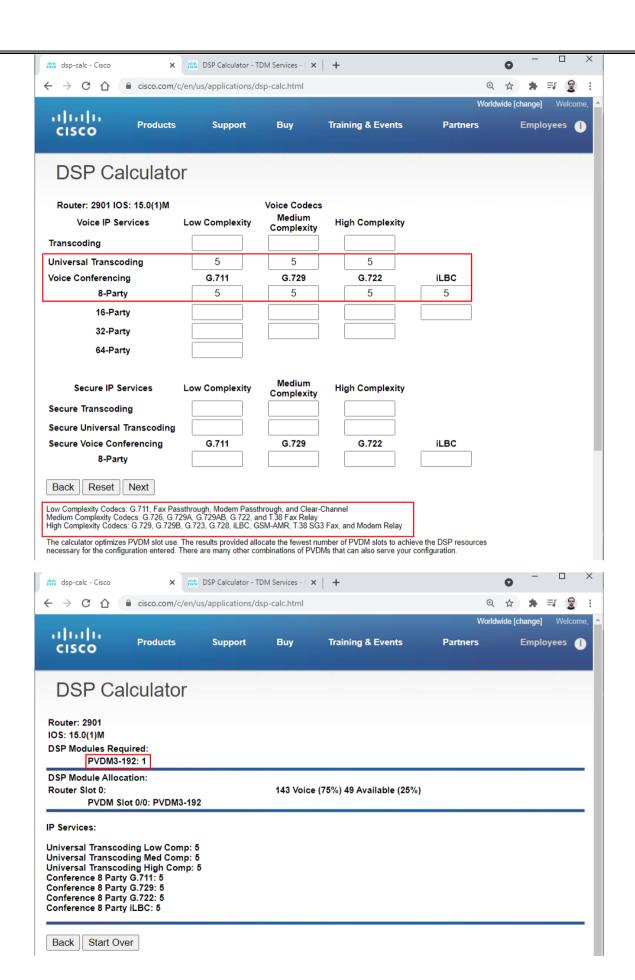
Shared credits: 480, reserved credits: 0

Signaling channels allocated: 0
Voice channels allocated: 0
Credits used (rounded-up): 0

Slot: 0

Device idx: 0 PVDM Slot: 0 Dsp Type: SP2600





- This is the way you calculate DSP requirement; in my lab I just have one PVDM3-32, hence we can have a smaller number of conferencing and transcoding sessions
- Make sure you have activated UC license in IOS to get the voice functionality

media-res#show version <--OUTPUT OMITTED-->

Technology Package License Information for Module: 'c2900'

Tachnology Tachnology-package Tachnology-package

Technology	Technology-package Current	Туре	Technology-package Next reboot
ipbase	ipbasek9	Permanent	ipbasek9
security	securityk9	Permanent	securityk9
<mark>uc</mark>	None	None	None
data	None	None	None

media-res#<mark>show license</mark>

Index 1 Feature: ipbasek9

Period left: Life time License Type: Permanent License State: Active, In Use License Count: Non-Counted License Priority: Medium

Index 2 Feature: securityk9
Period left: Life time
License Type: Permanent

License State: Active, In Use License Count: Non-Counted License Priority: Medium

Index 3 Feature: uck9

Period left: 8 weeks 3 days
Period Used: 2 hours 16 minutes
License Type: EvalRightToUse
License State: Active, In Use
License Suite: AdvUCSuiteK9
License Count: Non-Counted
License Priority: Low

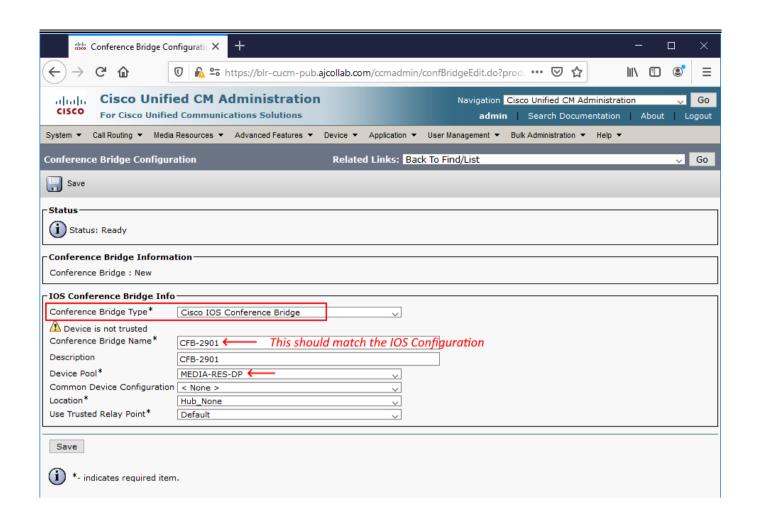
You can use below command and reboot to activate Evaluation license

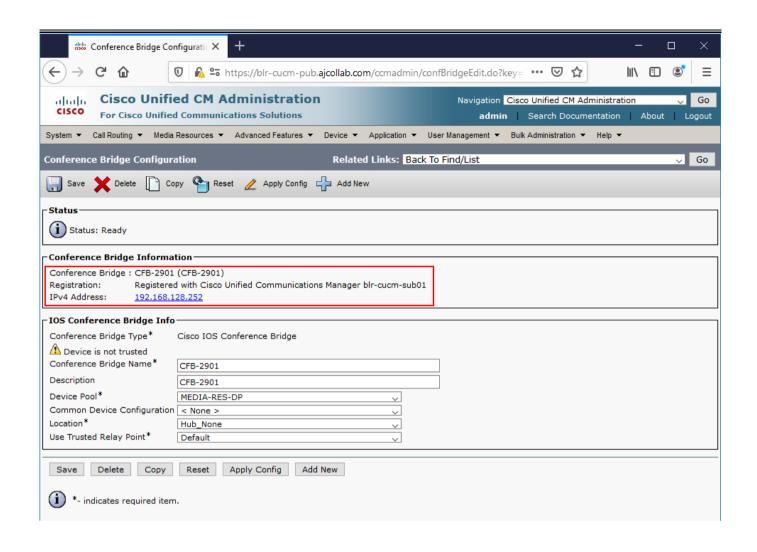
media-res(config)# license boot suite uck9 enable

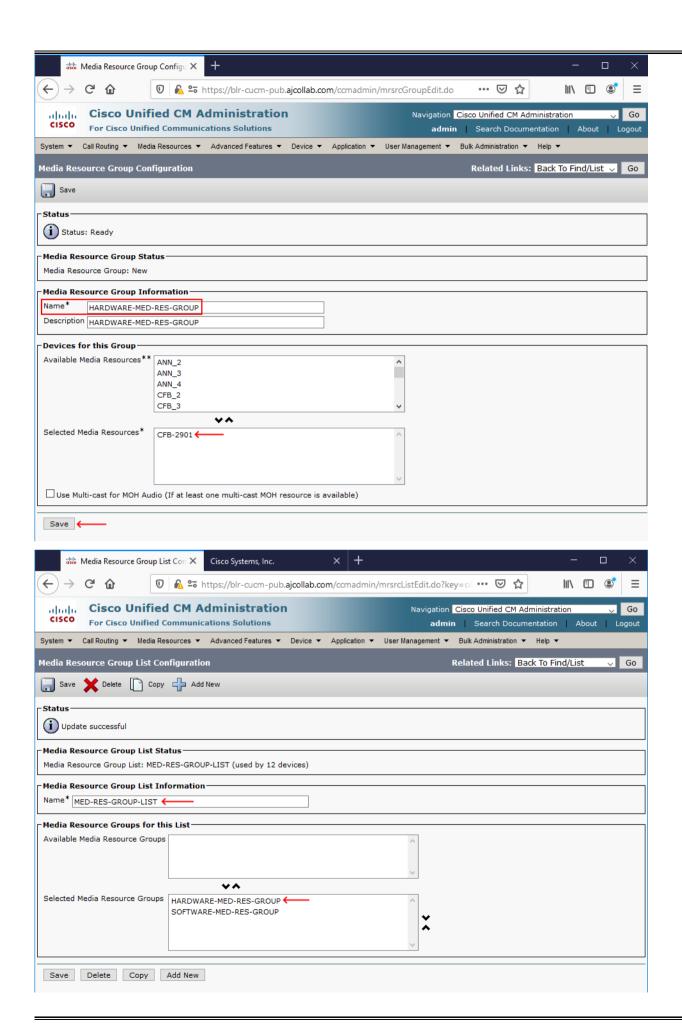
Conference Bridge Configuration voice-card 0 dspfarm dsp services dspfarm sccp local GigabitEthernet0/0 sccp sccp ccm 192.168.21.2 identifier 1 version 7.0 sccp ccm 192.168.21.3 identifier 2 version 7.0 dspfarm profile 1 conference codec g729br8 codec g729r8 codec g729ar8 codec g711alaw codec g711ulaw codec g729abr8 maximum sessions 2 associate application SCCP no shutdown sccp ccm group 1 associate ccm 1 priority 1 associate ccm 2 priority 2 bind interface gigabitEthernet 0/0 associate profile 1 register CFB-2901 Find and List Conference Bridge X ←) → C û The state of the s ≡ **Cisco Unified CM Administration** Go alada Navigation Cisco Unified CM Administration For Cisco Unified Communications Solutions admin | Search Documentation | About | Logout System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Annunciator Find and List Conferen Interactive Voice Response Add New Conference Bridge Media Termination Point **Conference Bridges** Music On Hold Audio Source Find Conference Bridges Find Clear Filter 🕂 😑 Fixed MOH Audio Source ase enter your search criteria using the options above Music On Hold Server Add New Video On Hold Server Media Resource Group

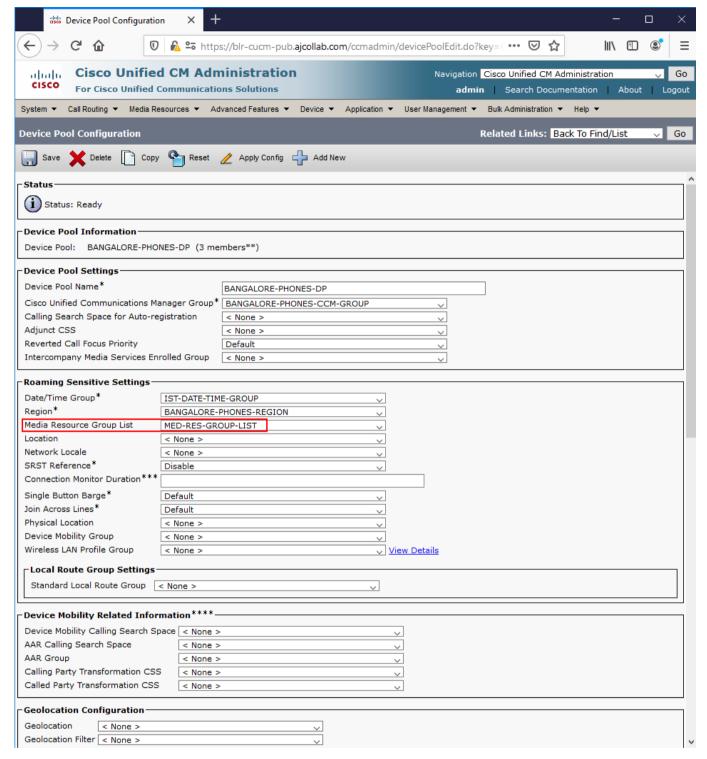
Refer this link to know more about the Conference bridge Types

Media Resource Group List MOH Audio File Management Mobile Voice Access Announcement









- To test the scenario, set the Region Relation between Texas and Media Resources to G.729.
 Previously the Phones at Texas location were unable to join / initiate meet-me conference since we had only the Software Conference Bridge available
- Now Texas phones will be able to join or initiate Meet-Me conference bridge

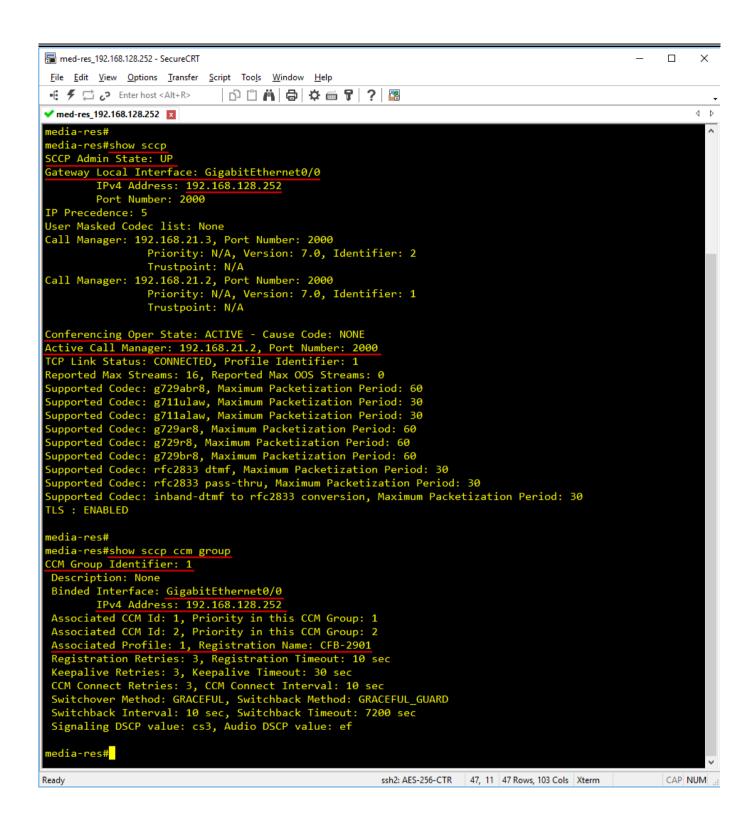
```
med-res_192.168.128.252 - SecureCRT
                                                                                                                                                                                                                             \underline{\text{File}} \quad \underline{\text{Edit}} \quad \underline{\text{V}} \text{iew} \quad \underline{\text{O}} \text{ptions} \quad \underline{\text{T}} \text{ransfer} \quad \underline{\text{S}} \text{cript} \quad \text{Too}\underline{\text{Is}} \quad \underline{\text{W}} \text{indow} \quad \underline{\text{H}} \text{elp}
                                                          D 🖺 🗥 🖨 🌣 📾 🎖 🖓 🌃
 ■ F 🖾 🕫 Enter host <Alt+R>
✓ med-res_192.168.128.252 🗵
                                                                                                                                                                                                                                        4 Þ
 media-res#
 media-res#show sccp connections
                                                                                                          sport rport ripaddr conn_id_tx
sess_id
                        conn_id
                                                                                        codec
33556442
33556442
33556442

      conf
      sendrecv g711u
      16404 21812 192.168.128.3

      conf
      sendrecv g711u
      16402 17714 192.168.128.6

      conf
      sendrecv g729← 16400 29384 192.168.129.1

                        50331662
                        33554436
50331661
Total number of active session(s) 1, and connection(s) 3
media-res#
media-res#
```

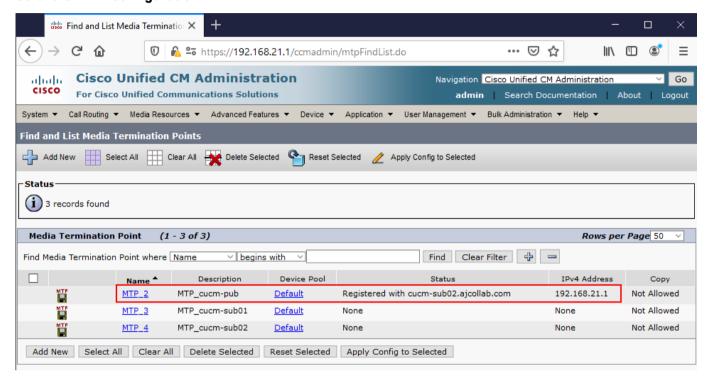


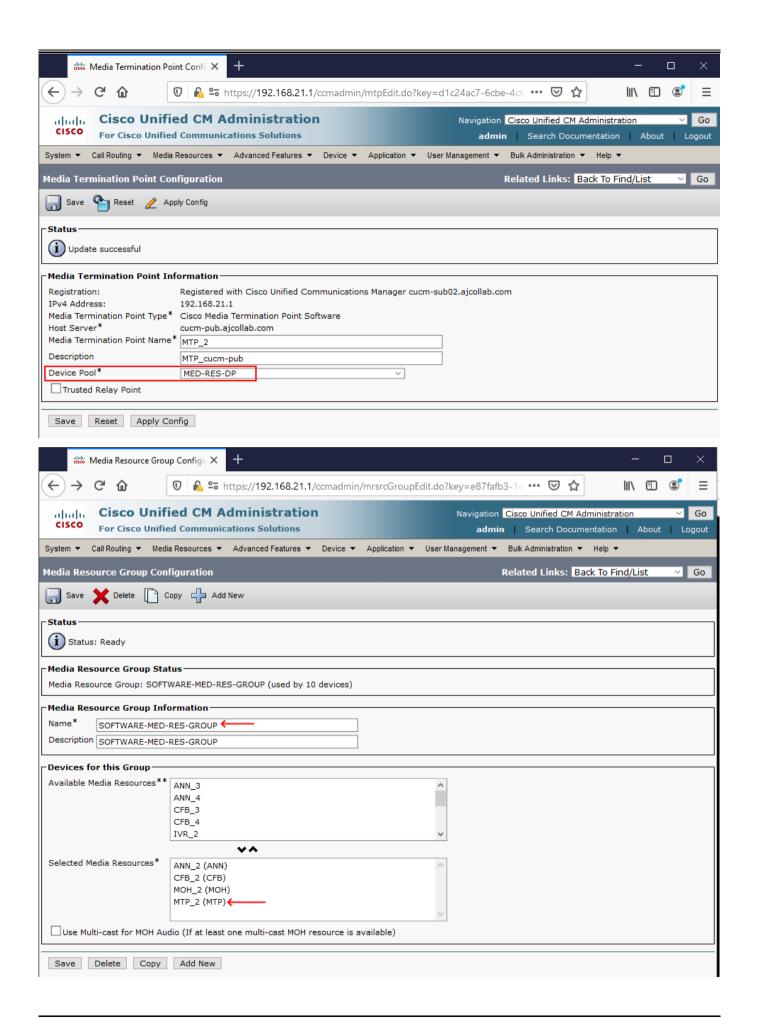
```
med-res_192.168.128.252 - SecureCRT
                                                                                                                                <u>F</u>ile <u>E</u>dit <u>V</u>iew <u>O</u>ptions <u>T</u>ransfer <u>S</u>cript Too<u>l</u>s <u>W</u>indow <u>H</u>elp
                                   | D L M | B | ❖ = T | ? | 8
■ F 🖾 🗗 Enter host <Alt+R>
✓ med-res_192.168.128.252 🗵
                                                                                                                                      4
                                                                                                                                        Þ
media-res#
media-res#show dspfarm profile 1
Dspfarm Profile Configuration
 Profile ID = 1, Service = CONFERENCING, Resource ID = 1
 Profile Description :
 Profile Service Mode : Non Secure
Profile Admin State : UP
 Profile Operation State : ACTIVE
 Application : SCCP Status : ASSOCIATED
 Resource Provider : FLEX_DSPRM Status : UP
Total Number of Resources Configured : 2
 Total Number of Resources Available : 2
 Total Number of Resources Out of Service : 0
 Total Number of Resources Active : 0
 Maximum conference participants : 8
 Codec Configuration: num of codecs:6
 Codec : g729abr8, Maximum Packetization Period : 60 , Transcoder: Not Required
 Codec : g711ulaw, Maximum Packetization Period : 30 , Transcoder: Not Required
 Codec : g711alaw, Maximum Packetization Period : 30 , Transcoder: Not Required
Codec: g729ar8, Maximum Packetization Period: 60, Transcoder: Not Required Codec: g729r8, Maximum Packetization Period: 60, Transcoder: Not Required Codec: g729br8, Maximum Packetization Period: 60, Transcoder: Not Required
 edia-res#
 edia-res#
```

[Lab] Software Media Termination Point (MTP) Configuration

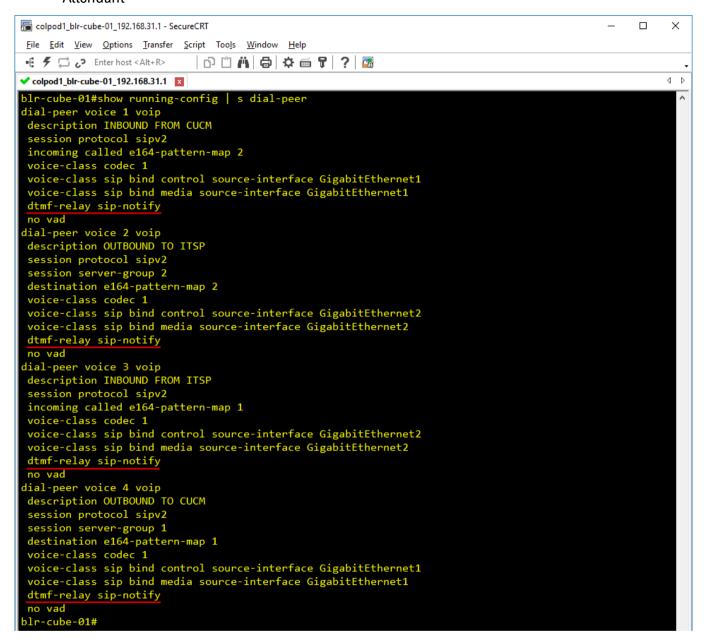
- IPVMS will enable Software MTP on the respective CUCM Nodes
- Software MTP used to interconnect G.711 alaw to G.711 ulaw and internetwork DTMF Relays (like SIP Notify to RTP NTE). MTP also used to interconnect the different sampling size media (G.711 alaw 20 ms to G.711 alaw 30 ms)
- We have a scenario when external users are calling the Main Line Number (CUC Auto Attendant),
 the DTMF inputs are not detecting, hence they are unable to navigate through the menu options

Software MTP Configuration

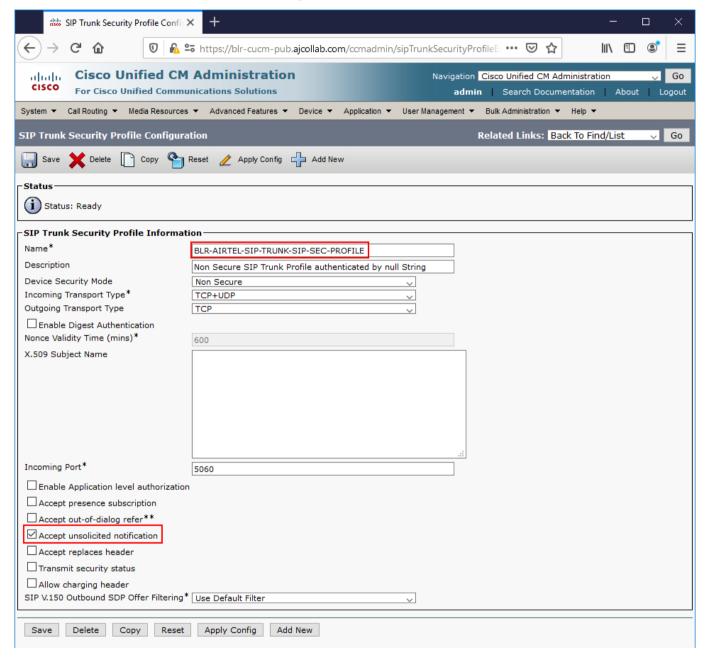




 Note: To implement this issue for the lab, I have configured dial-peers of the cube01 to use sipnotify (OOB) DTMF method. Now the DTMF won't be working from PSTN number to CUC Auto Attendant



- Make sure 'Accept unsolicited notification' is enabled on the CUBE SIP Trunk otherwise Call
 Manager itself reject the DTMF Notify from the CUBE
- Software MTP must be in the MRGL and it should be accessible for the CUC SIP Trunk since the conversion required while connecting to CUC



Make sure region relation between CUC and Media Resources set to G.711

[Lab] Hardware Media Termination Point (MTP) Configuration

 Cisco IOS hardware MTP provides multi codec support for DTMF Internetworking, Packetization difference

Note: We are not demonstrating the live scenario for the hardware MTP since it is difficult to set up packetization difference failures. We will look at the configurations only

We are assuming other DSP FARM configurations are already in place (we did the same for hardware conference bridge)

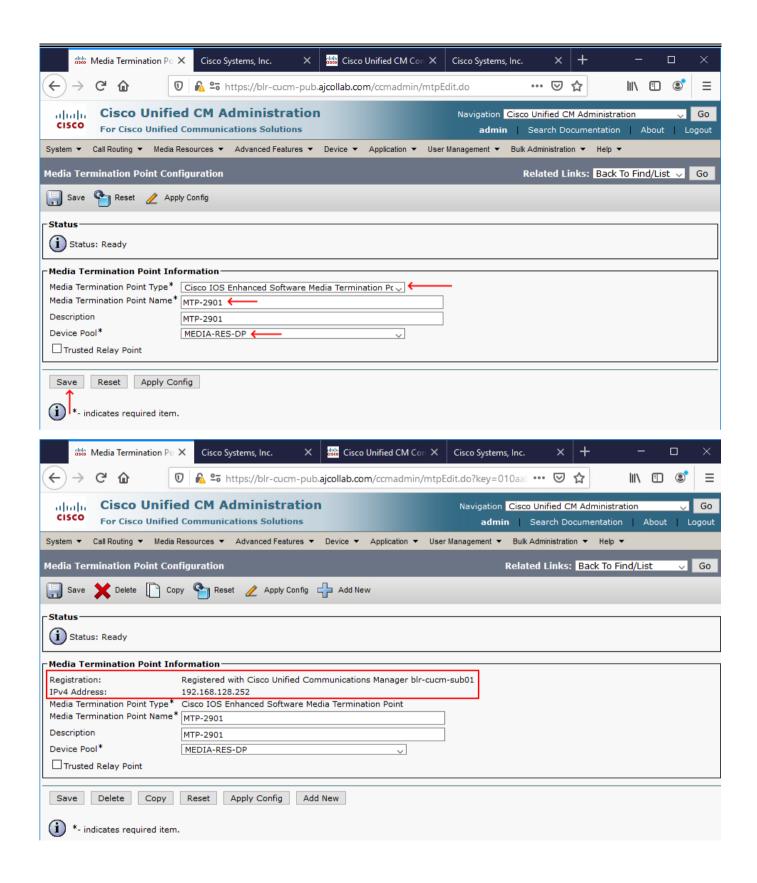
• g711 ulaw codec will be added to the MTP by default, hence use 'no codec g711ulaw' if you want the MTP for other codec. Only one codec supported per profile

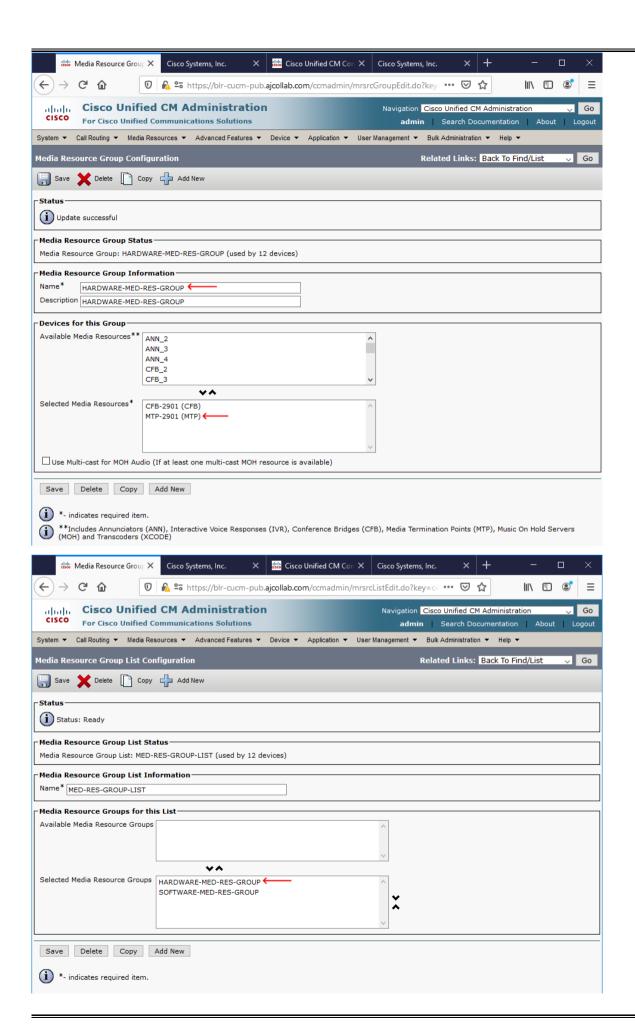
Hardware MTP Configuration

```
ļ
dspfarm profile 2 mtp
 no codec g711ulaw
 codec g729r8
 maximum sessions hardware 2
 associate application SCCP
 no shutdown
sccp ccm group 1
 associate profile 2 register MTP-2901
                                                                                                   🔚 med-res_192.168.128.252 - SecureCRT
                                                                                                        Х
 File Edit View Options Transfer Script Tools Window Help
 ■ # □ • Enter host < Alt+R>
                            | D 🖺 A | 🖨 🜣 📾 T | ? | 🝱

✓ med-res_192.168.128.252 

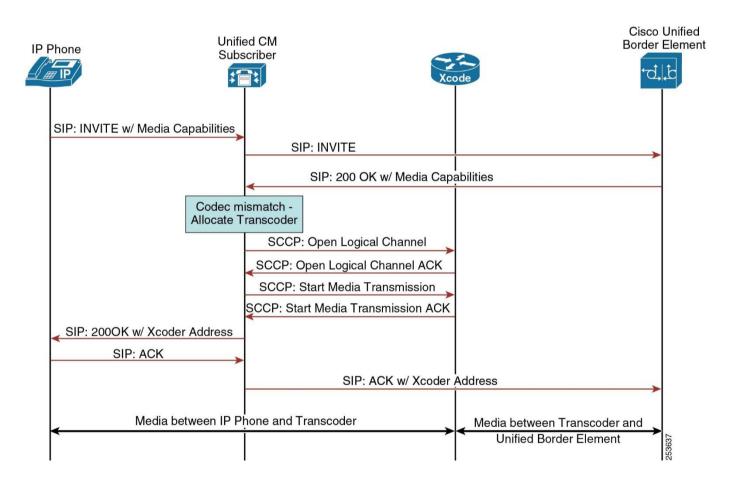
                                                                                                       4
 media-res#
 media-res#
 media-res#show running-config | s dspfarm profile 2 mtp
 dspfarm profile 2 mtp
  codec g729r8
 maximum sessions software 2
 associate application SCCP
 media-res#
 media-res#show running-config | s sccp ccm group 1
 bind interface GigabitEthernet0/0
 associate ccm 1 priority 1
  associate ccm 2 priority 2
  associate profile 2 register MTP-2901
  associate profile 1 register CFB-2901
  edia-res#
  edia-res#
  edia-res#
```



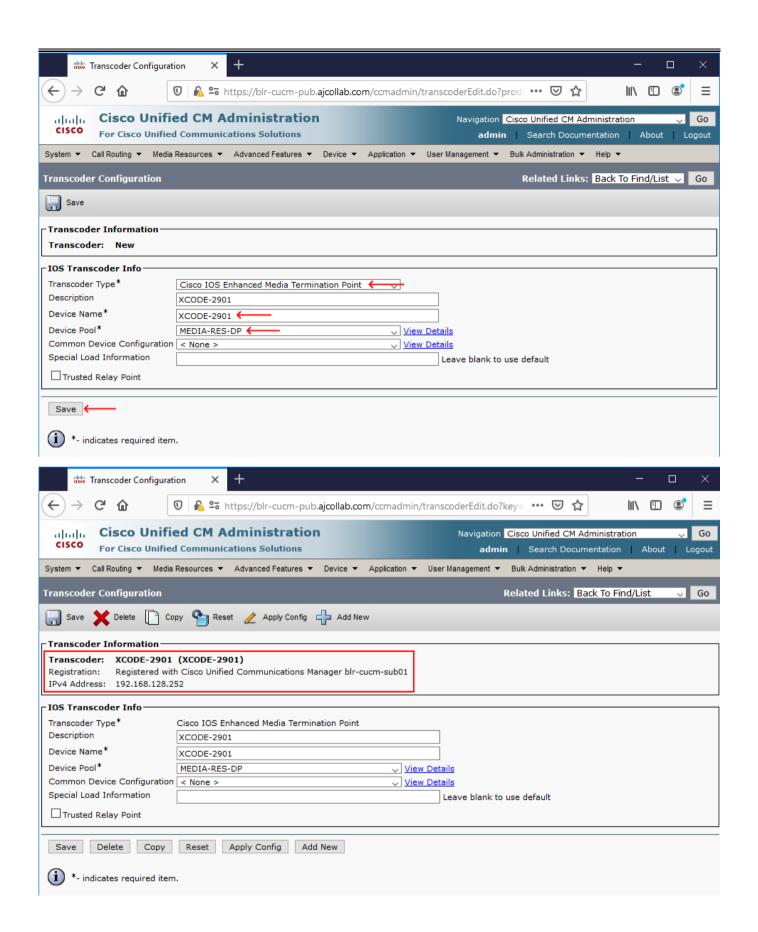


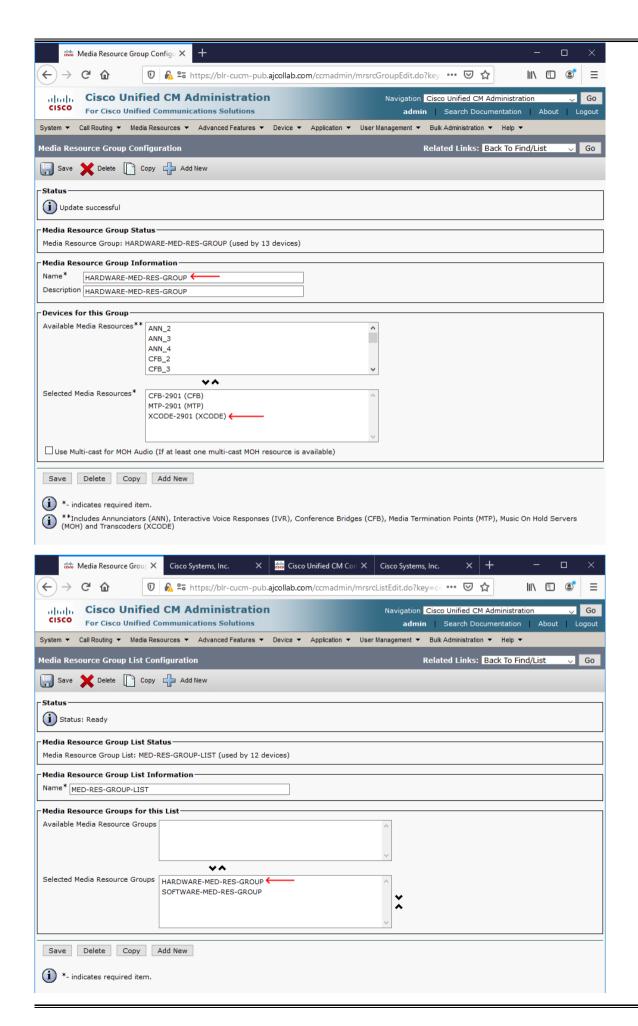
[Lab] Transcoder (XCODE) Configuration

- Transcoder used to convert between two codecs G.729 to G.711; G.711 to iLBC etc.
- Transcoders also does conversion codec packetization difference (20mSec to 30mSec)
- No Software Transcoder available, transcoders are always hardware based and requires PVDM/ Hardware resources
- Traditional Transcoding: G.711 to any other codec (e.g. G.711 <<->> iLBC or G.711 <<->> G.729,
 etc.) requires less DSPs. One leg has to be G.711 here
- Universal Transcoding: Transcoding between any codec to any other codec (e.g. G.729 <<->>iLBC), require more DSPs. This can also convert between G.711 to any other codec. This will an ideal fit for any complex scenarios

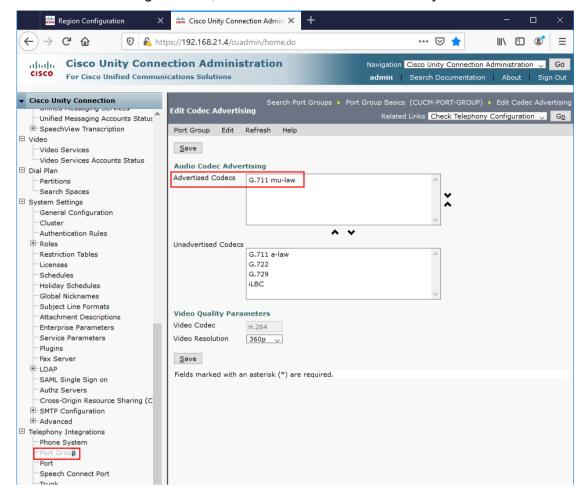


Transcoder Configuration dspfarm profile 3 transcode universal codec g729abr8 codec g729ar8 codec g711alaw codec g711ulaw codec g722-64 codec g729br8 codec g729r8 codec ilbc maximum sessions 5 associate application SCCP no shutdown sccp ccm group 1 associate profile 3 register XCODE-2901 med-res_192.168.128.252 - SecureCRT $\underline{\underline{F}} ile \quad \underline{\underline{F}} dit \quad \underline{\underline{V}} iew \quad \underline{\underline{O}} ptions \quad \underline{\underline{T}} ransfer \quad \underline{\underline{S}} cript \quad \underline{T} oo\underline{\underline{I}} s \quad \underline{\underline{W}} indow \quad \underline{\underline{H}} elp$ ■ F 🖾 ເ> Enter host < Alt+R> | D 🖺 A | 🖶 🛱 😭 🖀 ✓ med-res 192.168.128.252 🔣 media-res# media-res#show running-config | s dspfarm profile 3 transcode universal dspfarm profile 3 transcode universal codec g729abr8 codec g729ar8 codec g711alaw codec g711ulaw codec g722-64 codec g729br8 codec g729r8 codec ilbc maximum sessions 5 associate application SCCP media-res# media-res# media-res#show running-config | s sccp ccm group 1 sccp ccm group 1 bind interface GigabitEthernet0/0 associate ccm 1 priority 1 associate ccm 2 priority 2 associate profile 3 register XCODE-2901 associate profile 2 register MTP-2901 associate profile 1 register CFB-2901 media-res#

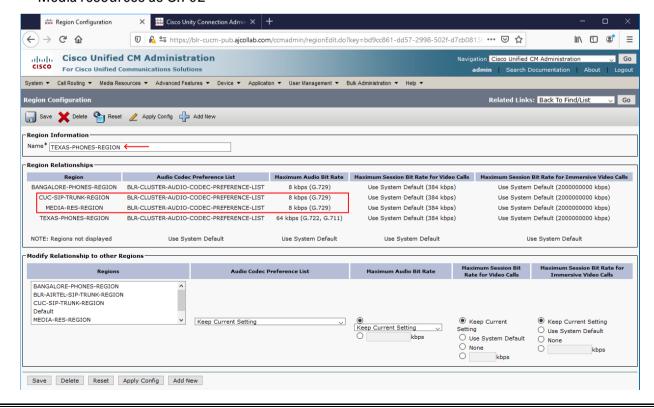




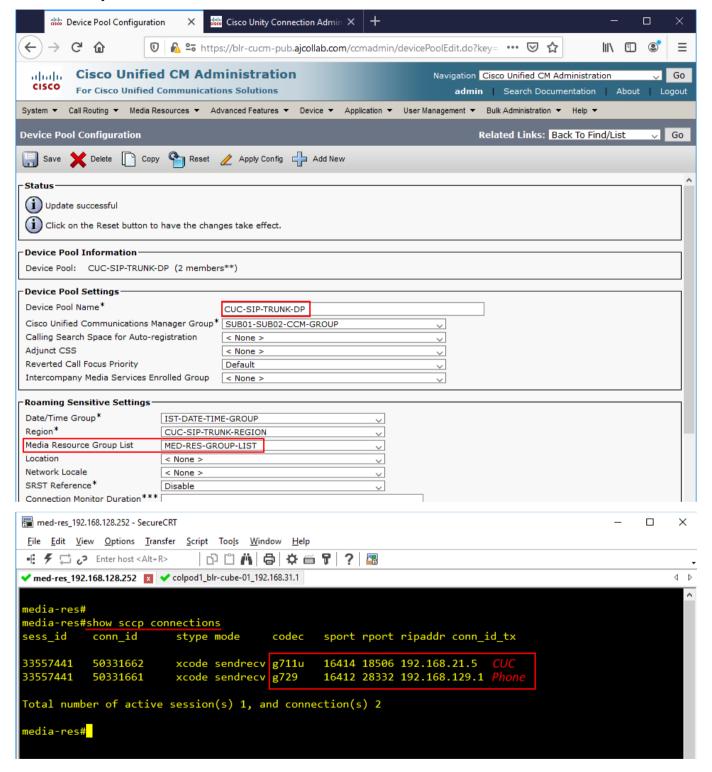
• To test the transcoding scenario, disable G.729 codec at Cisco Unity Connection

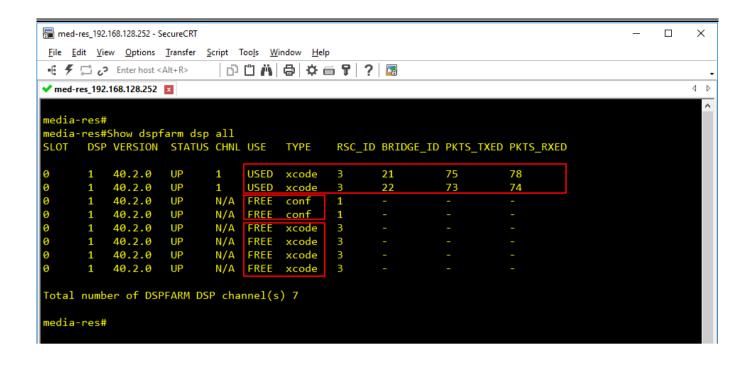


 Set the region relation between Texas Phones and CUC as G.729 and between Texas Phones and Media resources as G.792

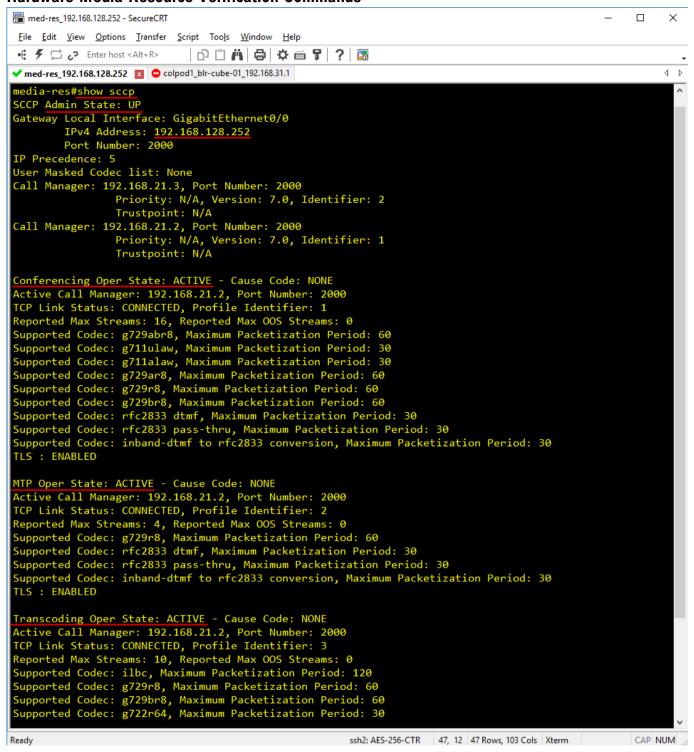


- Try to access voicemail from the Texas Phone and you will see a failure without transcoder since
 CUC accepts only G.711 Codec where Texas Phones can send only 8 Kbps
- Once you insert transcoder, this issue can be sorted

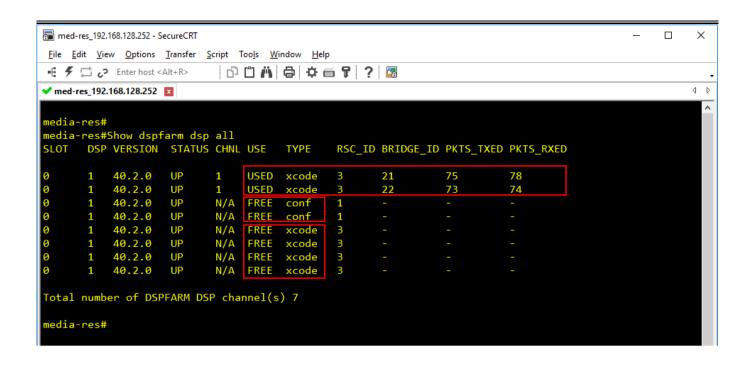




Hardware Media Resource Verification Commands



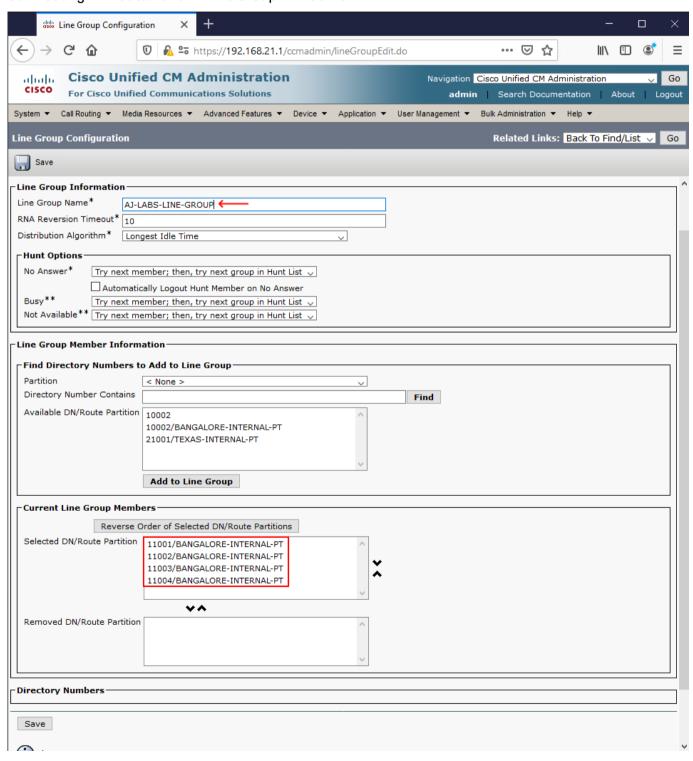
```
media-res#<mark>show dspfarm all</mark>
Dspfarm Profile Configuration
  Profile ID = 3, Service =Universal TRANSCODING, Resource ID = 3
  Profile Service Mode : Non Secure
  Profile Admin State : UP
  Profile Operation State : ACTIVE
  Application : SCCP Status : ASSOCIATED
  Total Number of Resources Configured : 5
Total Number of Resources Available : 5
Total Number of Resources Out of Service : 0
   Total Number of Resources Active : 0
 Codec Configuration: num_of_codecs:8
Codec : ilbc, Maximum Packetization Period : 120
Codec : g729r8, Maximum Packetization Period : 60
Codec : g729br8, Maximum Packetization Period : 60
Codec: g722-64, Maximum Packetization Period: 30
Codec: g711ulaw, Maximum Packetization Period: 30
Codec: g711alaw, Maximum Packetization Period: 30
Codec: g712alaw, Maximum Packetization Period: 30
Codec: g729ar8, Maximum Packetization Period: 60
Codec: g729abr8, Maximum Packetization Period: 60
Dspfarm Profile Configuration
  Profile ID = 1, Service = CONFERENCING, Resource ID = 1
  Profile Description :
  Profile Service Mode : Non Secure
Profile Admin State : UP
  Profile Operation State : ACTIVE
  Application : SCCP Status : ASSOCIATED
  Total Number of Resources Configured : 2
Total Number of Resources Available : 2
 Total Number of Resources Available : 2
Total Number of Resources Out of Service : 0
Total Number of Resources Active : 0
Maximum conference participants : 8
Codec Configuration: num_of_codecs:6
Codec : g729abr8, Maximum Packetization Period : 60 , Transcoder: Not Required Codec : g711alaw Maximum Packetization Period : 30 , Transcoder: Not Required Codec : g711alaw Maximum Packetization Period : 30 , Transcoder: Not Required
 Codec: g711alaw, Maximum Packetization Period: 30, Transcoder: Not Required Codec: g729ar8, Maximum Packetization Period: 60, Transcoder: Not Required Codec: g729r8, Maximum Packetization Period: 60, Transcoder: Not Required Codec: g729br8, Maximum Packetization Period: 60, Transcoder: Not Required Codec: g729br8, Maximum Packetization Period: 60, Transcoder: Not Required
 Dspfarm Profile Configuration
 Profile ID = 2, Service = MTP, Resource ID = 2
Profile Description :
Profile Service Mode : Non Secure
  Profile Admin State : UP
  Profile Operation State : ACTIVE
 Application: SCCP Status: ACSOCIATED
Resource Provider: NONE Status: NONE
Total Number of Resources Configured: 2
Total Number of Resources Available: 2
Total Number of Resources Out of Service: 0
Total Number of Resources Active: 0
  Hardware Configured Resources : 0
  Hardware Resources Out of Service: 0
Software Configured Resources : 2
  Number of Hardware Resources Active : 0
  Number of Software Resources Active : 0
Codec Configuration: num_of_codecs:1
Codec : g729r8, Maximum Packetization Period : 60
                                                                            TYPE
SLOT
             DSP VERSION STATUS CHNL USE
                                                                                            RSC_ID BRIDGE_ID PKTS_TXED PKTS_RXED
                                        UP
                      40.2.0
                                                                FREE
                                                                            conf
                                       UP
UP
                                                                FREE
FREE
                      40.2.0
0
                                                      N/A
                                                                            conf
0
                      40.2.0
                                                      N/A
                                                                            xcode
                                                               FREE
FREE
0
0
                                        UP
                                                      N/A
                      40.2.0
                                                                            xcode
                                                                                            3
                                        UP
                                                      N/A
                      40.2.0
                                                                            xcode
                                                                                            3
                      40.2.0
                                        UP
0
                                                      N/A
                                                                FREE
                                                                            xcode
                      40.2.0
                                                                FREE
 Total number of DSPFARM DSP channel(s) 7
```

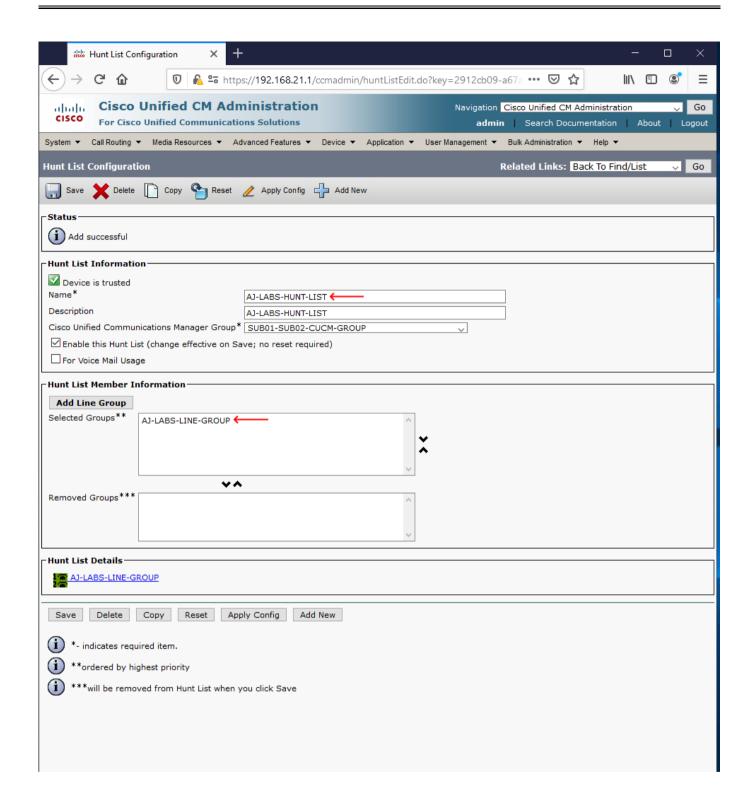


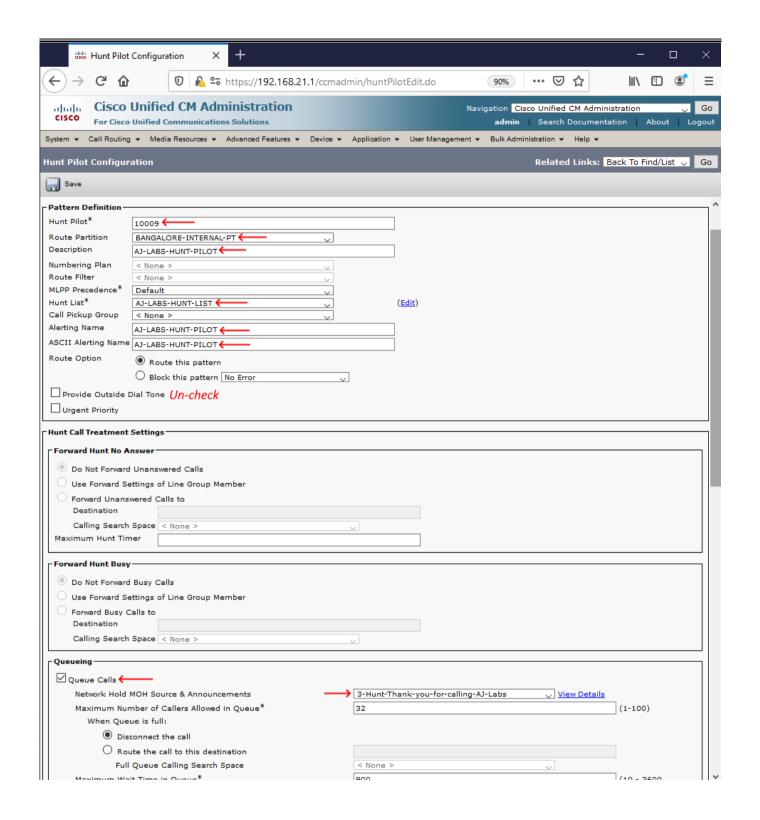
Hunt

- Hunt Group is a mechanism that helps business to manage inbound calls. It is a group of telephone lines that are associated with a common number
- When call comes to the hunt number, the call cycles through the group of lines until available line
 is found
- While hunting, the forwarding configuration of line group members is not considered

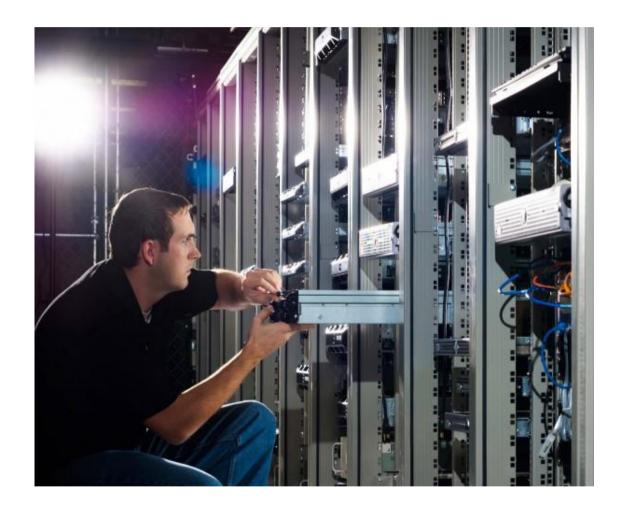
Call Routing >> Route/Hunt >> Line Group >> Add New





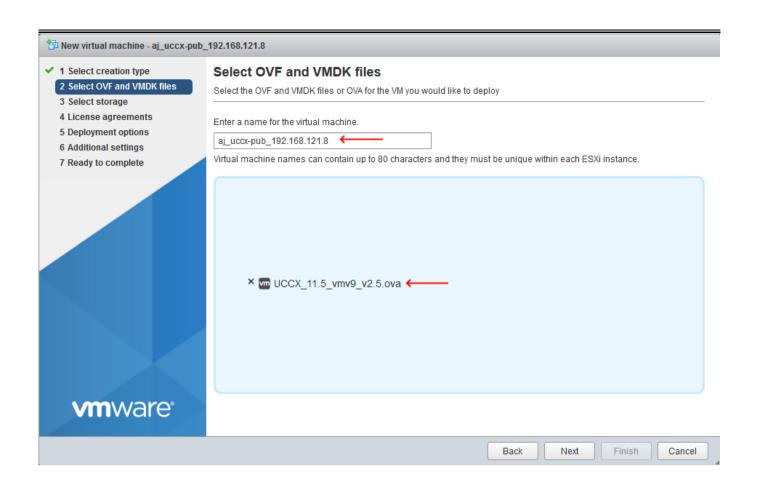


Chapter 1 Module 6 - UCCX Cisco On-Premise Collaboration Solution Cisco Unified Contact Center Express (UCCX)

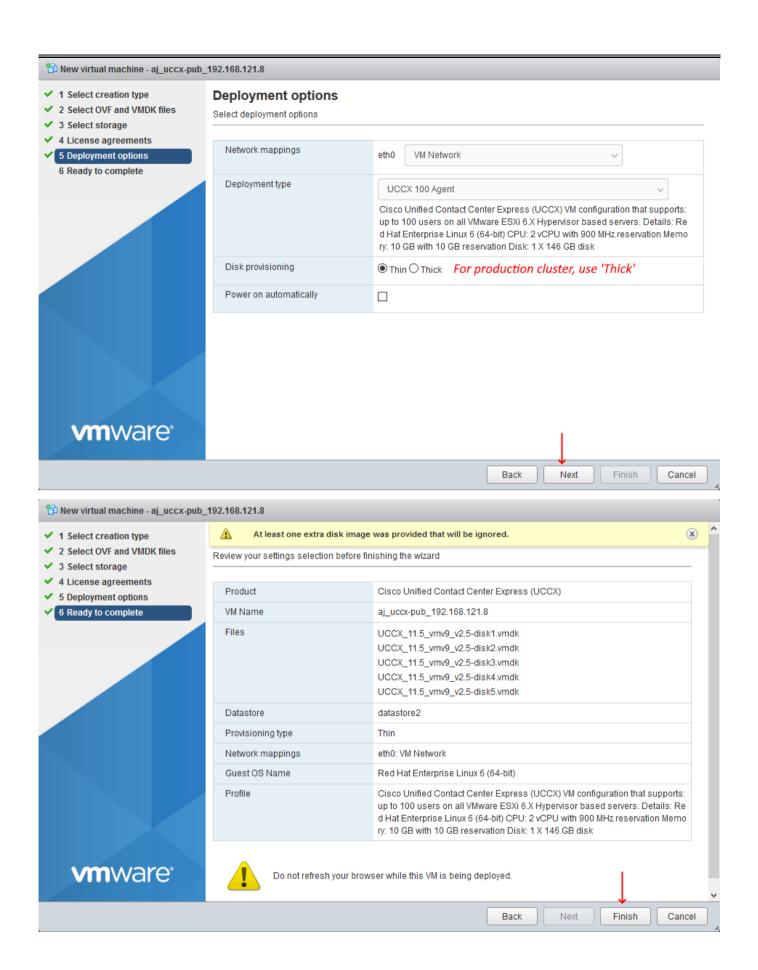


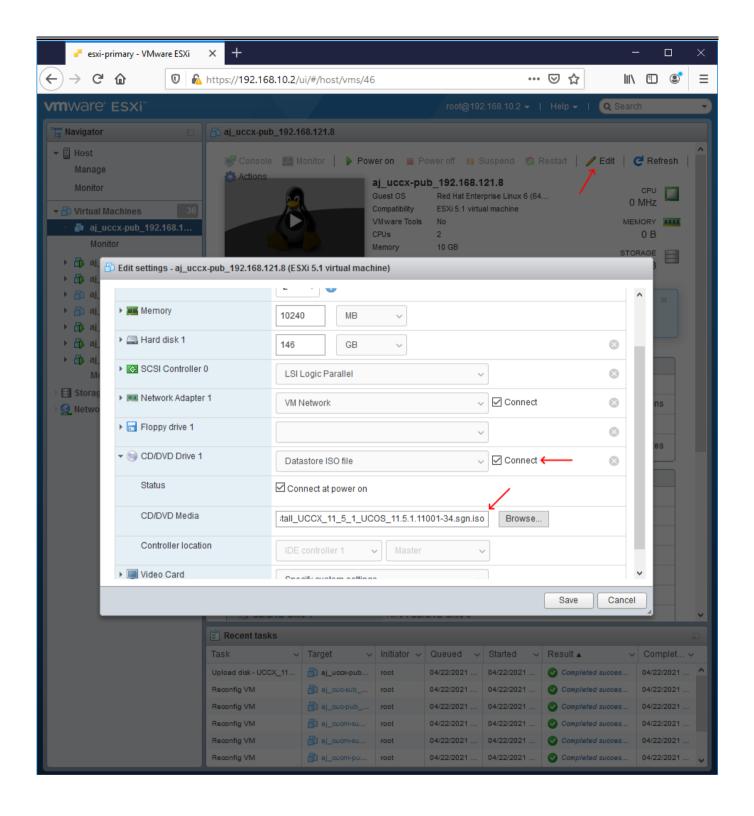
UCCX Publisher Installation A DNS Manager File Action View Help Name Туре Data Timestamp ✓ WIN-SERVER-01 _msdcs ▼ III Forward Lookup Zones _sites _msdcs.ajcollab.com ____tcp ajcollab.com _udp > Property Reverse Lookup Zones Domain Dns Zones > III Trust Points ForestDnsZones > Conditional Forwarders (same as parent folder) Start of Authority (SOA) [87], win-server-01.ajcolla... static (same as parent folder) Name Server (NS) win-server-01.ajcollab.com. static (same as parent folder) Host (A) 192.168.111.1 4/20/2021 3:00:00 AM 192.168.121.4 Host (A) cuc-pub static cuc-sub Host (A) 192.168.121.5 static cucm-pub Host (A) 192,168,121,1 static cucm-sub01 Host (A) 192.168.121.2 static cucm-sub02 Host (A) 192.168.121.3 static imp-pub Host (A) 192,168,121,6 static 192.168.121.7 imp-sub Host (A) static win-server-01 Host (A) 192.168.111.1 static uccx-pub Host (A) 192.168.121.8 uccx-sub Host (A) 192.168.121.9 1 New virtual machine ✓ 1 Select creation type Select creation type 2 Select OVF and VMDK files How would you like to create a Virtual Machine? 3 Select storage 4 License agreements This option guides you through the process of creating a Create a new virtual machine 5 Deployment options virtual machine from an OVF and VMDK files 6 Additional settings Deploy a virtual machine from an OVF or OVA file 7 Ready to complete Register an existing virtual machine **vm**ware Finish Cancel Back Next

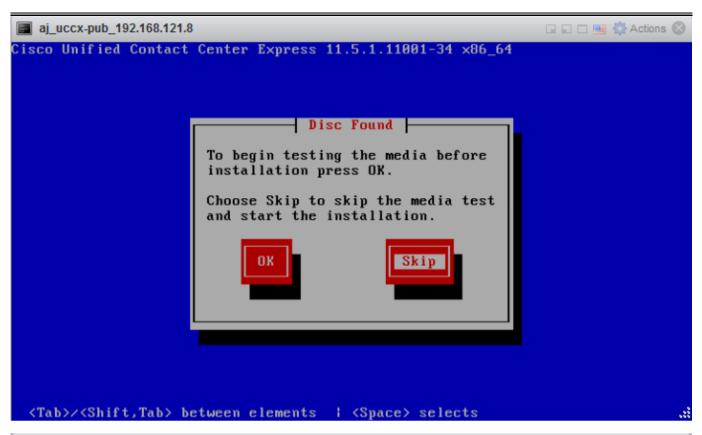
Please note, the IP Address in this demo is different from the lab topology given at the beginning of this document. I have taken this during a different deployment

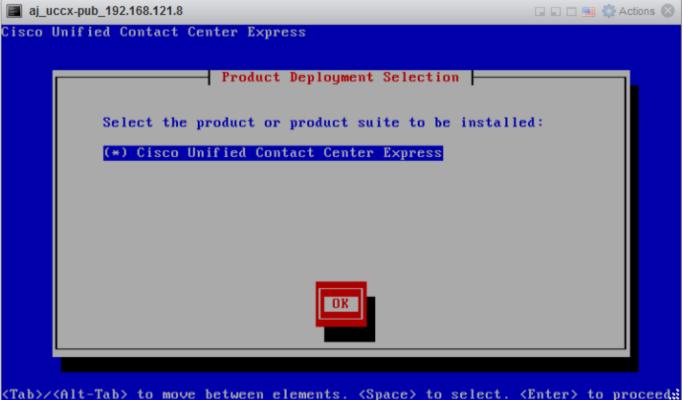


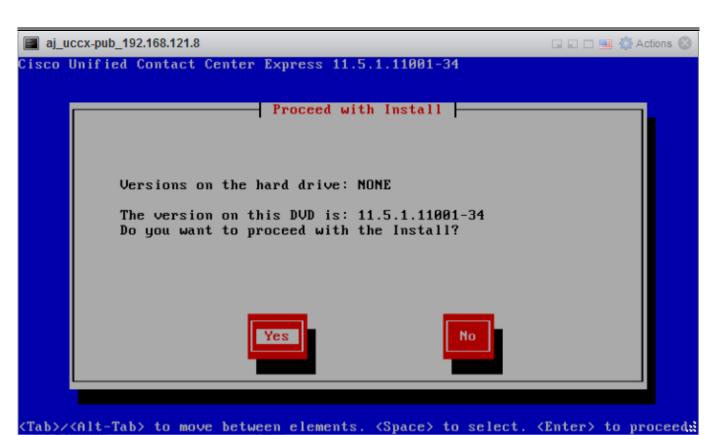


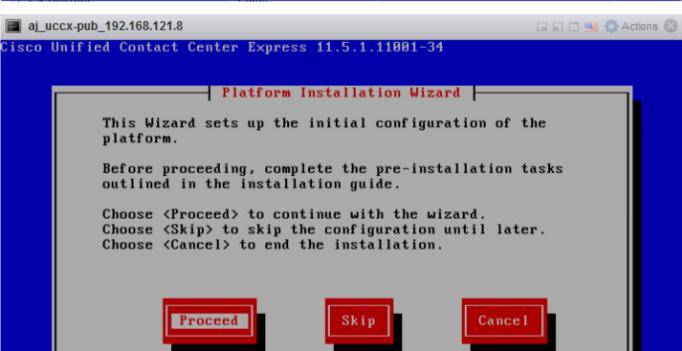




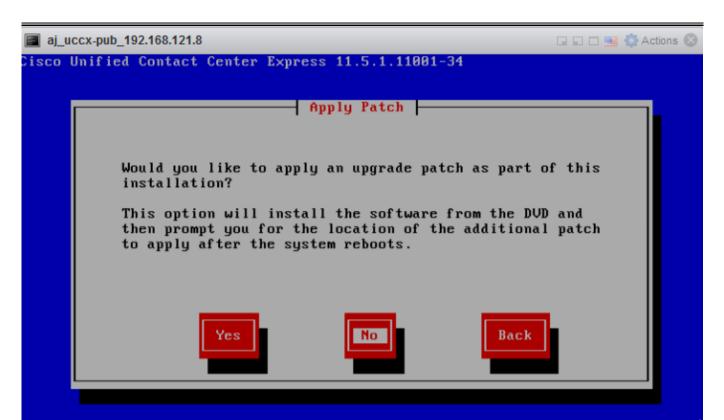




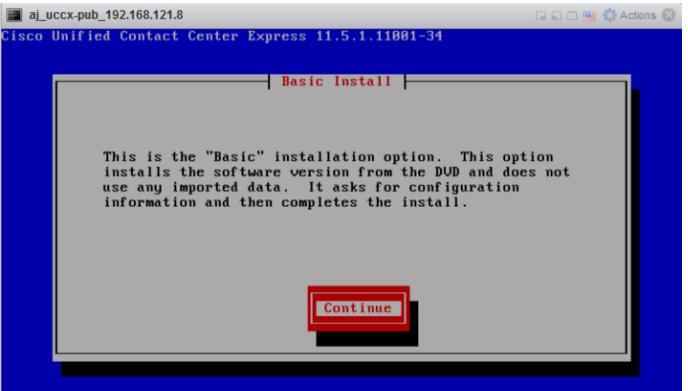




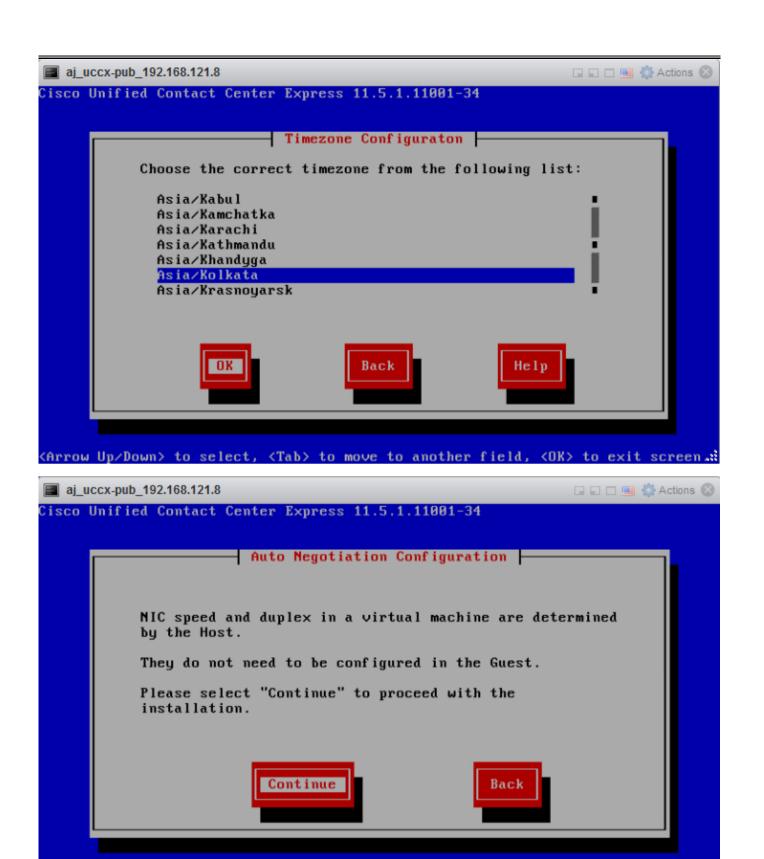
<Tab>/<Alt-Tab> to move between elements. <Space> to select. <Enter> to proceed:



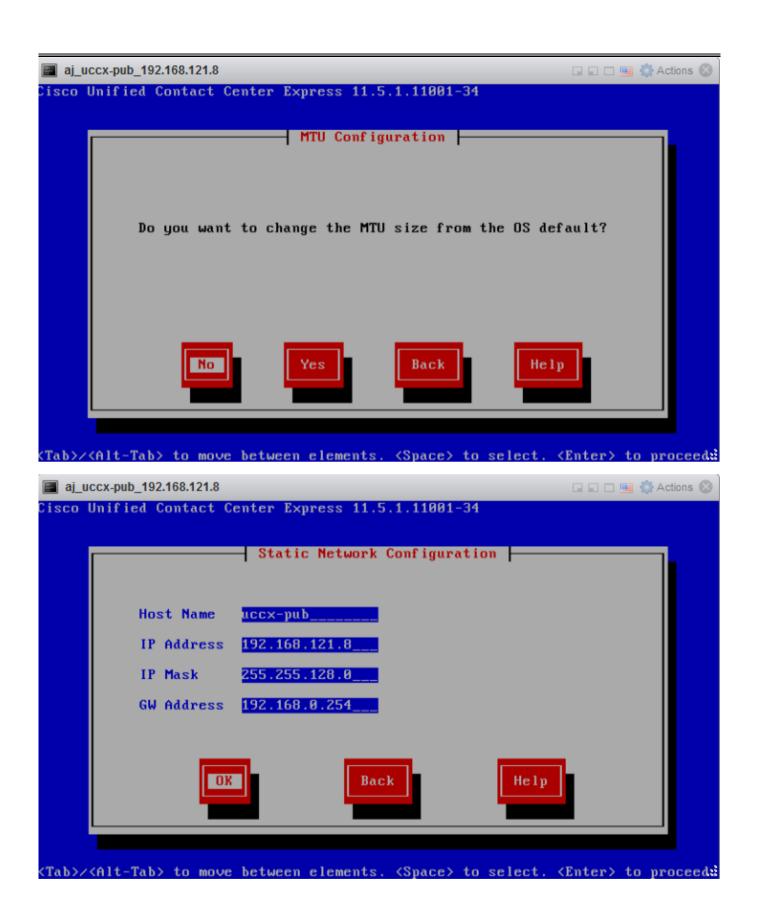
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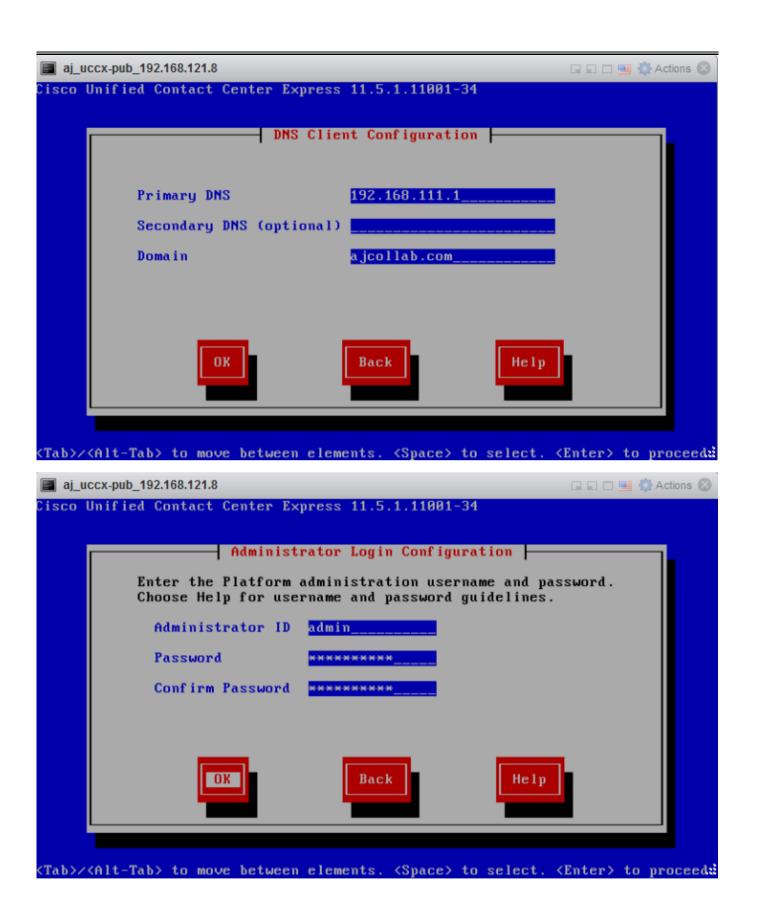


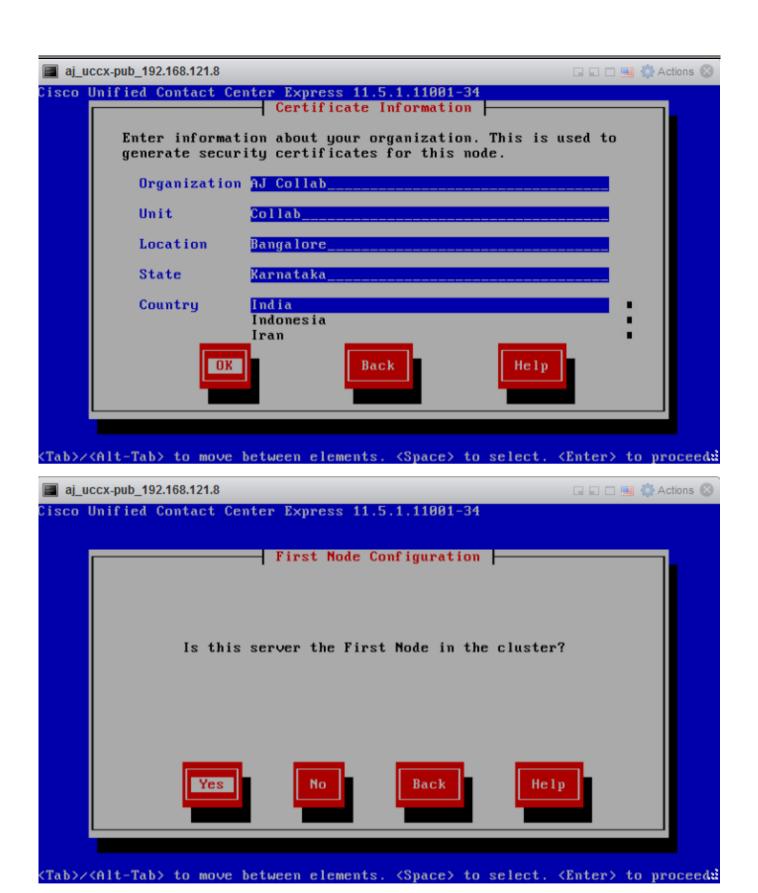
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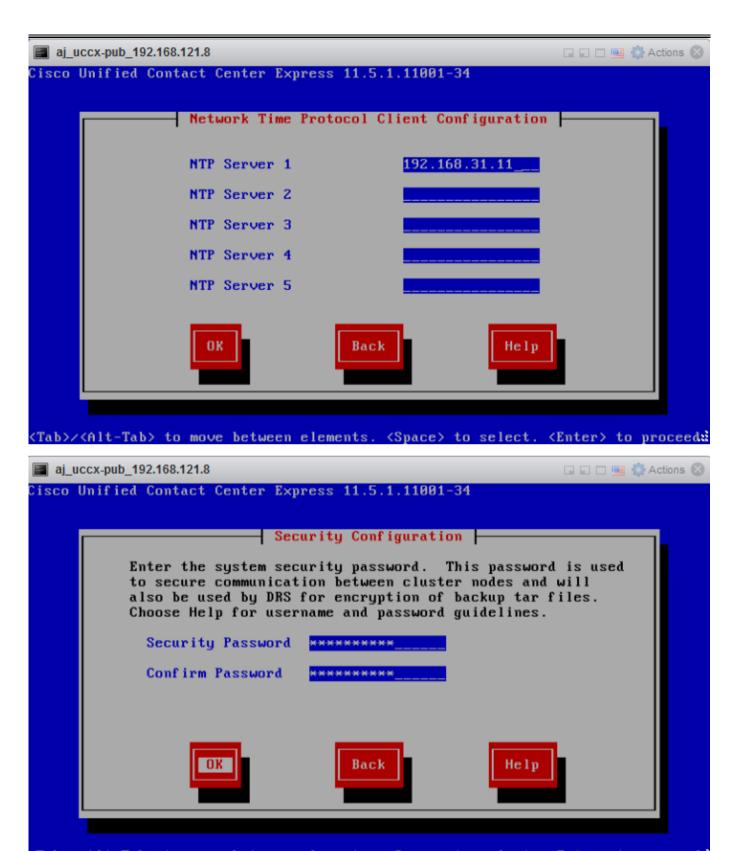


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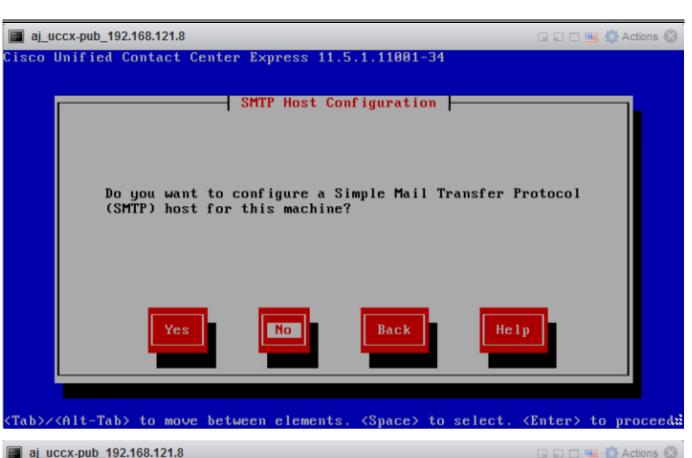








(Tab>/<Alt-Tab> to move between elements. <Space> to select. <Enter> to proceed:





<Tab>/<Alt-Tab> to move between elements. <Space> to select. <Enter> to proceed:

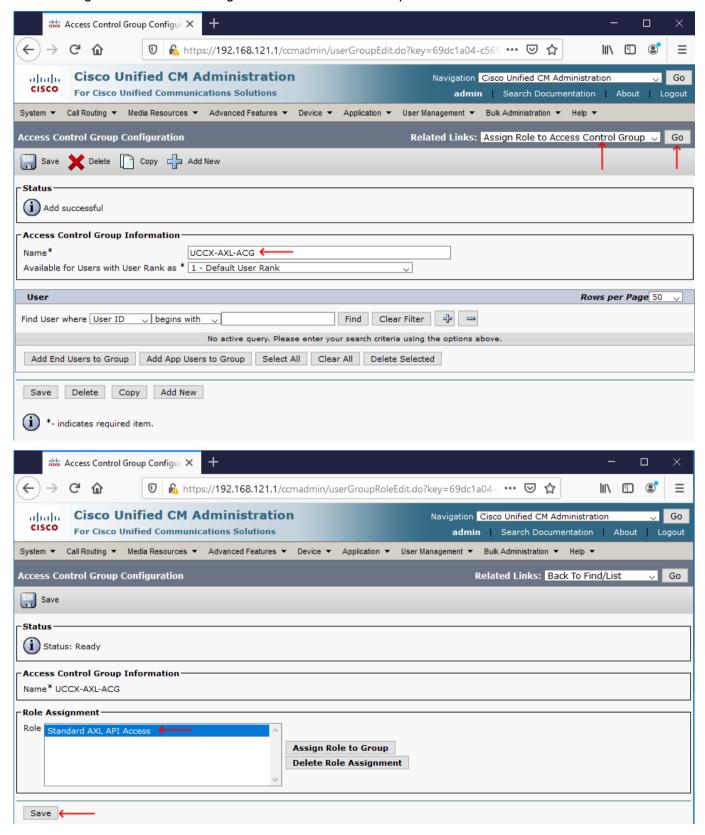


 UCCX Subscriber can be installed after UCCX Publisher has been initialized and integrated with CUCM

UCCX Initialization, Licensing and Integration with CUCM

Create an access control group with Standard AXL API Access Role

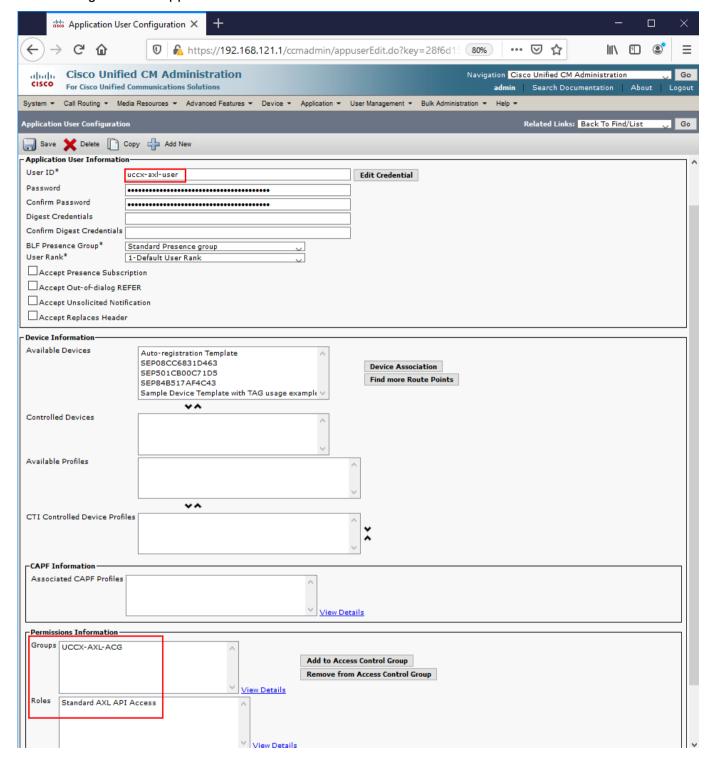
User Management >> User Settings >> Access Control Groups >> Add New

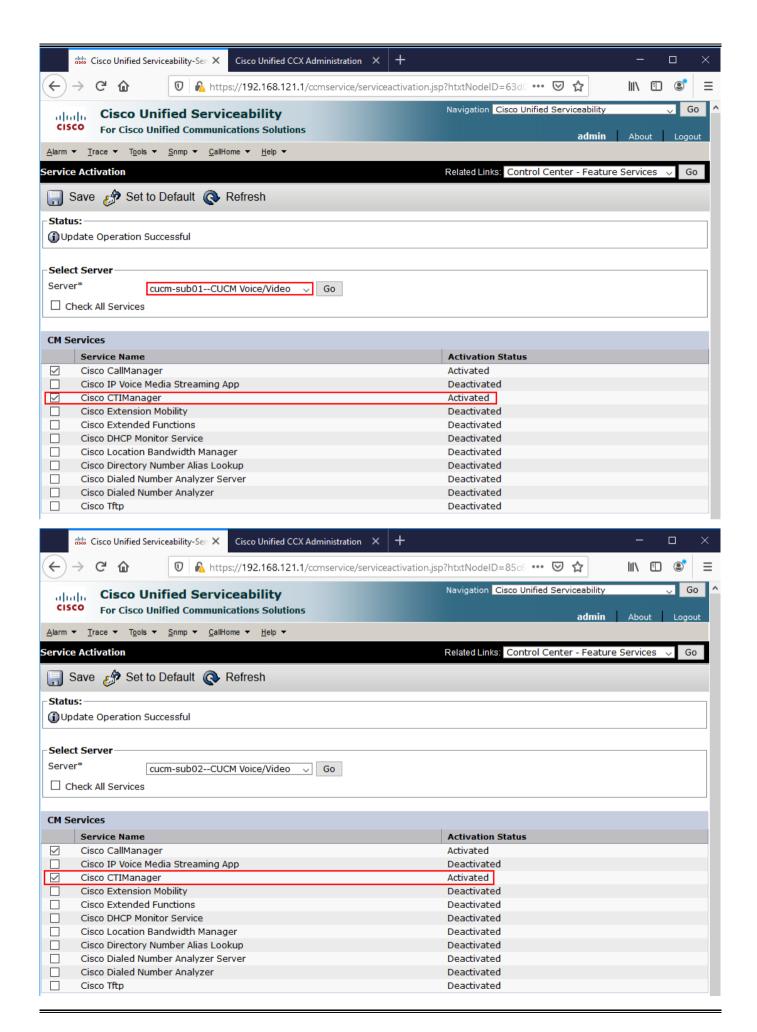


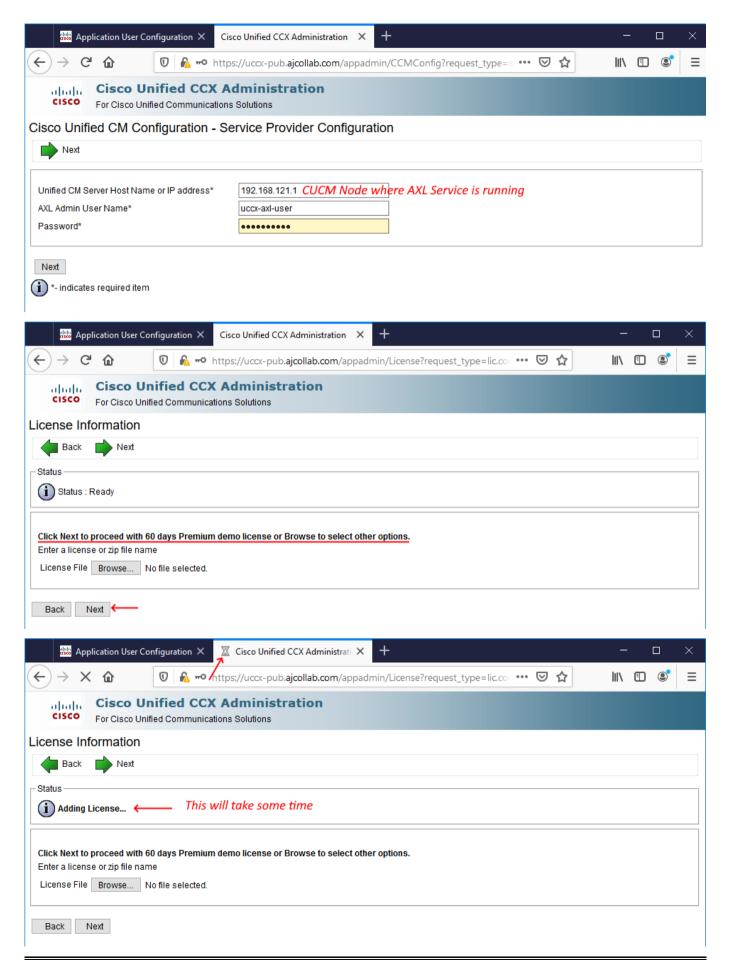
730

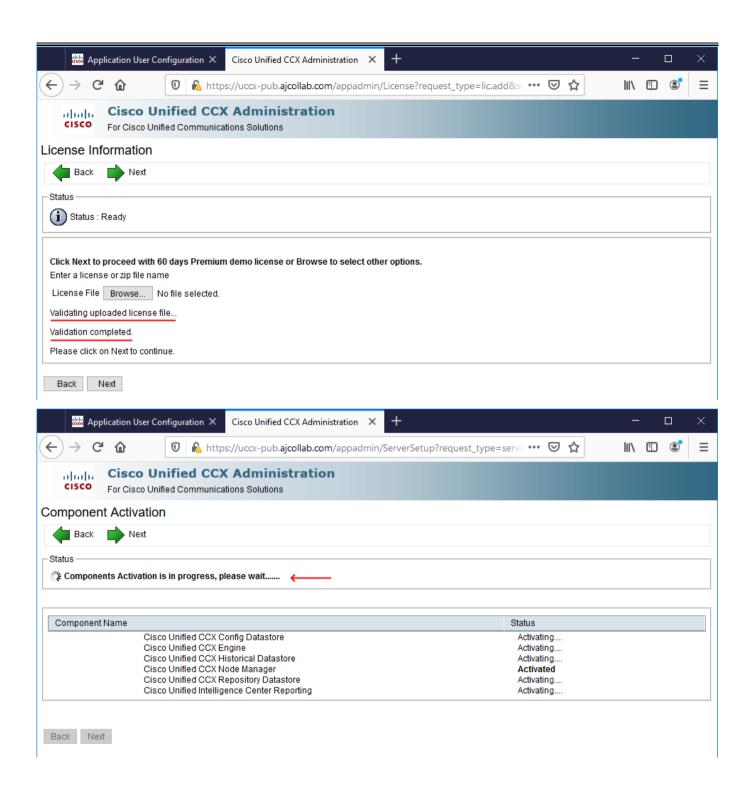
• Create an Application user in CUCM (uccx-axl-user)

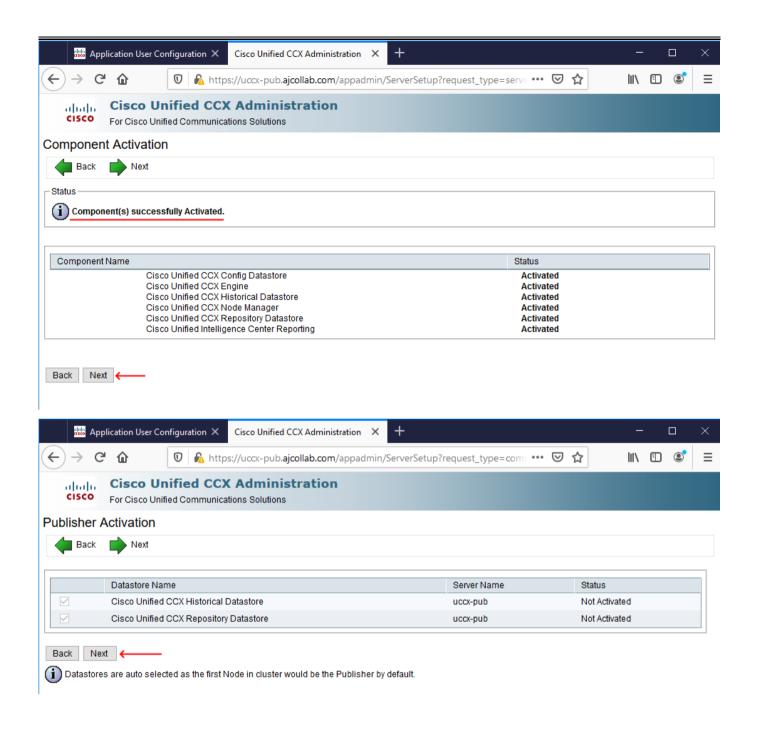
User Management >> Application User >> Add New >>

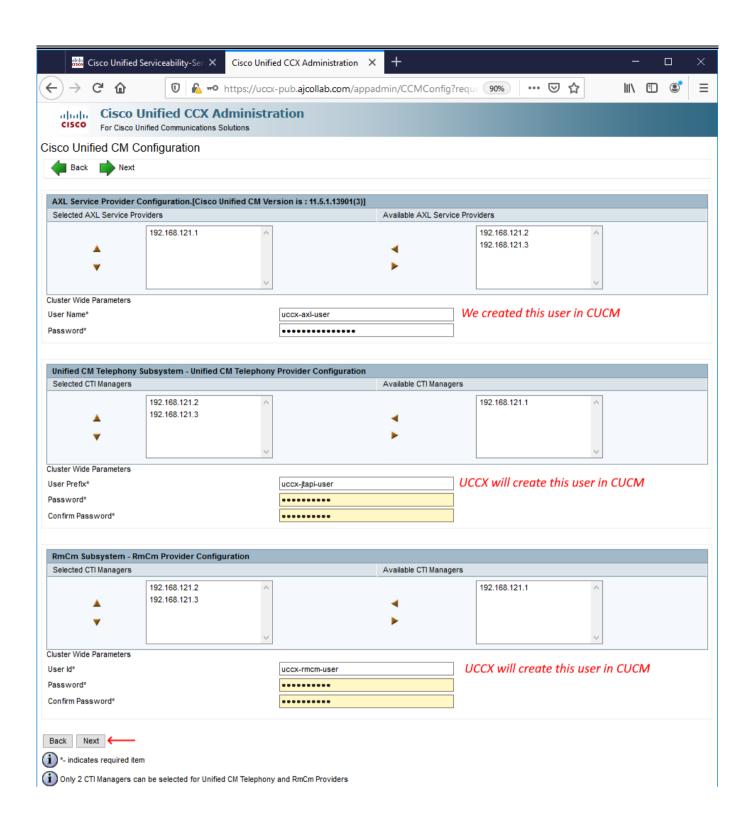


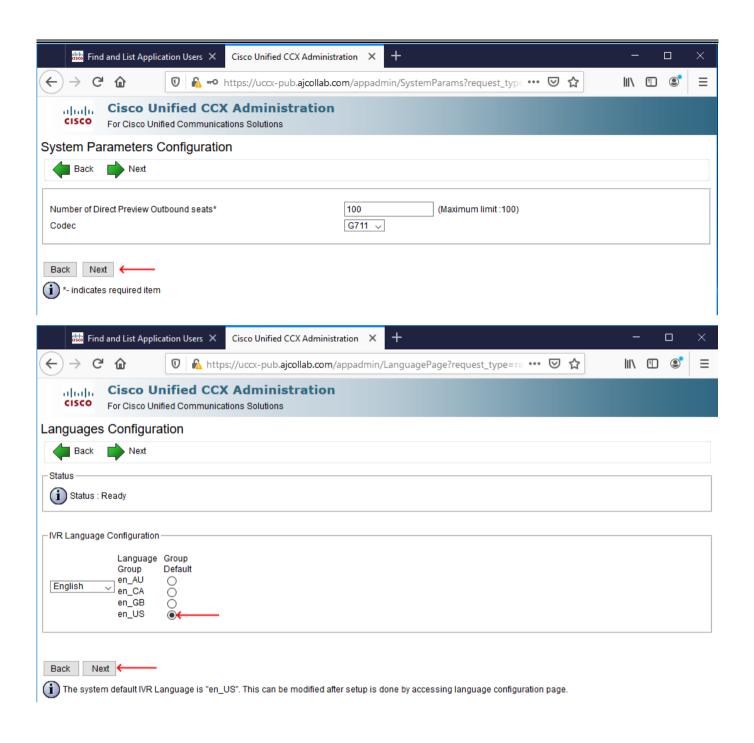




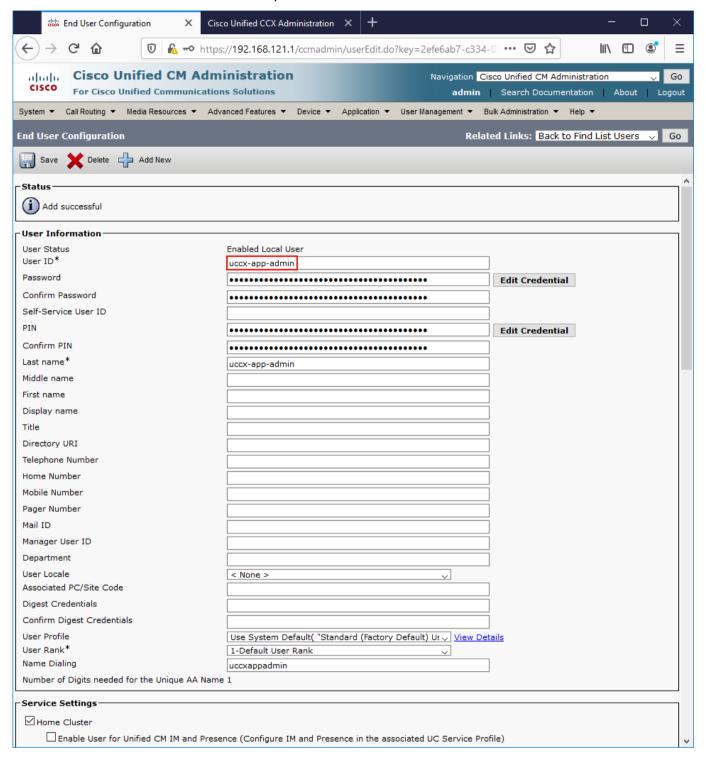


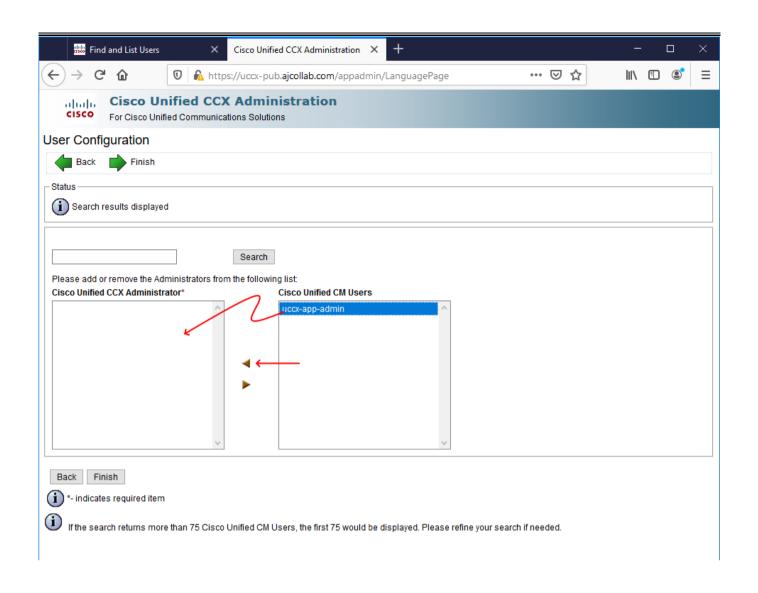


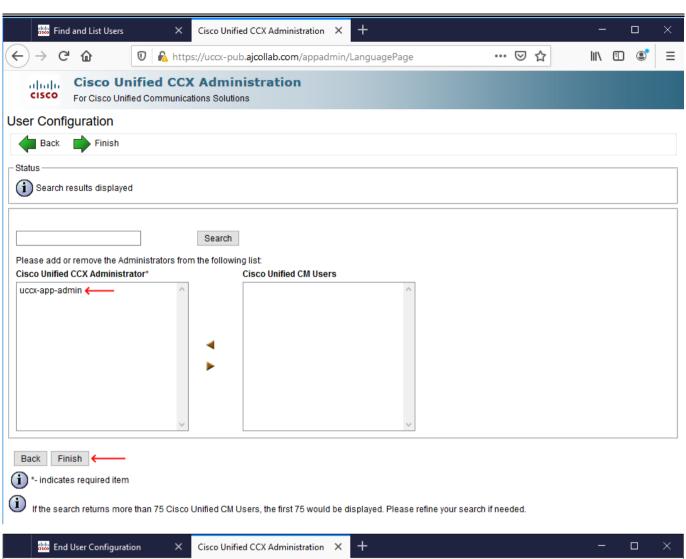


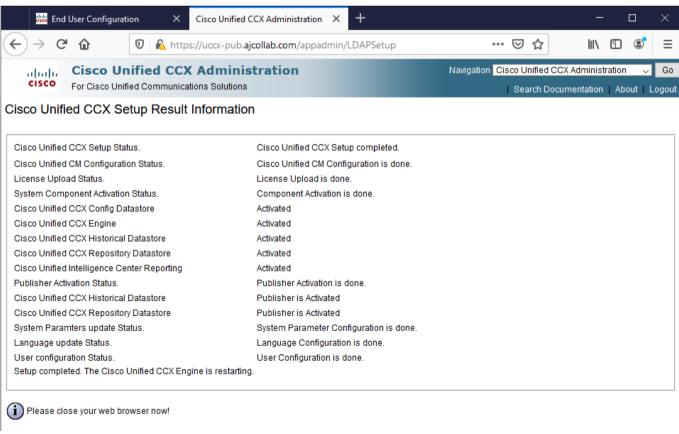


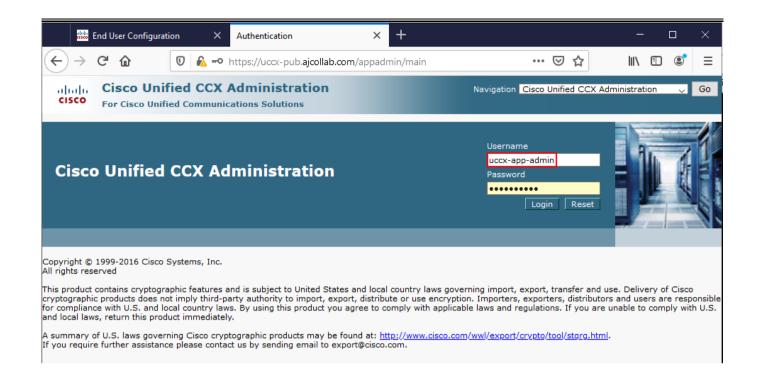
Create an end user in CUCM and promote that user as UCCX Administrator





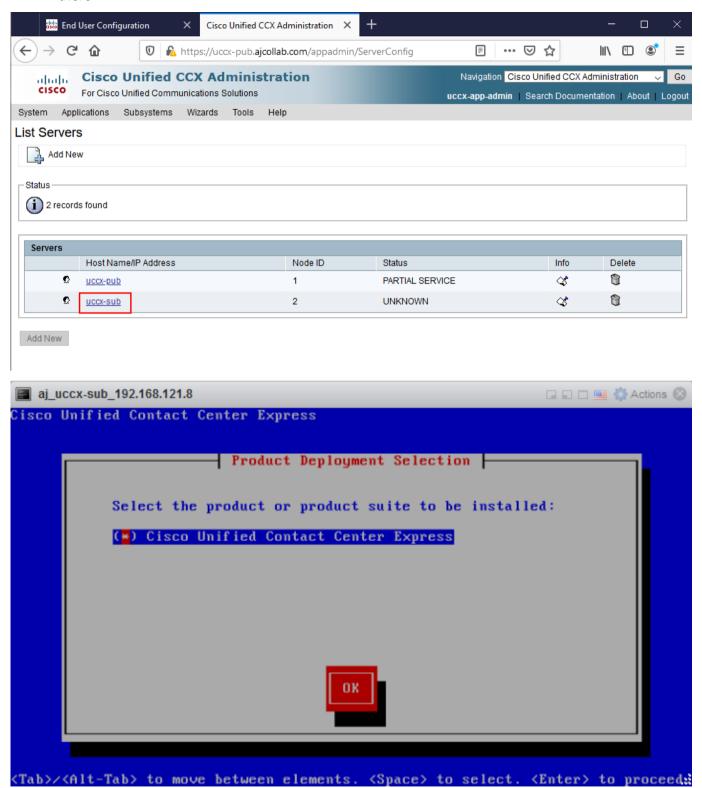






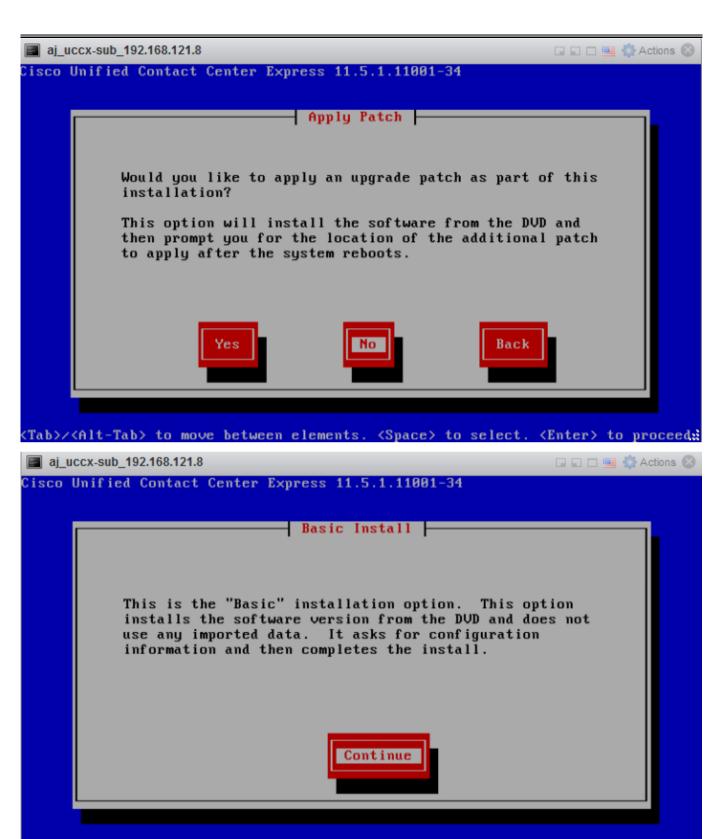
UCCX Subscriber Installation

- UCCS Subscriber can be installed after the initialization and integration of UCCX with CUCM
- After the initialization of UCCX Pub, go to System >> Server >> Add New and then add UCCX-SUB there

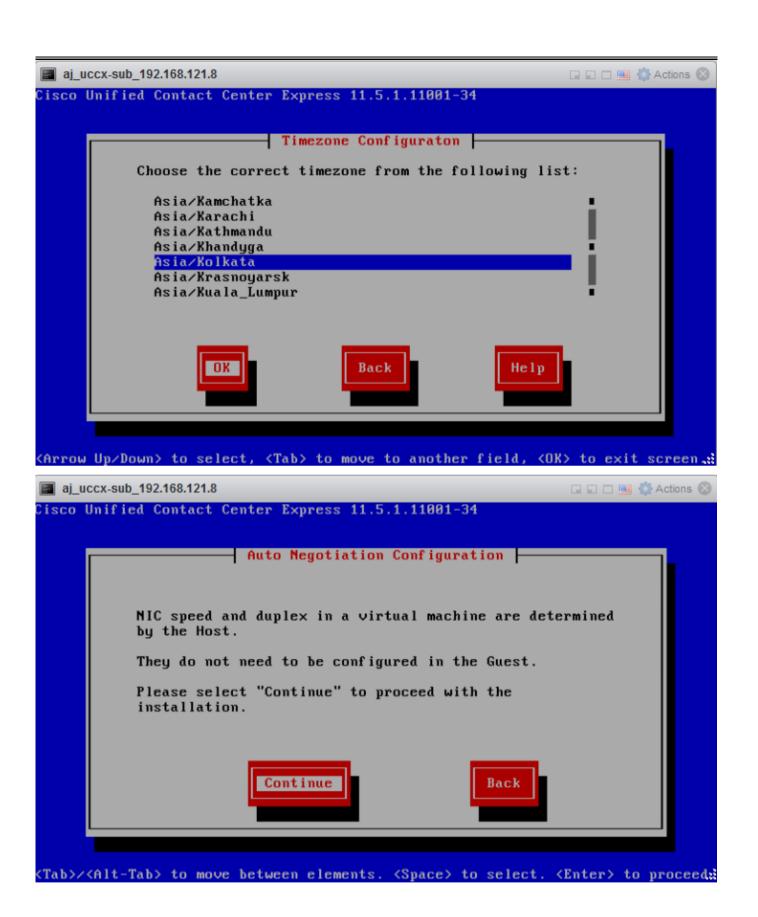


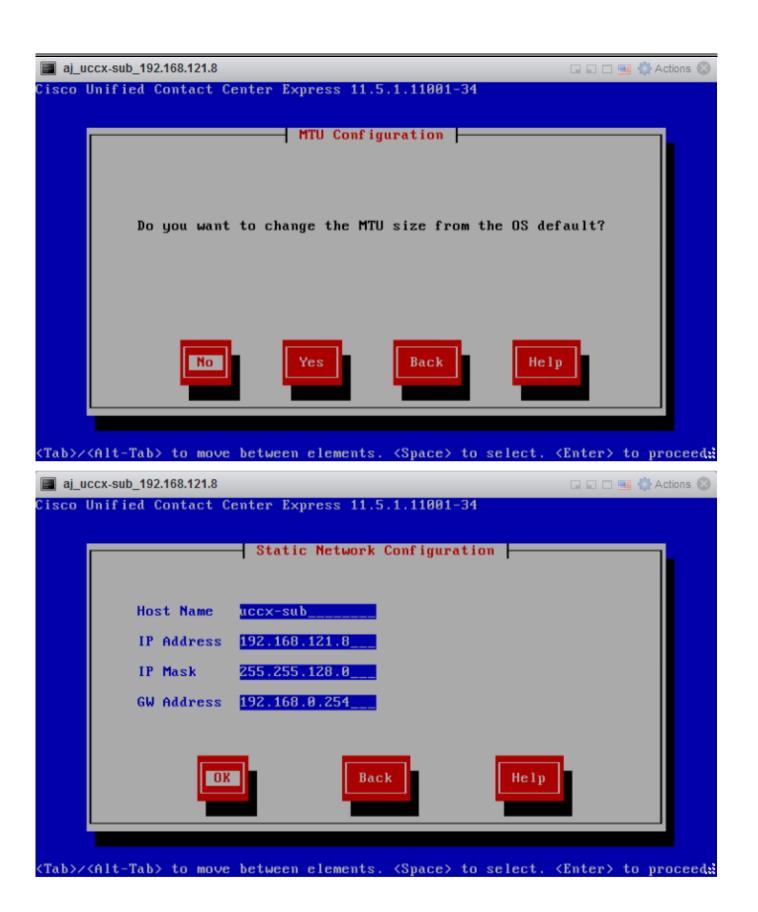


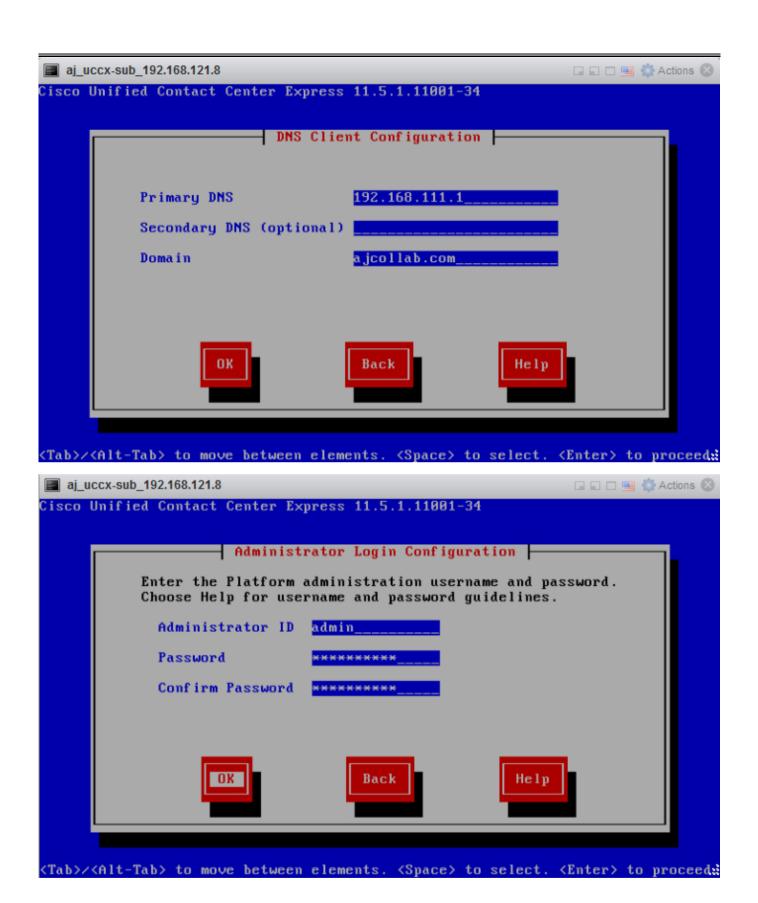
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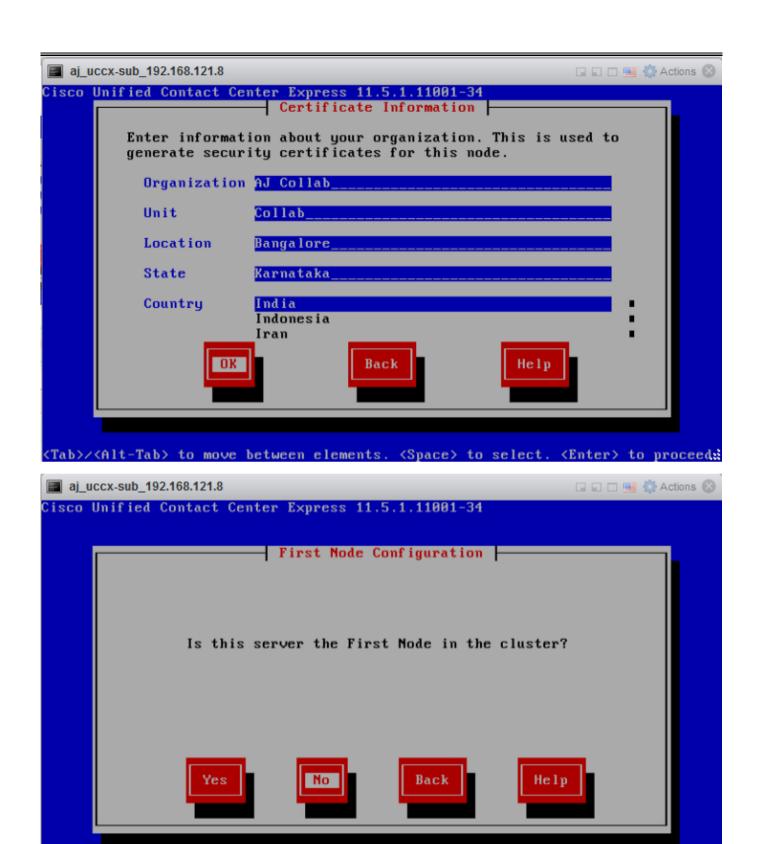


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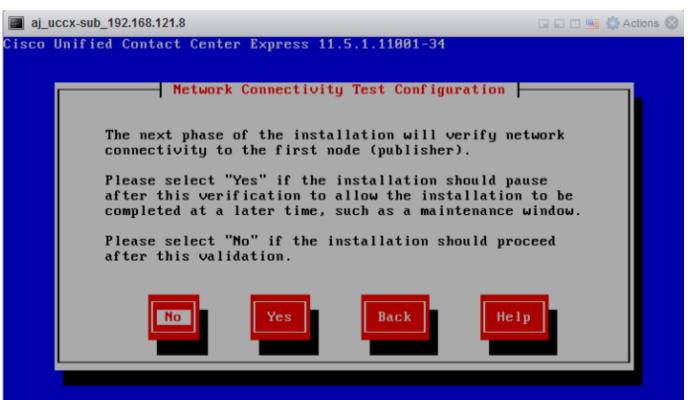




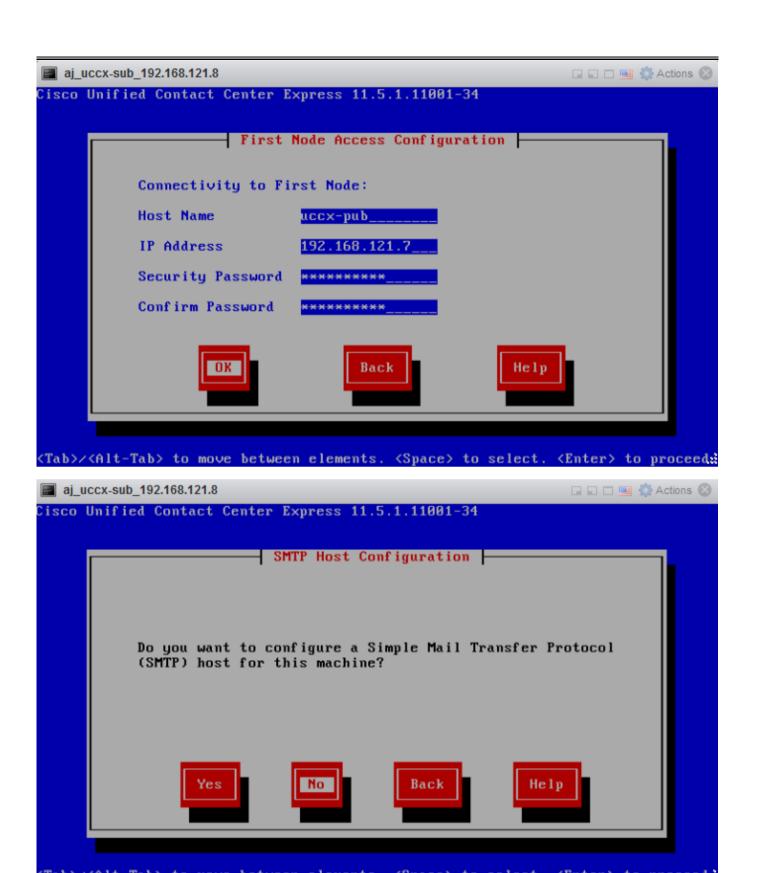
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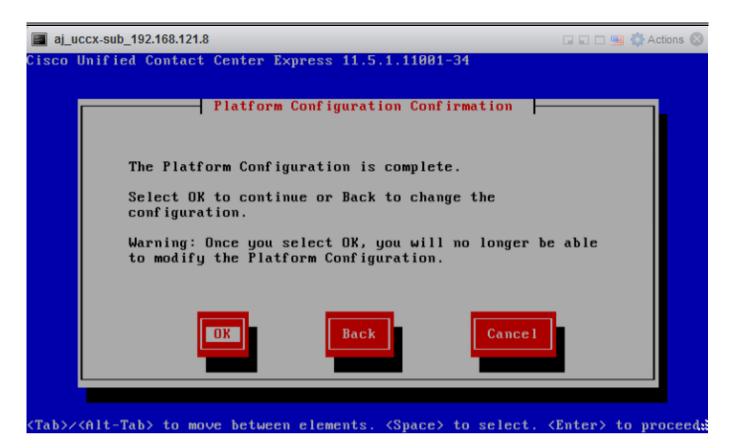
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<Tab>/<Alt-Tab> to move between elements. <Space> to select. <Enter> to proceed:



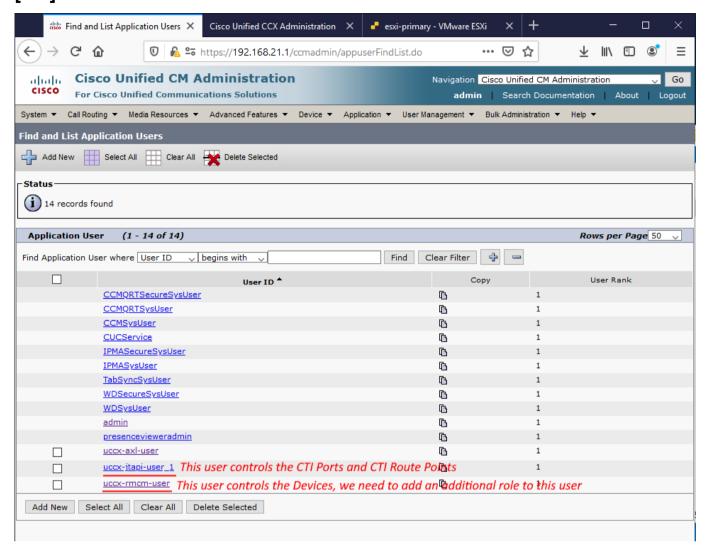
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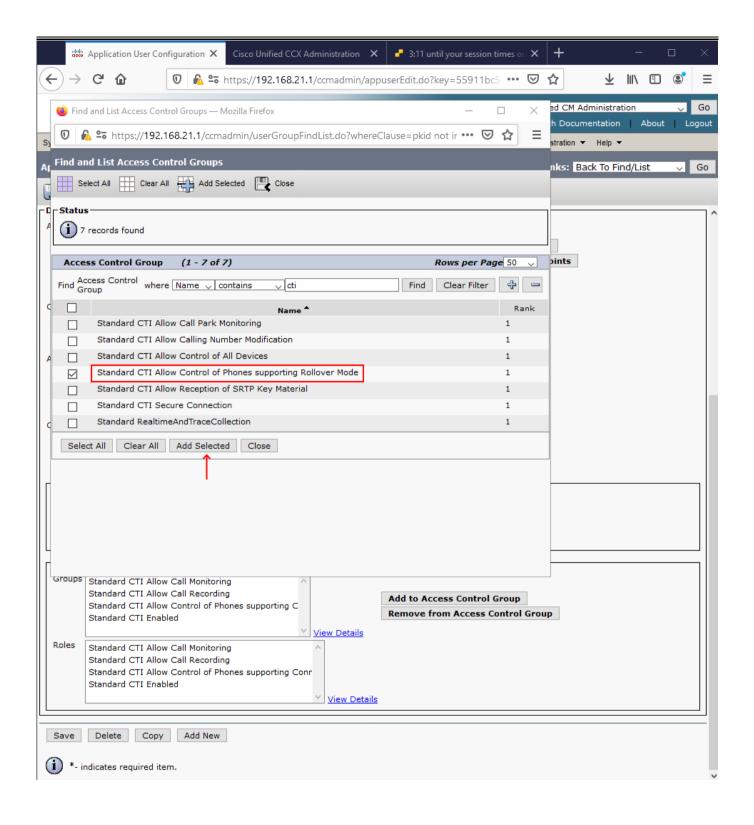


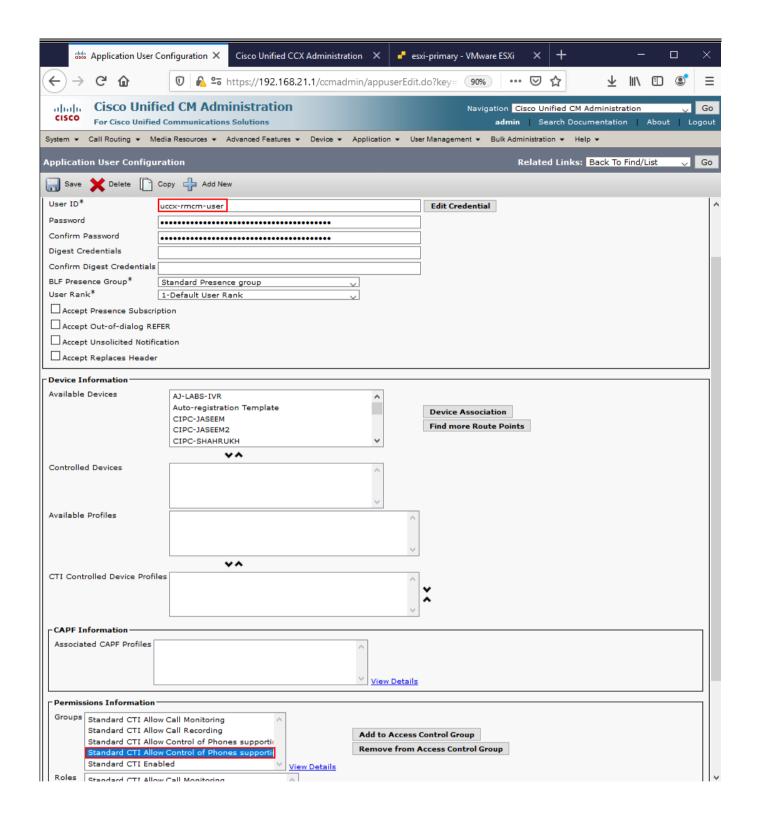
Note: By the way, I made a mistake in the VM Name, it is not aj_uccx-sub_192.168.121.8, it should be aj_uccx-sub_192.168.121.9

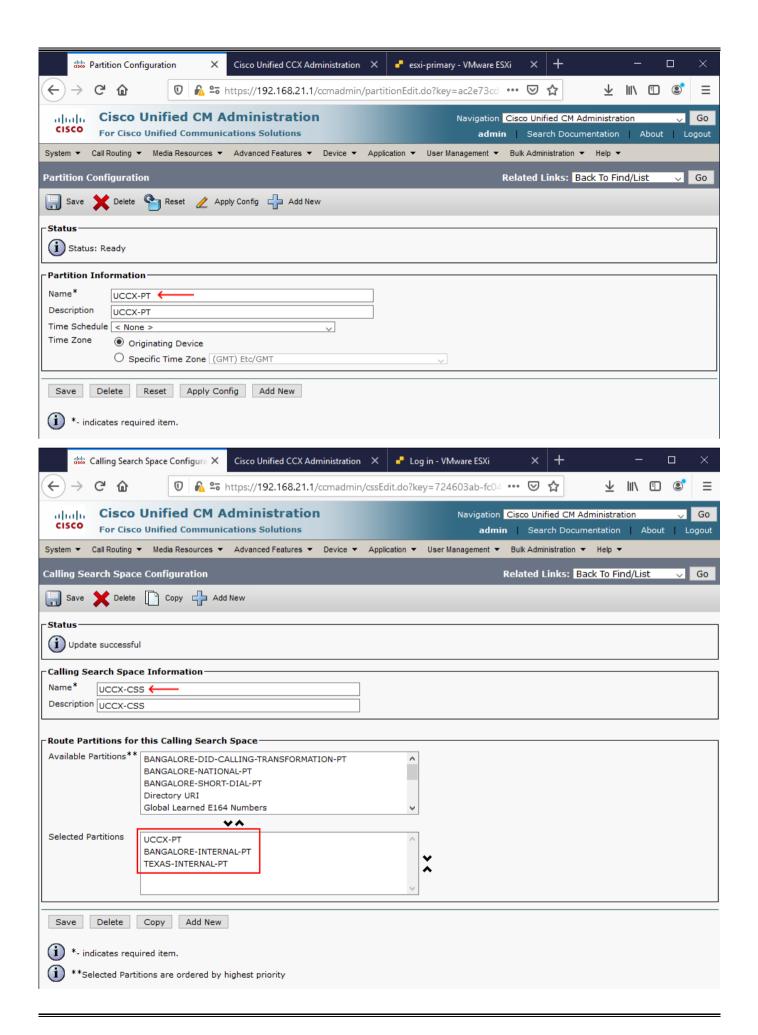
752

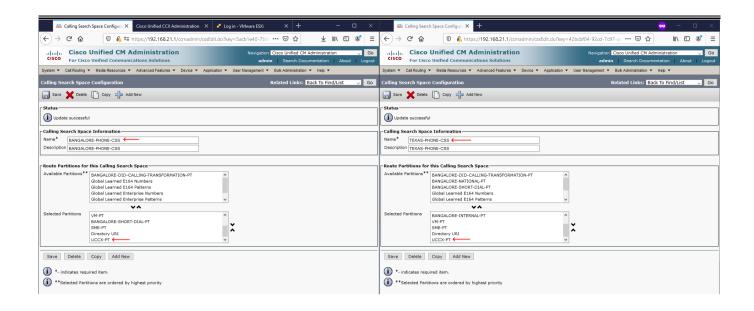
[Lab] UCCX Basic IVR Auto Attendant

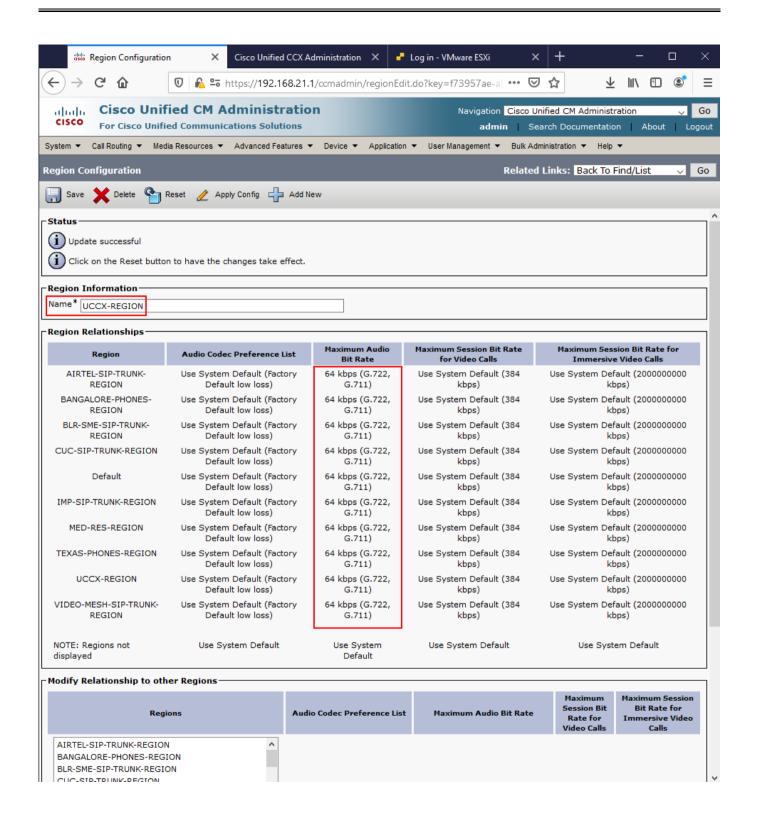


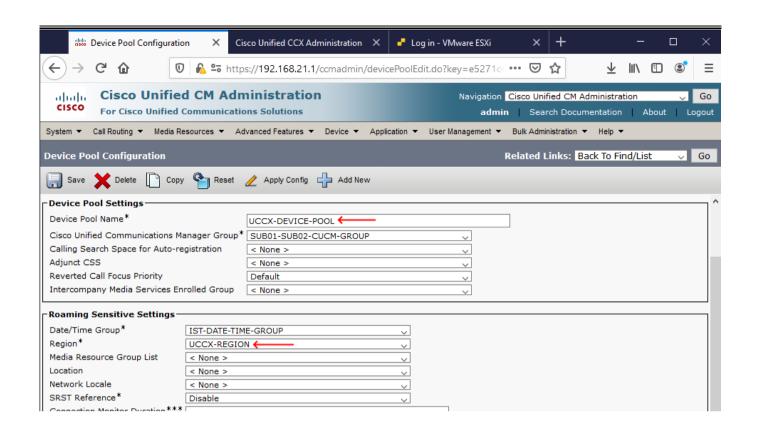




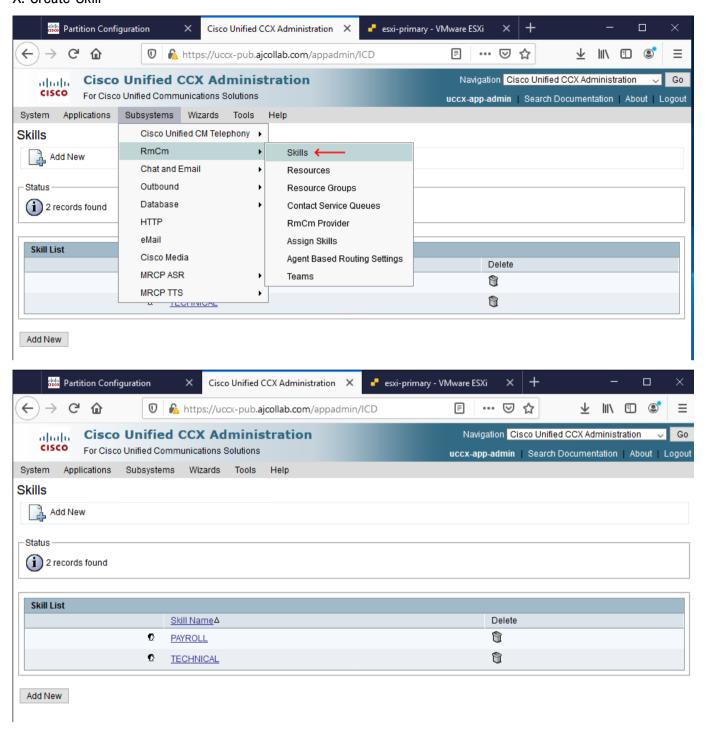




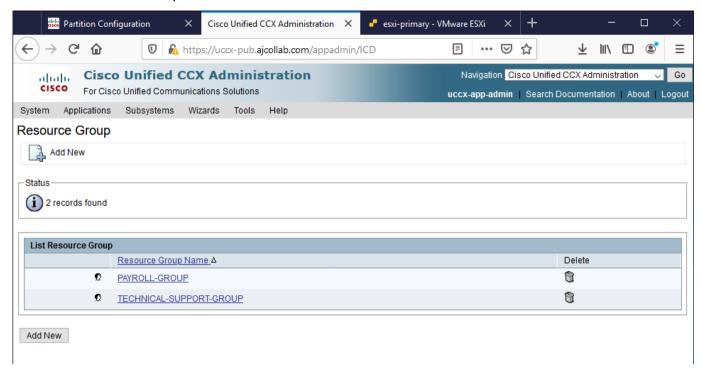




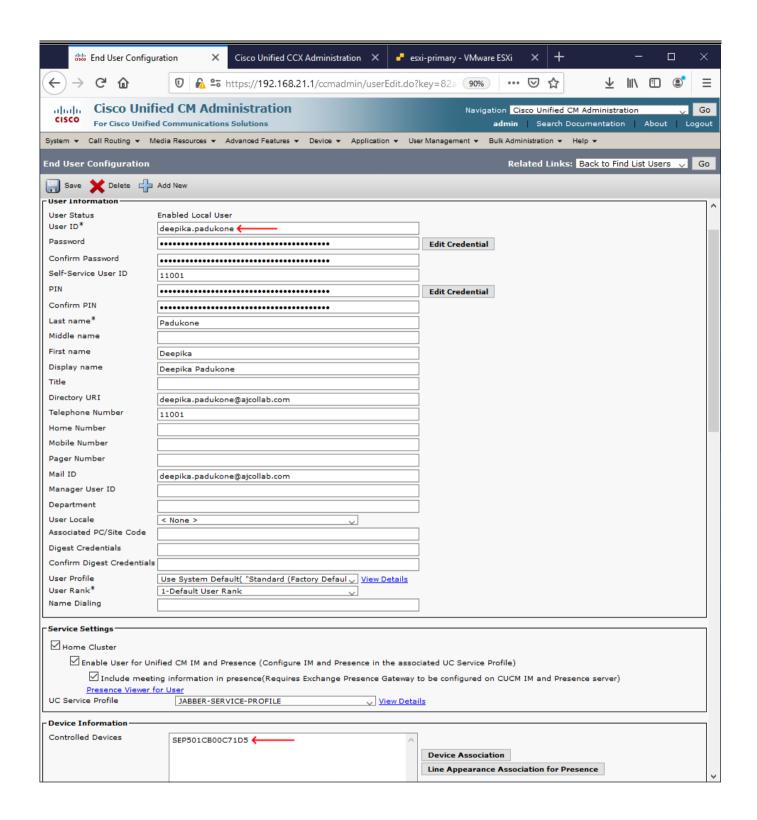
X. Create Skill

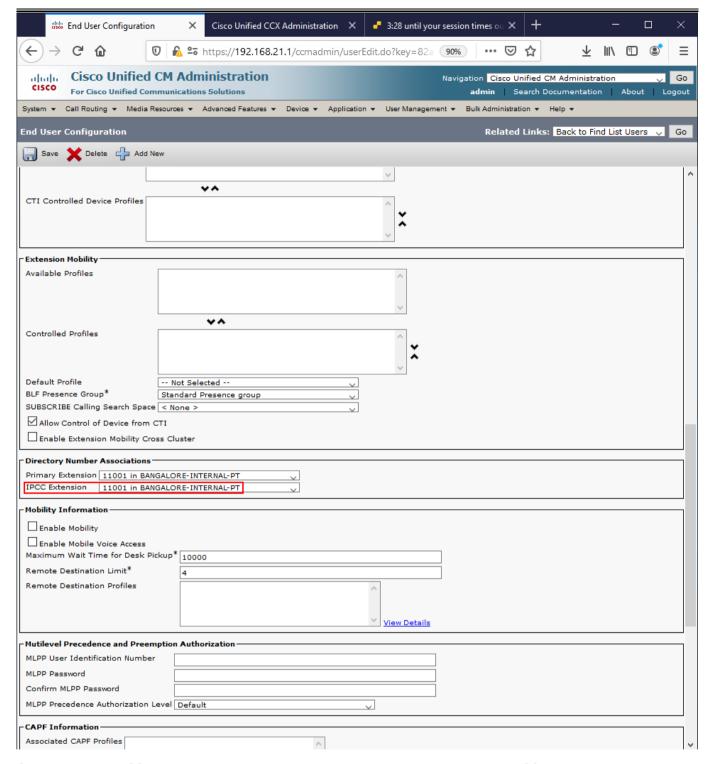


X. Create Resource Groups

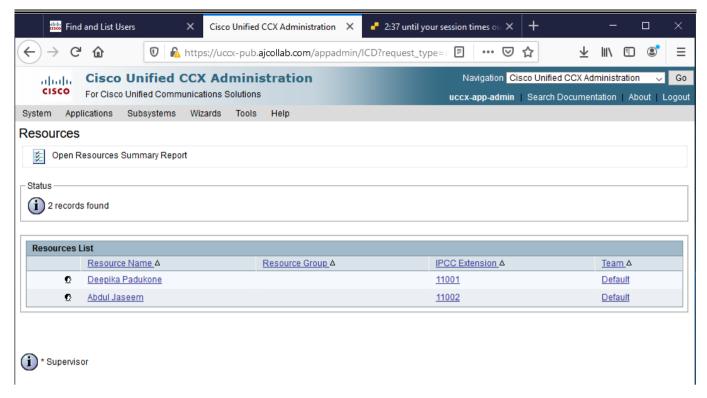


X. Configure Agents in CUCM

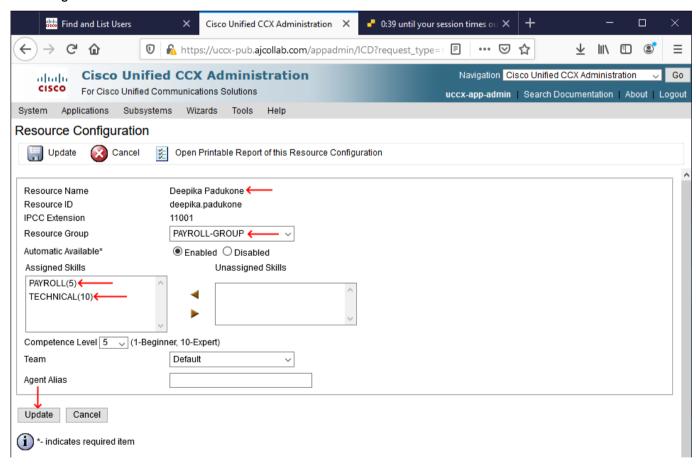


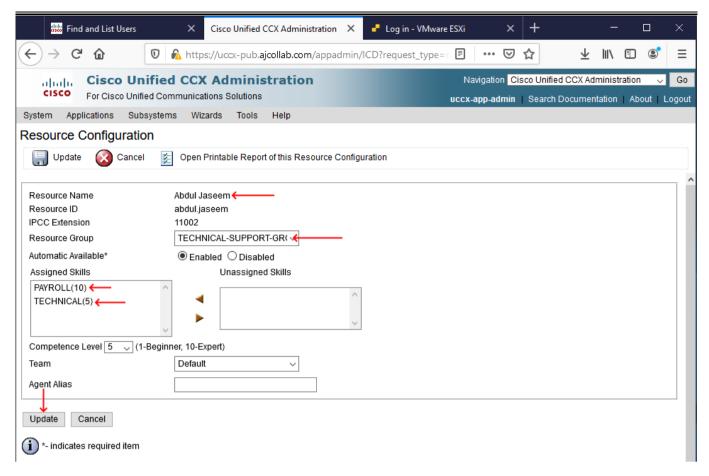


Once you add, IPCC Extension, those users will be automatically replicated to UCCX Resources

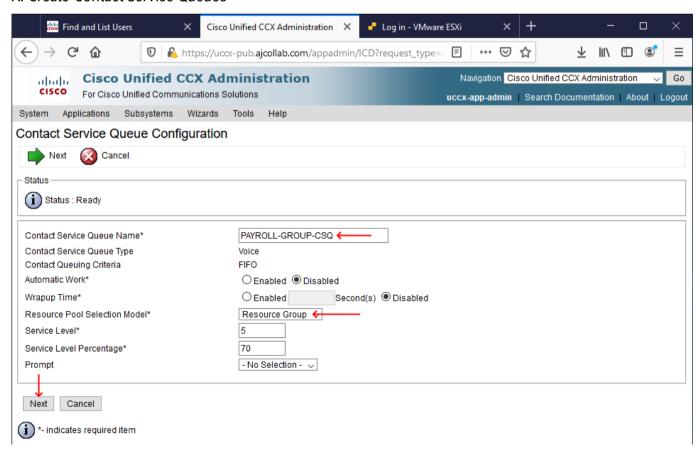


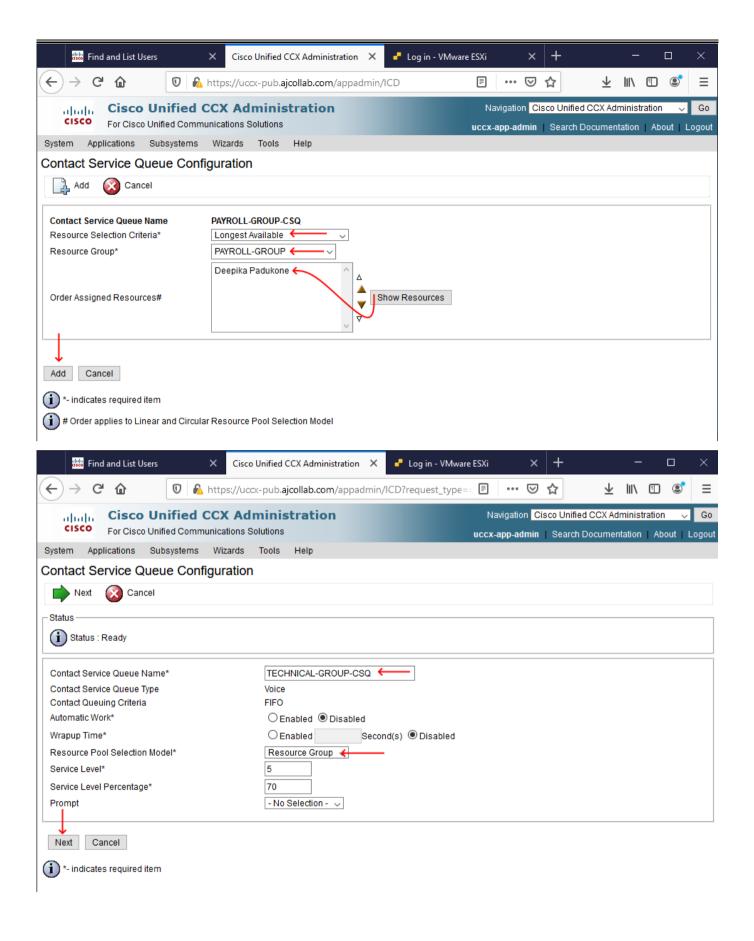
X. Configure the resource now

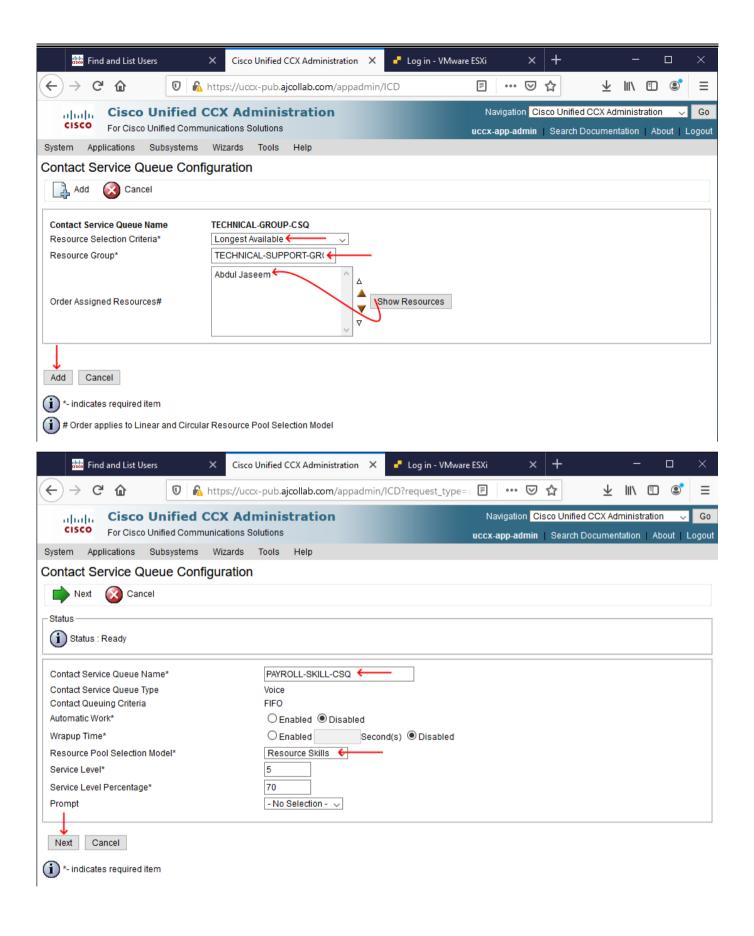


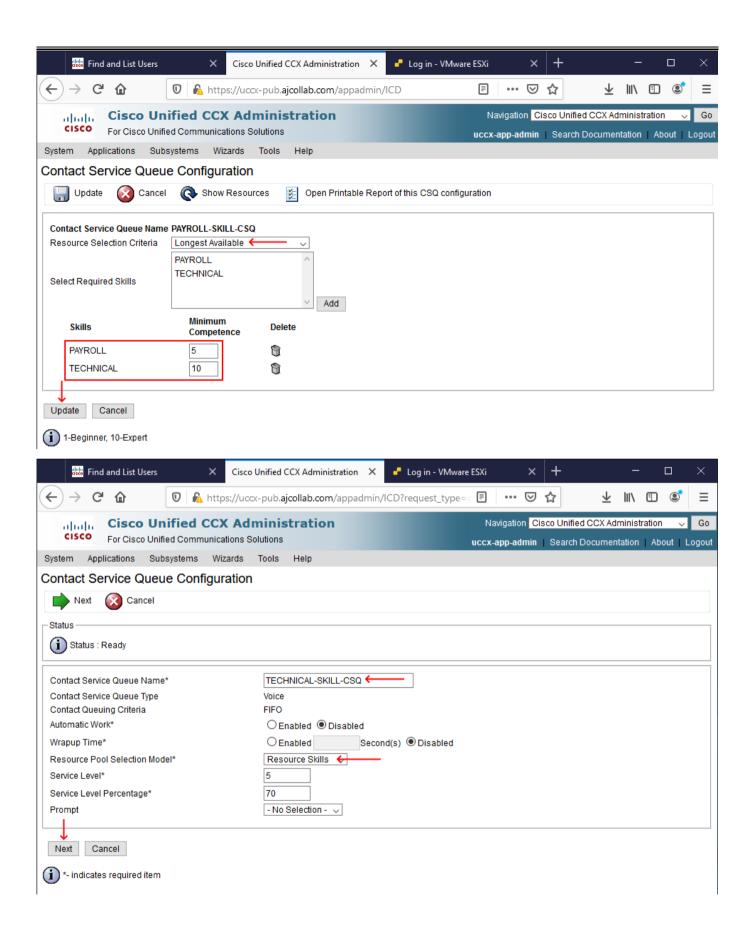


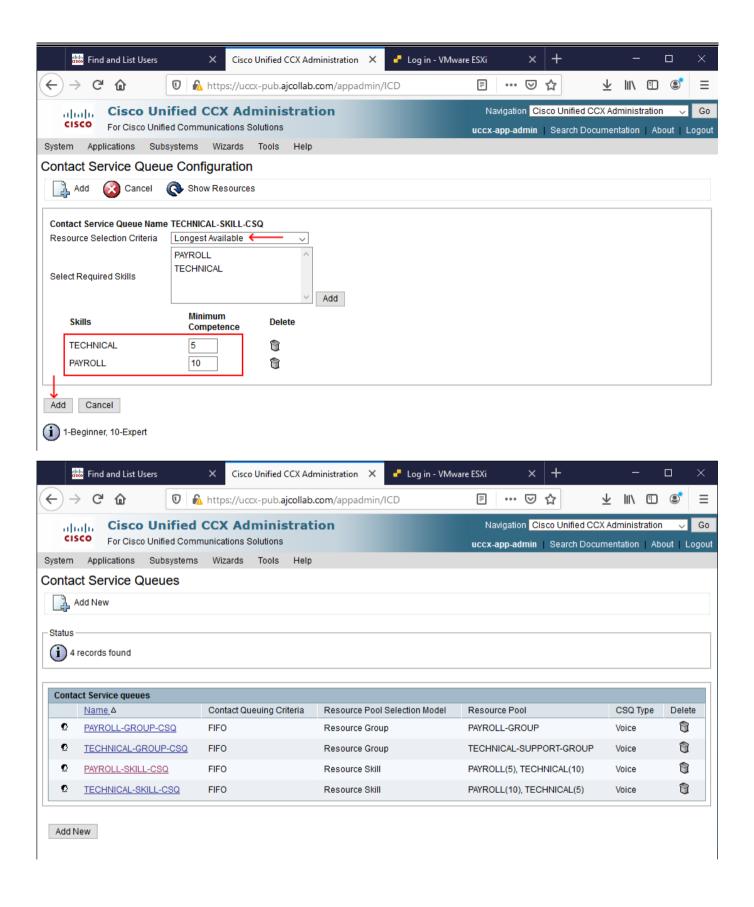
X. Create Contact Service Queues



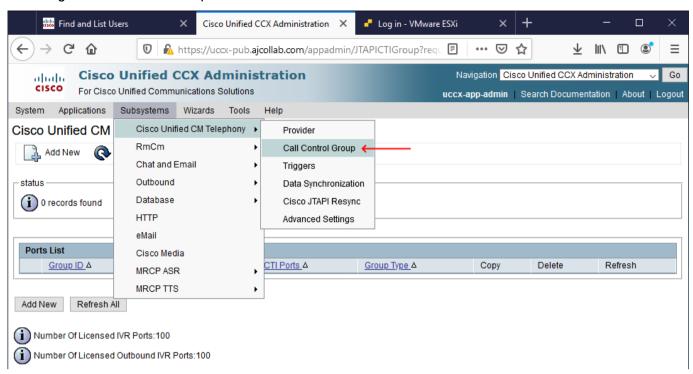


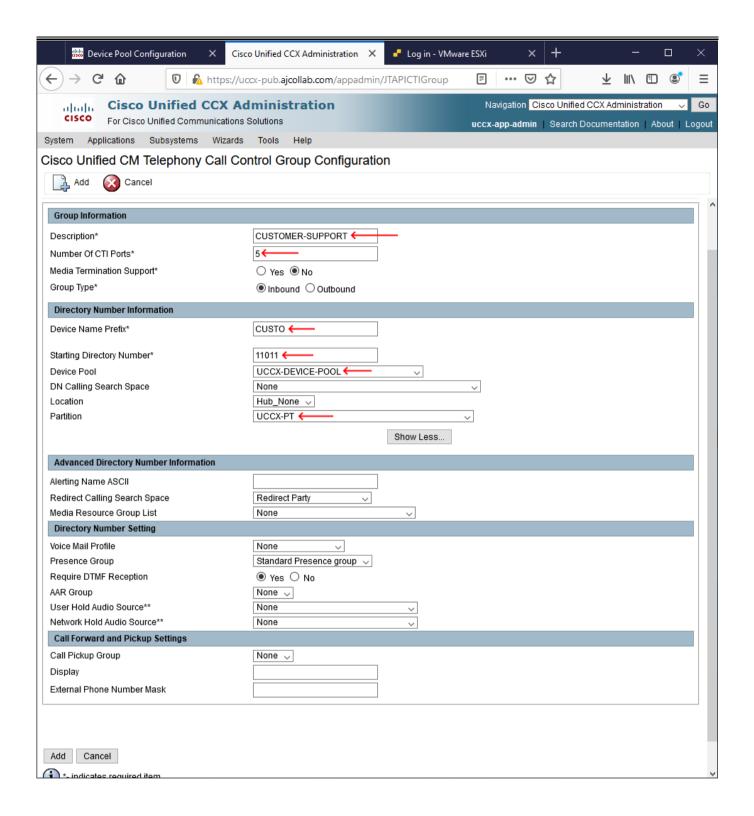


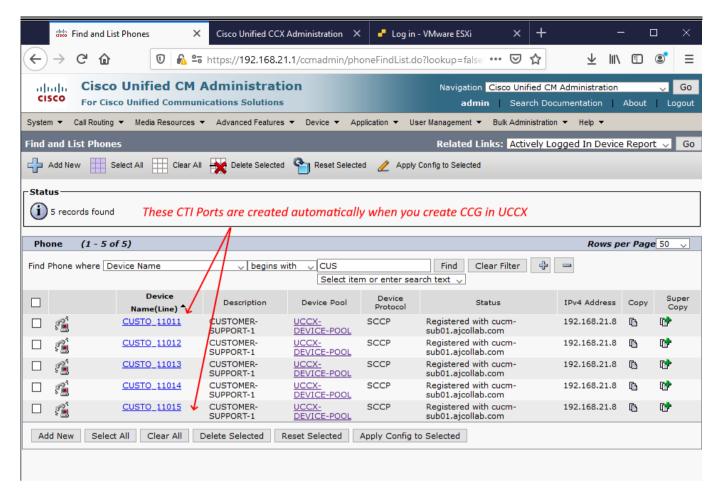




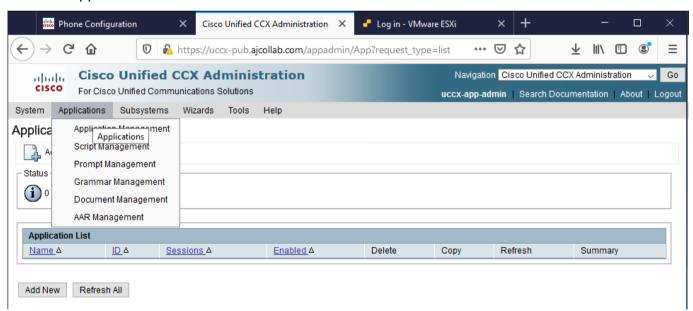
X. Configure Call Control Group

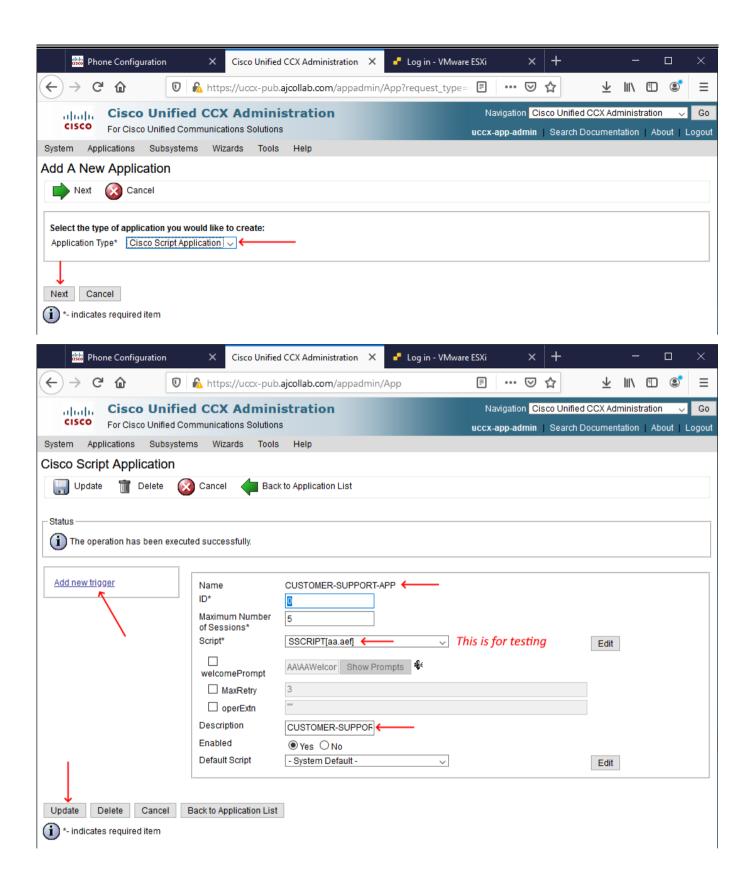


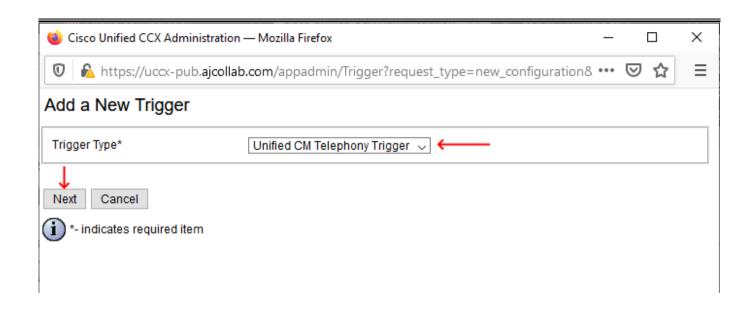


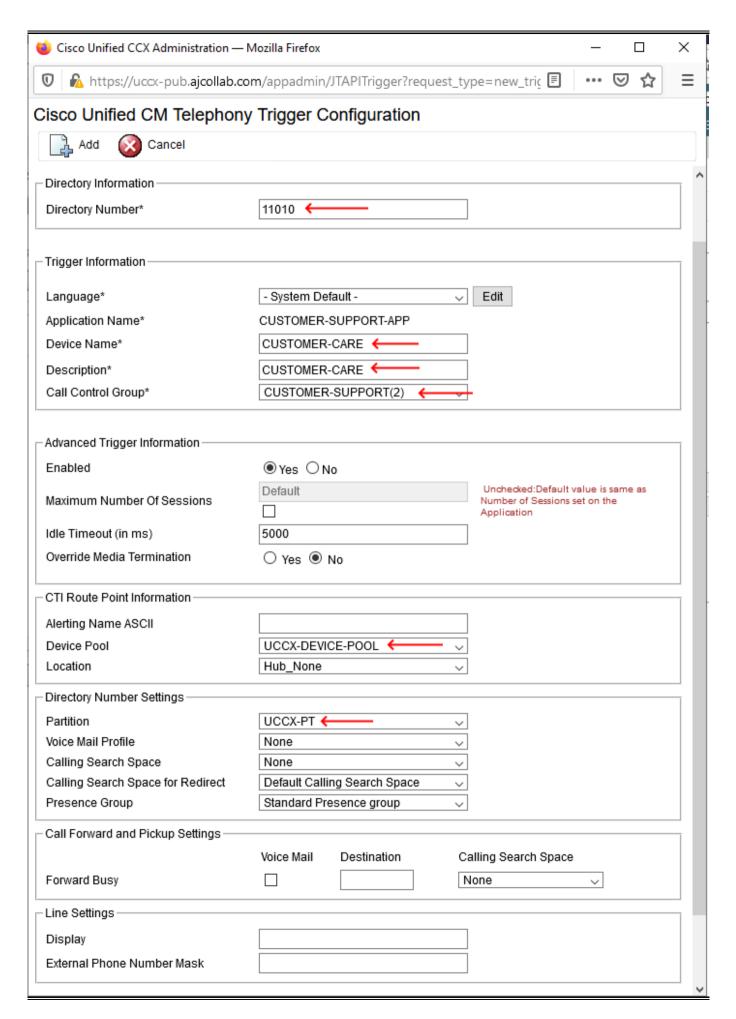


X. Create Application







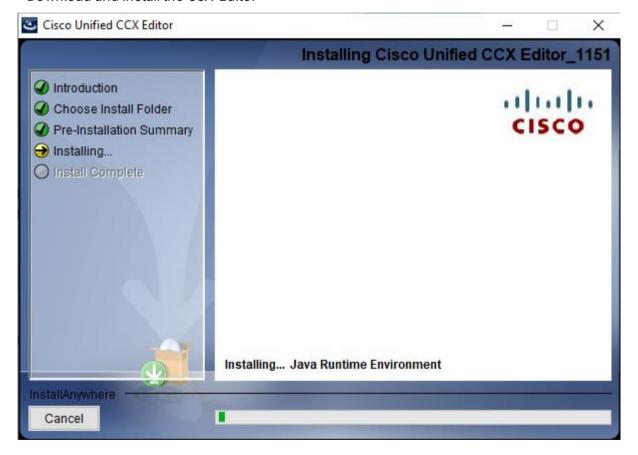


•	At this point when you dial 11010, you will be able to hear the Automated Attendant IVR. You can
	try navigating through the different options

Accessing CCX Script Editor

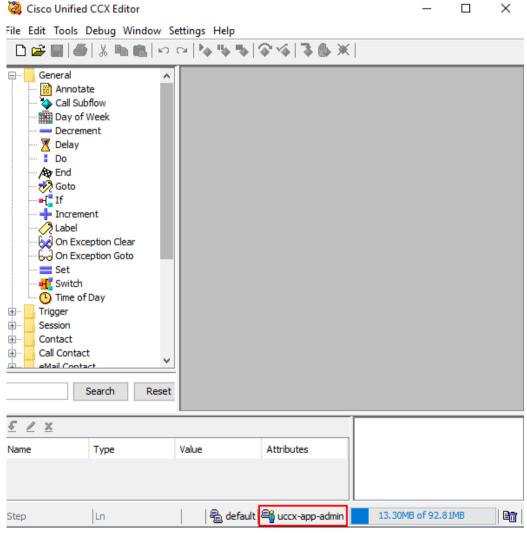


Download and install the CCX Editor

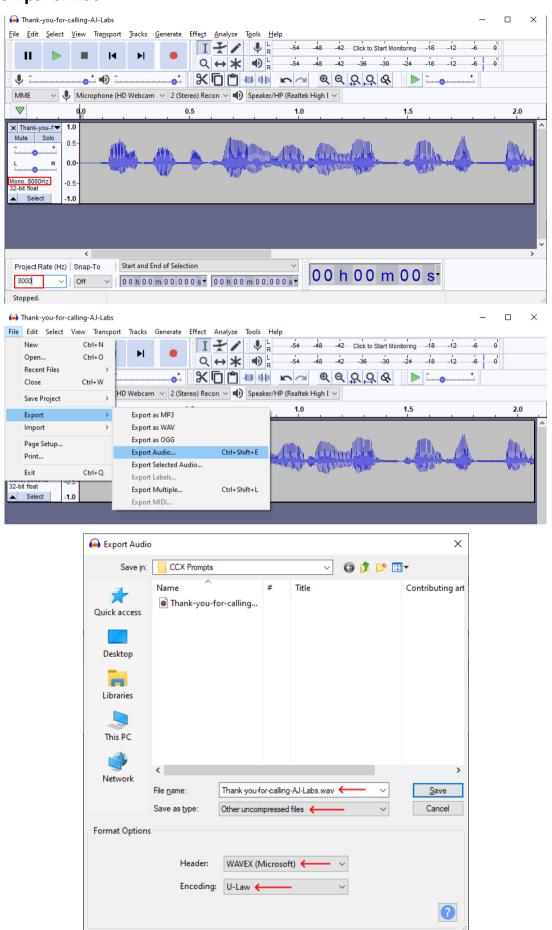


 We use the CUCM Promoted UCCX Administrator Account to login to CCX Editor. You can use the other 'admin' account as well, but some features (Reactive Debugging) won't be available



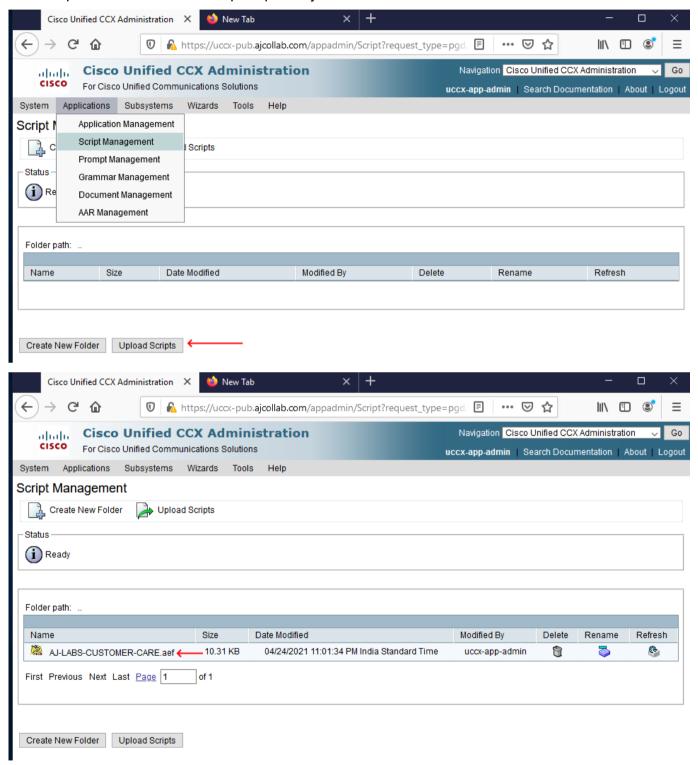


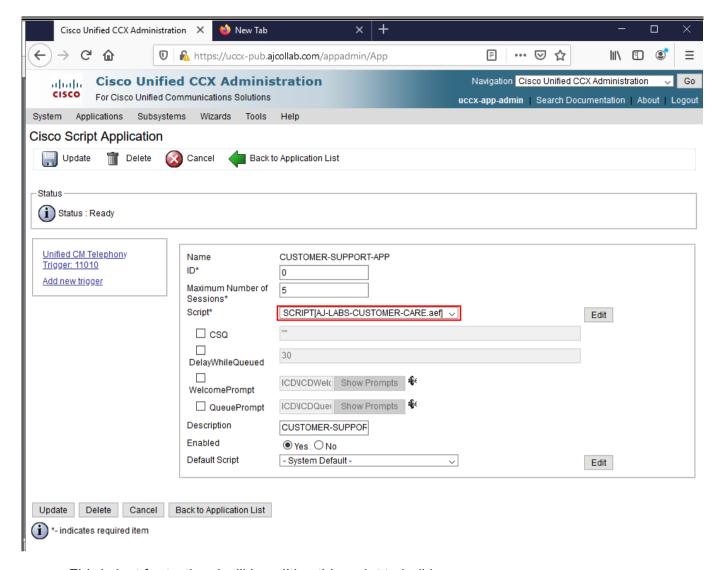
UCCX Prompt Format



[Lab] UCCX Scripting IVR and ACD

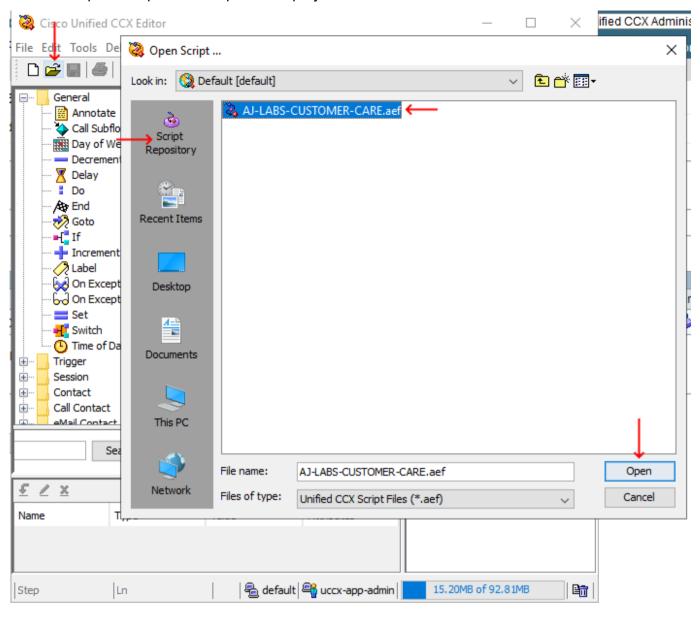
- Some Sample Scripts will be available in 'C:\Program Files (x86)\wfavvid_1151\Scripts\system\default'
- I have copied the icd.aef and renamed to AJ-LABS-CUSTOMER-CARE.aef
- Uploaded this to UCCX Script Repository

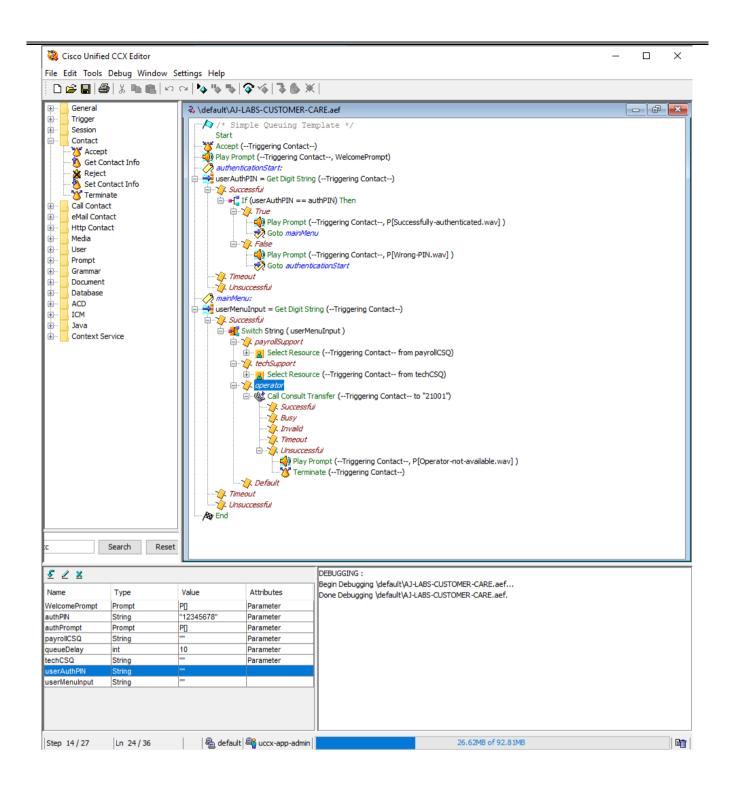




This is just for testing; I will be editing this script to build my own

Complete Script and Prompt for this project can be downloaded from here





Variable	Data Type	Default Value	Customizable over web
WelcomePrompt	Prompt	P[]	Yes
authPIN	String	12345678	Yes
userAuthPIN	String	Dynamic (Accept from caller)	No
authPrompt	Prompt	P[]	Yes
userMenuInput	String	Dynamic (Accept from caller)	No
payrollCSQ	String	un	Admin should provide
techCSQ	String	439	Admin should provide
queueDelay	Int	10	Yes
operator	String	(13)	Admin should provide

784

Accept

Here we accept the call that came to the Trigger

Play Prompt:

- Play the welcome prompt that is pulled from a variable WelcomePrompt "Thank you for calling AJ
 Labs Contact Center, we are happy to help you"
- The value of **WelcomePrompt** can me customized from the UCCX web portal

Get Digit String

- Plays a greeting that is pulled from a variable **authPrompt** "Please enter your PIN followed by pound key to authenticate"
- Accept the digits from user and saves to a variable userAuthPIN

lf

- Check whether userAuthPIN == authPIN that is provided in the UCCX web portal
- True > Play prompt 'Successfully-authenticated.wav' and go to Main Menu
- False > Play prompt 'Wrong-PIN.wav' and go to Get Digit String again

mainMenu

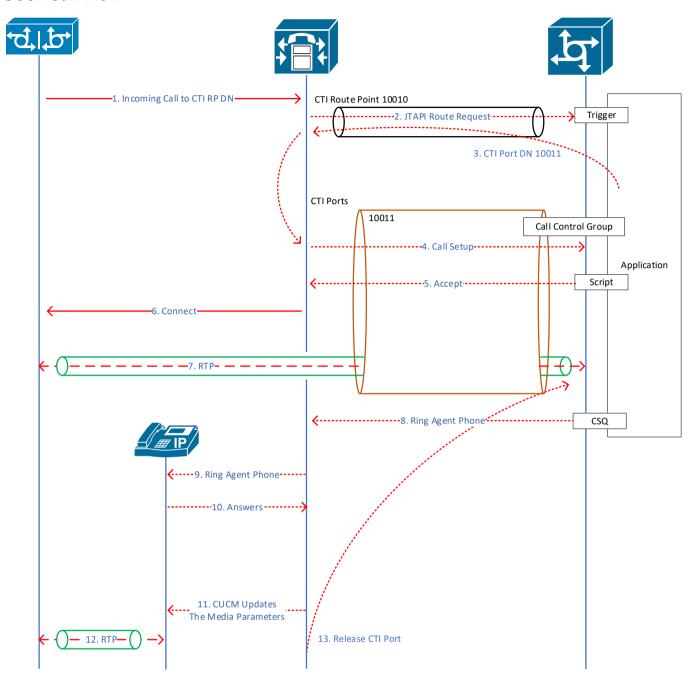
- Plays a prompt "Press 1 for Payroll support, Press 2 for technical support. For any other help, press 9"
- Accept digit from user and saves to a variable userMenuInput

Switch

- If user presses 1, route the call to **payrollCSQ**, if none of the agents are available stay in the queue. Value of **payrollCSQ** can be customized from UCCX web portal
- If user presses 2, route the call to techCSQ, if none of the agents are available stay in the queue.
 Value of techCSQ can be customized from UCCX web portal
- If user presses 3, Transfer the call to an **operator** variable whose value can be set from the UCCX web portal. If operator is not available, play a prompt saying that 'Sorry, the operator is not available now, please call us back later'

Note: Agent must login to https://<UCCX server>:8445/desktop and make the status available to get the calls.

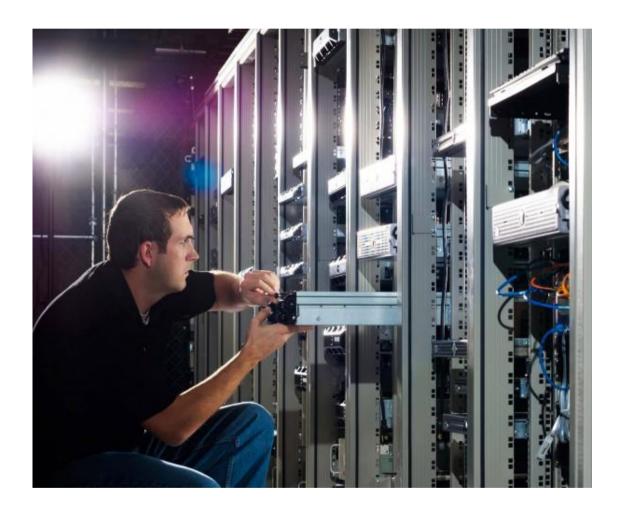
UCCX Call Flow



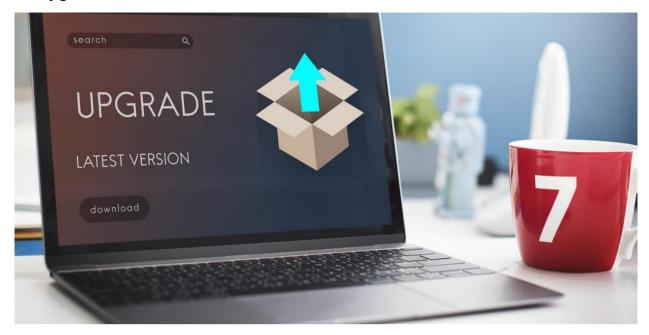
- 1. Call comes from CUBE / Voice gateway to CUCM with CTI Route point DN
- 2. CUCM sends JTAPI Route Request to UCCX and that reaches to UCCX Application. Application interconnects Trigger, Call Control Group, Script and CSQ
- 3. Application identifies free CTI Port DN and sends JTAPI response with CTI Port Number
- 4. CUCM Sends call setup to the CTI Port Number and UCCX will execute the Script
- 5. Script Accepts the call
- 6. CUCM Connects the CUBE leg to UCCX
- 7. RTP flows from CUBE to UCCX

- 8. Based on the caller interaction, UCCX identify a Contact Service Queue (CSQ) to route the call to an agent based on Agent's availability
- 9. Agent phone rings
- 10. Agent answers the call
- 11. CUCM Updates the media parameters to establish media from CUBE to the Phone
- 12. RTP flows from CUBE to Agent's IP Phone
- 13. CUCM releases the CTI Port and that will be available to serve upcoming calls

Chapter 1 Module 7 - Upgrade Cisco On-Premise Collaboration Solution UC Upgrade, Device Pack & Firmware Upgrade



CUCM Upgrade 11.5 to 12.5



- CUCM 11.5 and 12.5 has some major differences. CUCM 11.5 platform is based on RedHat Linux,
 it was RedHat Linux ever since version 5 came out (first Linux version of CUCM)
- CUCM 12.5 is built on CentOS platform
- Other change is about Licensing, 12.5 uses Smart License whereas 11.5 uses Prime License
 Manager
- Perform upgrade from vmware Console since 12.5 upgrade goes via OS new OS installation where you lose the connectivity for a long time
- It is always recommended to go through the Cisco official upgrade document before starting any upgrade

1. Pre-Upgrade Tasks

- 1.1 Take a full DRF cluster backup (we discussed already)
- 1.2 Create Smart Licensing Account
- 1.3 Convert Prime License to Smart License
- 1.4 Pre-Upgrade Check COP File
- 1.5 Free Common Space COP File
- 1.6 Delete Unused Firmware Files
- 1.7 Update Virtual Hardware (CPU, RMA, HDD, NIC)
- 1.8 Take Output of TFTP Content
- 1.9 Take output of show version active

2. Upgrade Tasks

- 2.1 Upgrade UCM Publisher
- 2.2 Upgrade CUCM Subscribers
- 2.3 Upgrade IMP Publisher
- 2.4 Upgrade IMP Subscriber
- 2.5 Verify Database Replication
- 2.6 Switch Version CUCM Publisher
- 2.7 Switch Version CUCM Subscribers, IMP Publisher and IMP Subscribers
- 2.8 Change VM Compatibility and Guest OS Version
- 2.9 Install Post-Upgrade COP File
- 2.10 Register CUCM 12.5 to Smart License Manager

3. Post Upgrade Task

- 3.1 Perform health check
- 3.2. Update TFTP Contents (Optional)
- 3.3 Install COP Files (Optional)

Post Upgrade Tasks

- 3.1 Install the Post-Upgrade Check COP File
- 3.2 Perform Regular Health Check that we covered in the beginning of the course
- 3.3 Update the TFTP Server

The custom file content (Custom Firmware, Ring Tones, Backgrounds, etc.) of TFTP server won't be replicated to new TFTP Node after the upgrade

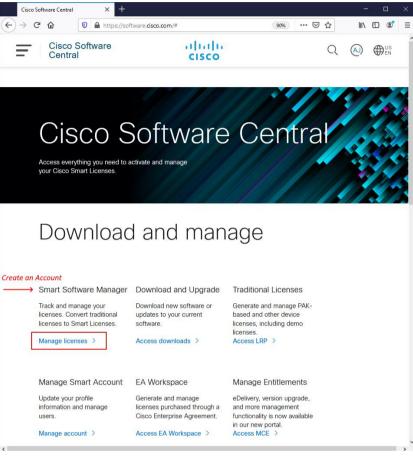
Install required files that was existed before

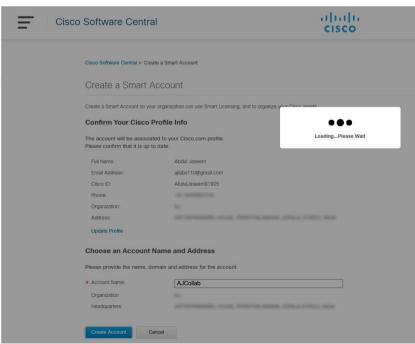
1.1 Take a full DRF cluster backup

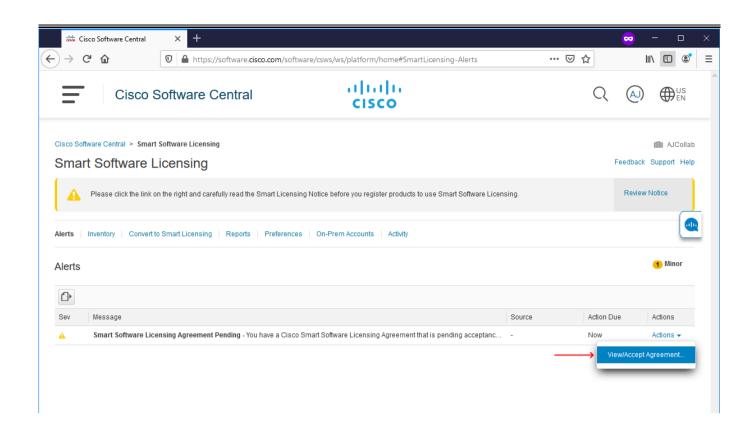
We have already discussed how to take a cluster backup to SFTP server in previous class

1.2 Creating Smart License Account

- Go to Cisco Software Central and create an account
- It will take some time to activate the account

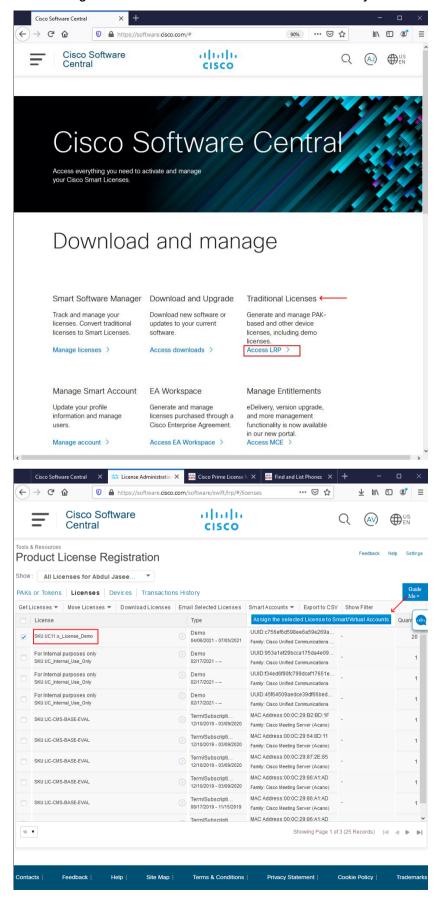


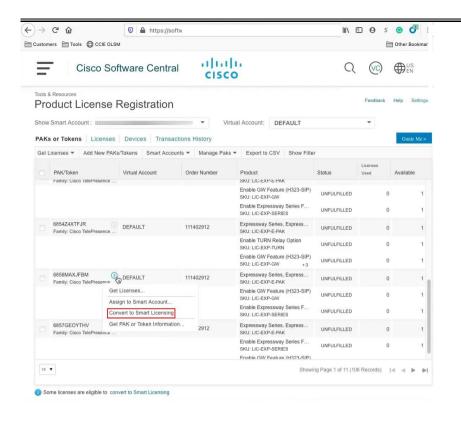




1.3 Convert Traditional License to Smart Licenses

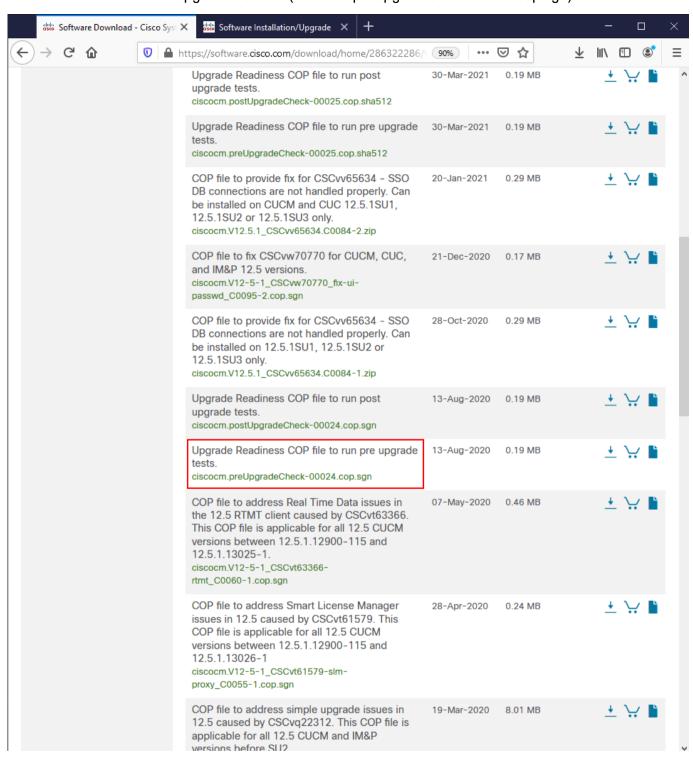
Login to Traditional License Portal and convert your old licenses to Smart Licenses



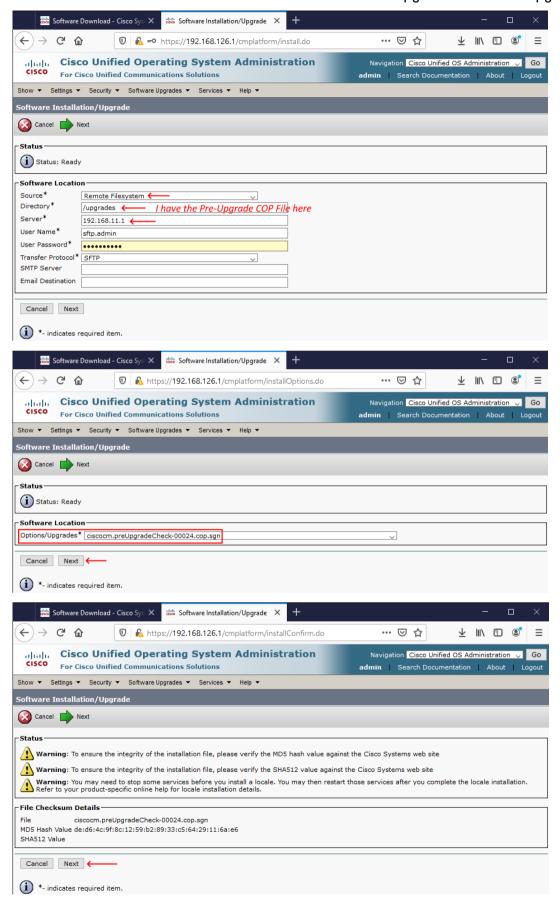


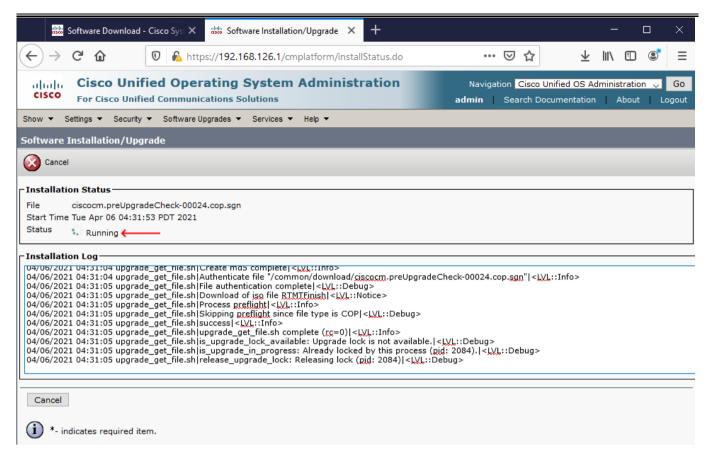
1.4 Pre-Upgrade Check COP File

• Download the Pre-Upgrade COP file (ciscocm.preUpgradeCheck-00024.cop.sgn) from the link



- Copy the file to SFTP Server
- Go to OS Administration of CUCM PUB >> Software Upgrades >> Install/ Upgrade





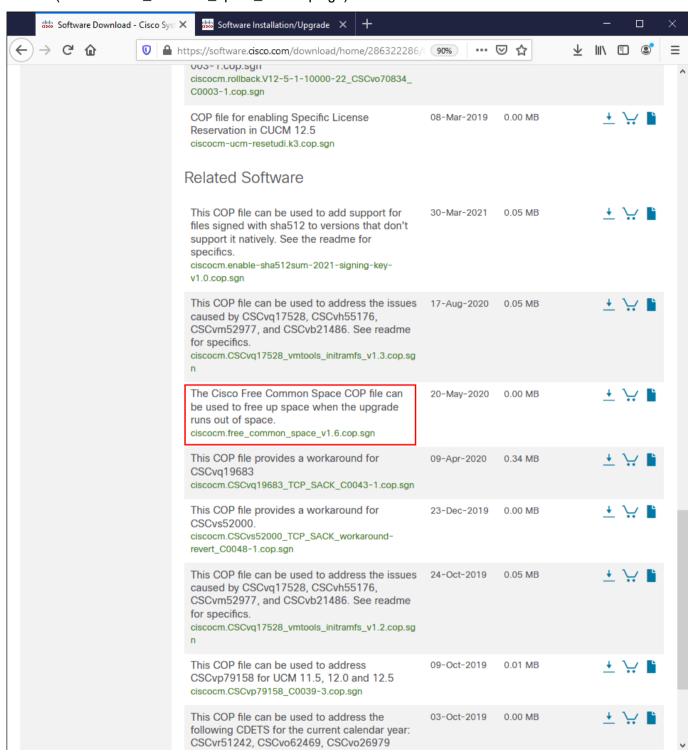
- It is going top take some time. You have to perform this in each node in the cluster
- This will give upgrade reediness summary, make sure you go through the summary and fix the failures and address the warnings

```
Date: 04/06/2021 11:32:00
                                                    Pre Upgrade Test
          Active Version: 11.5.1.13901-3
                             up-cucm-pub , CUCM Publisher
          Server:
          Result Test
1.1 PASS
                        Filesystem Checks
           *) 11.5
          Passed: You have 31 GB of required 22 GB common partition space required for L2 upgrade to 11.5.
          Passed: You have 31 GB of required 22 GB common partition space required for L2 upgrade to 12.0.
          12 5
          Passed: You have 31 GB of required 28 GB common partition space required for RU upgrade to 12.5.
                          PLM License Status
1.3
         PASS
                         Common Security Password Length
          System not in FIPS mode, Common Security Password's Minimum length requirement not enforced
                          Cluster Database Status
          FAIL Network status (NTP, DNS & Cluster node connectivity)
Check network reach ability. Use 'utils diagnose test' for more details
          ntp_reachability : Warning
The NTP service is restarting, it can take about 5 minutes.
         ntp_clock_drift : Warning
The local clock is not synchronised.
None of the designated NTP servers are reachable/functioning or legitimate.
                                                : Warning
         ntp_stratum : warning
The local clock is not synchronised.
None of the designated NTP servers are reachable/functioning or legitimate.
1.6 PASS
                          Deprecated Phone Models
1.7
         PASS
                          Test dataBase Sanity
1.8
         PASS
                          Network Adapter Type
          WARN DRS backup status
WARNING: No backup device is configured. This is required to recover your system in case of failure.
          System Status List
2.1 Cops Installed (PASS)
          No Installed Software Options Found.
                                                  VMTools Type (WARN)
          10.3.21.249
          10.3.21.249 native vm-tools WARNING: Update the native vmtools to latest for optimal system
          performance.
2.3 Upgrade Checks (WARN)
*) Smart Licensing requires you to have a Smart Account created and configured before you upgrade or migrate the Cisco Unified Communications Manager server to 12.0 or higher releases.
         Count
                                                   Phone Status (PASS)
                                                   Registered
                                                   Unregistere d
2.5 Status
                                                   Service Name (PASS)
          No Issues Found
_____
Summary:
    Total Test Run : 14
    Total Passed : 10
Total Warnings : 3
Total Failed : 1
Note: Please refer to the readme of Pre Upgrade cop for test details and
            pass/fail/warn/criteria
Duration for running tests: 0:08:00
Use "file view install PreUpgradeReport.txt" to view the report
04/06/2021:04:40:09 \ / common/download/TestManager/agents/CliAgent.py: 119 - INFO - Quitting CLI session and Destroying the CLI agent a
```

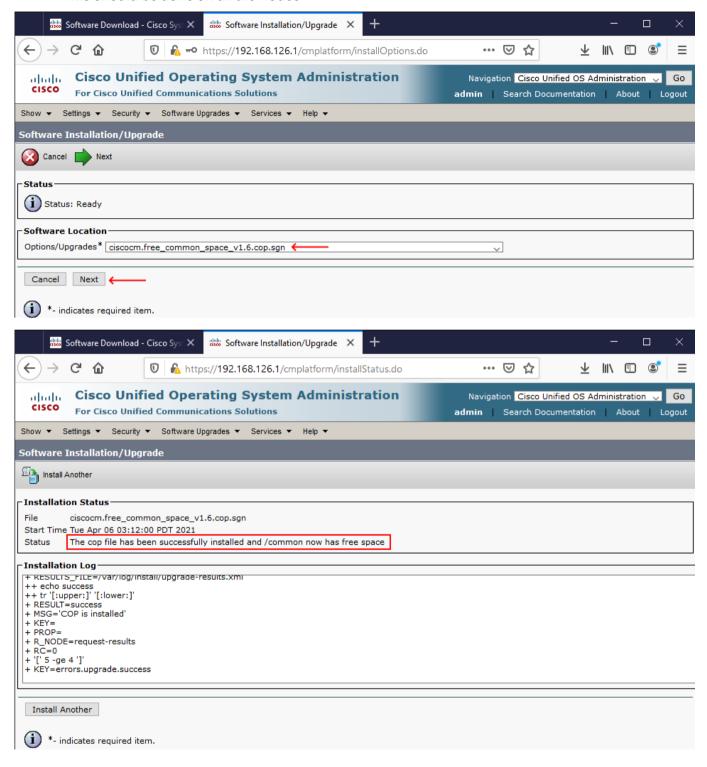
 I have got NTP error and DRF Warning, I have corrected the NTP and ran the pre-upgrade cop file again. There was no errors for the second time

1.5 Free Common Space COP File (Optional)

• If you face issues in the file system check, you can install the Free Common Space COP File (ciscocm.free common space v1.6.cop.sqn) from the link



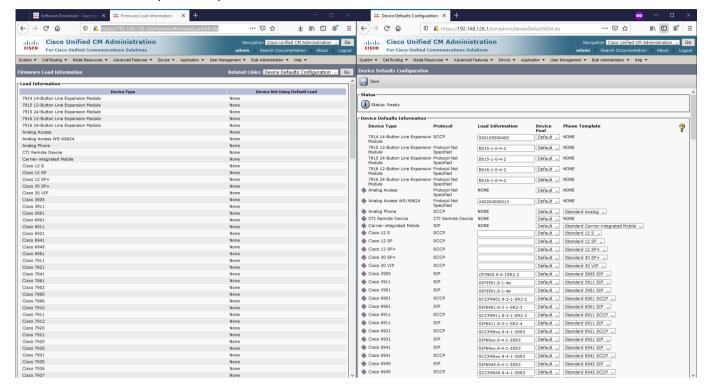
- Upload the file to SFTP Server just like you did before and install it from OS Admin Page
- This should be done on all the nodes



Post the Free Space COP tile, run the Pre-Upgrade Check COP File (Step 1.4) again and make sure
you have enough space available

1.6 Delete Unused Firmware Files (Optional)

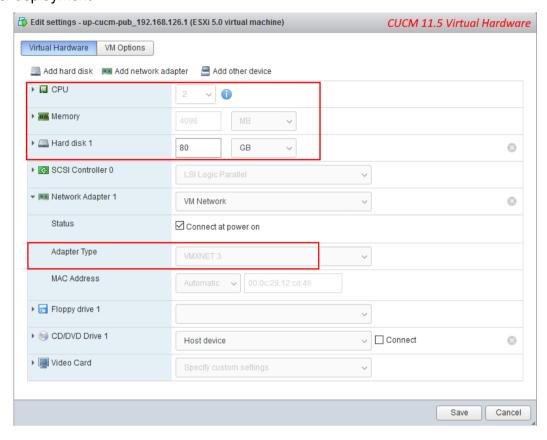
- Go to CM Administration >> Device >> Device Settings >> Firmware Load Information
- Find out the phones those are not using the default load and upgrade those phones to use the one that is available in Device Defaults
- This will free-up some space



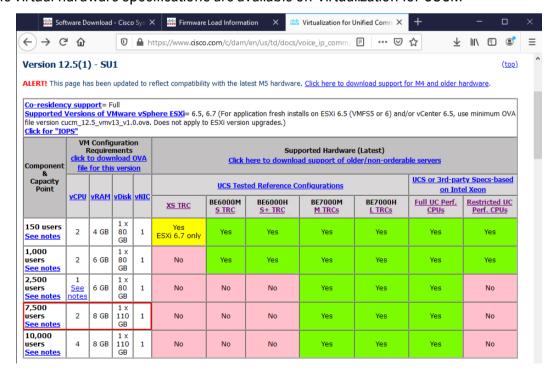
 Post deleting the unused firmware, run the Pre-Upgrade Check COP File (Step 1.4) again and make sure you have enough space available

1.7 Update Virtual Hardware (CPU, RAM, HDD, NIC)

 My current CUCM 11.5 installation has 2 vCPU, 4 GB RAM and 80 GB HDD and this is the 150user deployment



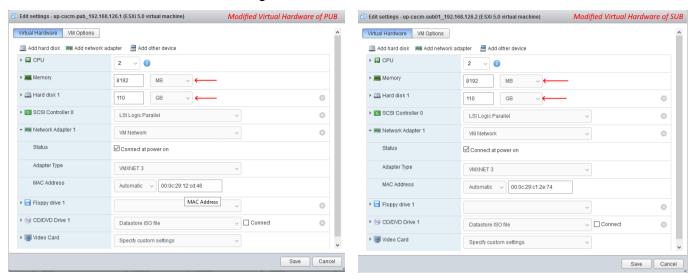
- CUCM 12.5 I'm planning 7500-user deployment and that requires 2 vCPU, 8 GB RAM, 110 GB HDD
- The virtual hardware specifications are available on Virtualization for CUCM



```
admin: show status
                   : up-cucm-pub
: Tue Apr 6, 2021 03:29:34
Host Name
Date
                     Pacific Daylight Time (America/Los_Angeles)
Time Zone
Locale
                   : en US.UTF-8
Product Ver
                     11.5.1.13901-3
Unified OS Version: 6.0.0.0-2
 03:29:35 up 2:31, 1 user, load average: 0.07, 0.03, 0.06
CPU Idle:
            92.93% System:
                              02.02%
                                        User:
                                                05.05%
           00.00%
 IOWAIT:
                      IRQ:
                              00.00%
                                        Soft:
                                                00.00%
Memory Total:
                     3925468K
        Free:
                      326072K
                     3599396K
        Used:
                      618540K
      Cached:
                      364500K
      Shared:
     Buffers:
                       24984K
                        Total
                                         Free
                                                         Used
Disk/active
                    14154228K
                                     1108608K
                                                    12900492K (93%)
Disk/inactive
                    14154228K
                                    13393148K
                                                       35424K (1%)
Disk/logging
                    49573612K
                                    33404808K
                                                    13643876K (29%)
admin: show hardware
HW Platform
                  : VMware Virtual Machine
Processors
                  : Intel(R) Xeon(R) CPU E5-2643 v2 @ 3.50GHz
Type
CPU Speed
                    3500
                    4096 MBytes
Memory
Object ID
                    1.3.6.1.4.1.9.1.1348
                    UCOS 6.0.0.0-2.i386
OS Version
Serial Number
                  : VMware-56 4d b5 09 5c 07 5f 30-45 b2 70 56 e0 12 cd 46
No RAID controller information is available
BIOS Information :
PhoenixTechnologiesLTD 6.00 12/12/2018
RAID Details
No RAID information is available
Physical device information
Number of Disks : 1
Hard Disk #1
Size (in GB) : 80
Partition Details :
Disk /dev/sda: 10443 cylinders, 255 heads, 63 sectors/track
Units = sectors of 512 bytes, counting from 0 \,
   Device Boot
                  Start
                              End
                                    #sectors Id
                                                  System
                   2048 29028351
/dev/sda1
                                    29026304 83
                                                  Linux
/dev/sda2
               29028352 58054655
                                    29026304
                                              83
                                                  Linux
/dev/sda3
               58054656 58578943
                                      524288 83
                                                  Linux
/dev/sda4
               58578944 167772159 109193216
                                               5
                                                  Extended
/dev/sda5
               58580992 66772991
                                     8192000
                                                  Linux swap / Solaris
/dev/sda6
               66775040 167772159 100997120 83
                                                  Linux
```

804

Here I would need a virtual hardware upgrade, for that shutdown the VM (utils system shutdown)
 from CLI and edit the VM Settings from vCenter or ESXi



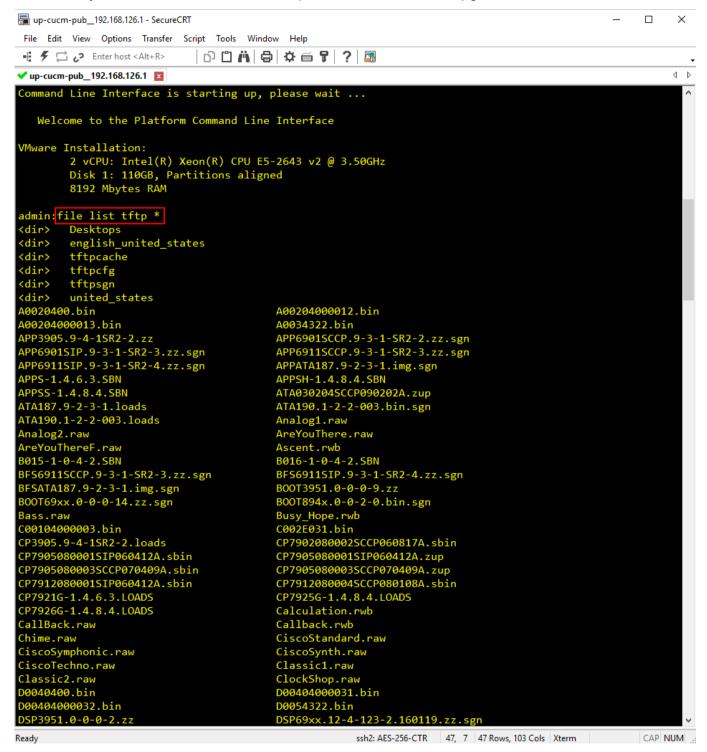
- Now power on the VM, the Hard Disk will automatically expand to 110 GB
- Verify the new Virtual Hardware has been updated after powering on using below commands

```
admin: show status
                     up-cucm-pub
Tue Apr 6, 2021 03:46:38
Host Name
Date
                     Pacific Daylight Time (America/Los_Angeles)
Time Zone
Locale
                   : en US.UTF-8
Product Ver
                     11.5.1.13901-3
Unified OS Version: 6.0.0.0-2
 03:46:39 up 2 min, 1 user, load average: 1.01, 0.68, 0.28
CPU Idle:
            46.73% System:
                              26.13%
                                        User:
                                                 27.14%
 IOWAIT:
            00.00%
                       IRQ:
                              00.00%
                                        Soft:
                                                00.00%
Memory Total:
                     8062404K
                     6902008K
        Free:
        Used:
                     1160396K
                      542632K
      Cached:
                      172984K
      Shared:
     Buffers:
                       94692K
                        Total
                                         Free
                                                          Used
Disk/active
                    14154228K
                                     1157112K
                                                     12851988K
                                                               (92%)
Disk/inactive
                    14154228K
                                    13393148K
                                                        35424K
Disk/logging
                    80537376K
                                    62841832K
                                                     13602896K (18%)
admin: show hardware
HW Platform
                    VMware Virtual Machine
Processors
                    Intel(R) Xeon(R) CPU E5-2643 v2 @ 3.50GHz
Tvpe
CPU Speed
                    3500
Memory
                    8192 MBytes
Object ID
                    1.3.6.1.4.1.9.1.1348
OS Version
                  : UCOS 6.0.0.0-2.i386
Serial Number
                  : VMware-56 4d b5 09 5c 07 5f 30-45 b2 70 56 e0 12 cd 46
RAID Version
No RAID controller information is available
BIOS Information :
PhoenixTechnologiesLTD 6.00 12/12/2018
RAID Details
No RAID information is available
Physical device information
Number of Disks : 1
Hard Disk #1
Size (in GB)
Partition Details :
Disk /dev/sda: 14359 cylinders, 255 heads, 63 sectors/track
Units = sectors of 512 bytes, counting from 0 \,
   Device Boot
                  Start
                              End
                                    #sectors Id
                                                  System
/dev/sda1
                   2048
                         29028351
                                    29026304
                                              83
                                                  Linux
/dev/sda2
               29028352
                         58054655
                                     29026304
                                                  Linux
/dev/sda3
               58054656
                         58578943
                                      524288
                                              83
                                                  Linux
/dev/sda4
               58578944 230686719
                                   172107776
                                                  Extended
/dev/sda5
               58580992 66772991
                                     8192000
                                              82
                                                  Linux swap / Solaris
/dev/sda6
               66775040 230686719 163911680 83 Linux
//
```

• I could see that the virtual hardware is updated and match with 7500-user node of CUCM 12.5

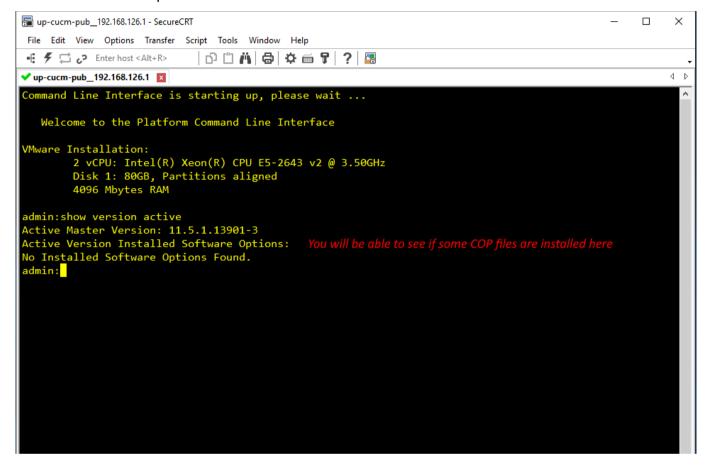
1.8 Take the output of TFTP Contents

- This is applicable to TFTP Servers only
- It is just for a reference; we can compare the result after the upgrade if needed



1.9 Take the output of show version active

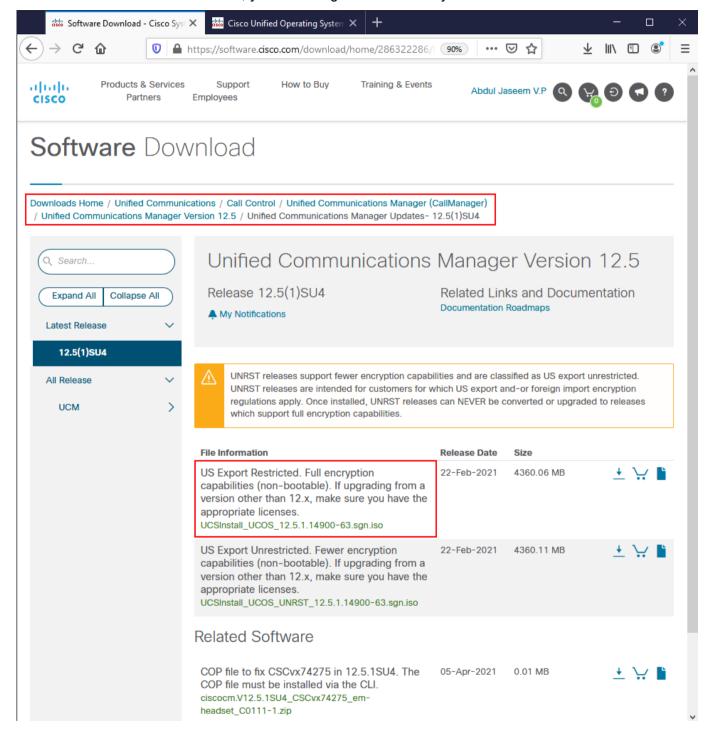
- Any COP files, or device packages, etc. will be listed in show version active
- This becomes handy if you want to install those files post the upgrade
- Upgrade will never install the custom COP files
- Collect the output from all the nodes in the cluster



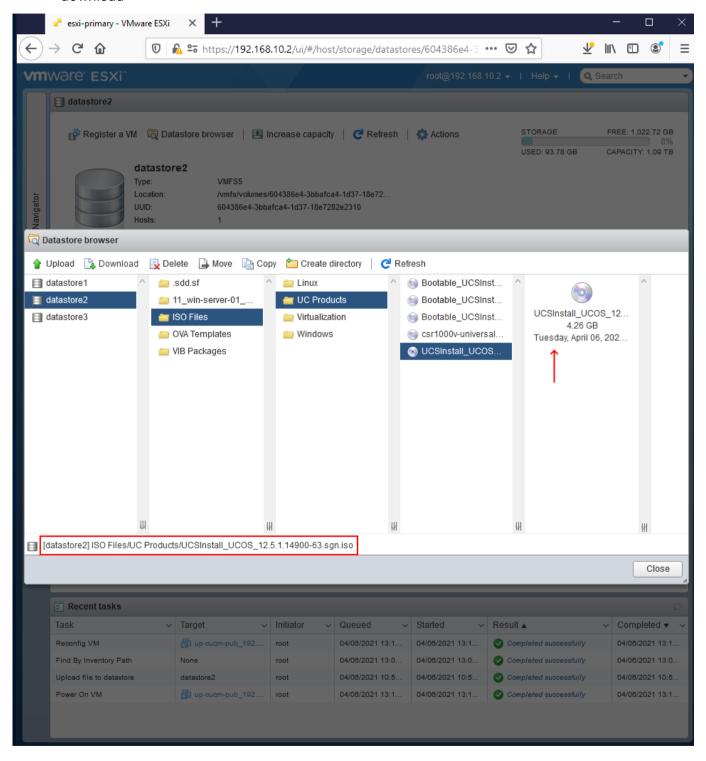
- In my case, I don't have any COP files here, but you do see that in production cluster
- Take note of this output for future reference

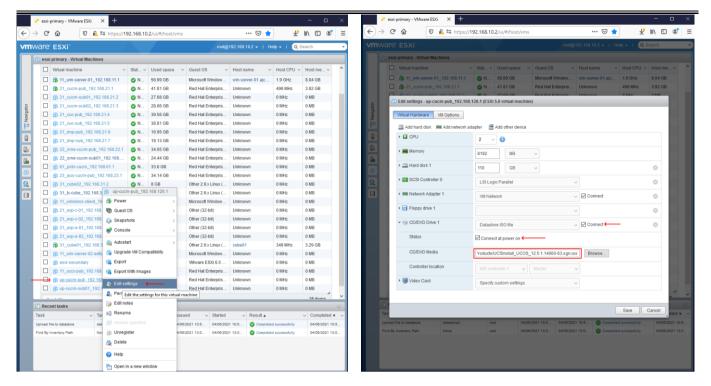
2.1 Upgrade CUCM Publisher

- Before starting anything, go ahead and reboot the entire cluster (utils system restart). This is not documented anywhere but it is my recommendation
- Download the CUCM 12.5 Upgrade file from the link
- This file will be non-bootable, we can use this only to upgrade not for fresh install
- For bootable CUCM ISO File, you need to get in touch with your Accounts Team or TAC



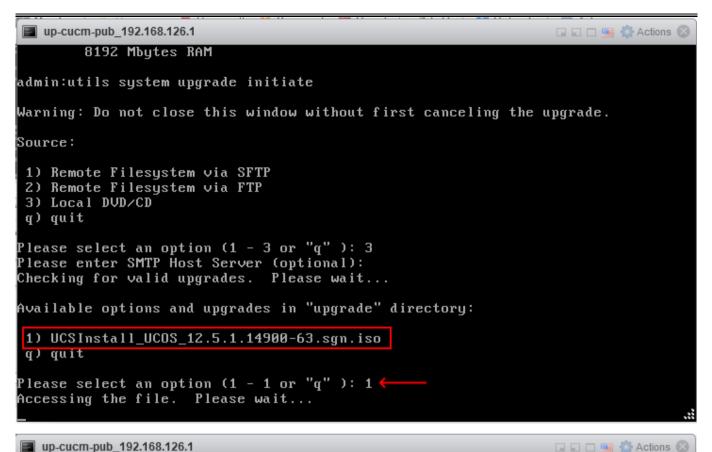
- Upload the non-bootable ISO File to the ESXi Data Store and mound to the CD/DVD of the CUCM
 11.5 PUB VM
- Note: We can use bootable ISO as well for the upgrade, but bootable ISO is not directly available to download

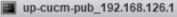




- Instead of connecting to CD/DVD to CUCM 12.5 ISO, you can place it in an SFTP server and use
 Install/Upgrade option in OS Administration page (like the way you installed COP files)
- You can use CLI to perform upgrade, but I recommend using vmware Console for upgrade task since you lose network connectivity for a long time while upgrading
- Enter utils system upgrade initiate on the vmware Console CLI

```
up-cucm-pub_192.168.126.1
                                                                🗔 🔲 🧰 🥋 Actions 🚷
ast login: Tue Apr 6 13:21:13 on tty1
Command Line Interface is starting up, please wait ...
  Welcome to the Platform Command Line Interface
JMware Installation:
        2 ∨CPU: Intel(R) Xeon(R) CPU E5-2643 ∨2 0 3.50GHz
        Disk 1: 110GB, Partitions aligned
       8192 Mbytes RAM
admin:utils system upgrade initiate
Warning: Do not close this window without first canceling the upgrade.
Source:
1) Remote Filesystem via SFTP
2) Remote Filesystem via FTP
3) Local DVD/CD
q) quit
Please select an option (1 - 3 or "q" ): 3🗲
Please enter SMTP Host Server (optional):
Checking for valid upgrades. Please wait...
```





This is a Refresh Upgrade. Refresh Upgrades require an extended service outage a nd multiple reboots. Please refer to the Software Upgrades section of the Cisco Unified Communication Operating System Administration Guide for more information

If there exists any weak ciphers (like 1DES,null_encryption, blowfish448, rijnda el, md5) in IPSEC policies then they will be converted, 1DES as encryption ciph er will be converted to AES128 , MD5 as hash will be converted to SHA256 and nul l_encryption,blowfish448, rijndael as ESP to AES128.

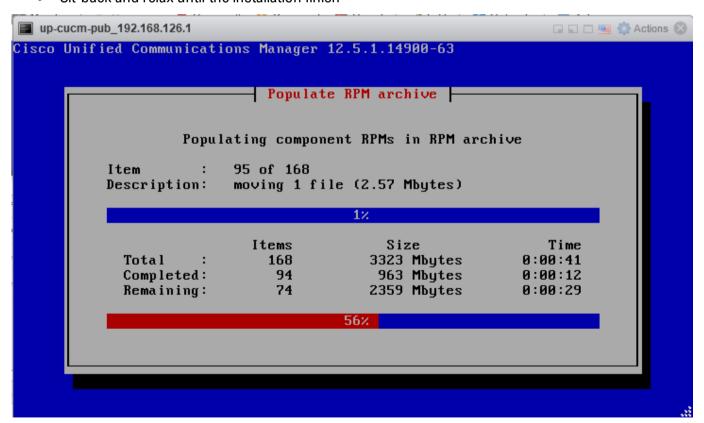
In order to use Certificate-based authentication with IPsec both sides of the co nnection must use certificates signed by the same root CA in the trust chain. Se lf-signed IPsec certificates are no longer supported and IPSec connections using self-signed certificates will fail.

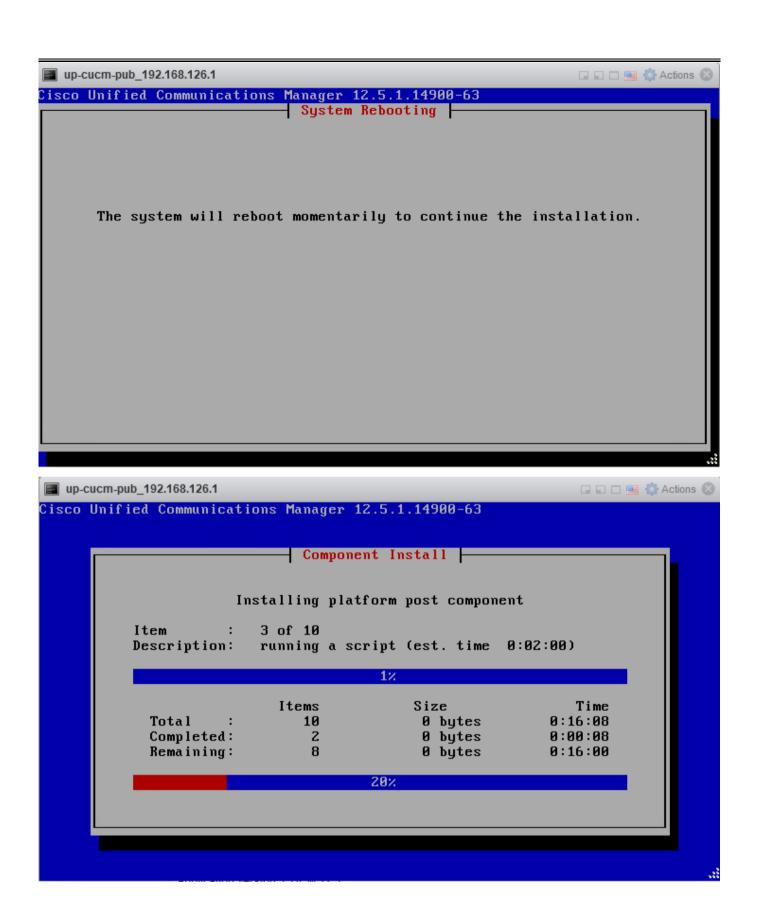
Switch to new version if the upgrade is successful (yes/no): no ←

Start Refresh Upgrade (yes/no): yes 🗲 The upgrade log is install log 2021-04-06.13.31.32.log Upgrading the system. Please wait...

```
up-cucm-pub_192.168.126.1
04/06/2021 13:37:22 component_install!Got version 12.5.1.14900-63|<LVL::Debug>
04/06/2021 13:37:22 component_install¦Initialize ucplatform_cluster "to" side ve
rsion complete¦<LVL::Info>
04/06/2021 13:37:22 component install¦Initialize ucplatform cluster "from" side
version¦<LVL::Info>
04/06/2021 13:37:22 component_install¦Component ucplatform_cluster available on
active side¦<LVL::Debug>
04/06/2021 13:37:22 component_install|Access "from" side API|<LVL::Debug>
04/06/2021 13:37:22 component_install|Got version 11.5.1.13901-3|<LVL::Debug>
04/06/2021 13:37:22 component_install¦Initialize ucplatform_cluster "from" side
version completel<LVL::Info>
04/06/2021 13:37:22 component install¦Initialize global data complete¦<LVL::Info
04/06/2021 13:37:22 component_install¦Build the command list for ucplatform_clus
ter¦<LUL::Info>
04/06/2021 13:37:22 component_install¦Initialize command list¦<LVL::Debug>
04/06/2021 13:37:22 component_install¦Build ucplatform_cluster export phase comm
ands:<LVL::Debug>
04/06/2021 13:37:22 component_install:Processing ucplatform_cluster export eleme
nt:<LVL::Debug>
04/06/2021 13:37:22 component_install¦Processing ucplatform_cluster script eleme
nt:<LVL::Debug>
04/06/2021 13:37:22 component_install¦calculating timeout from script¦<LUL::Deb
սց>
```

- Upgrade has started, you will see most of the time the screen like above and some other times the Blue-Red windows just like you seen in the new installation
- Sit back and relax until the installation finish.

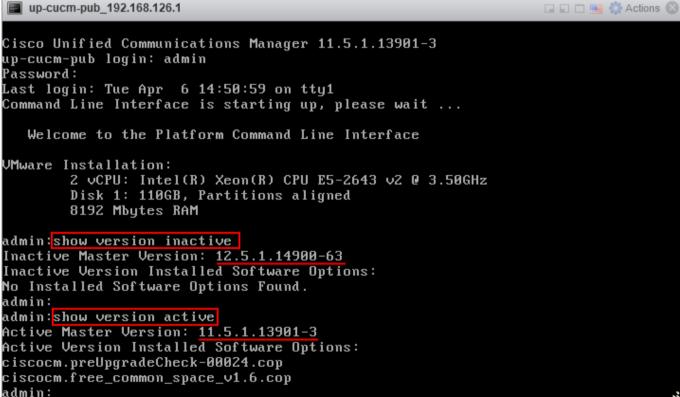




```
■ up-cucm-pub_192.168.126.1

Cisco Unified Communications Manager 12.5.1.14988-63

up-cucm-pub login: _
```



- The active version is still 11.5 because 12.5 installed at the inactive partition, we perform switch version after some task and that will make the active version as 12.5
- It is not recommended to perform any switch version of CUCM PUB until all the nodes are upgraded (Even if you do switch version, there won't be a problem)

2.2 Upgrade CUCM Subscribers

 Perform the same steps on CUCM Subscriber servers. You can upgrade all subscribers at one time

```
up-cucm-sub01 192.168.126.2
                                                                      🗔 🔲 🧰 🧌 Actions 🔕
        2 ∨CPU: Intel(R) Xeon(R) CPU E5-2643 ∨2 @ 3.50GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbutes RAM
admin:utils system upgrade initiate
Warning: Do not close this window without first canceling the upgrade.
Source:
 1) Remote Filesystem via SFTP
2) Remote Filesystem via FTP
3) Local DVD/CD
q) quit
Please select an option (1 - 3 or "q" ): 3 🕻
Please enter SMTP Host Server (optional):
Checking for valid upgrades. Please wait...
Available options and upgrades in "upgrade" directory:
 1) UCSInstall_UCOS_12.5.1.14900-63.sgn.iso
q) quit
Please select an option (1 - 1 or "q" ): 1\leftarrow
```

• Since the steps are similar, I'm not going to repeat it here

```
up-cucm-sub01_192.168.126.2
                                                                 🗔 🔲 🥅 🧰 Actions 🔕
Cisco Unified Communications Manager 11.5.1.13901-3
up-cucm-sub01 login: admin
Password:
Last login: Tue Apr 6 15:11:16 on tty1
Command Line Interface is starting up, please wait ...
   Welcome to the Platform Command Line Interface
UMware Installation:
        2 ∨CPU: Intel(R) Xeon(R) CPU E5-2643 ∨2 @ 3.50GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
admin:show version inactive
Inactive Master Version: 12.5.1.14900-63
Inactive Version Installed Software Options:
No Installed Software Options Found.
admin:
admin:show version active
Active Master Version: 11.5.1.13901-3
Active Version Installed Software Options:
No Installed Software Options Found.
admin:_
```

Now the subscriber also having 11.5 in the active partition and 12.5 in the inactive partition

2.3 Upgrade IMP Publisher

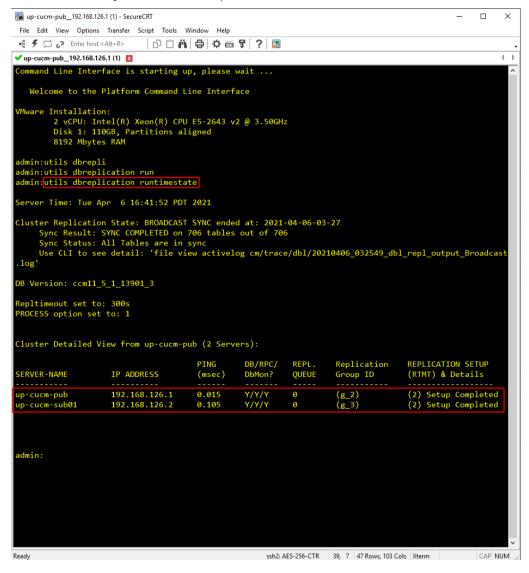
- Use the CUP Upgrade file to upgrade IM and Presence Server, steps are exactly the same
- You must upgrade IMP Pub first and then IMP SUB
- IMP Pub Upgrade can be parallelly done during CUCM Subscriber upgrades
- Do not perform Switch version

2.4 Upgrade IMP Subscriber

- Perform the similar steps to upgrade IMP Subscriber
- Do not perform switch version

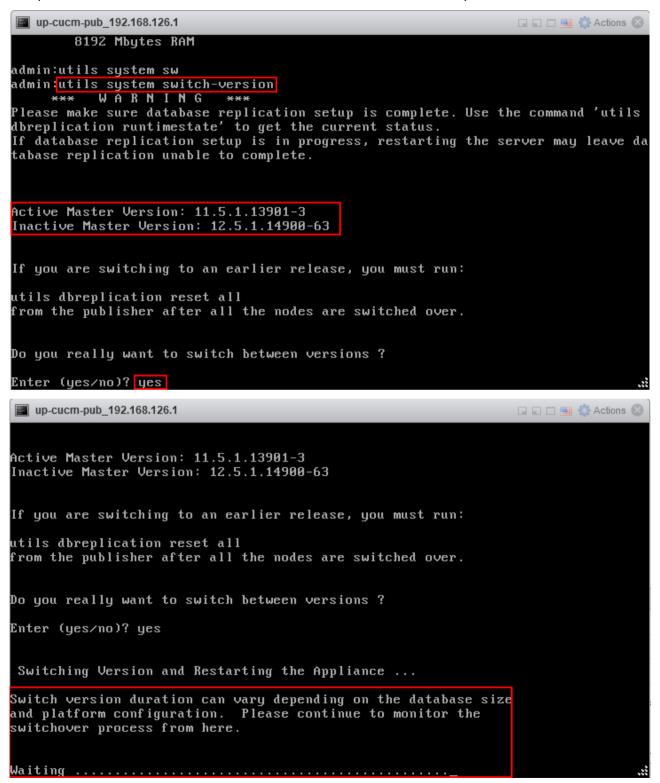
2.5 Verify Database Replication

- Make sure the database replication is complete on all the nodes
- There shouldn't be any issues in DB replication



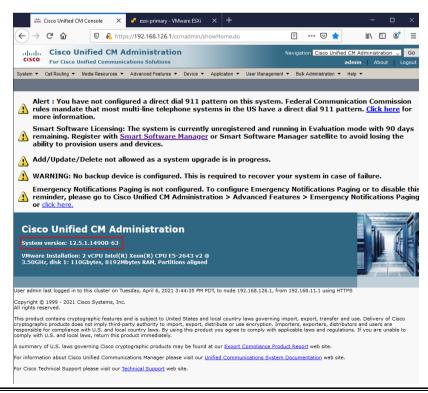
2.6 Switch Version CUCM Publisher

- After all Subscribers are upgraded, switch CUCM PUB 11.5 to 12.5 using utils system switchversion command
- You can do this from SSH session, since we are doing everything from the vmware Console, I
 prefer the same for this as well. Wait for all the services to come up in CUCM Publisher



```
🗔 🔲 🧰 🚳 Actions 🔕
 up-cucm-pub_192.168.126.1
Cisco Unified Communications Manager 12.5.1.14900-63
up-cucm-pub login: admin
Password:
Last login: Wed Apr 7 00:34:13 from win-server-01.ajcollab.com
Command Line Interface is starting up, please wait ...
  Welcome to the Platform Command Line Interface
UMware Installation:
        2 vCPU: Intel(R) Xeon(R) CPU E5-2643 v2 @ 3.50GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
admin:show version active
Active Master Version: 12.5.1.14900-63
Active Version Installed Software Options:
No Installed Software Options Found.
admin:
admin:show version inactive
Inactive Master Version: 11.5.1.13901-3
Inactive Version Installed Software Options:
ciscocm.preUpgradeCheck-00024.cop
ciscocm.free_common_space_v1.6.cop
admin:
```

- Now the active version is 12.5. You still have the version 11.5 on the inactive partition. For some reason if we want to go back to 11.5, do one more switch version
- Switch version will flip the files between active and inactive partitions
- You should wait for some time to get all the services are up, Cisco Tomcat Service will take quite some time to come up. Make sure you can to the Web GUI and login before proceeding to the next step



2.7 Switch Version CUCM Subscribers, IMP Publisher and IMP Subscribers

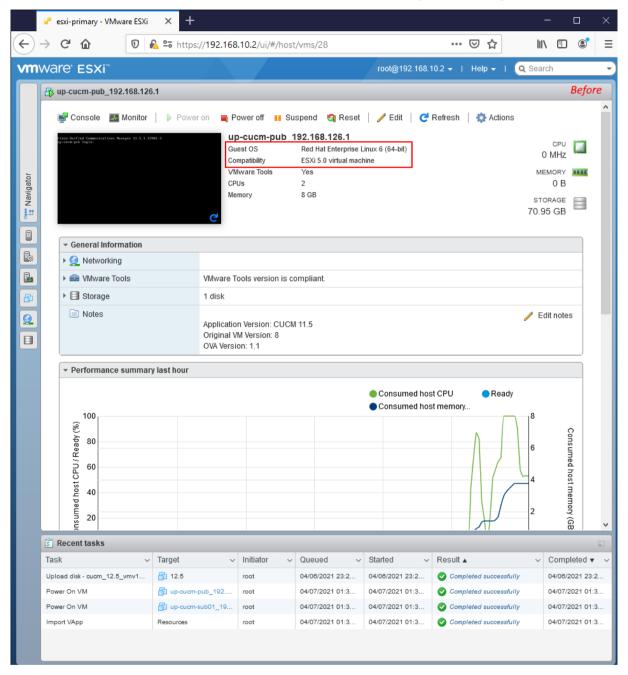
 Once the CUCM Publisher is completely up and running, perform the switch version of CUCM Subscribers. IMP Publisher and IMP Subscriber

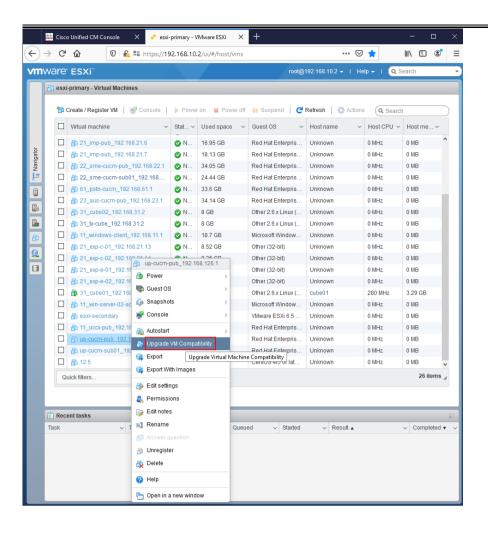
```
p-cucm-sub01_192.168.126.2
                                                                     🗔 🔲 🧰 🥋 Actions 🔕
        8192 Mbytes RAM
admin:utils system sw
admin:utils system switch-version
           WARNING
     ***
Please make sure database replication setup is complete. Use the command 'utils dbreplication runtimestate' to get the current status.
If database replication setup is in progress, restarting the server may leave da
tabase replication unable to complete.
Active Master Version: 11.5.1.13901-3
Inactive Master Version: 12.5.1.14900-63
If you are switching to an earlier release, you must run:
utils dbreplication reset all
from the publisher after all the nodes are switched over.
Do you really want to switch between versions ?
Enter (ues/no)? ues
■ up-cucm-sub01 192.168.126.2
The installation of Cisco Unified Communications Manager has completed successfu
lly.
Cisco Unified Communications Manager 12.5.1.14900-63
up-cucm-sub01 login: admin
Password:
Command Line Interface is starting up, please wait ...
   Welcome to the Platform Command Line Interface
UMware Installation:
        2 vCPU: Intel(R) Xeon(R) CPU E5-2643 v2 @ 3.50GHz Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
admin:show version active
Active Master Version: 12.5.1.14900-63
Active Version Installed Software Options:
No Installed Software Options Found.
admin:
admin:show version inactive
Inactive Master Version: 11.5.1.13901-3
Inacti∨e Version Installed Software Options:
No Installed Software Options Found.
admin:
```

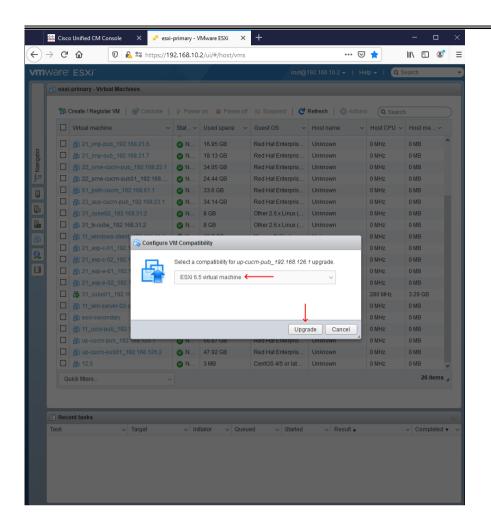
- You can do it all at once for CUCM Subscribers along with IMP Publisher. IMP Subscribers can be switched after the IMP Publisher is up
- I do not have IMP as part of the cluster hence not adding the screenshot

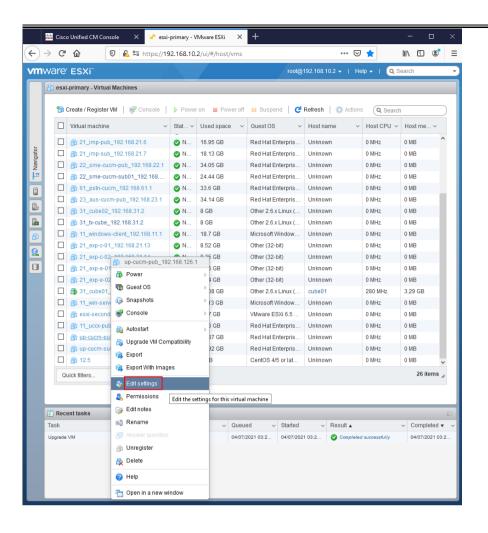
2.8 Change VM Compatibility and Guest OS Version

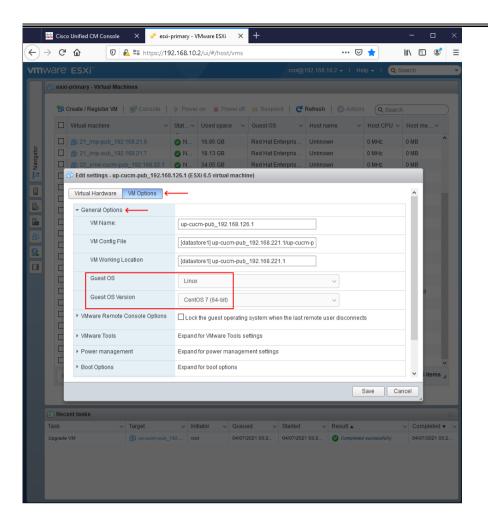
 After the complete switch version of the cluster, power off each node and change VM Compatibility to ESXi 6.5 virtual machine and Guest OS is CentOS 4/5 one by one starting from CUCM Publisher

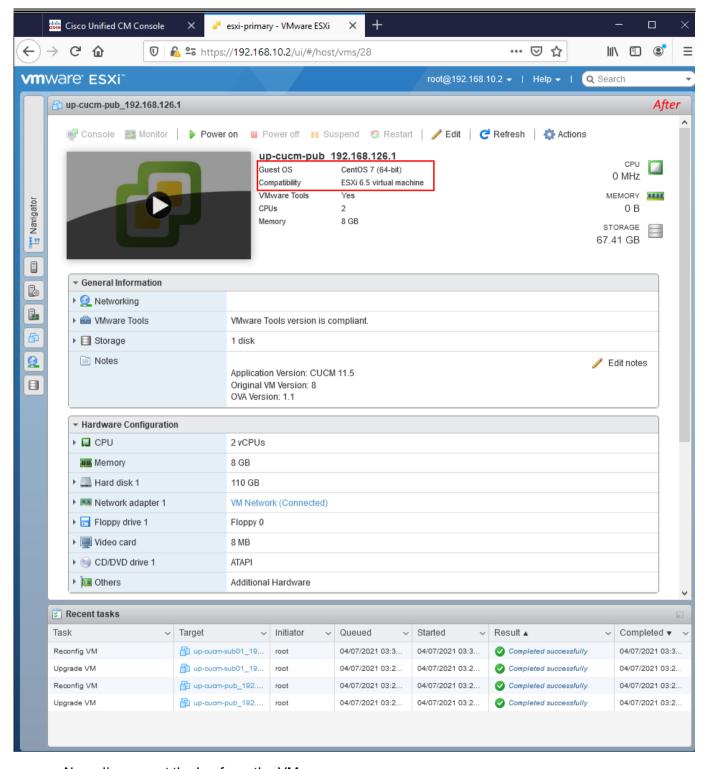






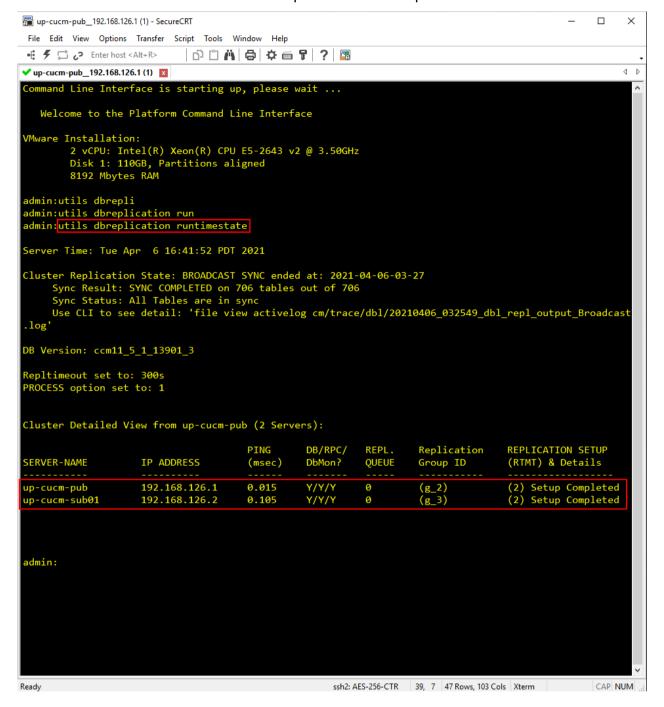






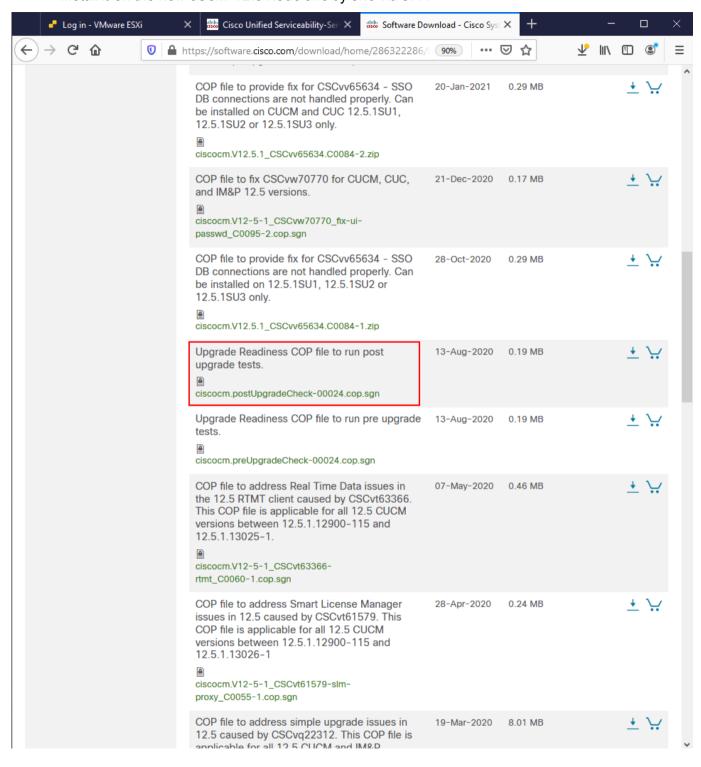
Also, disconnect the Iso from the VM

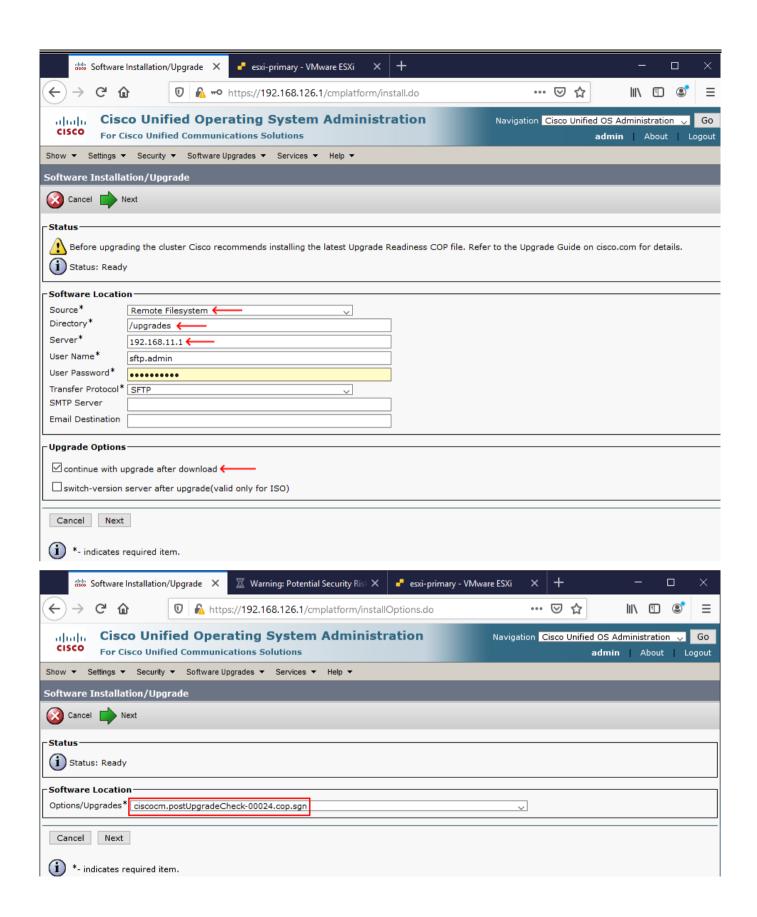
Power on all the VM and wait for DB Replication to come up



2.9 Install the Post-Upgrade Check COP File

- Download the Post-Upgrade reediness COP file from the link
- Install it on the new CUCM 12.5 Node one by one via SFTP





Post Upgrade Test Date: 04/08/2021 10:15:49 ______ Active Version: 12.5.1.14900-63 Inactive Version: 11.5.1.13901-3 Server: up-cucm-pub.ajcollab.com , CUCM Publisher Pre Upgrade Date: 04/06/2021 10:01:00 Result Test WARN SLM License Status License Status is UNREGISTERED.Register the system with Cisco Smart Software Manager or satellite. <mark>Evaluation Period Remaining: 89 days</mark>, 18 hr, 0 min, 0 sec. 1.2 PASS Cluster Database Status PASS Cluster Database Status
FAIL Network status (NTP, DNS & Cluster node connectivity) Check network reach ability. Use 'utils diagnose test' for more : Warning detailsntp_reachability The NTP service is restarting, it can take about 5 minutes. ntp_clock_drift : Warning The local clock is not synchronised. None of the designated NTP servers are reachable/functioning or legitimate. : Warning ntp stratum The local clock is not synchronised. None of the designated NTP servers are reachable/functioning or legitimate. 1.4 PASS Test dataBase Sanity 1.5 PASS Network Adapter Type ______ System Status List 2.1 Cops Installed (PASS) ciscocm.postUpgradeCheck-00024.cop 2.2 Version VMTools Type (WARN) 10.3.21.249 native vm-tools WARNING: Update the native vmtools to latest for optimal system performance. Tests Comparison Active Inactive _____ Enterprise Parameter (FAIL) 3.1 Value Value DeviceTokenExpiryTimer 60 Unknown NULL Unknown HloginAccessNumber None Unknown FeaturePreviewEnablement PhoneMigrationUserIdentificationPrompt Unknown ${\tt PhoneTemplateSelection}$ Unknown 1 Unknown ForceIVRtobeNonSecure NULL ${\tt TrustedServers}$ Unknown 5 headsetBasedEMAutoLogoutTimer Unknown 1 Unknown EnableAutoRecoveryForICPeerPeriodicSyncingFa... 1 Unknown TerminateUserSession Unknown CCMUserShowPhonesReadyToActivate Т 12 CDRPurgeWindow 21999 22001 ServicePort EnableAutoRegistrationforFXSports Unknown Unknown AuthenticatedPhoneRecording 5131abbb-... 122e40ba-... softkeyManager Unknown makeCallProxyAPI c5172a6e-... d48ba279-... softkeyAssistant $Real \verb|TimeMonitoring| Tool \verb|Display| Preference$ Unknown

EndpointEncryptionAlgorithms

EnableUserSearchWithCustomer

SecurityProfileforMigratedPhone

EnableSettingOfICPeerPeriodicSyncing

a

30

1

1

Unknown

Unknown

Unknown

Unknown

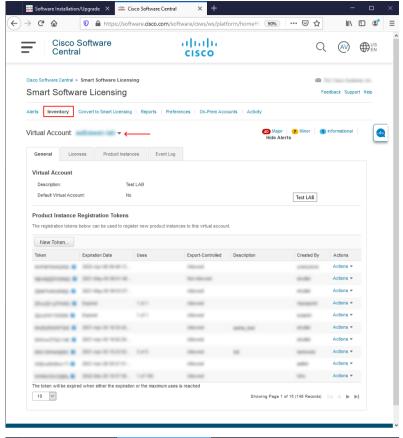
```
G7221andG7222CodecEnabled
     1
                   Unknown
     0
                   Unknown
                                autoLoginForHeadsetBasedEM
     a
                  Unknown
                                ProvisioningReplacementPhoneforEndUser
                                HeadsetAssociation
     0
                   Unknown
     30
                   Unknown
                                headsetBasedAutoLoginTimer
                                headsetBasedEM
                  Unknown
     0000-00-0...
                  Unknown
                                UserCustomerMapAuditTime
     b27df280-...
                  9092b14d-...
                                softkeyManagerShared
     8003
                  8002
                                SdlListeningPort
                  Unknown
                                 EnableDirectoryPartitionSearch
                                DisplayExternalPresentationNameandNumber
                  Unknown
     30
                   Unknown
                                 QuietClearReleaseStopTimer
     1
                                DisableCertSyncForPeriodicSync
                  Unknown
                                FIPSModeExchangeServerAuthentication
     a
                  Unknown
     NULL
                  Unknown
                                HlogoutAccessNumber
                   Unknown
                                MOHPortAssignmentBasedOnAudioSource
                                ActivationCodeExpiry
     168
                  Unknown
     0
                  Unknown
                                ClusterSIPOAuthMode
3.2 Value
                  Value
                                CTI EndPoint Registration Status (PASS)
     No Issues Found
3.3 Value
                  Value
                                Service Parameter (PASS)
     No Issues Found in: MaximumServingThreadCount/Cisco Tftp
3.4 Value
                                Trunk Status (PASS)
                  Value
     No Issues Found
3.5 Count
                  Count
                                Phone Status (PASS)
     0
                  0
                                Unregistered
     a
                  0
                                Registered
3.6 Status
                  Status
                                Service Name (FAIL)
     STARTED
                  STOPPED
                                Cisco CallManager
     STARTED
                                Cisco Device Activation Service
                  Unknown
     STARTED
                  STOPPED
                                Cisco Dialed Number Analyzer
     STARTED
                  Unknown
                                 *Cisco Smart License Manager
     STARTED
                                 *Platform Communication Web Service
                  Unknown
     STARTED
                   STOPPED
                                Cisco Dialed Number Analyzer Server
     STARTED
                  STOPPED
                                Cisco Tftp
     (* Network Services)
______
Summary:
  Total Test Run : 13
  Total Passed
  Total Warnings: 2
  Total Failed
Note: Please refer to the readme of Post Upgrade cop for test details and
     pass/fail/warn/criteria
     Phones and services take time finish setting up.
     Rerunning the COP will give latest status.
     Some values may be truncated due to column width size. Please refer the
     \verb|pre_upgrade_readiness_cmds.log|| post_upgrade_readiness_cmds.log||
      for exact values.
     Use "file view install pre_upgrade_readiness_cmds.log/
     post_upgrade_readiness_cmds.log" to view the command output with
      exact values.
Duration for running tests: 0:07:26
//
```

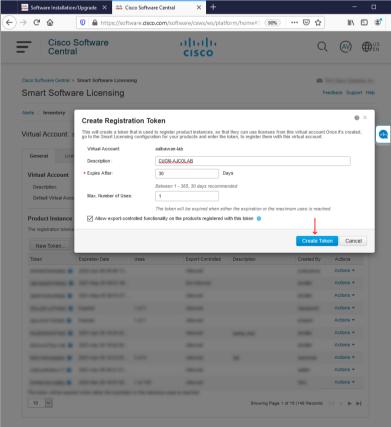
- I have fixed all the Failed section
- The failure 3.1 Enterprise Parameter is expected, looks like a cosmetic bug, you can ignore this

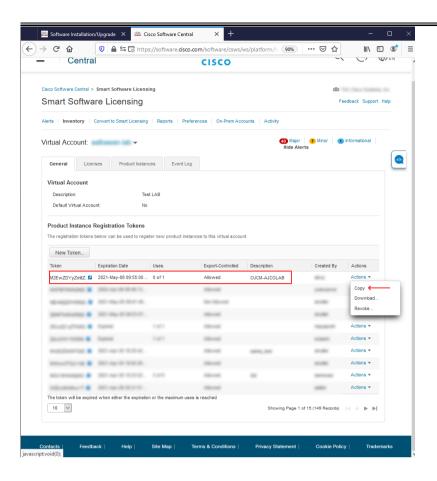
832

2.10 Register CUCM 12.5 to Smart License Manager

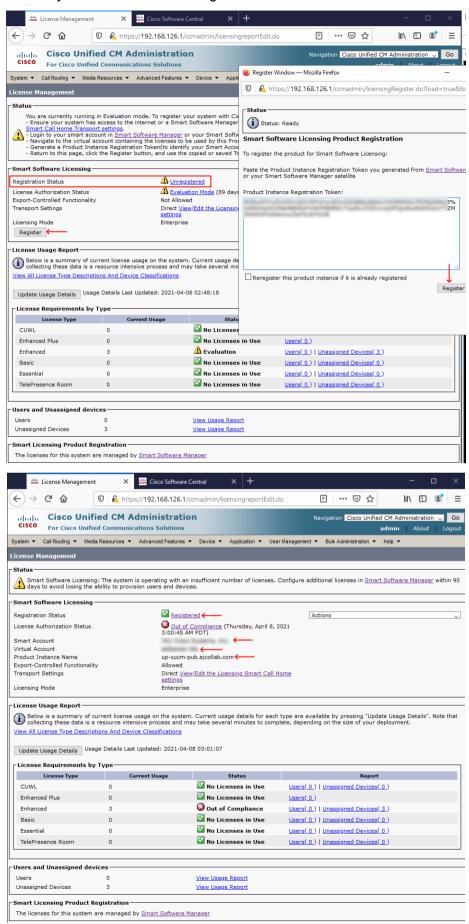
- Login to Smart License Account and create Token
- You CUCM must have internet connectivity to use Smart Licensing







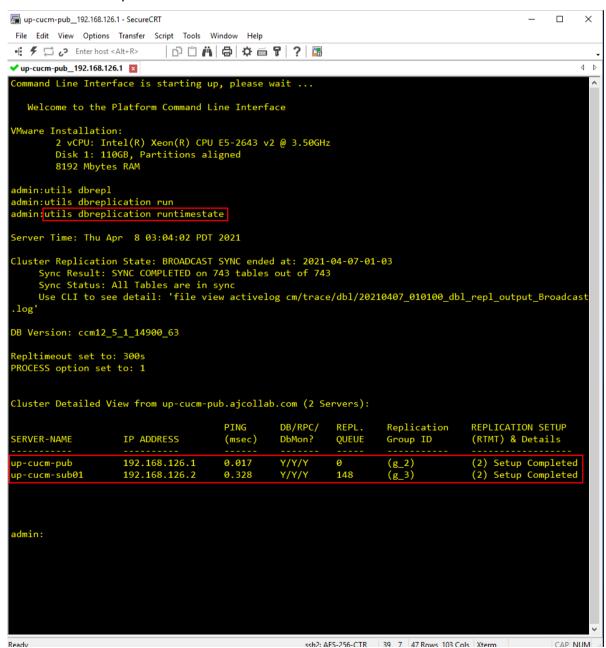
• Go to CUCM >> System >> License Management



- If you are not able provide internet access to CUCM, you can install Stellate server as a proxy for Smart License. CUCM talks to Satellite Server internally and Satellite Server talks to internet
- If the above options are not available, License Reservation can be used (customer like Govt.
 Security, etc.). They can generate a string from the Smart License account and enter it in CUCM to get the license reserved. If there is any change, we have to regenerate new string and do update CUCM

3.1 Perform Health Check

- We covered UC Health check in the beginning of the course
- Make sure DB replication is fine



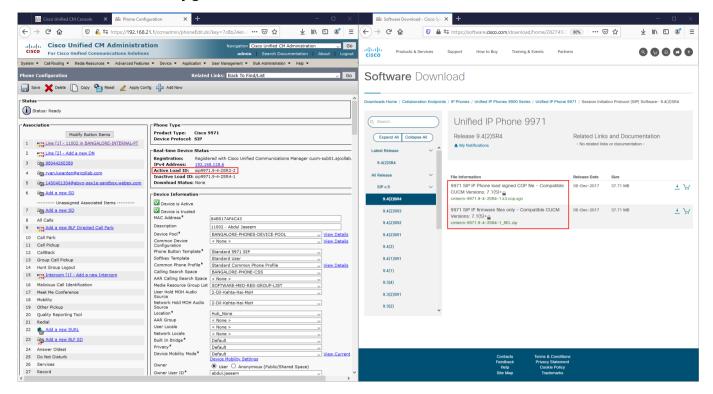
3.2 Update the TFTP Server

- The custom file content (Custom Firmware, Ring Tones, Backgrounds, etc.) of TFTP server won't be replicated to new TFTP Node after the upgrade
- Upload required files that was existed before in the cluster
- If you have two TFTP Server, you have to manually upload files to both nodes, DB Replication will not replicate Custom TFTP Files
- We know the previous TFTP contents from Step 1.8

3.3 Install other COP Files if needed

- If you find some COP files are needed post the upgrade, you can go ahead and install it
- We know the previous COP file details from Step 1.9

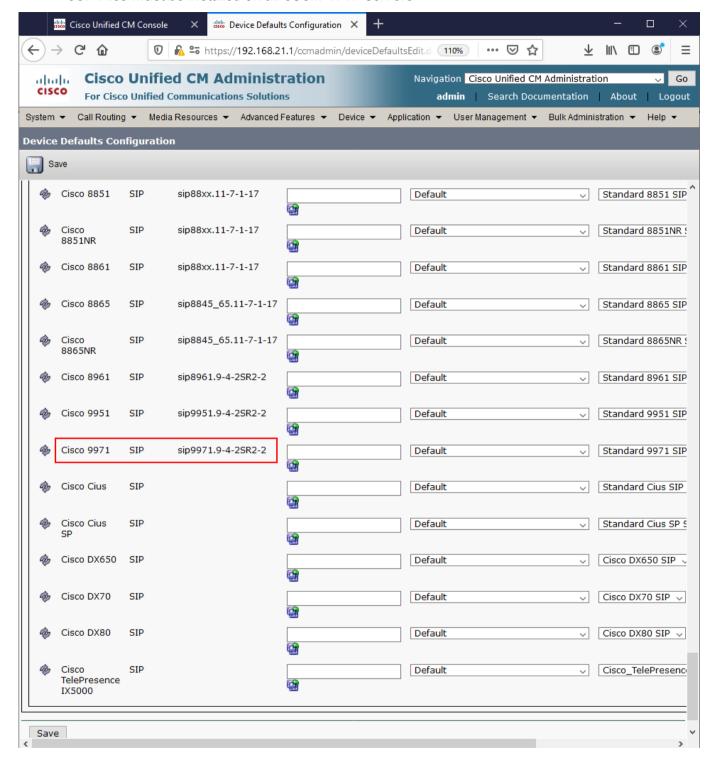
IP Phone Firmware Upgrade



- On Cisco Downloads you will see 2 types of firmware files. '9971 SIP IP Phone load signed COP file' and '9971 SIP IP firmware files only'
- The first file is the standard firmware COP (Cisco Options Package) file that uses SFTP method of installation
- Second file is just the firmware files and uses manual upload method to CUCM TFTP Server

COP File Based Firmware Upgrade

- Please note the Device Default Firmware from Device >> Device Settings >> Device Defaults
- COP Files must be installed on all CUCM TFTP Servers



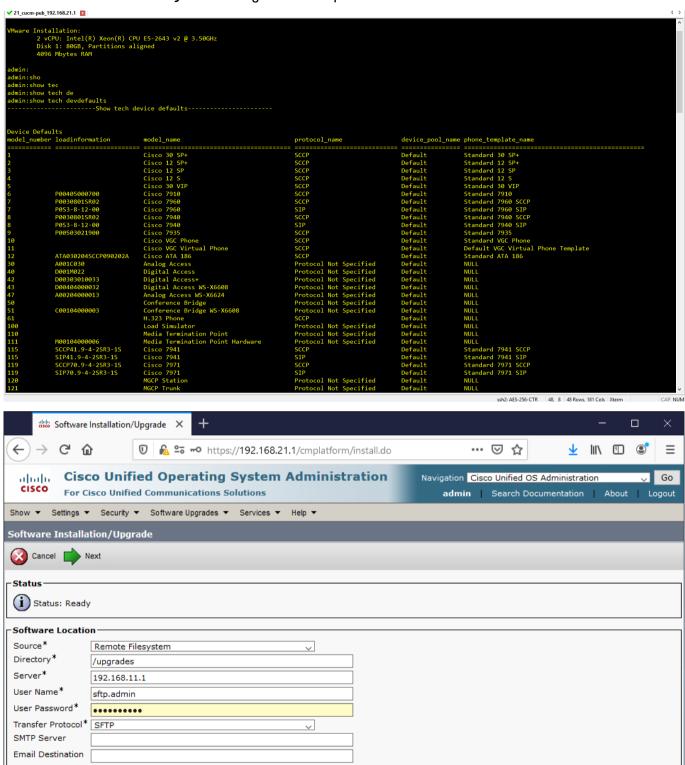
- COP Files or Device Pack installation on CUCM Publisher will override the default firmware, hence when the phone reboots, all the Phones will go for upgrade
- Here our CUCM PUB and TFTP are the same server, hence recommended to return the device default to old firmware after installing COP file on CUCM Publisher

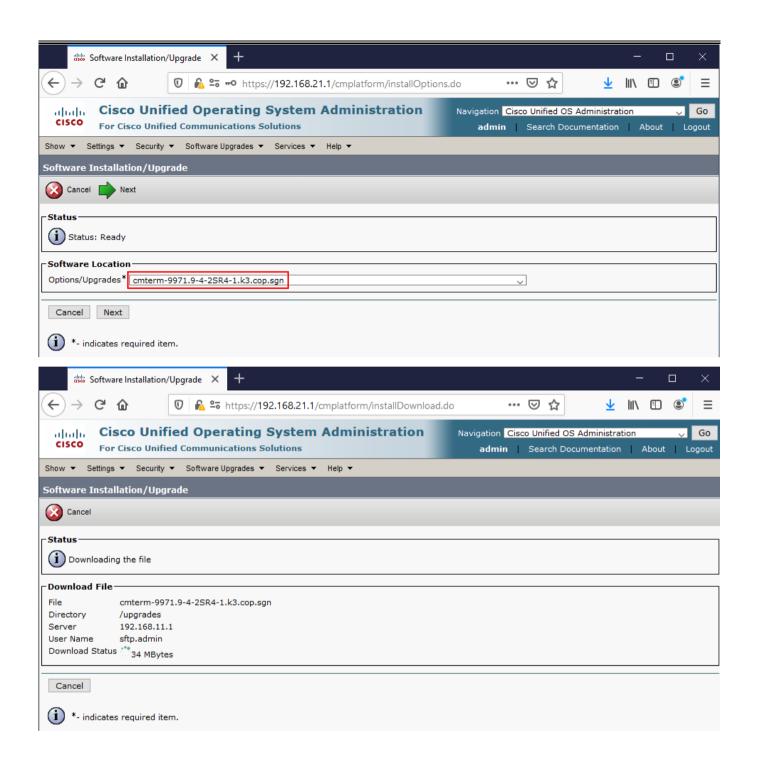
· Only change the device default if it is necessary

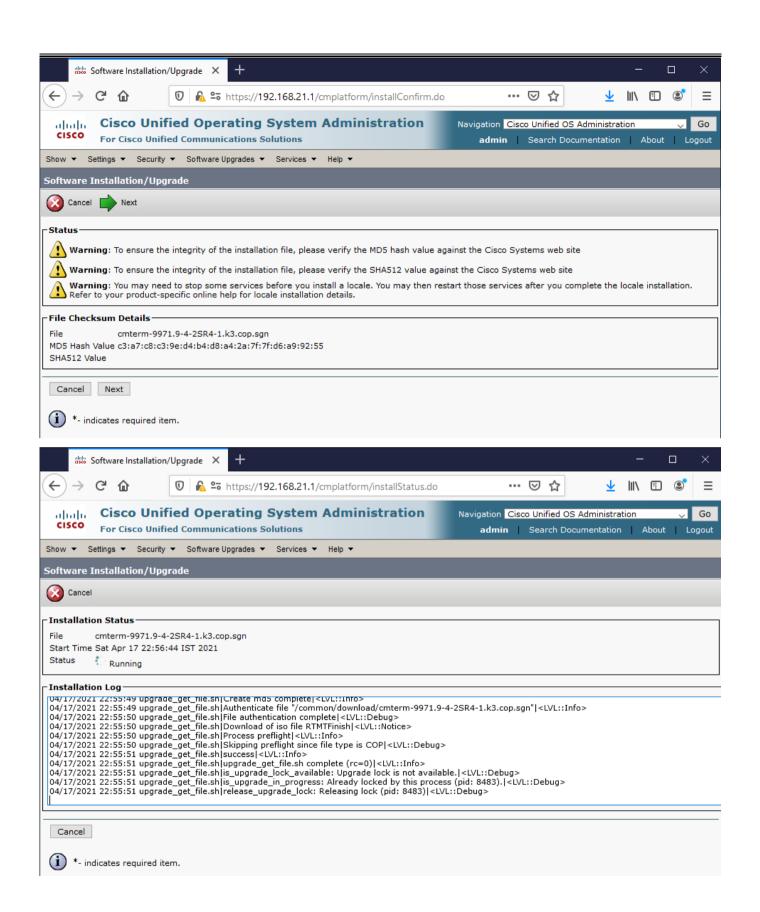
Cancel Next

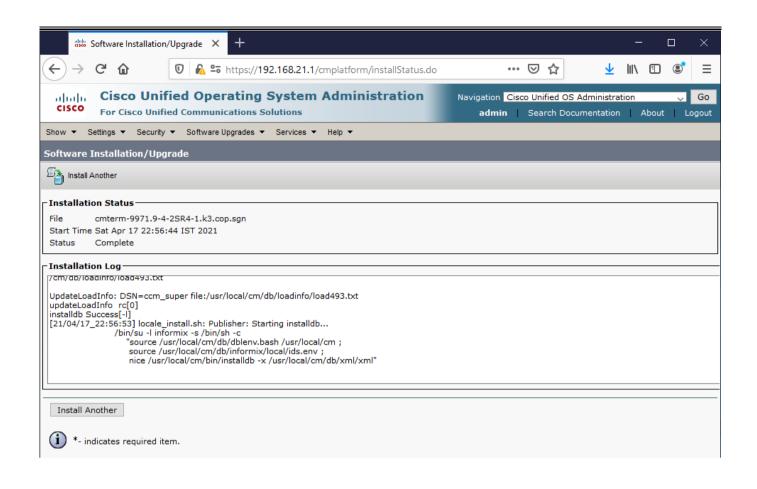
i *- indicates required item.

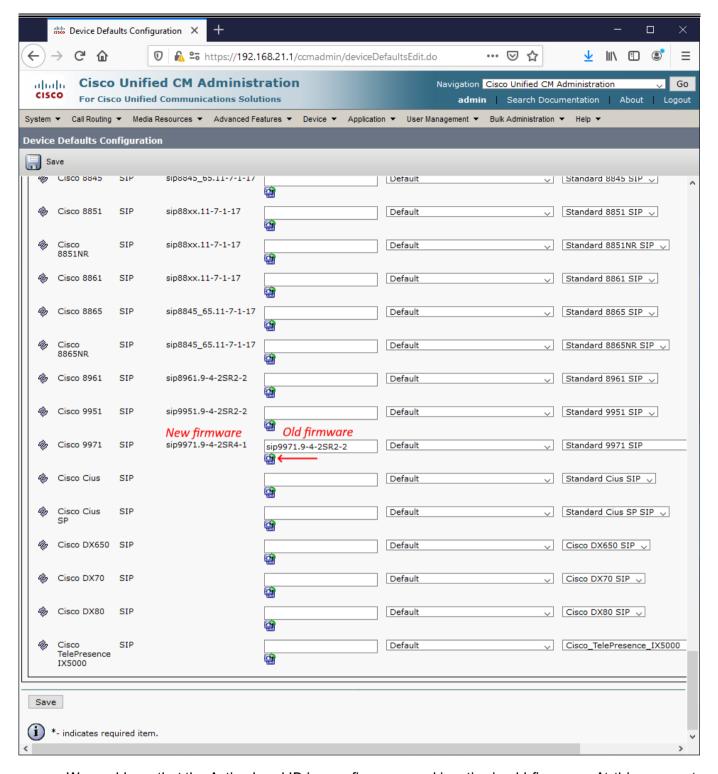
• show tech devdefaults will give the complete list of default firmware from CLI



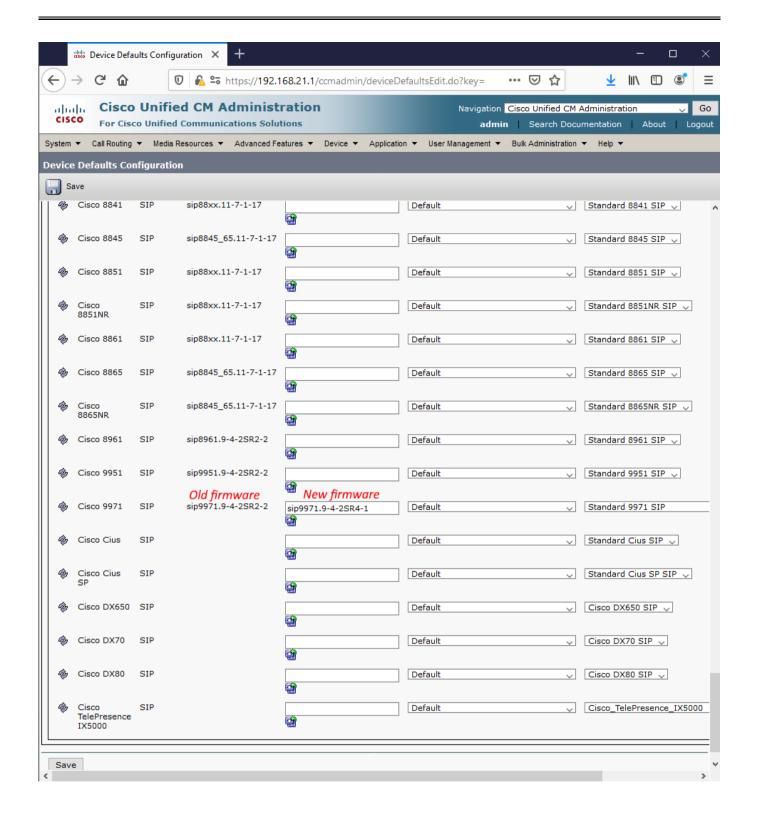




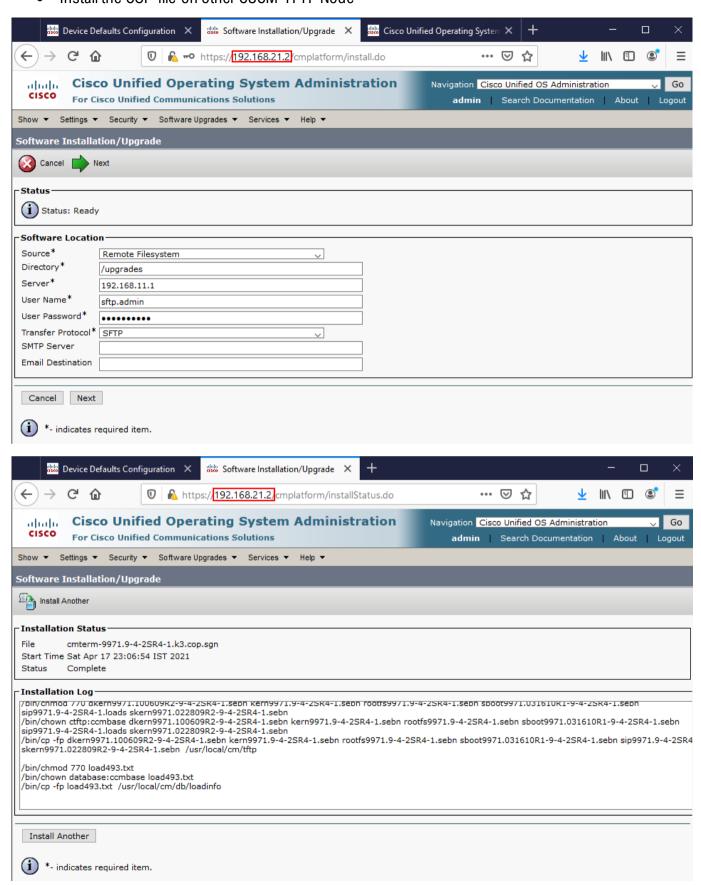




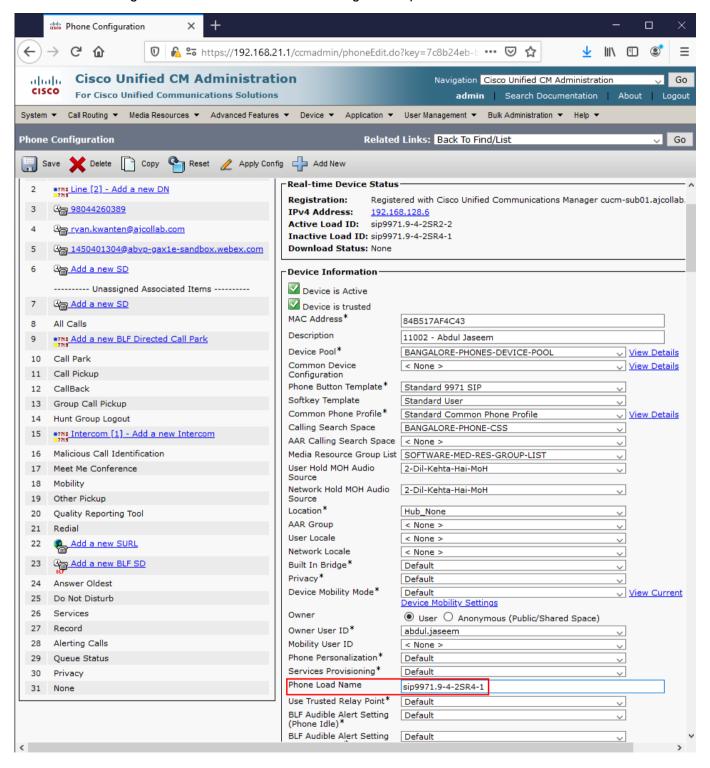
 We could see that the Active Load ID is new firmware and inactive is old firmware. At this moment any 9971 Phone reboots, it goes for an upgrade. Let's flit the active and inactive load to make sure Active is always the old firmware

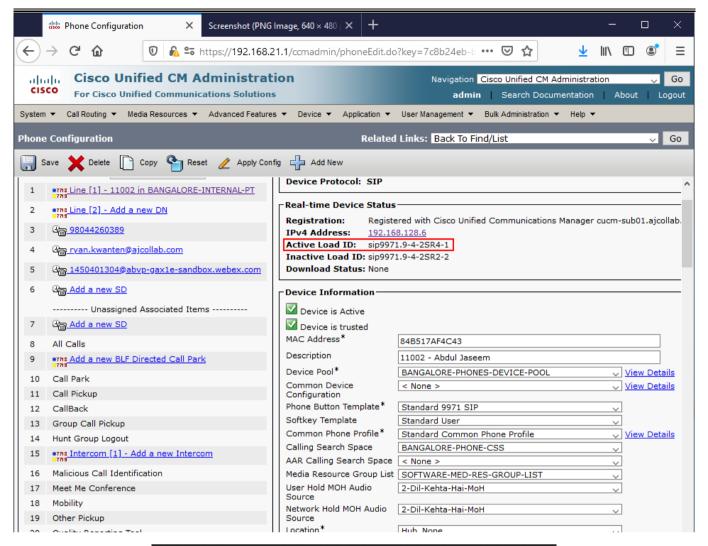


• Install the COP file on other CUCM TFTP Node



- Now copy the new load name 'sip9971.9-4-2SR4-1' and populate in the 'Phone Load Name' of device page
- Save configurations and reboot the Phone to get the updated firmware



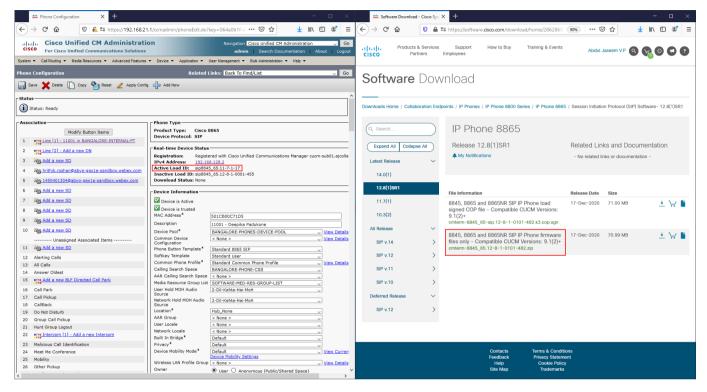




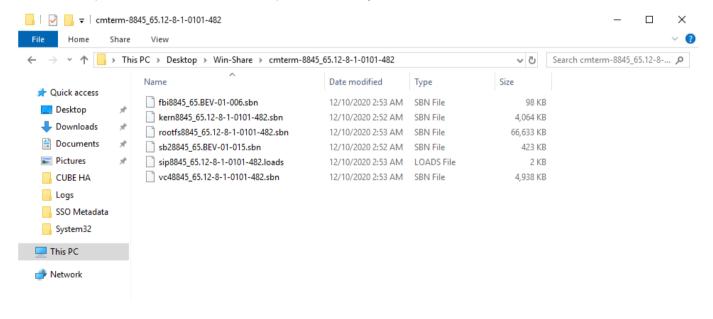
- Now you can delete the values in the 'Phone Load Name' if needed
- The 'Phone Load Name' take preference over Device Default firmware name
- You can downgrade the firmware by giving the old firmware name on 'Phone Load Name'

Firmware Files only Upgrade

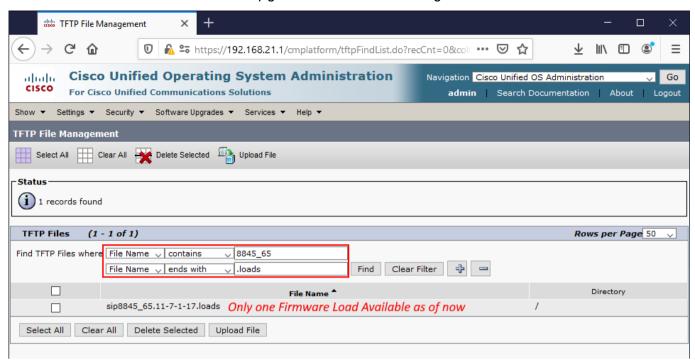
- If you do not prefer this method, we can go for TFTP based firmware upgrade process. It doesn't require any SFTP Server
- This method never affects anything on the Device Defaults. Individual firmware files must be installed on all the CUCM TFTP Nodes

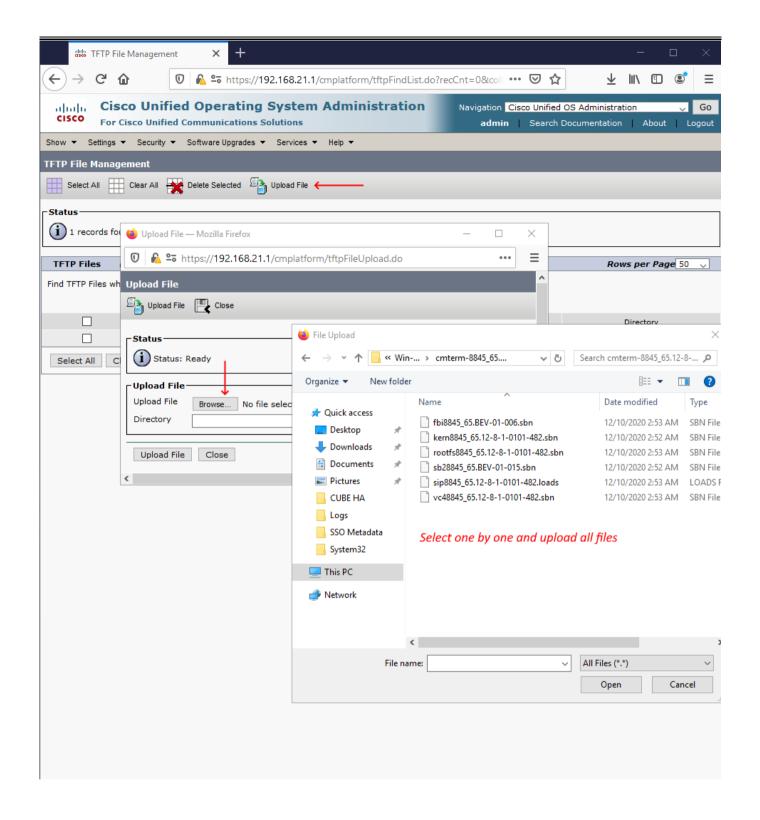


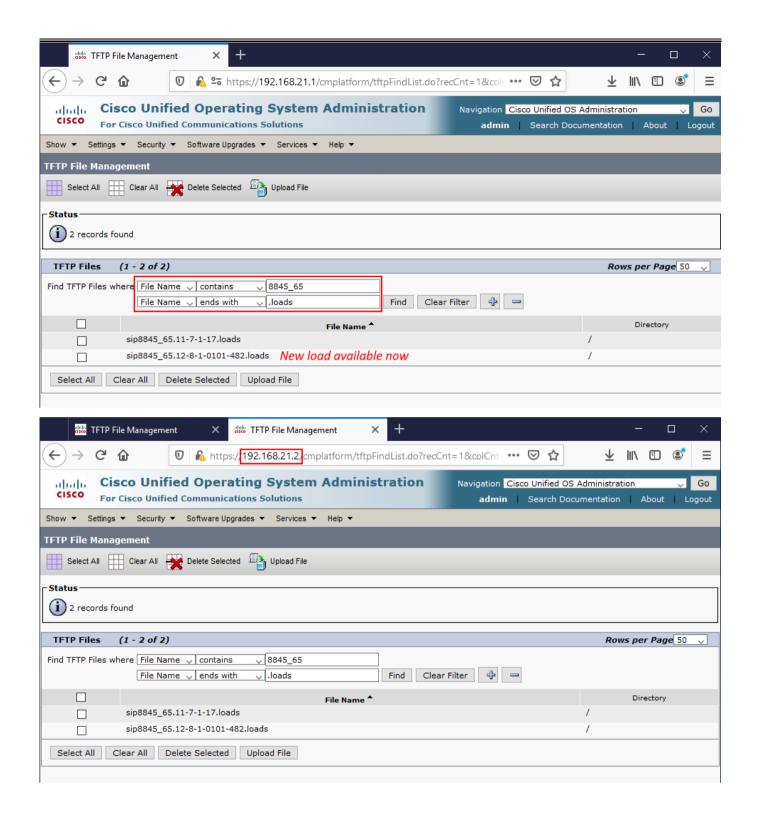
 The firmware files will be a Zipped folder, extract it and you will find multiple firmware file components. Each file must be uploaded one by one to all CUCM TFTP Servers



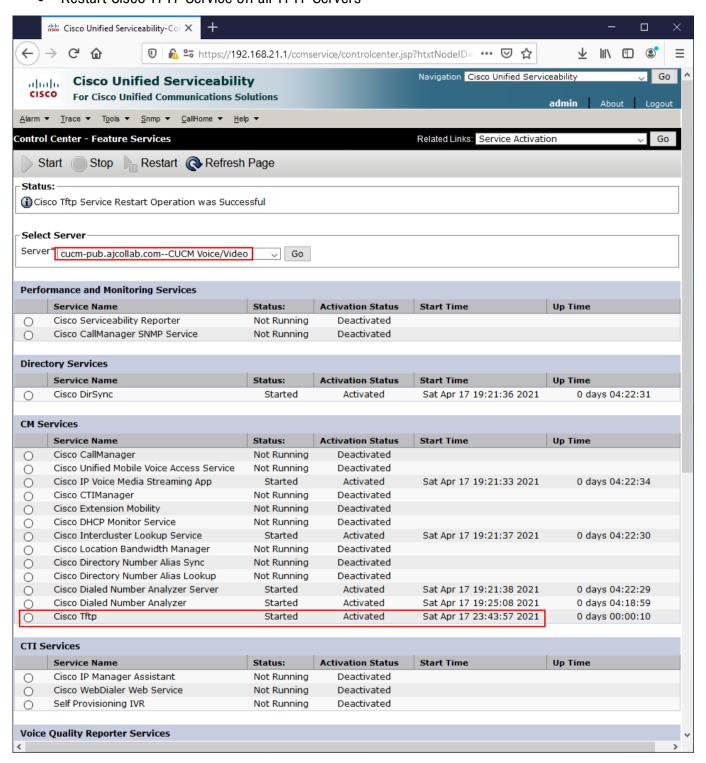
Go to OS Administration >> Software Upgrades >> TFTP File Management



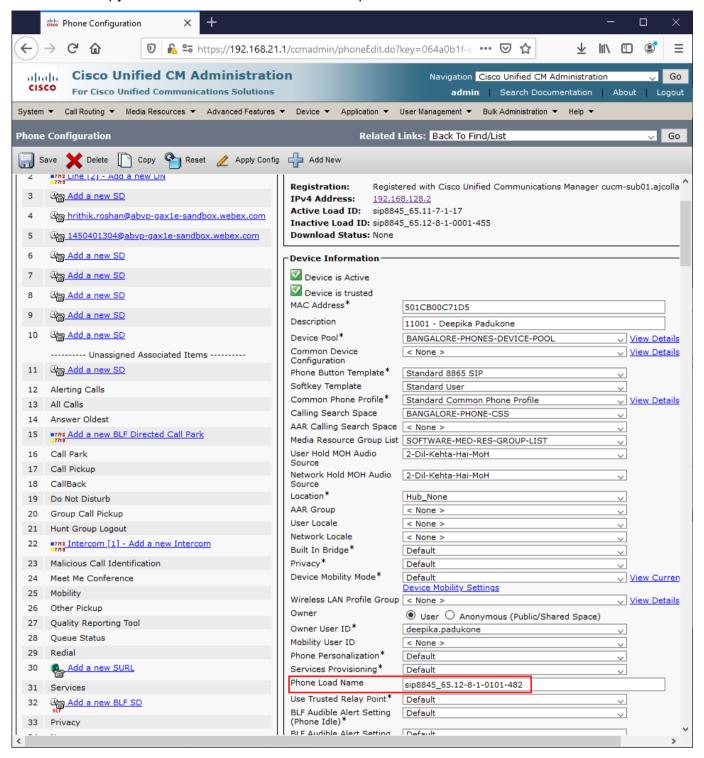


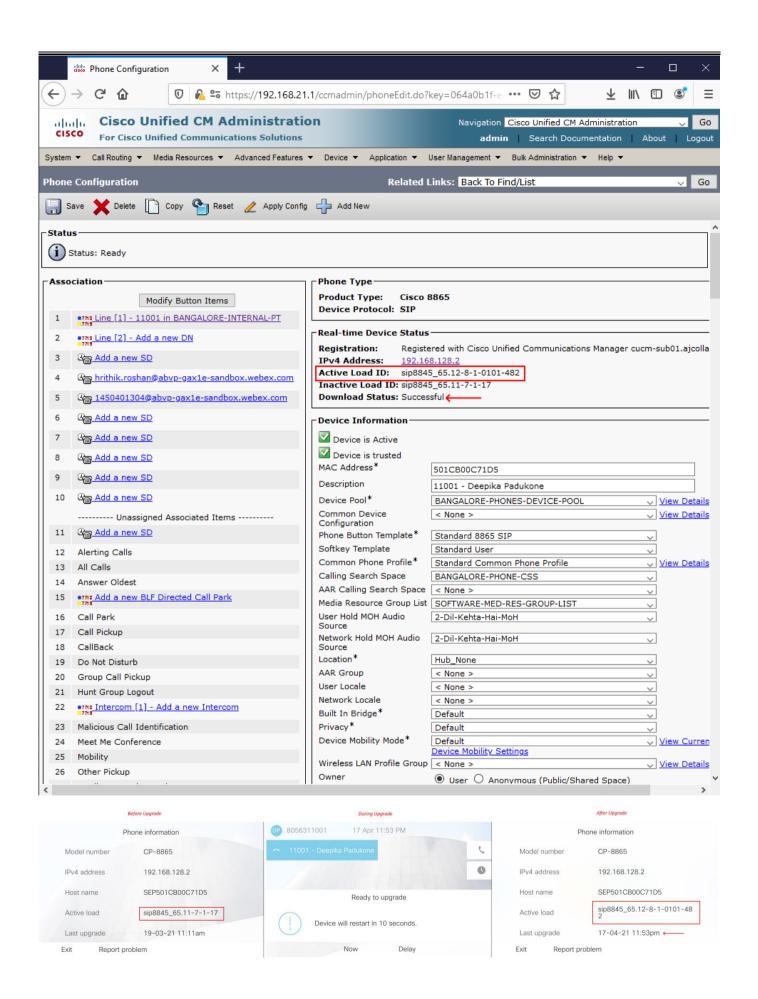


Restart Cisco TFTP Service on all TFTP Servers



Now copy the load name without '.loads' and update in the Phone Load Name

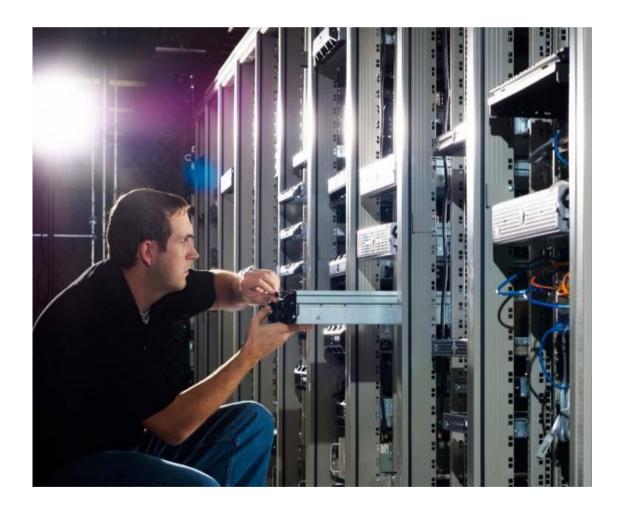




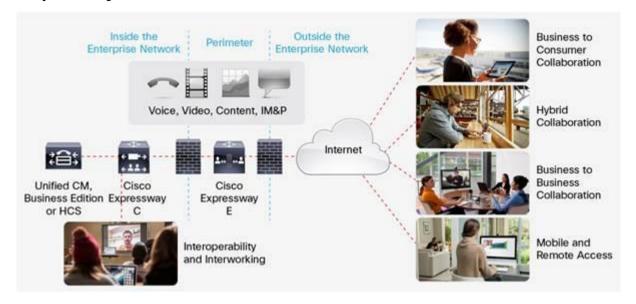
Installing Device Packs

- CUCM Device Pack is a packaged firmware for multiple devices, some old versions of CUCM, it enables latest phone model support as well
- This can be installed only via SFTP method, make sure you take the Device Default values before
 installing Device Pack since Device Pack will update many default firmware
- Device Pack has to be installed on all CUCM Nodes (Publisher and Subscribers) and a Cluster wide reboot is also need in many situations (Read the documentation of Device Pack to know more about the Reboot)
- It is just like another COP file installation; I'm not adding the steps here since we are much familiar with COP file installation
- List Device Pack for CUCM 11.5 available here: CUCM 11.5 Device Packs

Chapter 2 - Expressway Cisco Edge Collaboration Solution Cisco VCS or Expressway Series



Cisco Expressway

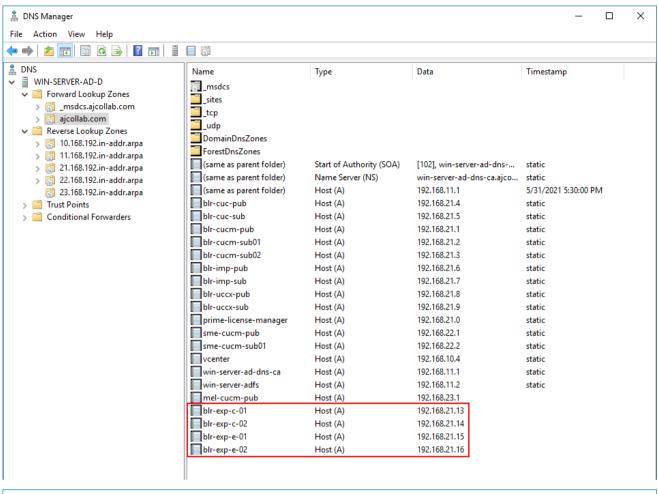


- It enables simple and safe, session-based collaboration outside your firewall without extra registration, accounts, passwords, or VPN
- Cisco Expressway provides mobile users and guests with unlimited access to all collaboration workloads, including video, voice, instant messaging and content sharing, and thus companies with the opportunity to improve and extend their organizational reach and collaboration with external employees, customers, or partners
- Cisco Expressway supports a wide range of IP phones, endpoints, smartphones, tablets, and provides video interoperability with standards-based H.323, H.264 SVC, or SIP systems
- Older name of was VCS (Video Communication Server), it can do end point registration and firewall traversal
- Cisco removed the endpoint registration capability and released the same product in different name called Expressway. Primary use was just firewall traversal
- Prior to X8.9, we can't register any devices in Expressway, but now (post X8.9) there is no
 difference in VCS and Expressway in terms of functionality. Expressway can do device registration
 and firewall traversal. There is a slight difference in the licensing
- VCS is a full-featured call control system that can interface with many kinds of system. The
 Expressway Series shares many VCS features, but is more appropriate for deployments where
 Cisco Unified Communications Manager is managing the call control

Type of calls in Expressway

- B2B (Business to Business) Calls: One domain to another domain calls
- MRA (Mobile and Remote Access)
- Microsoft Integration (Skype for Business)
- Hybrid Calls (Cisco Meeting Server CMS)

[Lab] DNS Entries for Expressway Series Servers



```
×
Administrator: Command Prompt
dicrosoft Windows [Version 10.0.14393]
(c) 2016 Microsoft Corporation. All rights reserved.
:\Users\Administrator>nslookup blr-exp-c-01
Server: win-server-ad-dns-ca.ajcollab.com
Address:
           192.168.11.1
          blr-exp-c-01.ajcollab.com
lame:
Address: 192.168.21.13
:\Users\Administrator>nslookup blr-exp-c-02
Server: win-server-ad-dns-ca.ajcollab.com
Address: 192.168.11.1
Name: blr-exp-c-02.ajcollab.com
Address: 192.168.21.14
C:\Users\Administrator>nslookup blr-exp-e-01
Server: win-server-ad-dns-ca.ajcollab.com
Address: 192.168.11.1
          blr-exp-e-01.ajcollab.com
lame:
Address: 192.168.21.15
:\Users\Administrator>nslookup blr-exp-e-02
Server: win-server-ad-dns-ca.ajcollab.com
Address: 192.168.11.1
Vame:
         blr-exp-e-02.ajcollab.com
Address: 192.168.21.16
```

Note: You may find some screenshots without 'blr' in the hostname, those are taken from my old lab.

[Lab] Deploy Expressways

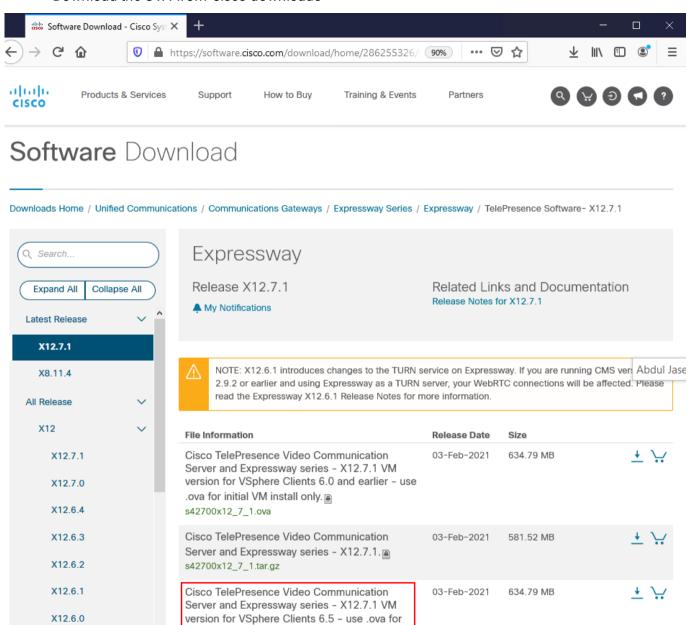
- Expressway C and E uses same software, initialization and License makes it Edge or Core
- While deploying we use same OVA file for Expressway E and C

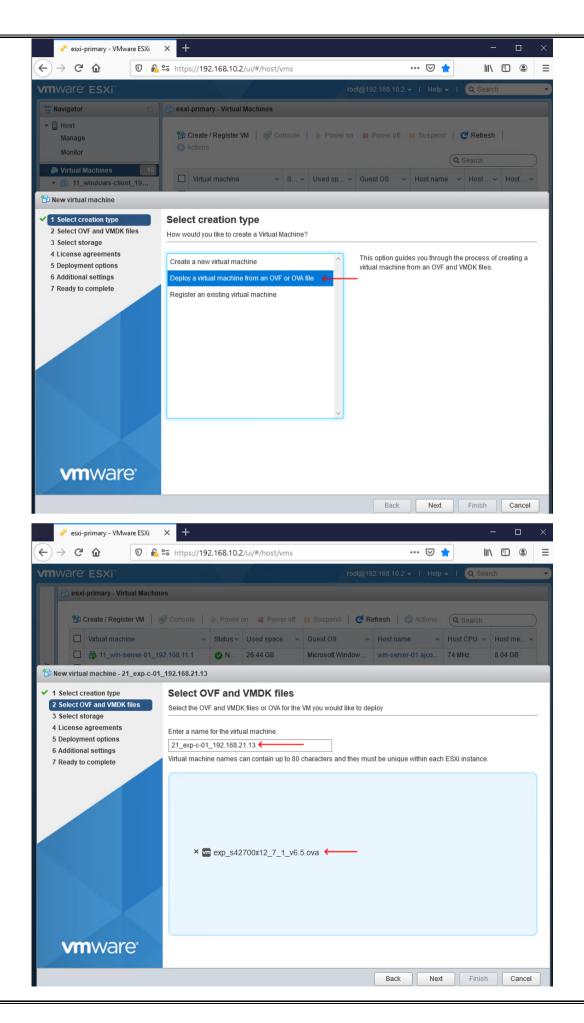
initial VM install only.

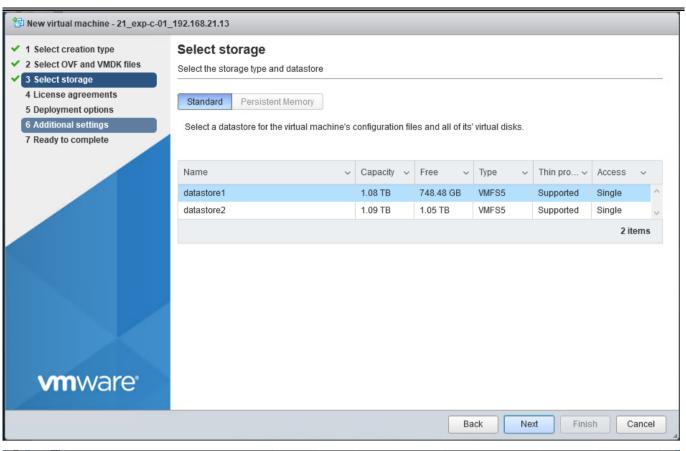
s42700x12_7_1_v6.5.ova

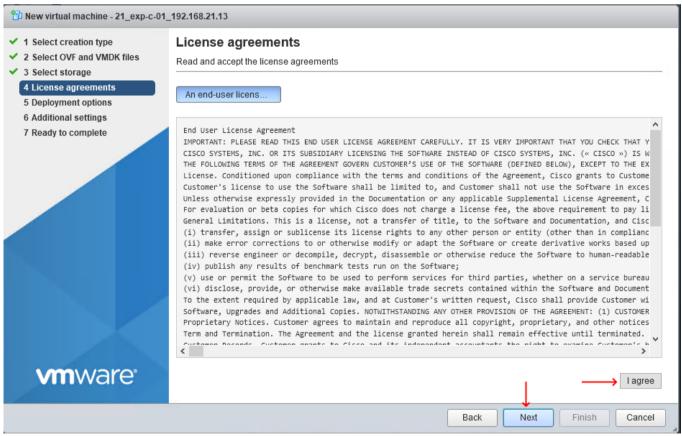
X12.5.9

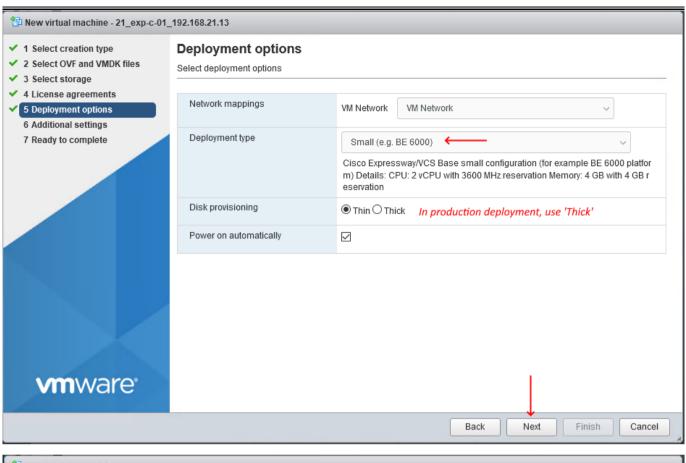
Download the OVA from Cisco downloads

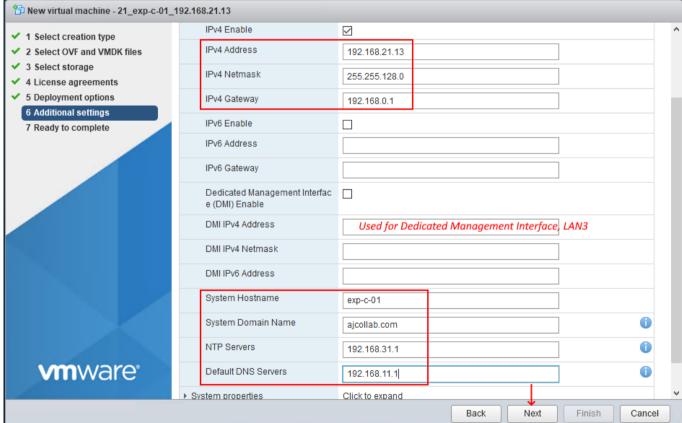




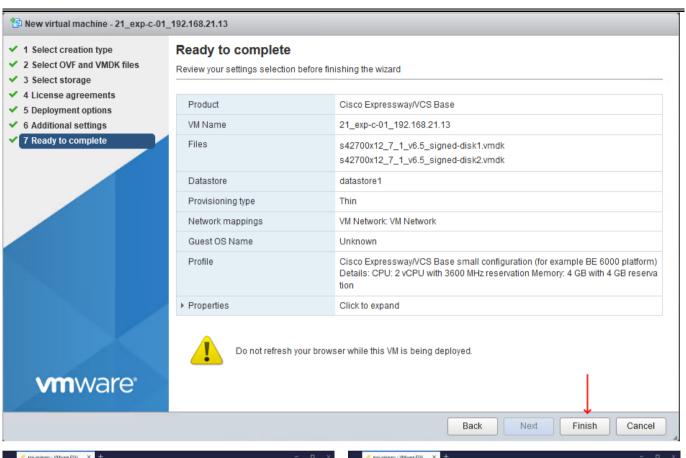


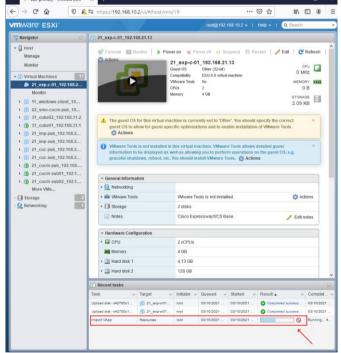


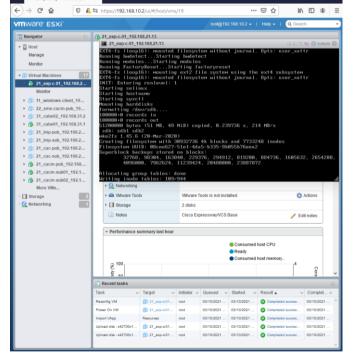


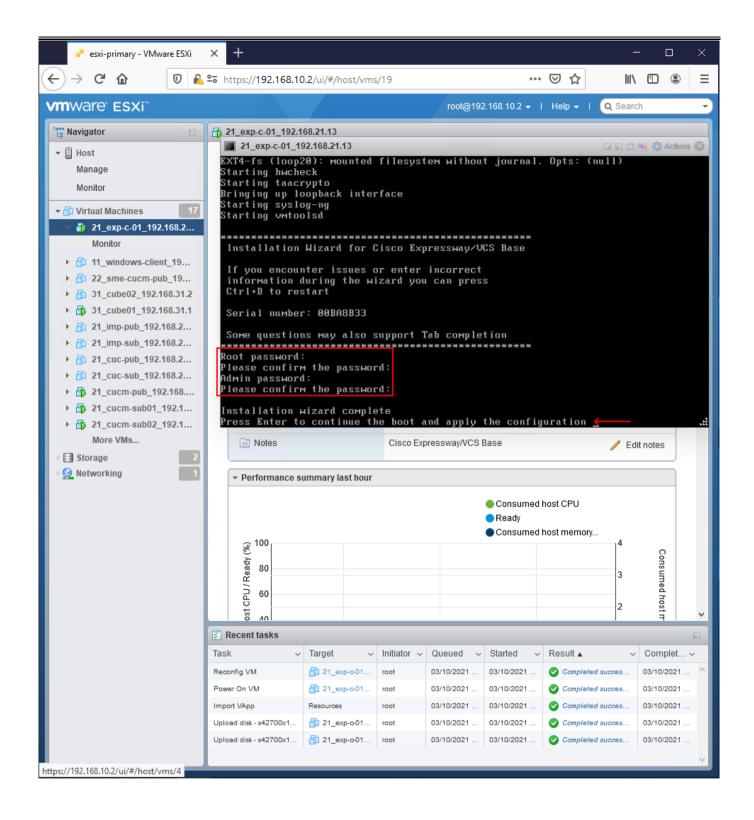


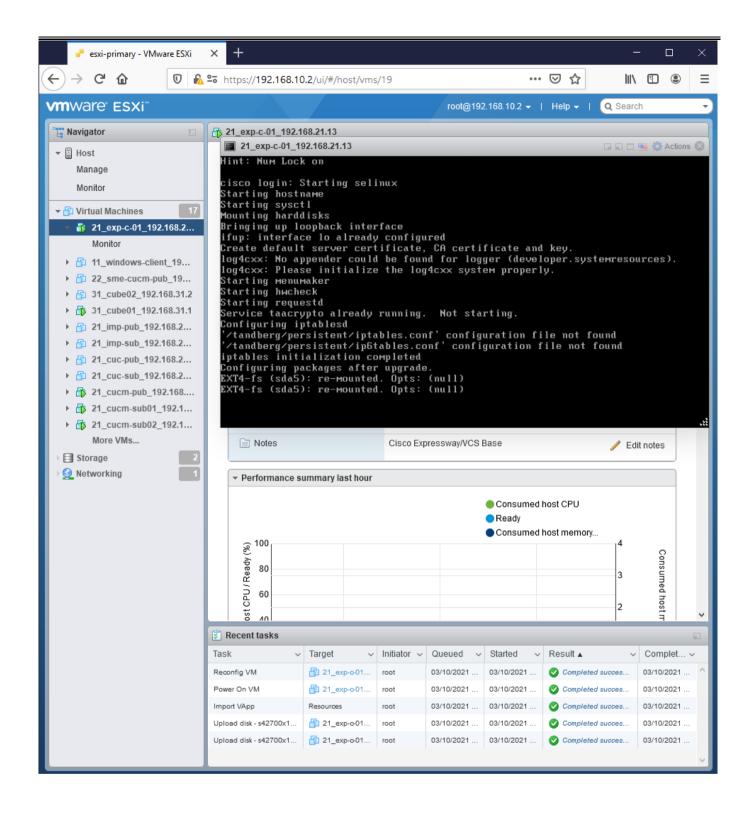
Note: If you miss to configure IP here, you can do in in the vmware console after the VM boot up

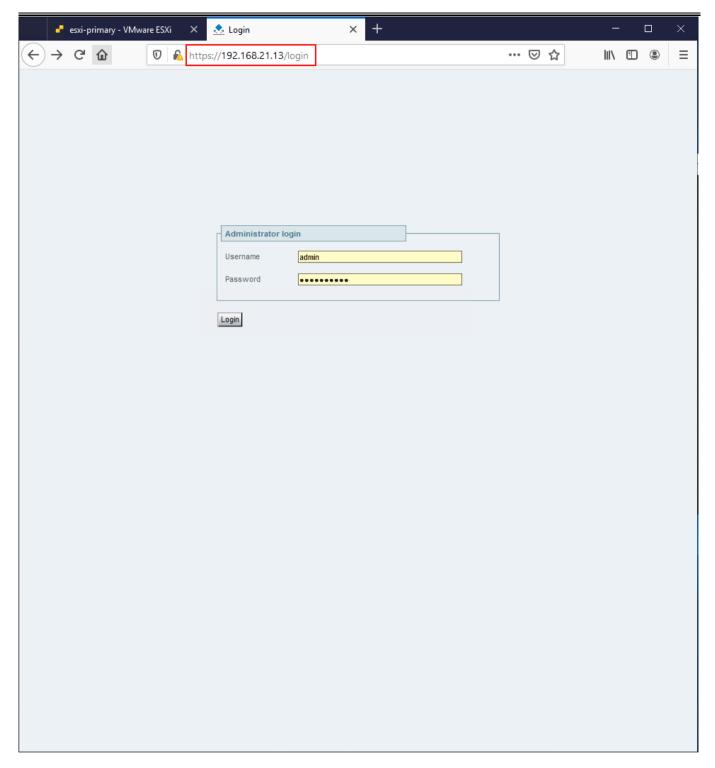












• I have deployed 4 OVAs and will be converting 2 to Expressway C and other 2 to Expressway E

Different Types of License Expressway - C

- After deploying the Expressway OVA, we need to license it. Licenses are in the form of options keys
- Below are some major options keys used in Expressway C

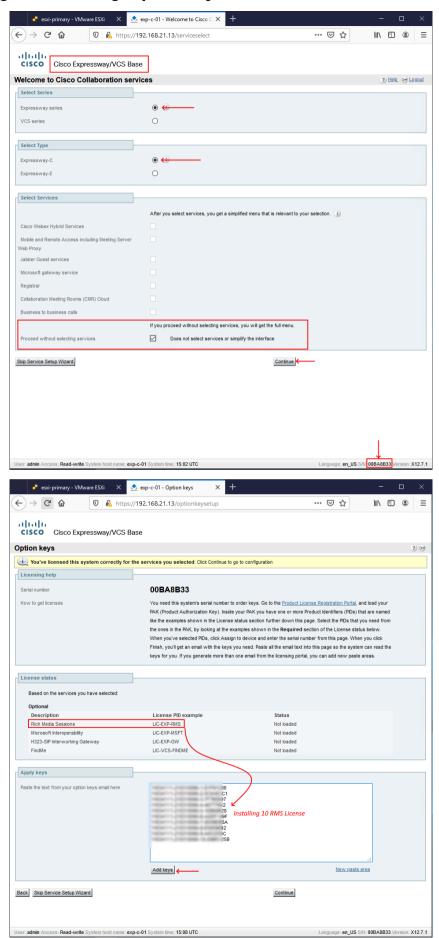
License Name	Purpose
LIC-EXP-RMS	Expressway Rich Media Session – For specific calls like B2B and LYNC integration
LIC-EXP-ROOM	Enable the registration of Room endpoints to the Expressway-C. Remote devices are proxied
	to Expressway-C by Expressway-E
LIC-EXP-DSK	Enable the registration of Desk endpoints to the Expressway-C. Remote devices are proxied
	to Expressway-C by Expressway-E
LIC-EXP-GW	Enable GW Feature (H323-SIP) – For H323-SIP interworking
LIC-EXP-MSFT	Microsoft Interoperability Option – For LYNC integration

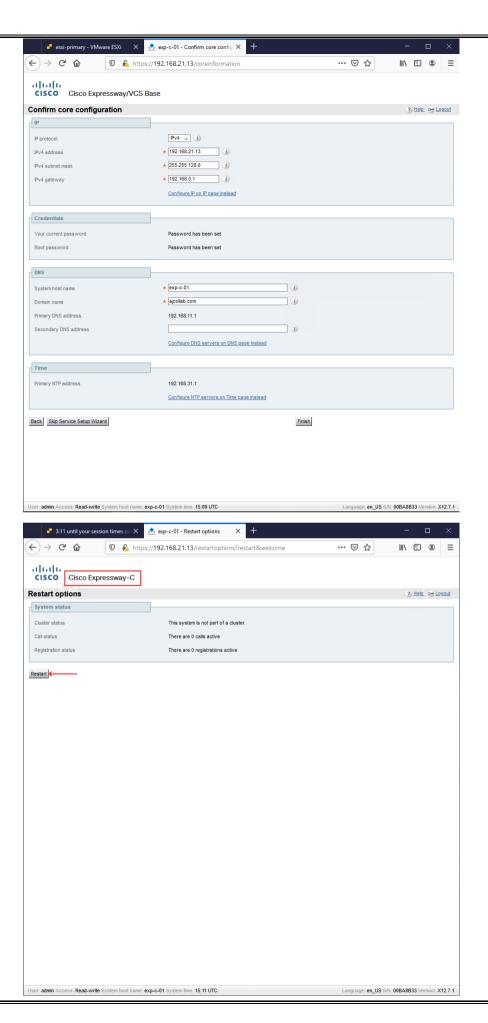
Different Types of License Expressway - E

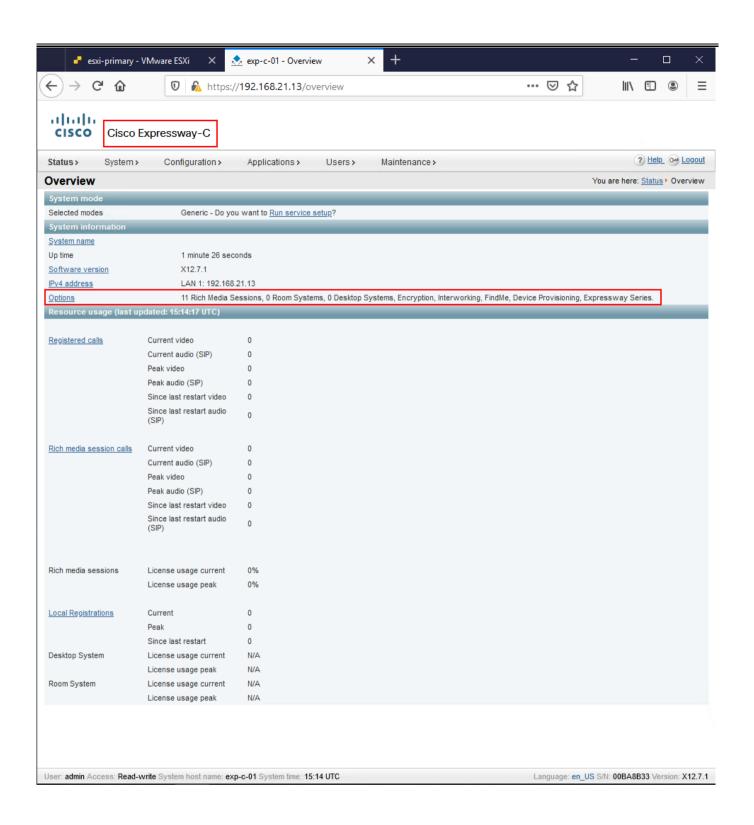
• Below are some major options keys used in Expressway E

License Name	Purpose
LIC-EXP-RMS	Expressway Rich Media Session – For specific calls like B2B and LYNC integration
LIC-EXP-AN	Enable Advanced Networking Option – Enable Dual-NIC and Static NAT address

[Lab] Initializing and Licensing Expressway - C

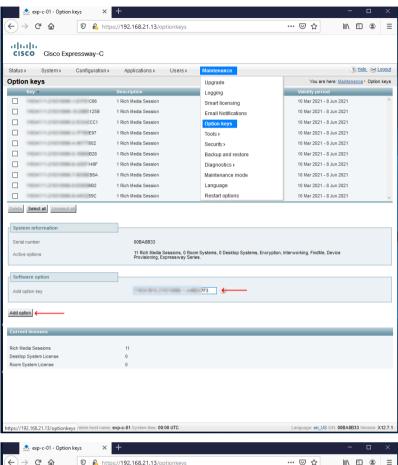


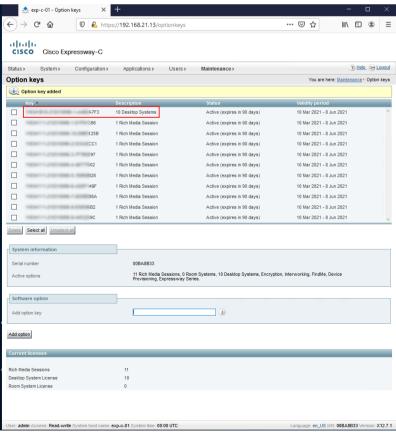




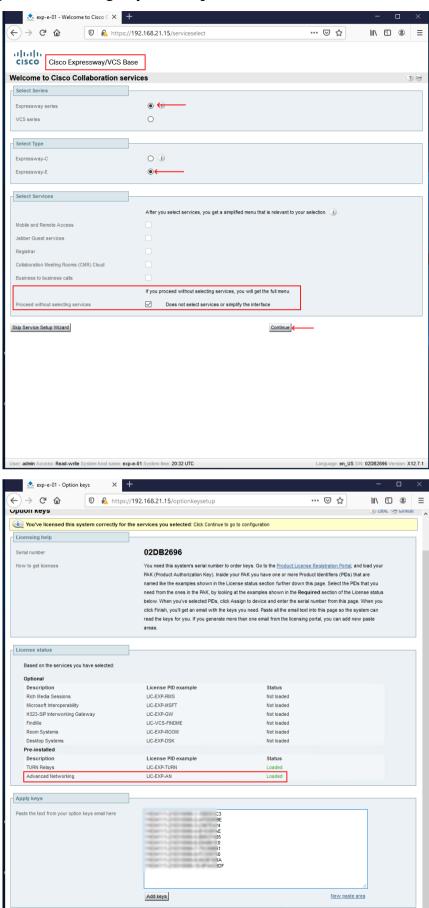
[Lab] Installing Additional License on Expressway - C

- I wanted to perform Desk Endpoint (DX70, DX80, etc.) registration in Expressway C
- Hence, I should install 'LIC-EXP-DSK' on Expressway C



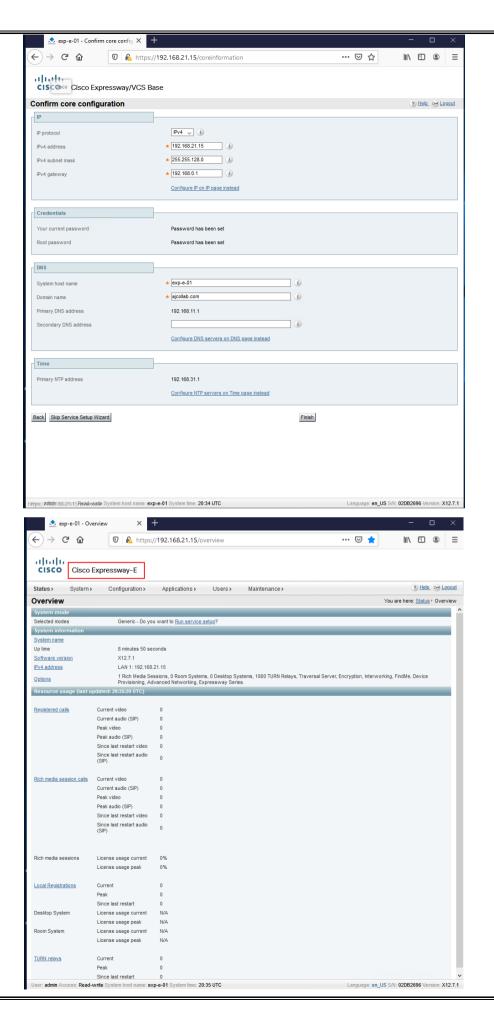


[Lab] Initializing and Licensing Expressway - E

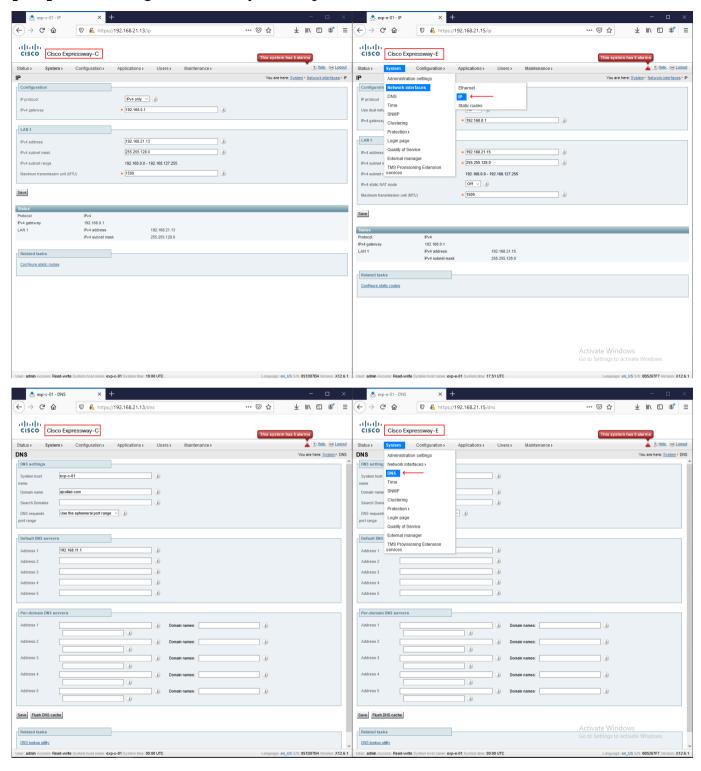


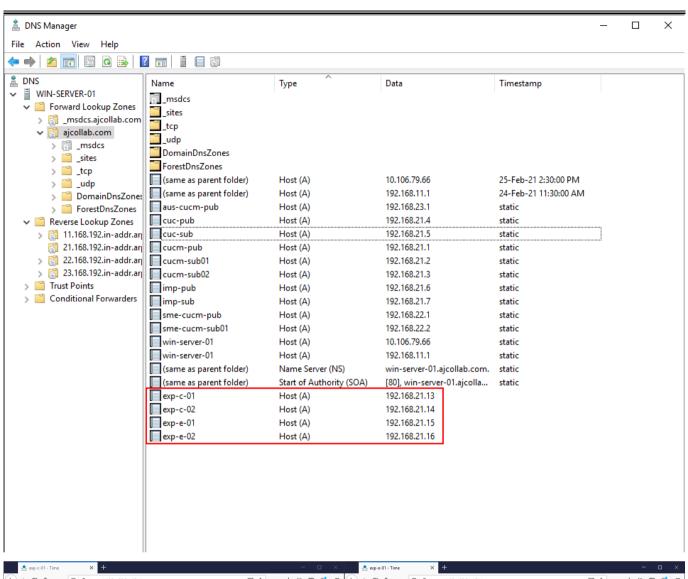
Continue

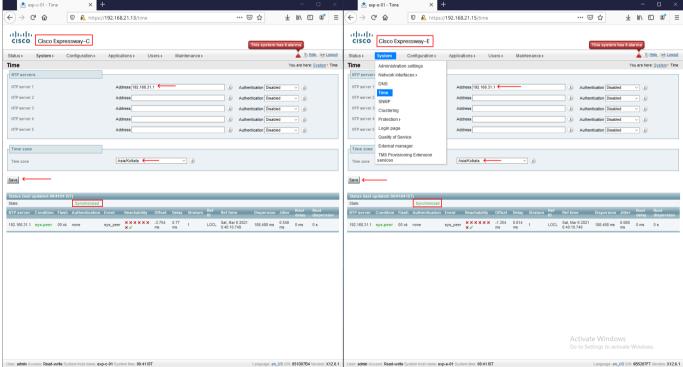
Back Skip Service Setup Wizard



[Lab] Basic Configuration of Expressway - C and E

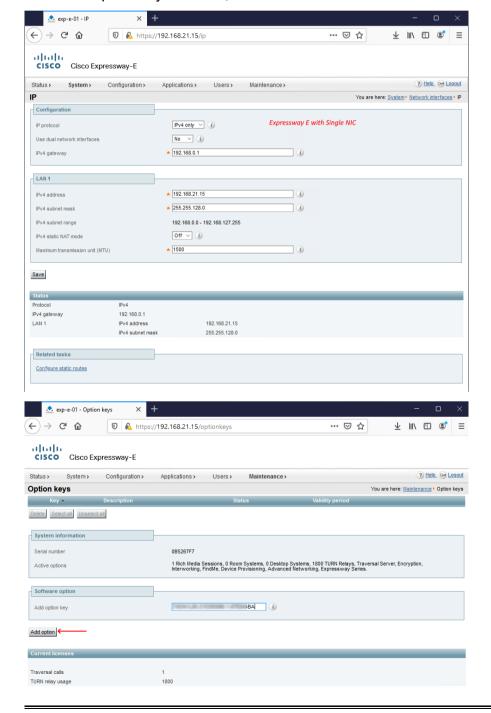


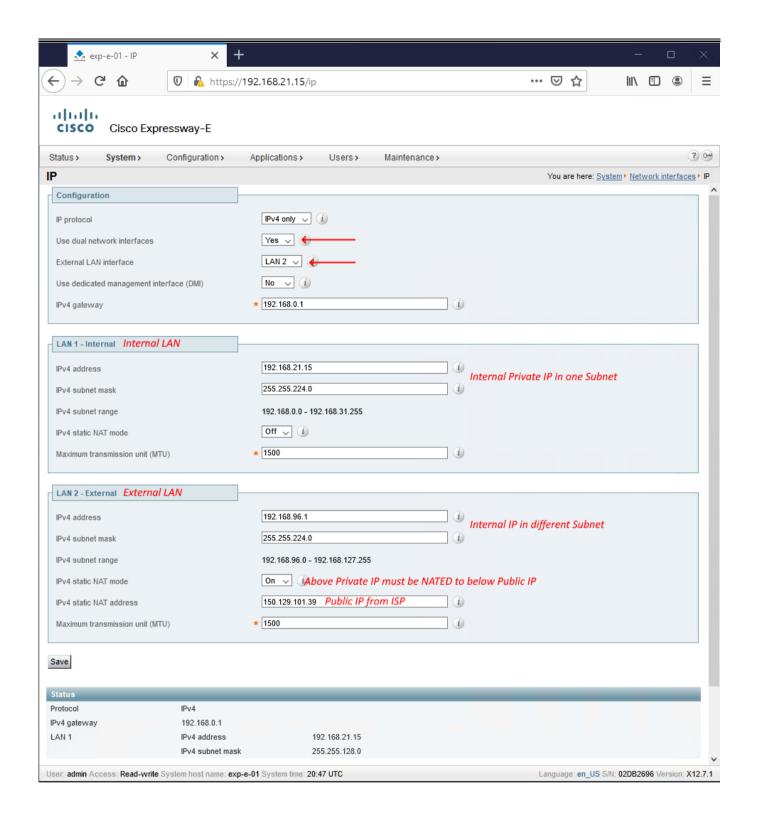


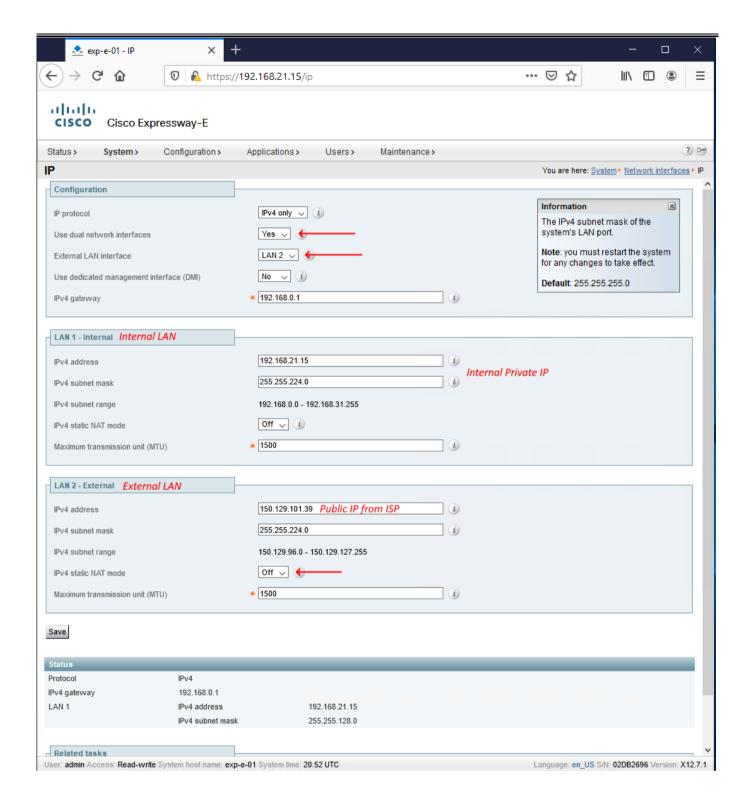


Expressway - E Single NIC vs Dual NIC

- Single NIC, assign Private IP Address on the Expressway E and configure a static one to one NAT with a Public IP or directly configure public IP on Expressway E
- Single NIC requires NAT reflection on the firewall, otherwise Expressway E's private IP address will be advertised via C IN of SDP to the far end
- Dual NIC, internal NIC with private IP range and External NIC with Private IP (different subnet than NIC1) NATEd to Public IP or direct Public IP
- We need 'Advanced Networking' option keys to enable Dual NIC on Expressway E
- If Expressway E is a VM, add additional vNIC to the VM







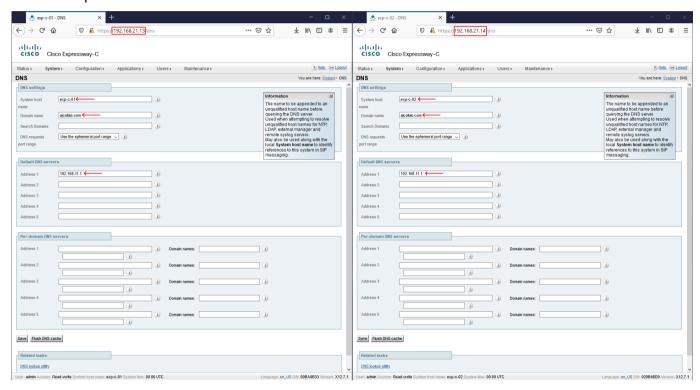
Clustering of Expressway

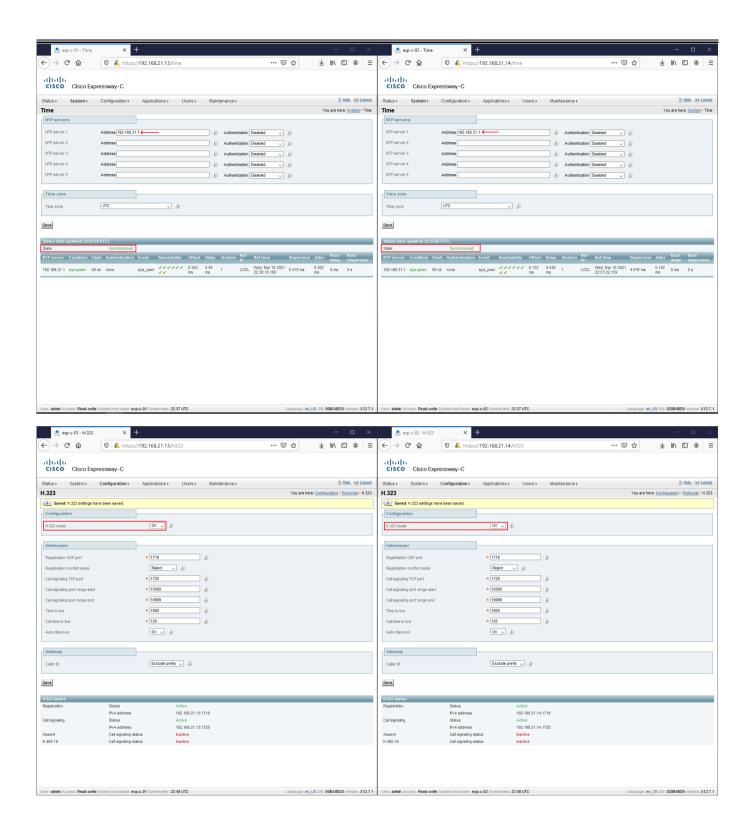


- We have 2 Expressway E and 2 Expressway C servers deployed (total 4 Expressways)
- An Expressway can be part of a cluster of up to six Expressways
- Clustering can provide redundancy while an Expressway is in maintenance mode, or in case it becomes inaccessible due to a network or power outage.
- Endpoints can register to any of the peers in the cluster, so if an endpoint loses connection to its initial peer, it can re-register to another one in the cluster
- Each Expressway in the cluster is a peer of every other one in that cluster
- When you create a cluster, you nominate one peer as the primary, from which its configuration is replicated to the other peers
- We must configure a Cluster FQDN for the Expressway Cluster. This should resolve to IP Address
 of each peers in the cluster

Clustering Prerequisites

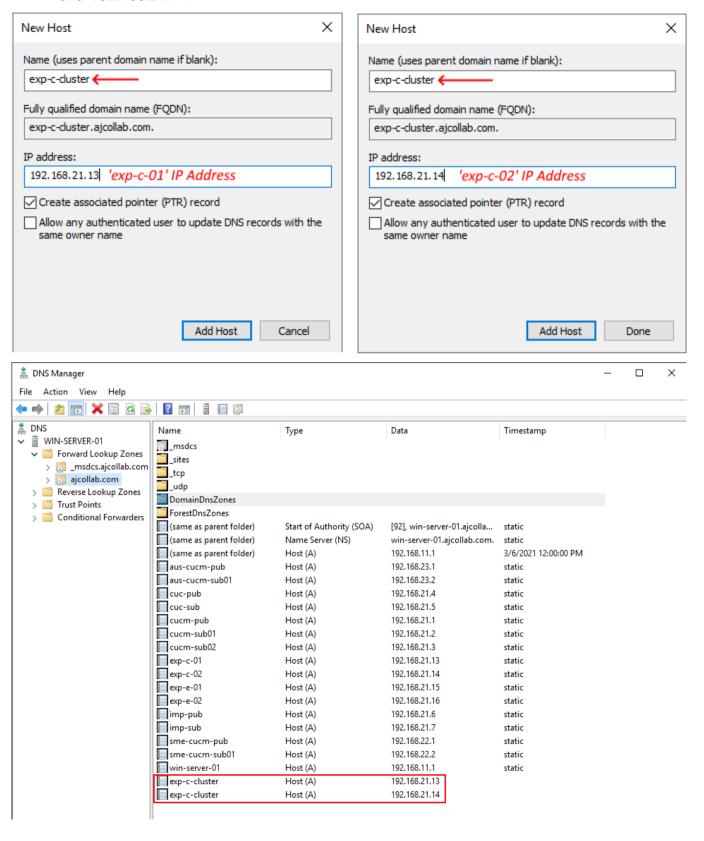
- All cluster peers are running the same Expressway version
- Each peer is using a hardware platform (appliance or virtual machine) with equivalent capabilities
- Each Expressway in the cluster must be within a 40ms hop round trip delay of all other peers in the cluster
- All peers have the same set of option keys installed, with exception Traversal and non-traversal call licenses, RMS, Room and Desktop System. Rest all the options keys must match
- H.323 mode is enabled on each peer (Configuration > Protocols > H.323). The cluster uses H.323 signaling between peers to determine the best route for calls, even if all endpoints are SIP endpoints

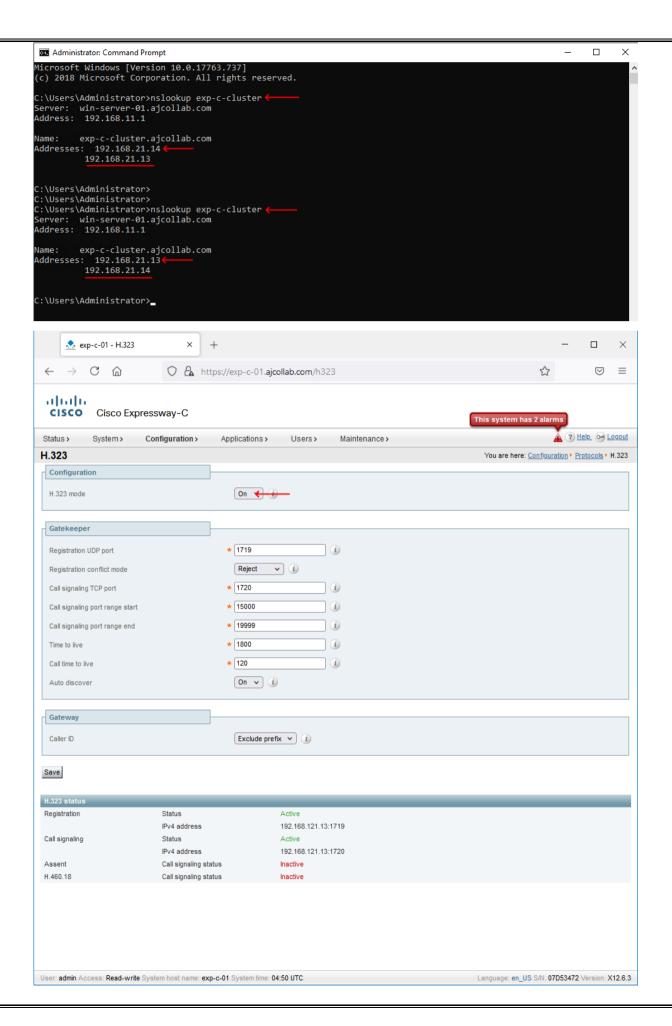


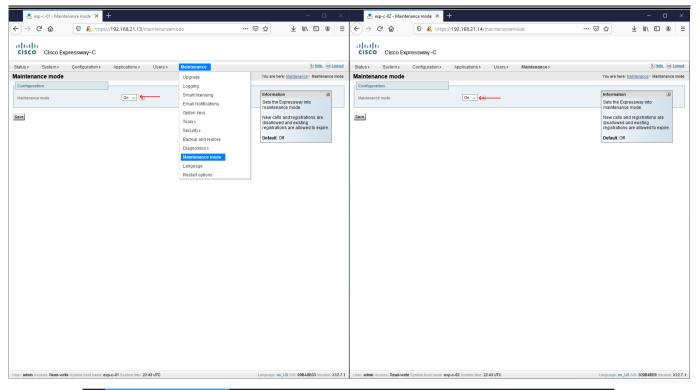


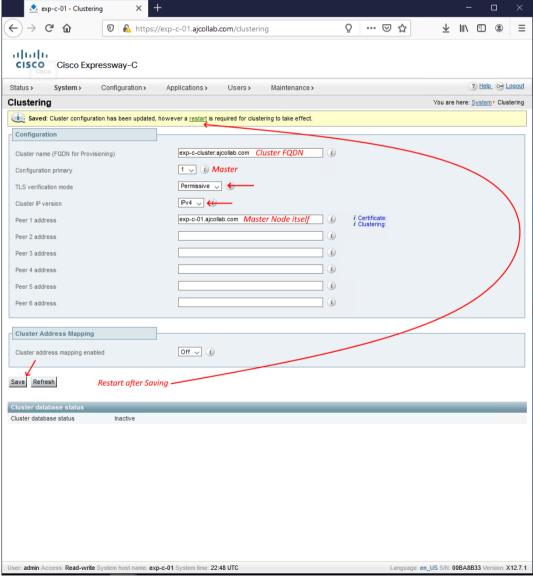
[Lab] Expressway - C Clustering

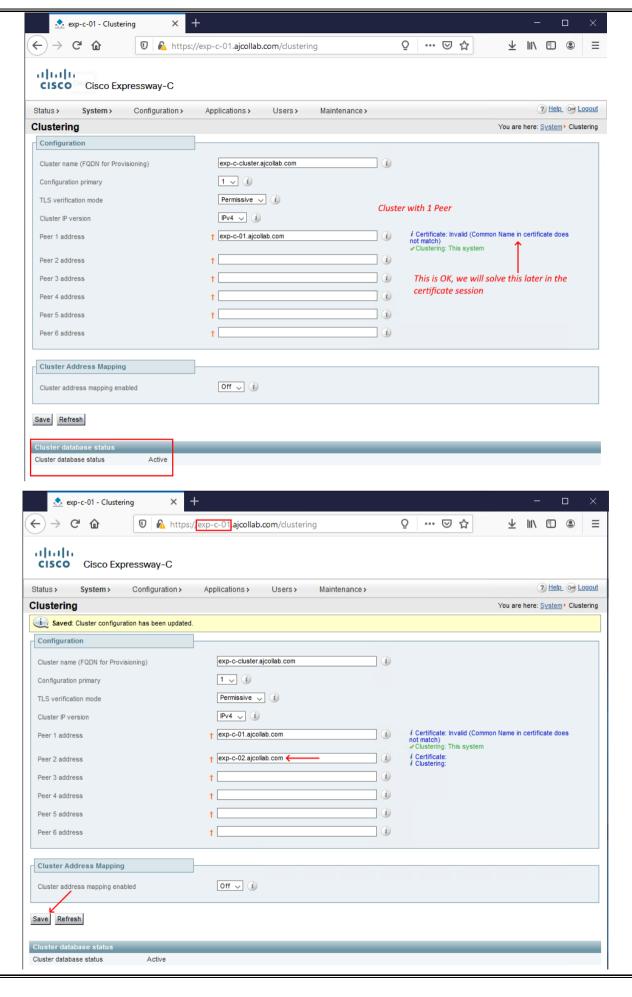
In my case, I will configure two A Record with name 'exp-c-cluster' and resolves to '192.168.21.13'
 and '192.168.21.14'

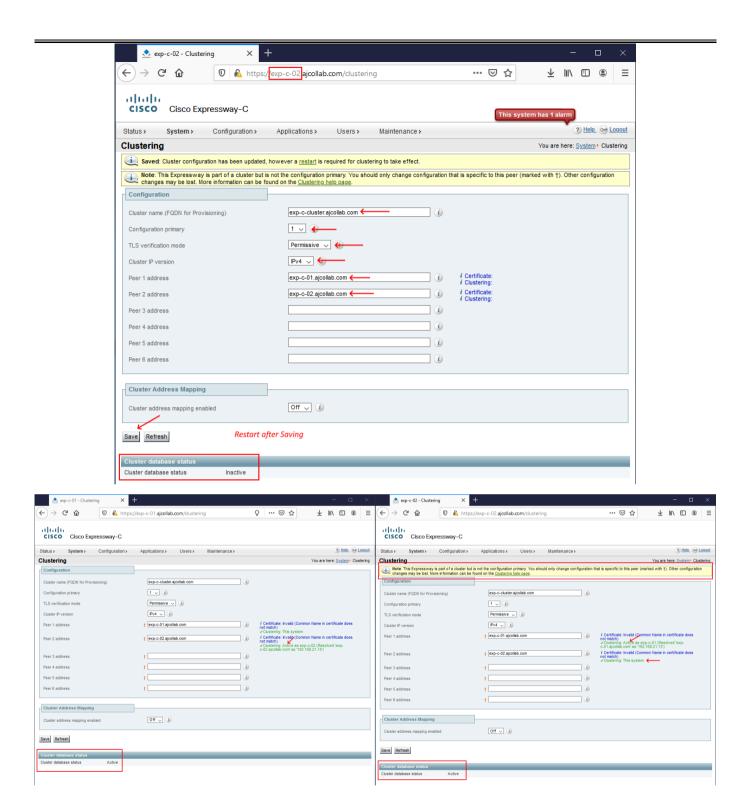






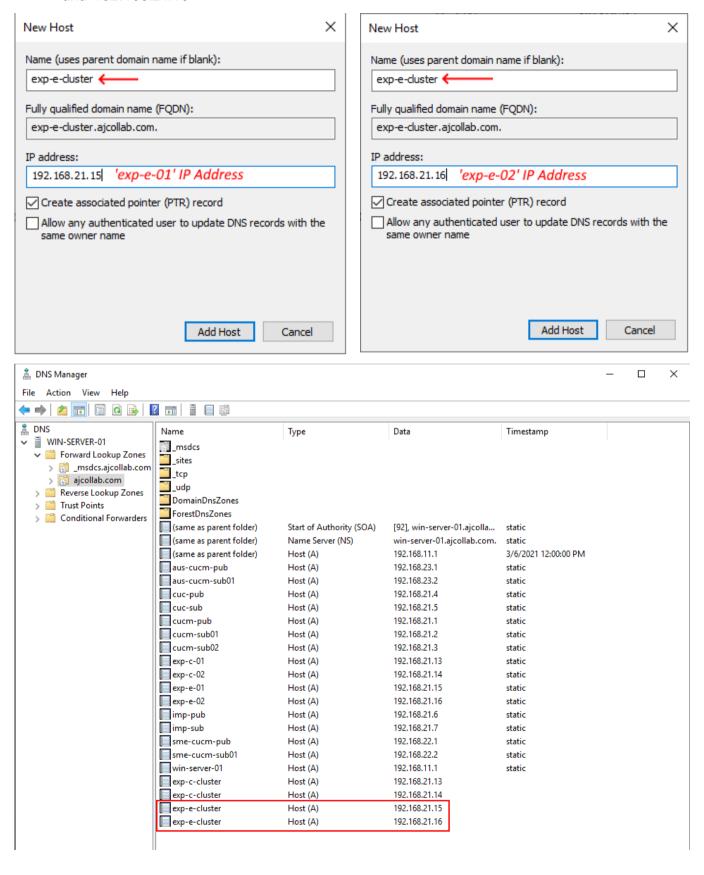






[Lab] Expressway - E Clustering

In my case, I will configure two A Record with name 'exp-c-cluster' and resolves to '192.168.21.15'
 and '192.168.21.16'



```
Administrator Command Prompt

Microsoft Windows [Version 10.0.17763.737]
(c) 2018 Microsoft Corporation. All rights reserved.

C:\Users\Administrator>nslookup exp-e-cluster
Server: win-server-01.ajcollab.com
Address: 192.168.11.1

Name: exp-e-cluster.ajcollab.com
Addresses: 192.168.21.15

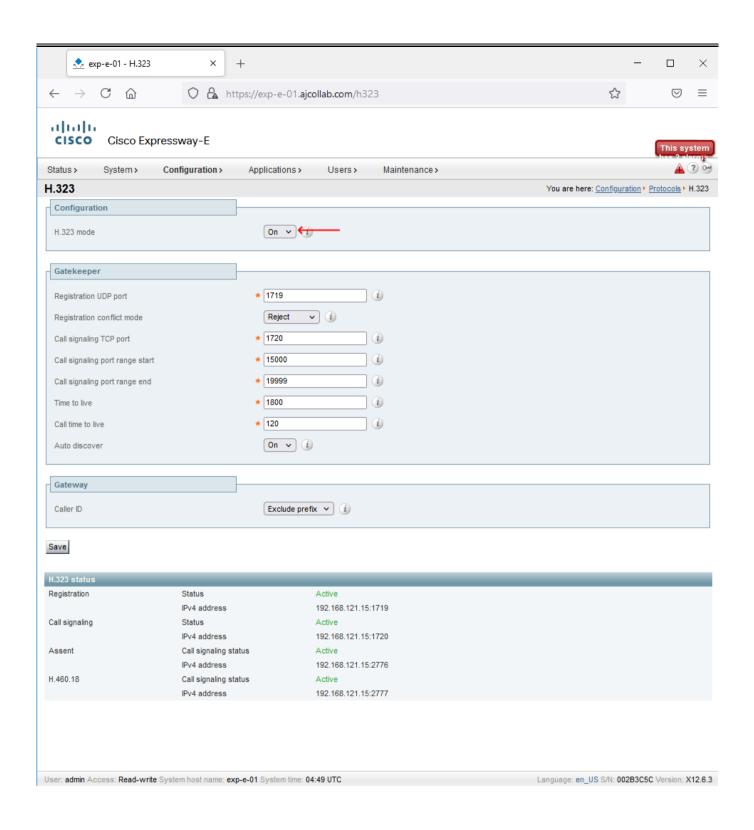
192.168.21.16

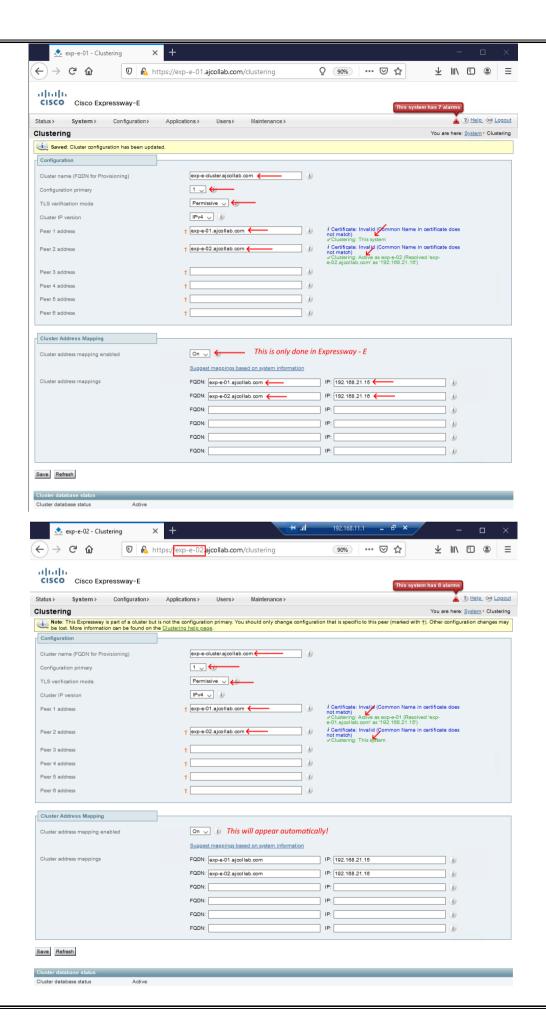
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
Addresses: 192.168.11.1

Name: exp-e-cluster.ajcollab.com
Addresses: 192.168.11.6

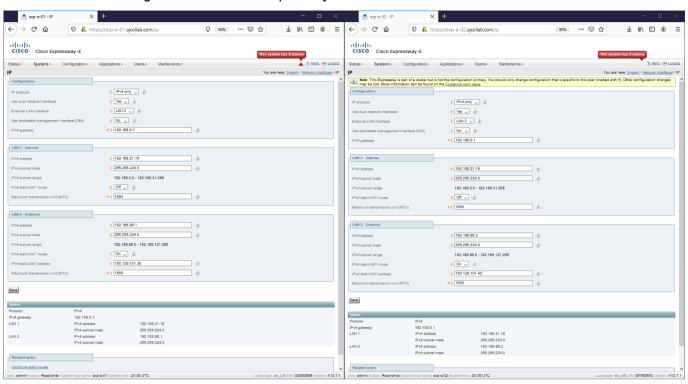
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
C:\Users\Administrator>
```

- Follow the similar procedure that you have done for Expressway C to get the cluster up
- We need to enable 'Cluster address mapping enabled' on Expressway E so that if forms cluster using internal lps. Note that Expressway E will have Public IP as well



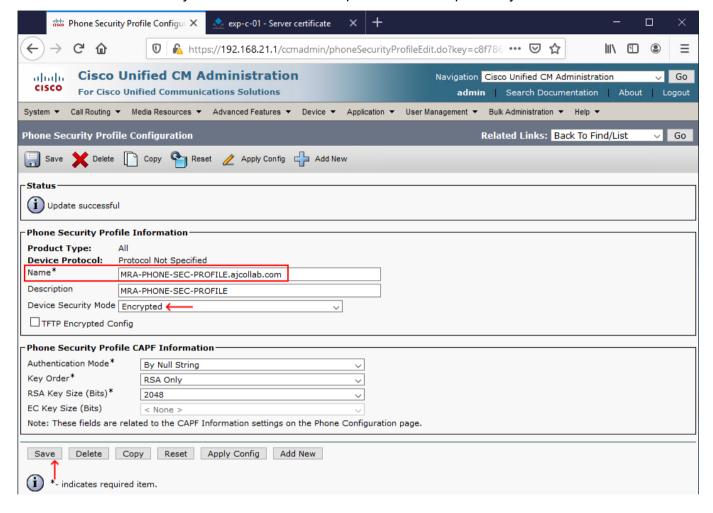


• Dual NIC configurations must be separately enabled on the other Peer

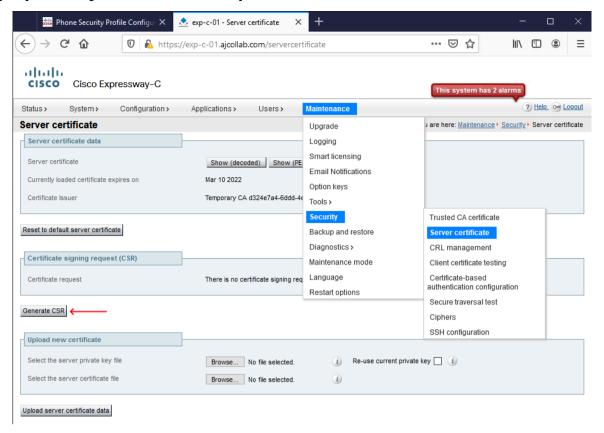


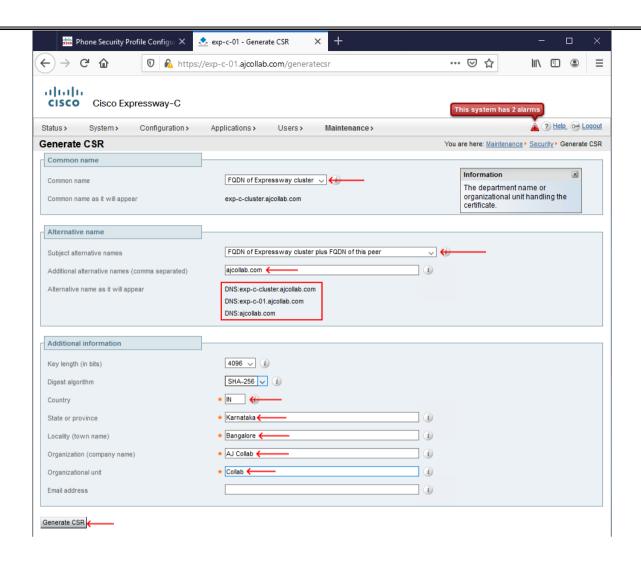
Phone Security Profile

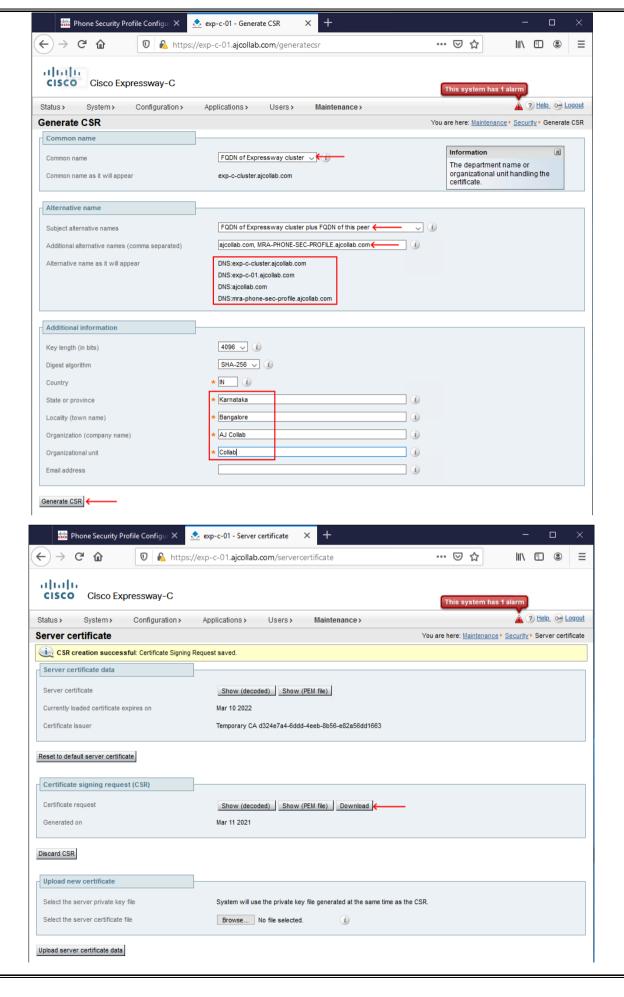
- For phone registration via MRA, we should have a Phone Security Profile
- The Phone Security Profile name should be present in the Expressway E' SAN field

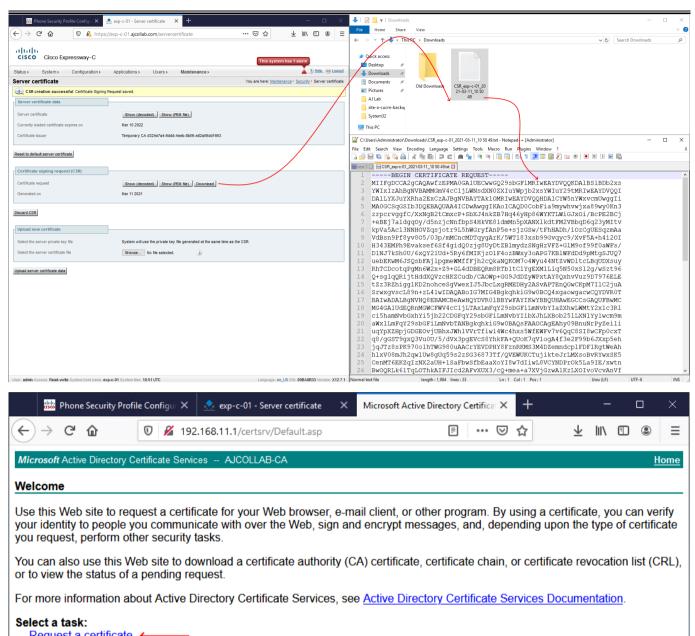


[Lab] Expressway - C Certificate Requirement for MRA





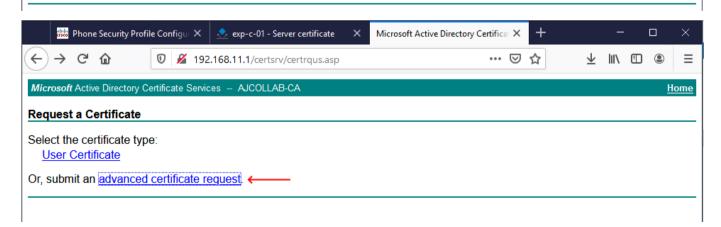


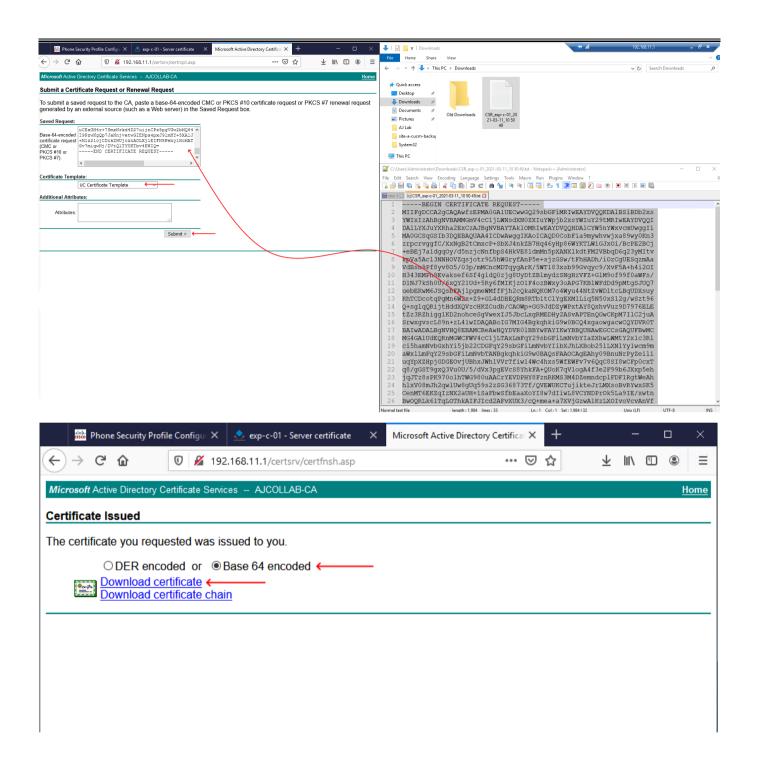


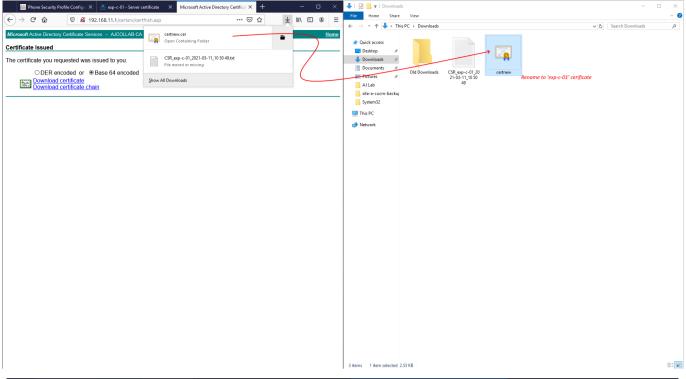
Request a certificate

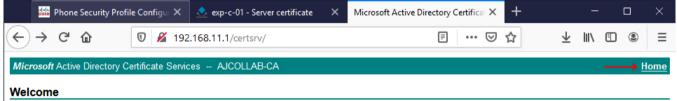
View the status of a pending certificate request

Download a CA certificate, certificate chain, or CRL









Use this Web site to request a certificate for your Web browser, e-mail client, or other program. By using a certificate, you can verify your identity to people you communicate with over the Web, sign and encrypt messages, and, depending upon the type of certificate you request, perform other security tasks.

You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CRL), or to view the status of a pending request.

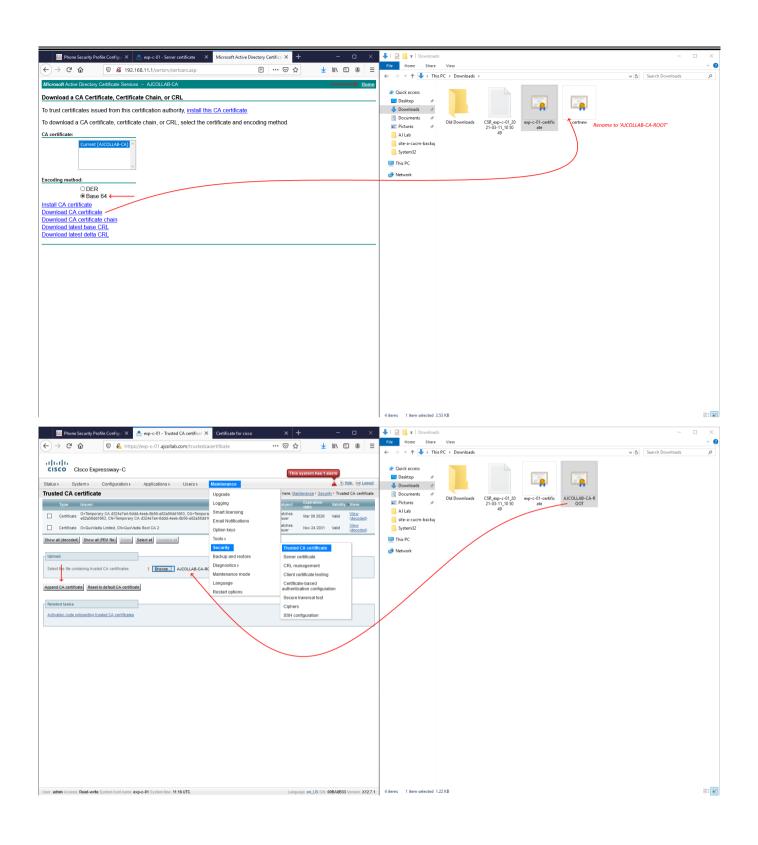
For more information about Active Directory Certificate Services, see Active Directory Certificate Services Documentation.

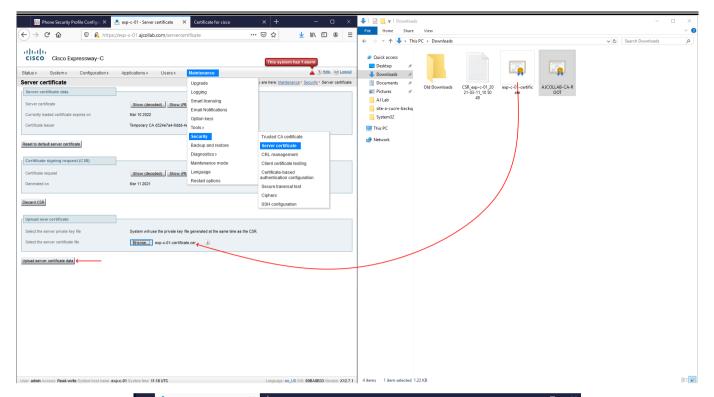
Select a task:

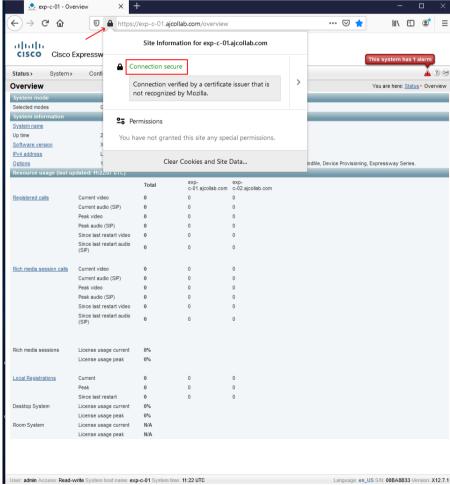
Request a certificate

View the status of a pending certificate request

Download a CA certificate, certificate chain, or CRL +

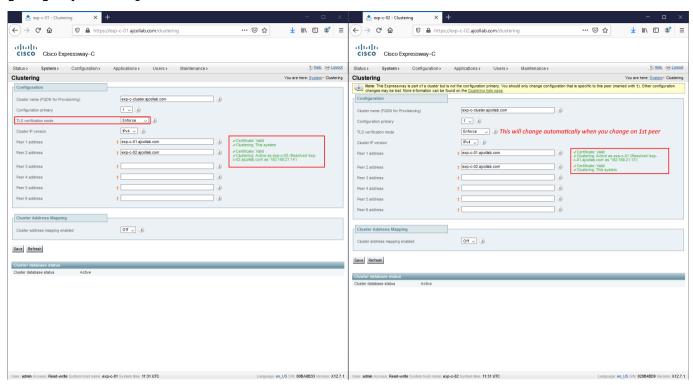




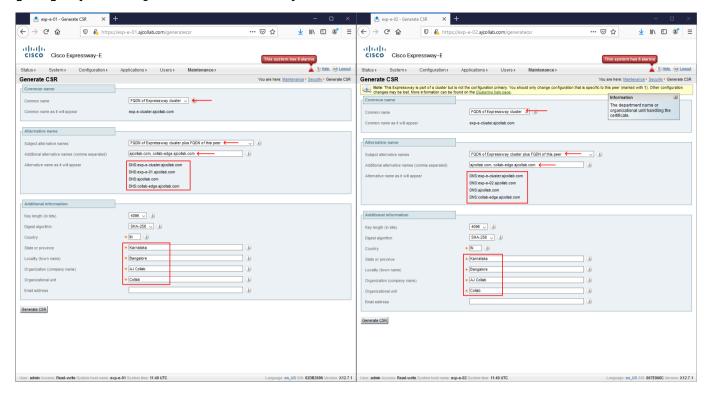


- Perform the same steps to upload certificate on another Expressway Peer
- Once the certificates are uploaded in all the peers of the cluster, reboot and then we can set the cluster 'TLS verification mode' to 'Enforce'

[Lab] Expressway - C Cluster to TLS Enforce

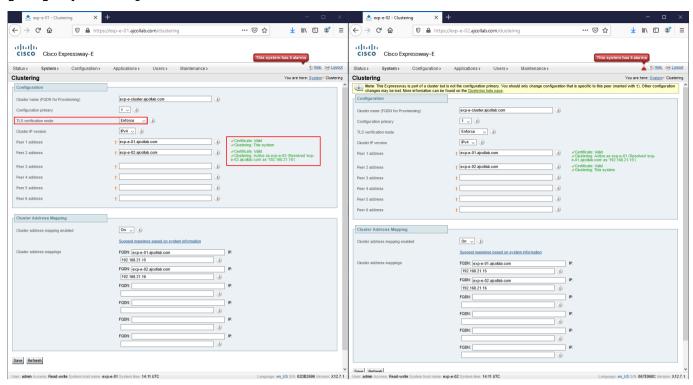


[Lab] Expressway - E Certificate Requirement for MRA



- Expressway E certificate is signed by Public CA like GoDaddy, DigiCert, etc. not internal Enterprise
 CA
- Since it is bit costly, I have used my internal CA to sign the certificate for the lab. I followed same steps that we did for Expressway C (You will get a certificate warning on Jabber while login)
- In production design, always get the Expressway E certificate signed by public CA

[Lab] Expressway - E Cluster to TLS Enforce

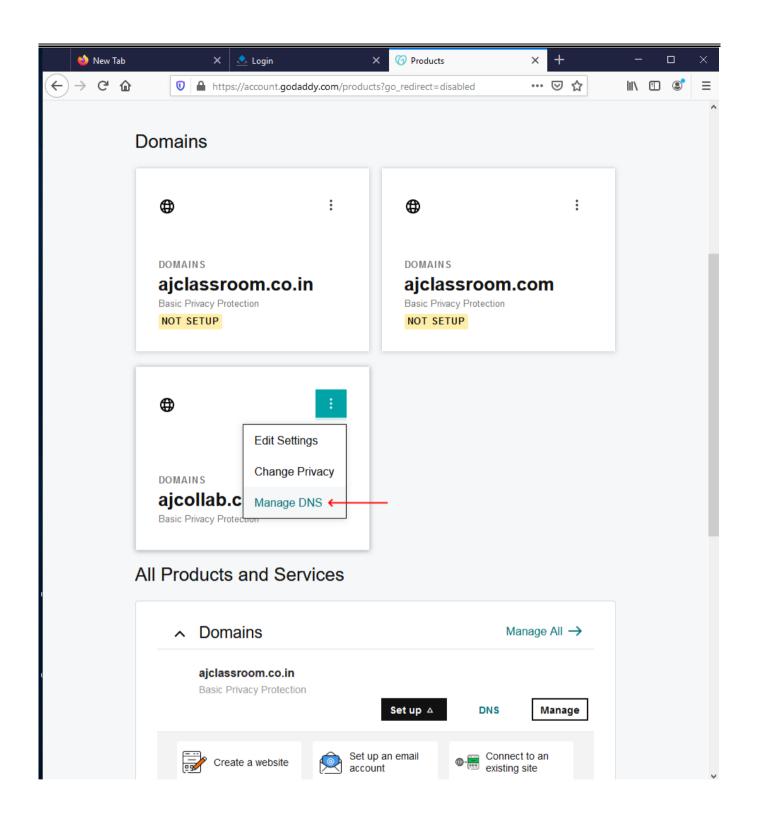


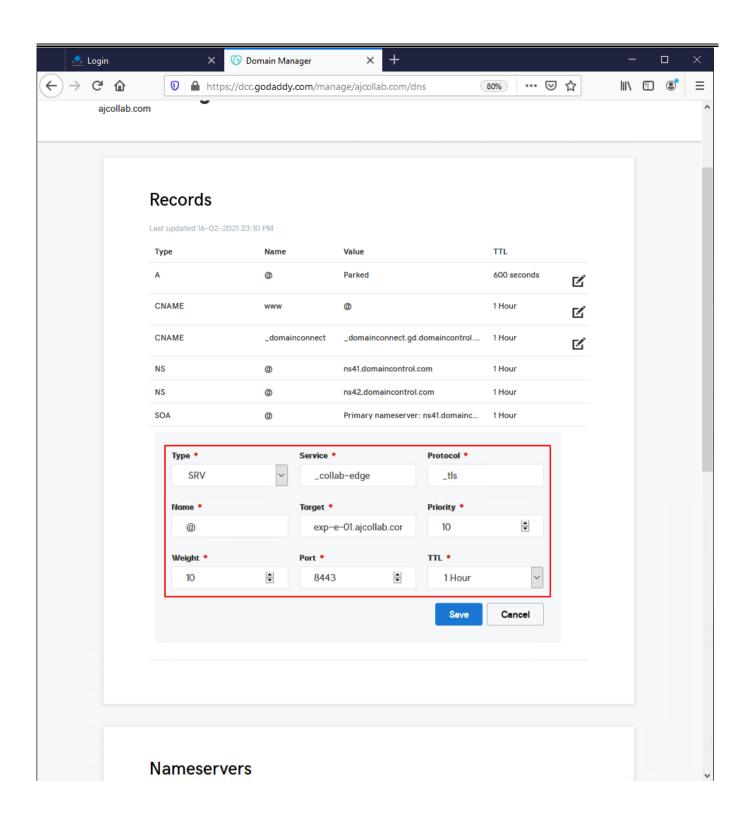
[Lab] Configure MRA (Mobile Remote Access)

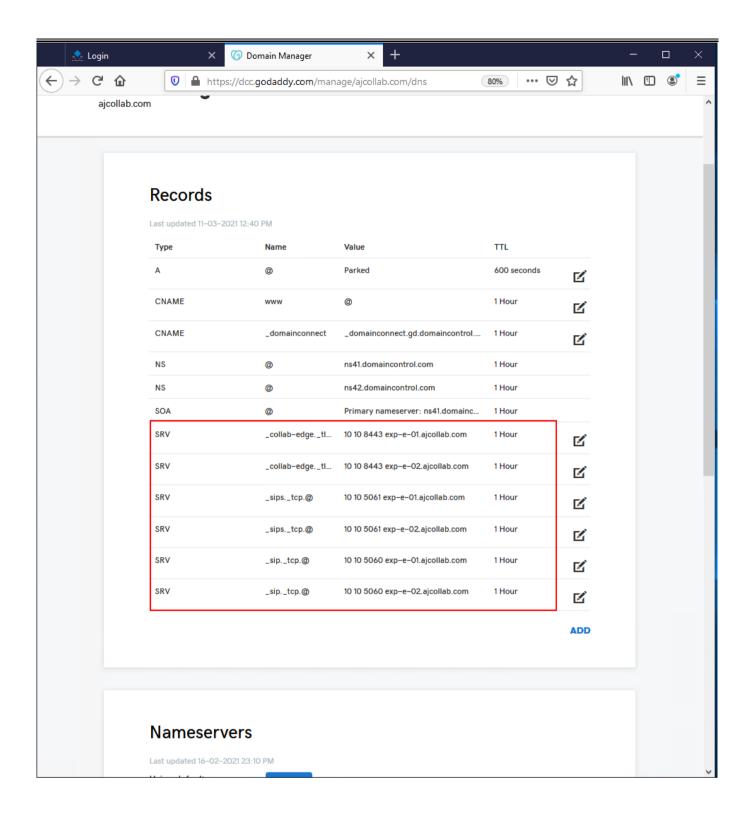
- External DNS SRV Records must be configured on the public DNS domain provider
- I have purchased 'ajcollab.com' from GoDaddy, let's add below records GoDaddy
- Assume that the Expressway E FQDN resolves to the public IP but here we use private IP since in the lab I don't have public IP assigned to Expressways

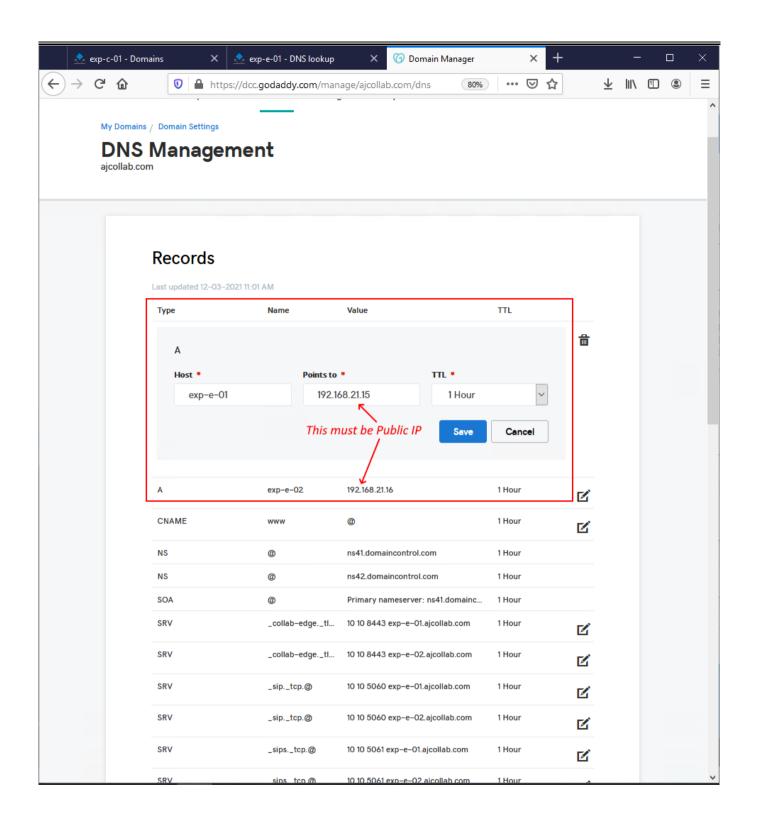
Service	Protocol	Name	Target Host	Priority	Weight	Port
_collab-edge	_tls	@	exp-e-01.ajcollab.com	10	10	8443
_collab-edge	_tls	@	exp-e-02.ajcollab.com	10	10	8443
_sips	_tcp	@	exp-e-01.ajcollab.com	10	10	5061
_sips	_tcp	@	exp-e-02.ajcollab.com	10	10	5061
_sip	_tcp	@	exp-e-01.ajcollab.com	10	10	5060
_sip	_tcp	@	exp-e-02.ajcollab.com	10	10	5060

A Record	IP Address
exp-e-01.ajcollab.com	192.168.21.15
exp-e-02.ajcollab.com	192.168.21.16
exp-c-cluster.ajcollab.com	192.168.21.15
exp-c-cluster.aicollab.com	192.168.21.16





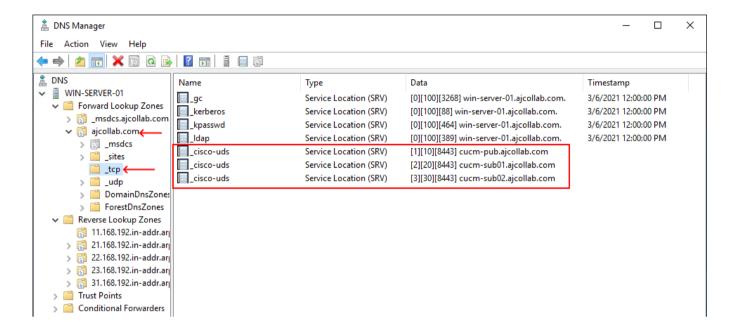




```
Command Prompt - nslookup
Microsoft Windows [Version 10.0.19041.508]
(c) 2020 Microsoft Corporation. All rights reserved.
C:\Users\User>nslookup •
Default Server: UnKnown
Address: 192.168.0.1
 set type=srv
 _collab-edge._tls.ajcollab.com 	
Server: UnKnown
Address: 192.168.0.1
Non-authoritative answer:
_collab-edge._tls.ajcollab.com SRV service location:
priority = 10
          weight
                          = 10
                          = 8443
          port
                          = blr-exp-e-02.ajcollab.com
          svr hostname
collab-edge._tls.ajcollab.com SRV service location:
          priority
                          = 10
          weight
                          = 10
          port
                          = 8443
          svr hostname
                          = blr-exp-e-01.ajcollab.com <
ajcollab.com
                nameserver = ns41.domaincontrol.com
ajcollab.com
                nameserver = ns42.domaincontrol.com
bĺr-exp-e-01.ajcollab.com
                                  internet address = 192.168.21.15
internet address = 192.168.21.16
blr-exp-e-02.ajcollab.com
```

• Internal DNS Records are already configured for Jabber On-Premise Registration lab

Service	Protocol	Priority	Weight	Port	Target Host
_cisco-uds	_tcp	1	10	8443	cucm-pub.ajcollab.com
_cisco-uds	_tcp	2	20	8443	cucm-sub01.ajcollab.com
_cisco-uds	_tcp	3	30	8443	cucm-sub02.ajcollab.com

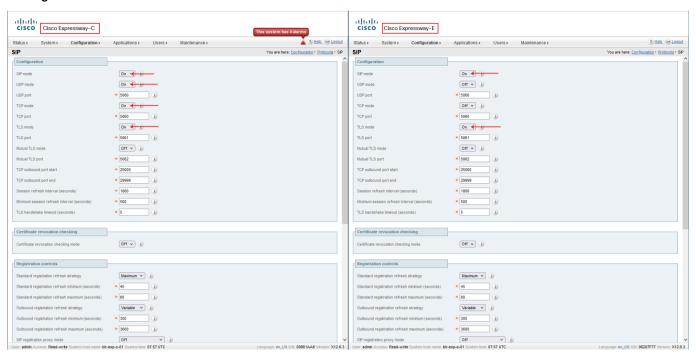


```
Administrator: Command Prompt
Microsoft Windows [Version 10.0.17763.737]
(c) 2018 Microsoft Corporation. All rights reserved.
C:\Users\Administrator>nslookup
Default Server: win-server-01.ajcollab.com
Address: 192.168.11.1
 set type=srv←——
_cisco-uds._tcp.ajcollab.com
Server: win-server-01.ajcollab.com
Address: 192.168.11.1
cisco-uds._tcp.ajcollab.com
                               SRV service location:
         priority
          weight
                        = 10
         port
                        = 8443
          svr hostname
                        = cucm-pub.ajcollab.com
cisco-uds._tcp.ajcollab.com
                               SRV service location:
         priority
          weight
                        = 20
         port
                        = 8443
          svr hostname = cucm-sub01.ajcollab.com
cisco-uds._tcp.ajcollab.com
                               SRV service location:
         priority
          weight
                        = 30
         port
                        = 8443
                       = cucm-sub02.ajcollab.com 	
          svr hostname
cucm-sub01.ajcollab.com internet address = 192.168.21.2
cucm-sub02.ajcollab.com internet address = 192.168.21.3
C:\Users\Administrator>
C:\Users\Administrator>nslookup cucm-pub.ajcollab.com
Server: win-server-01.ajcollab.com
Address: 192.168.11.1
        cucm-pub.ajcollab.com
Address: 192.168.21.1
C:\Users\Administrator>
C:\Users\Administrator>nslookup cucm-sub01.ajcollab.com
Server: win-server-01.ajcollab.com
Address: 192.168.11.1
        cucm-sub01.ajcollab.com
Address: 192.168.21.2
C:\Users\Administrator>
C:\Users\Administrator>nslookup cucm-sub02.ajcollab.com
Server: win-server-01.ajcollab.com
Address: 192.168.11.1
        cucm-sub02.ajcollab.com
Address: 192.168.21.3
:\Users\Administrator>_
```

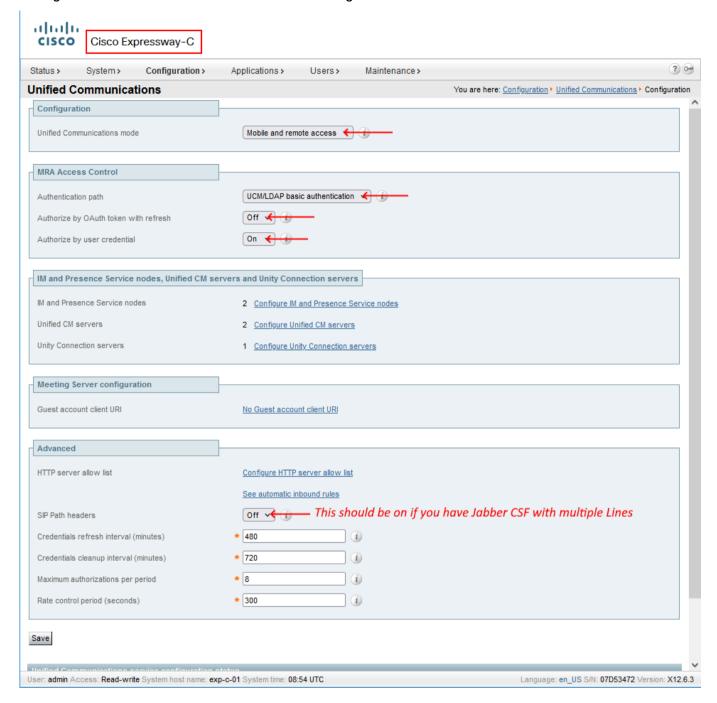
- Make sure internal Jabber Login works fine
- CUCM Cluster nodes, CUC Nodes and Expressway C should be signed with internal Enterprise CA (Certificate cross import is required otherwise)
- CUCM, CUC and Expressway C must have Enterprise CA as Trust Certificate
- Refer CUCM Certificate Section to know more about multi SAN CUCM Certificates
- Expressway E must be signed with Public CA
- Expressway E and C must have Public Root CA as Trust Certificate

-	
	915

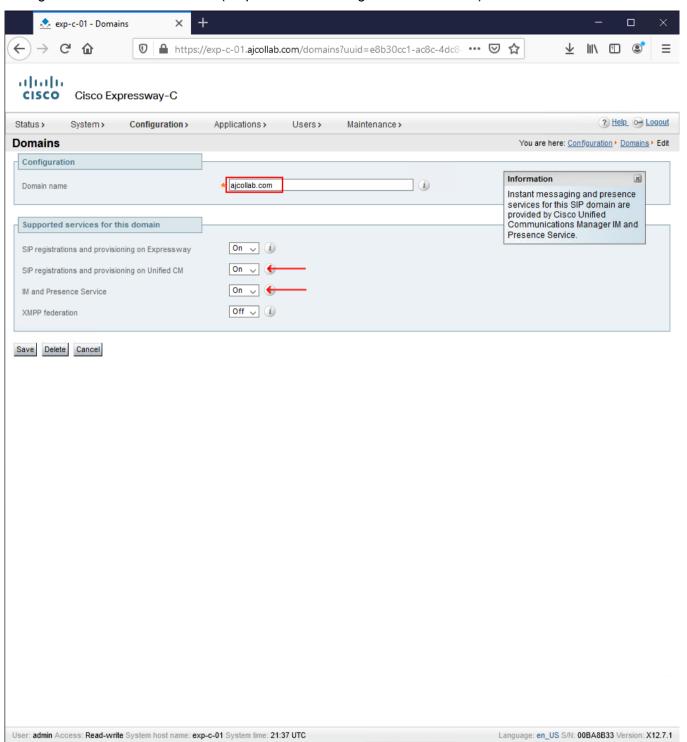
Configuration >> Protocol >> SIP >>



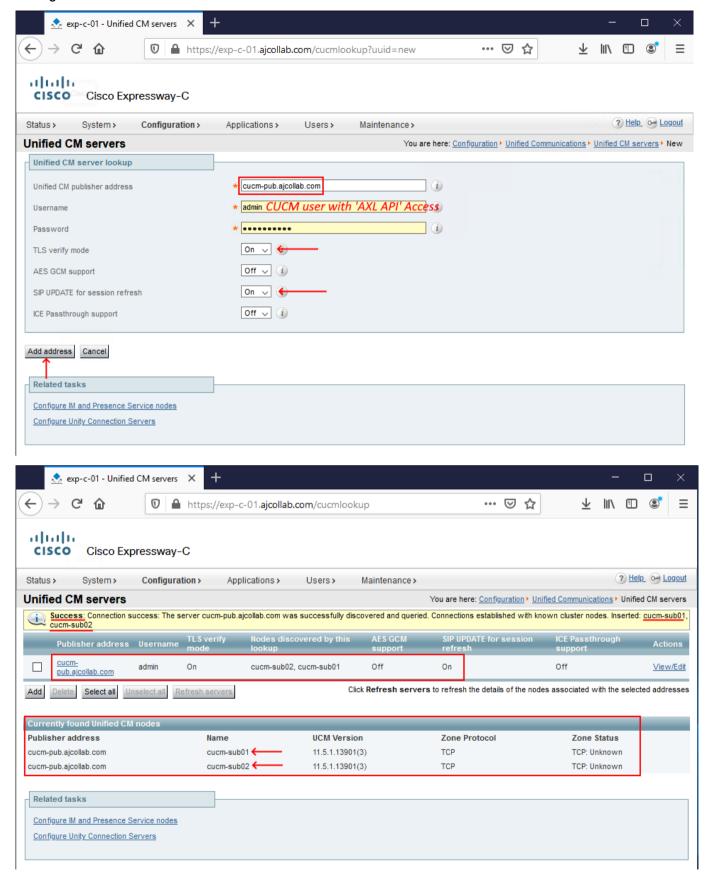
Configuration >> Unified Communications >> Configuration



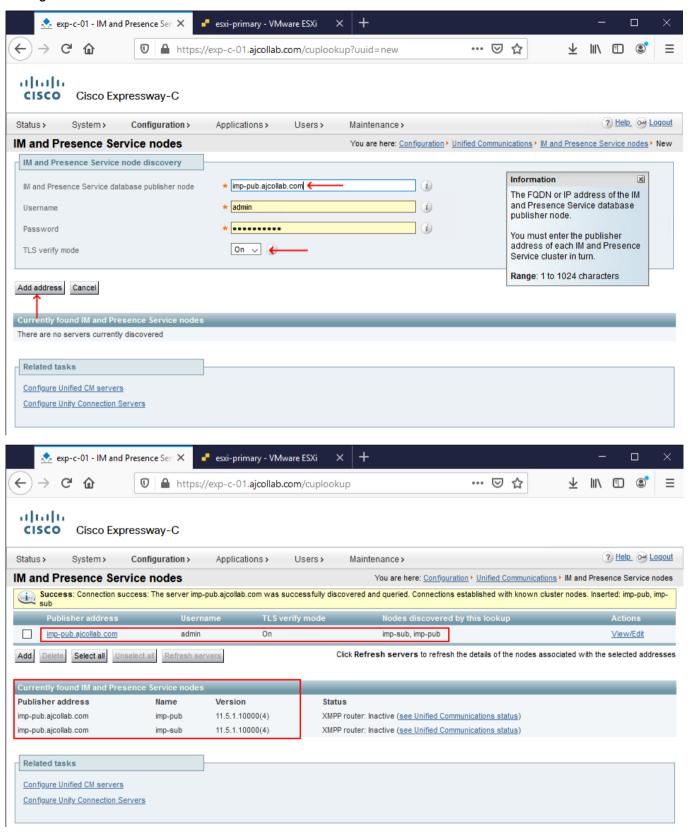
Configuration >> Domain >> New (Helps to route the registration to CUCM)



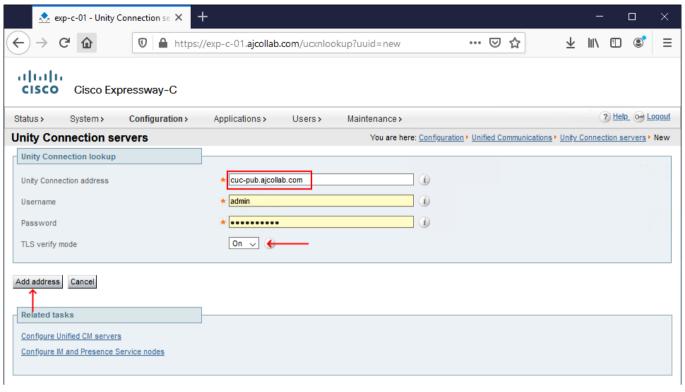
Configuration >> Unified Communications >> Unified CM Servers

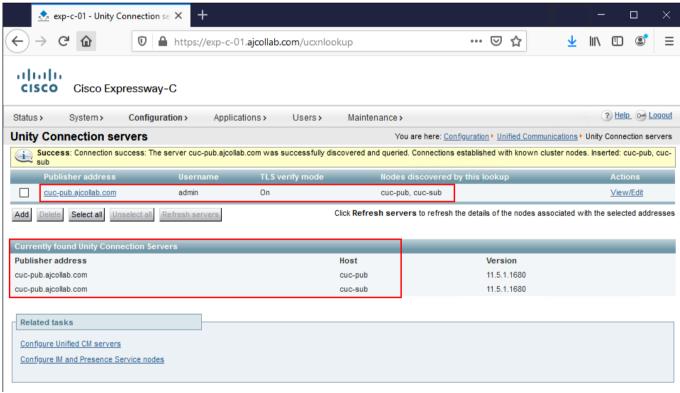


Configuration >> Unified Communications >> IM and Presence Service nodes



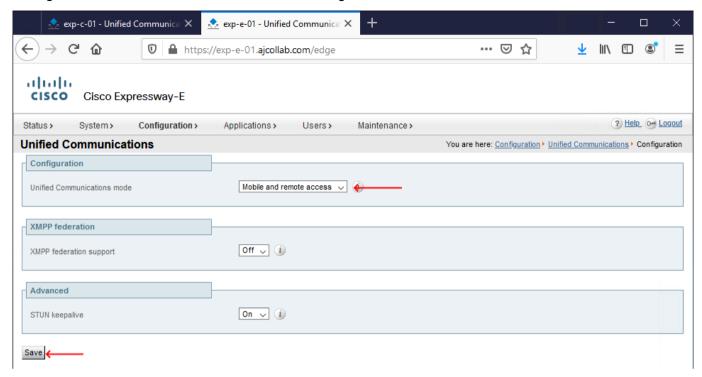
Configuration >> Unified Communications >> Unity Connection servers



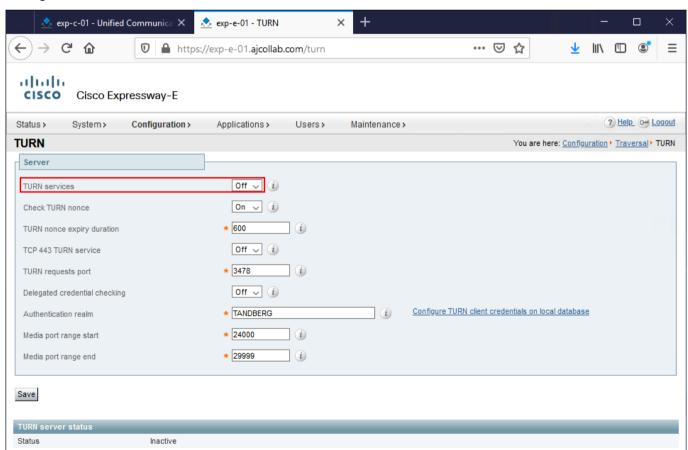


Expressway - E

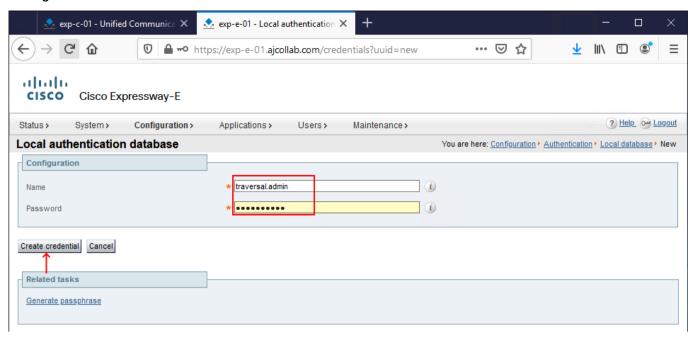
Configuration >> Unified Communications >> Configuration



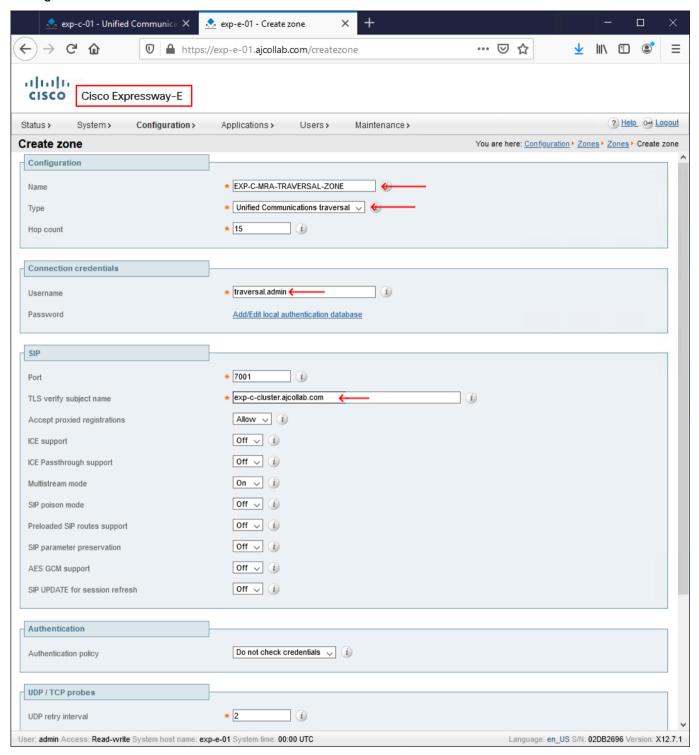
Configuration >> Traversal >> TURN



Configuration >> Authentication >> Devices >> Local database >> New

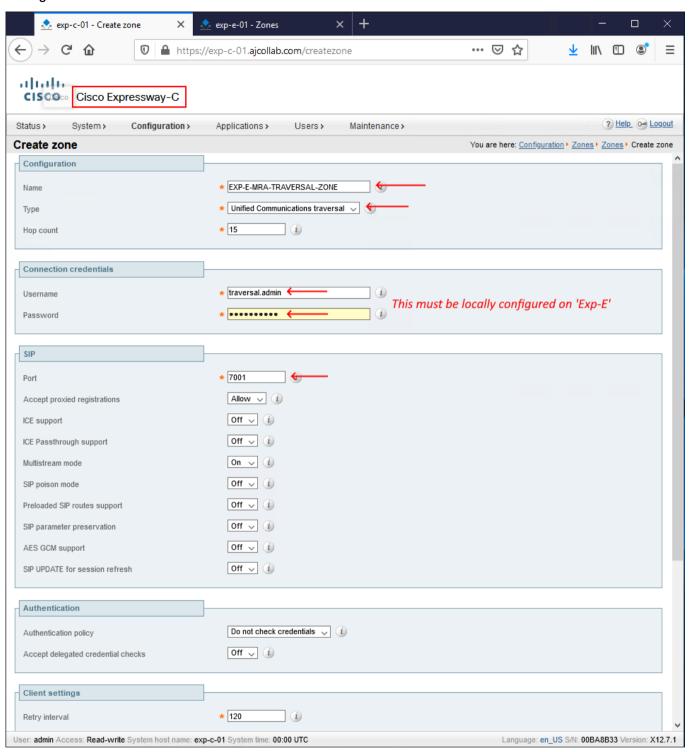


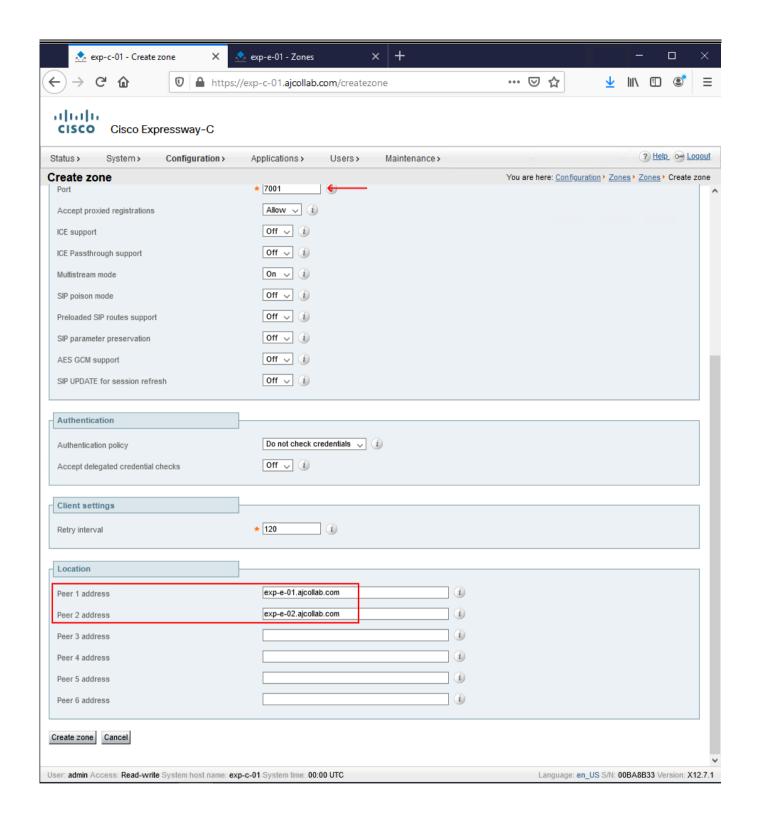
Configuration >> Zones >> Zones >> New

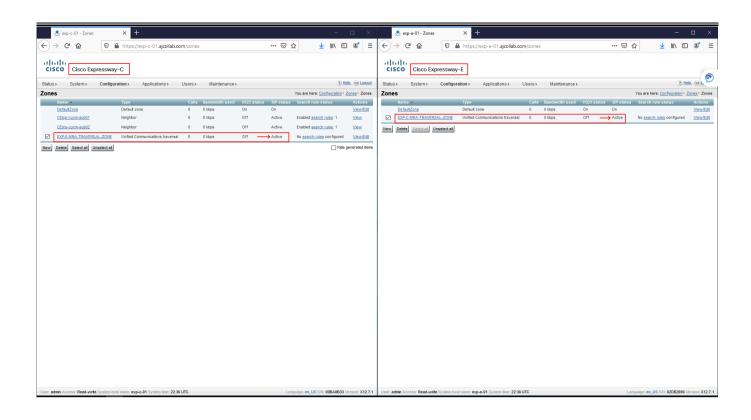


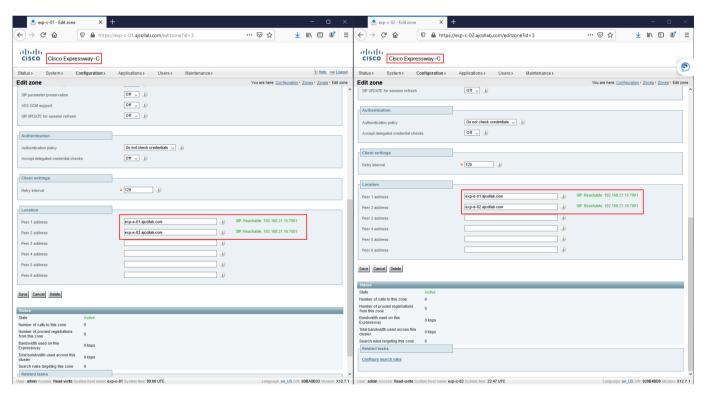
Expressway - C

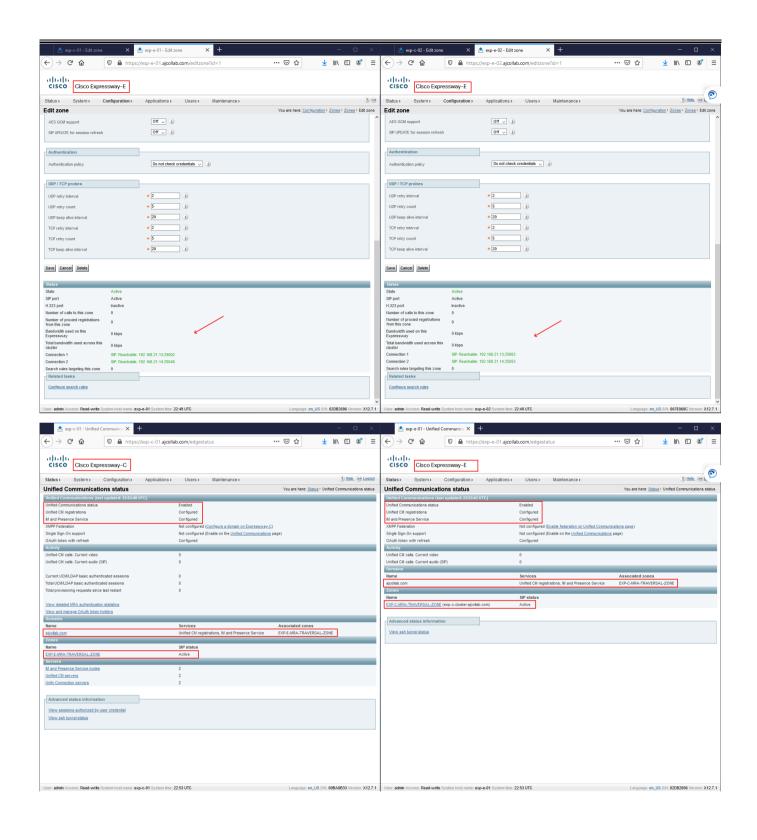
Configurations >> Zones >> Zones >> New

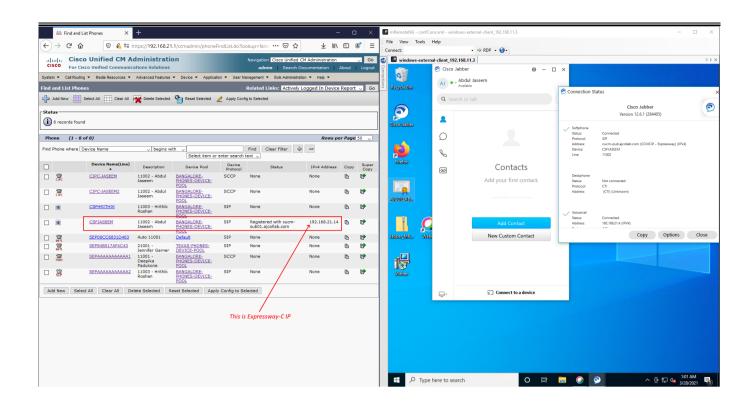






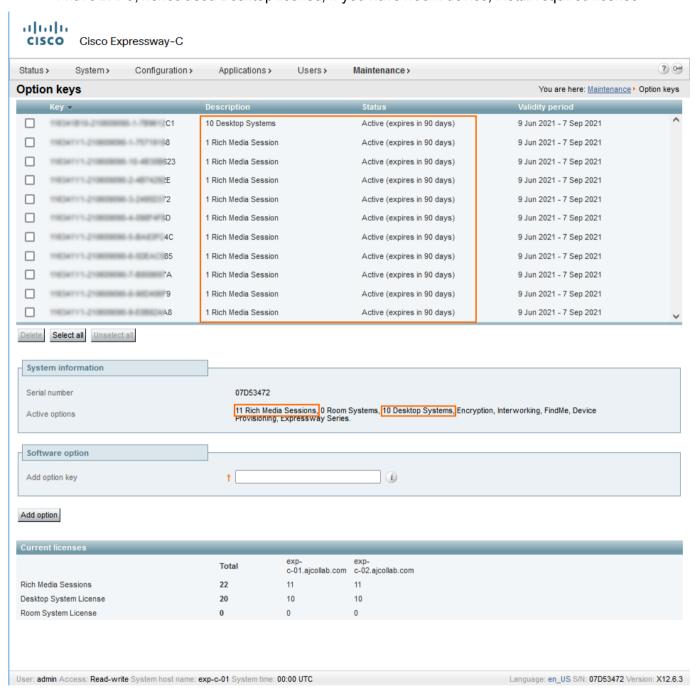




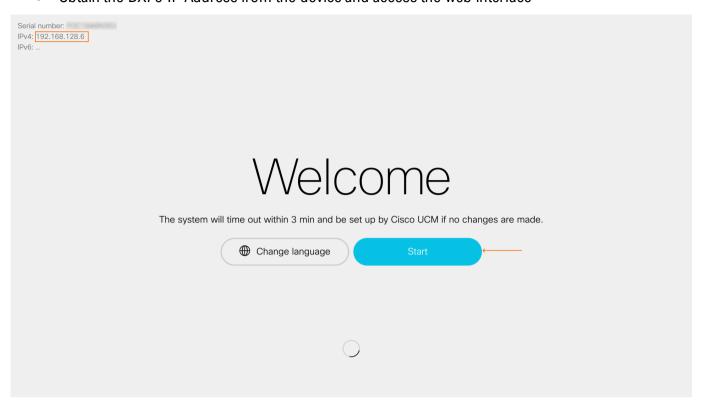


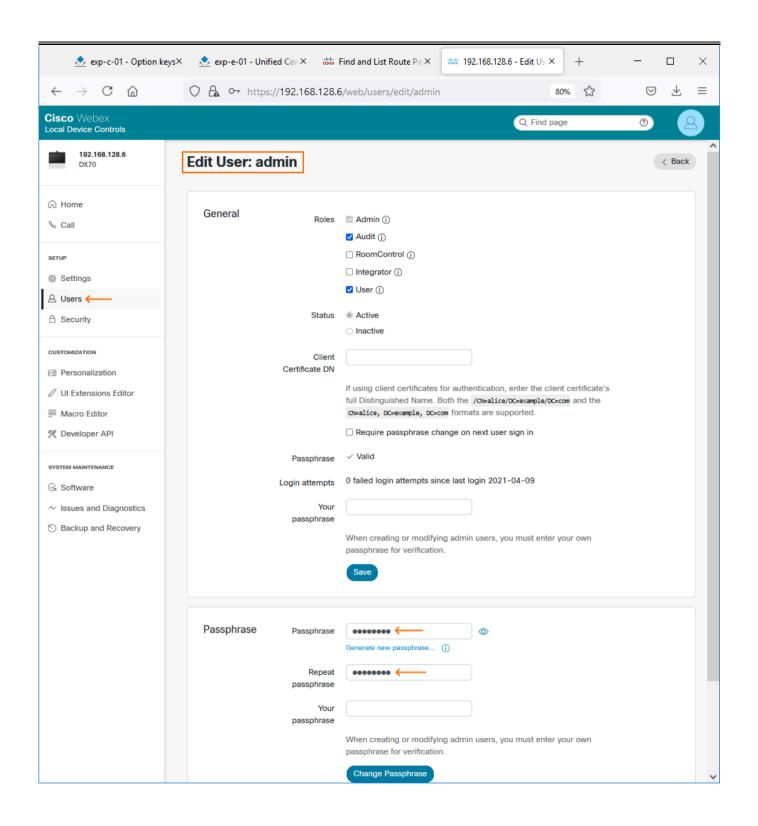
[Lab] Register DX70 in Expressway C

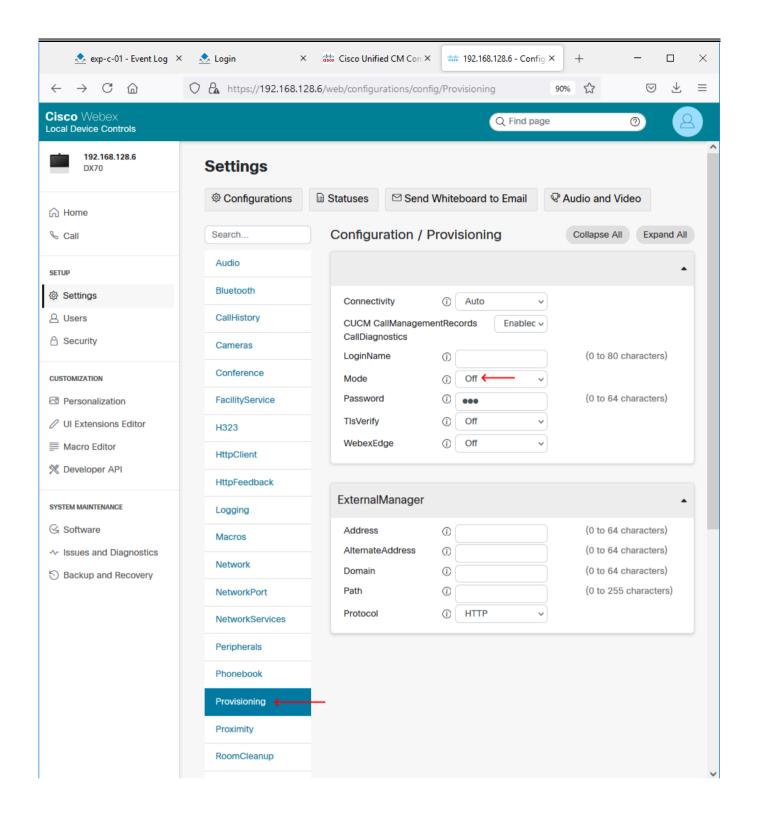
- Make sure you have Desktop System and RMS licenses installed
- I have DX70, hence used Desktop license, if you have Room device, install required license

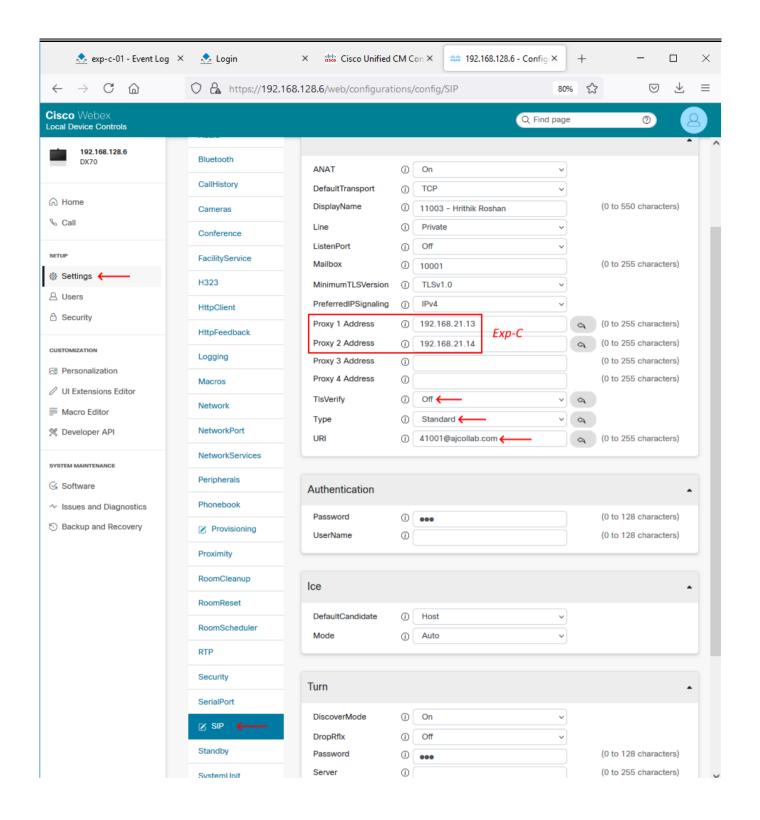


• Obtain the DX70 IP Address from the device and access the web interface

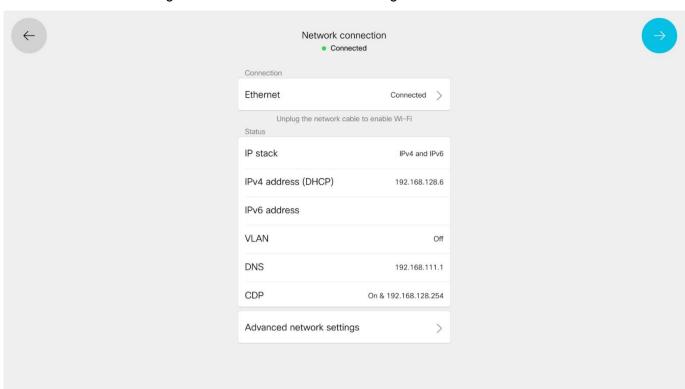


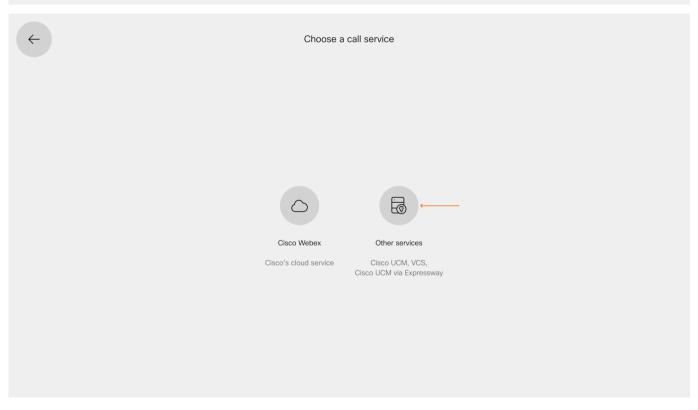






• You can do the registration via the device initial configuration wizard as well



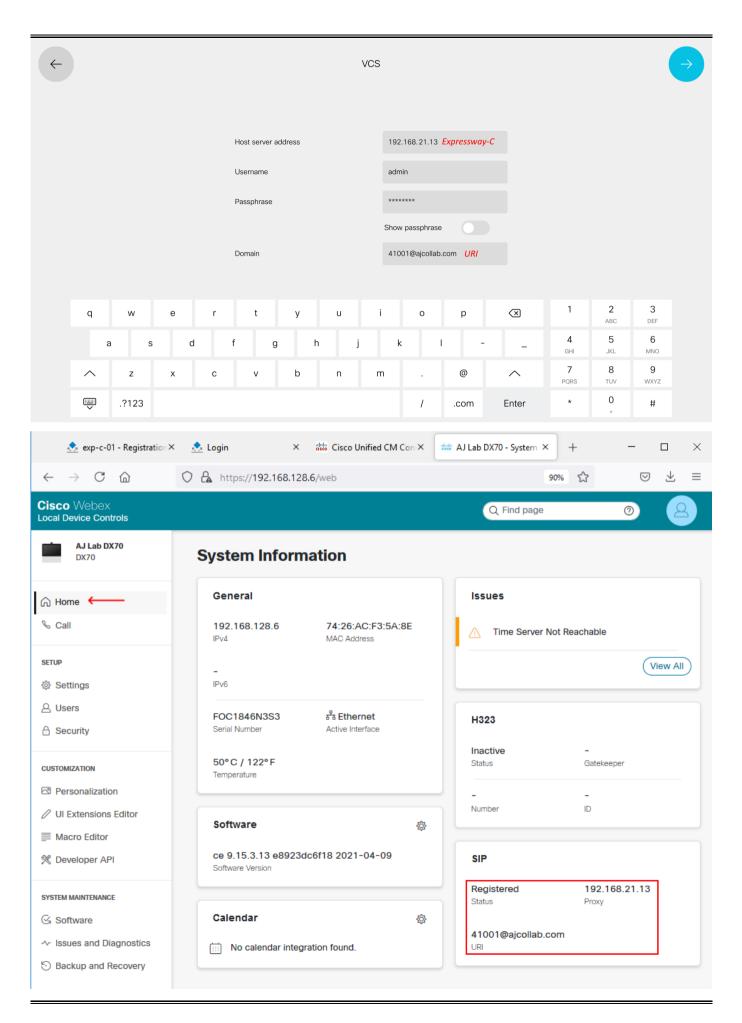


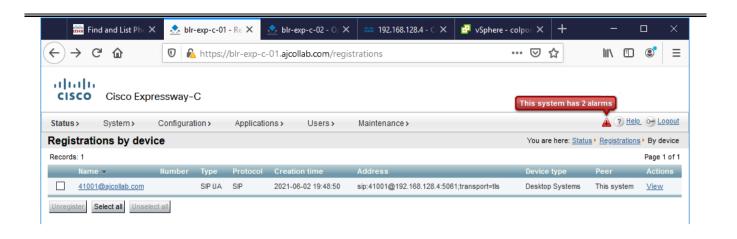
This option comes only if the TP device has TFTP Server details from DHCP

A Cisco UCM service at 192.168.121.1 is available in your network. Continuing will activate this device to that service. Alternatively, choose another service.

Change service -

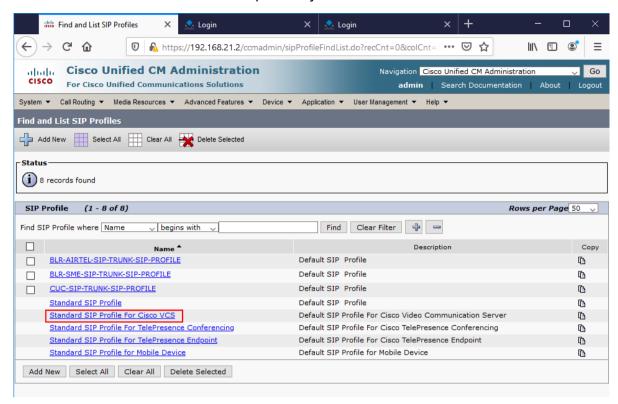


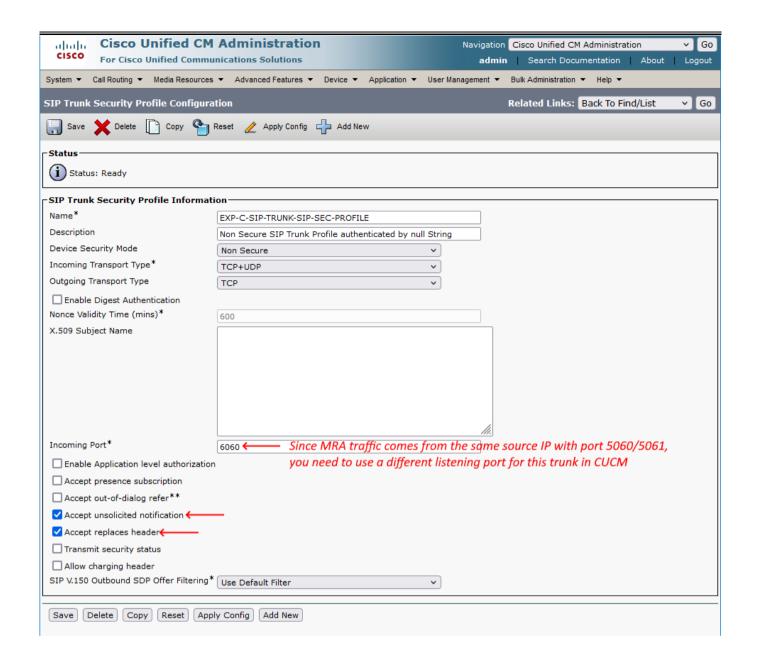


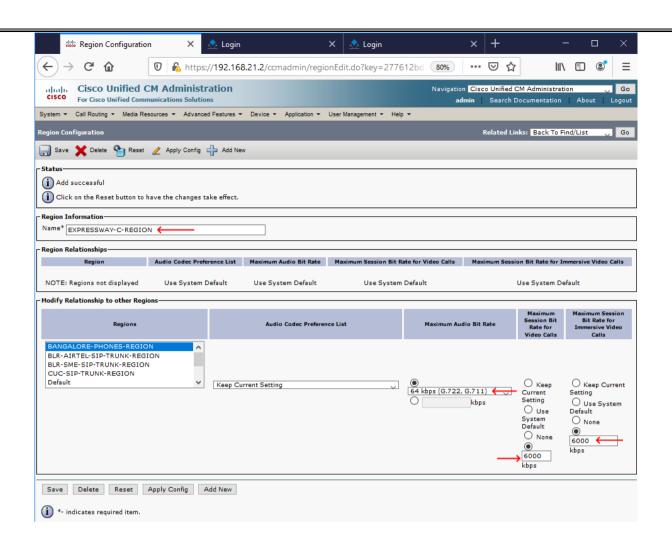


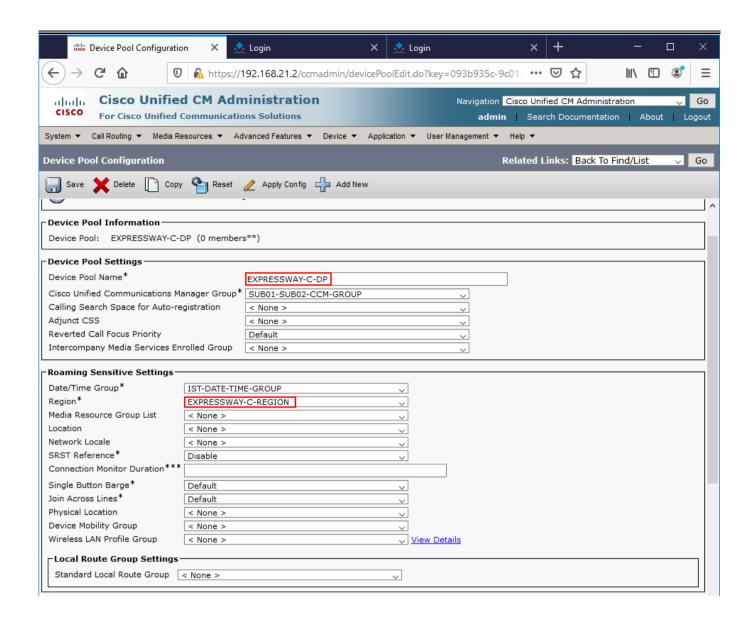
[Lab] CUCM and Expressway C Integration

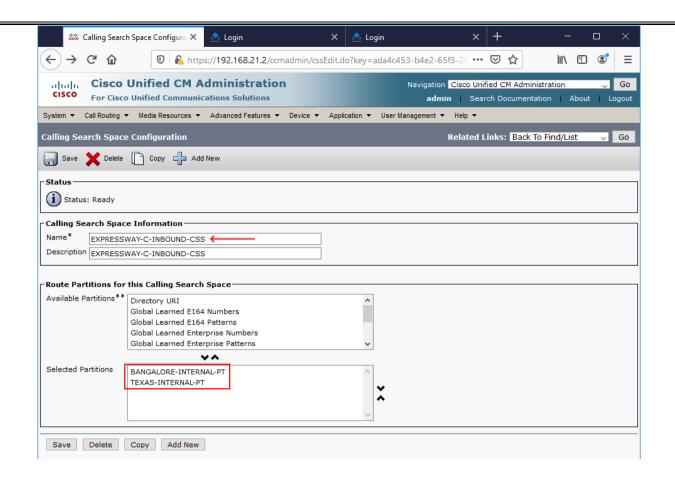
• Create a SIP Trunk from CUCM to Expressway C

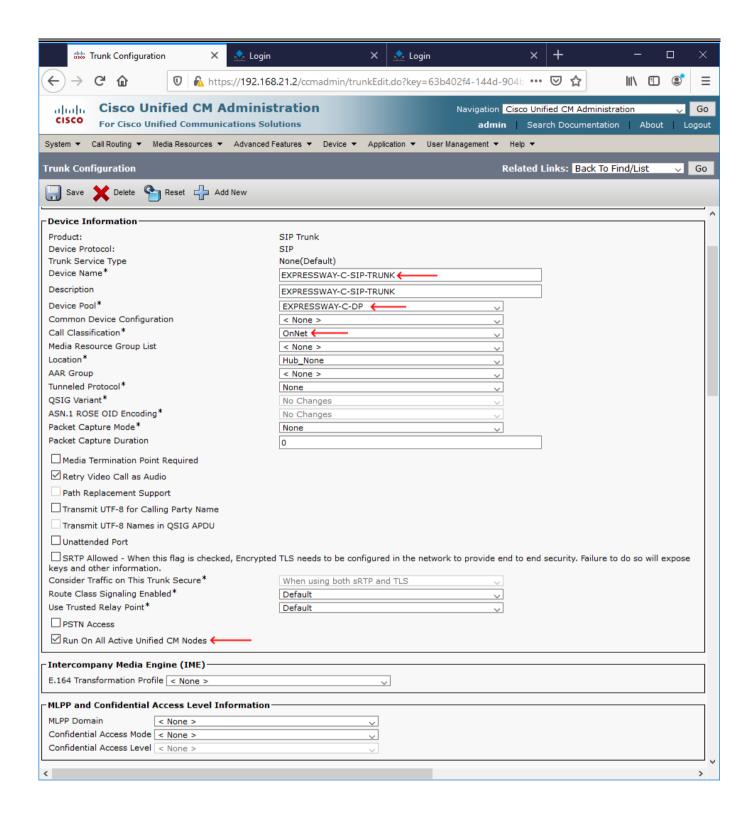


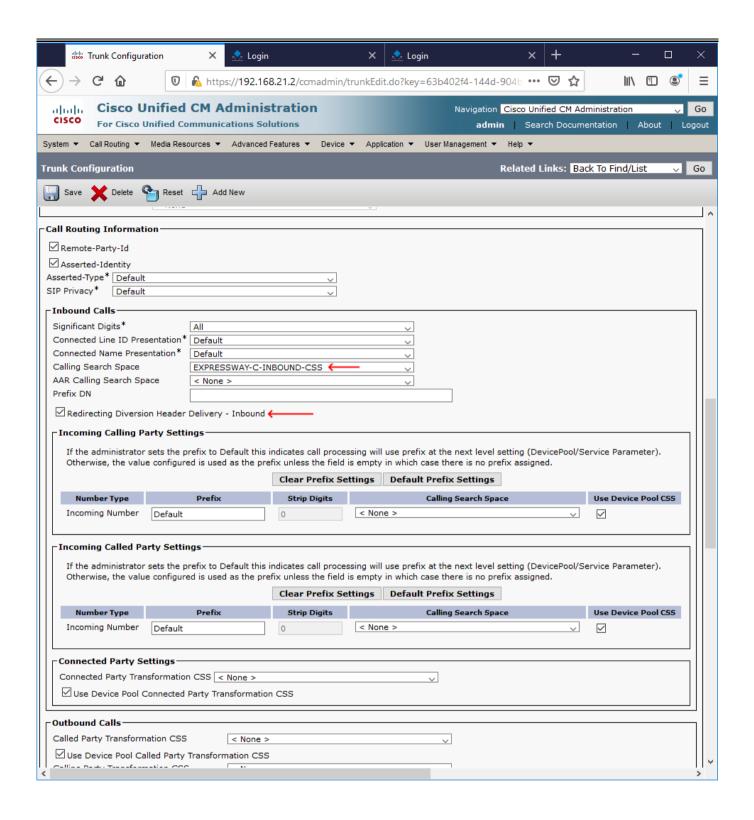


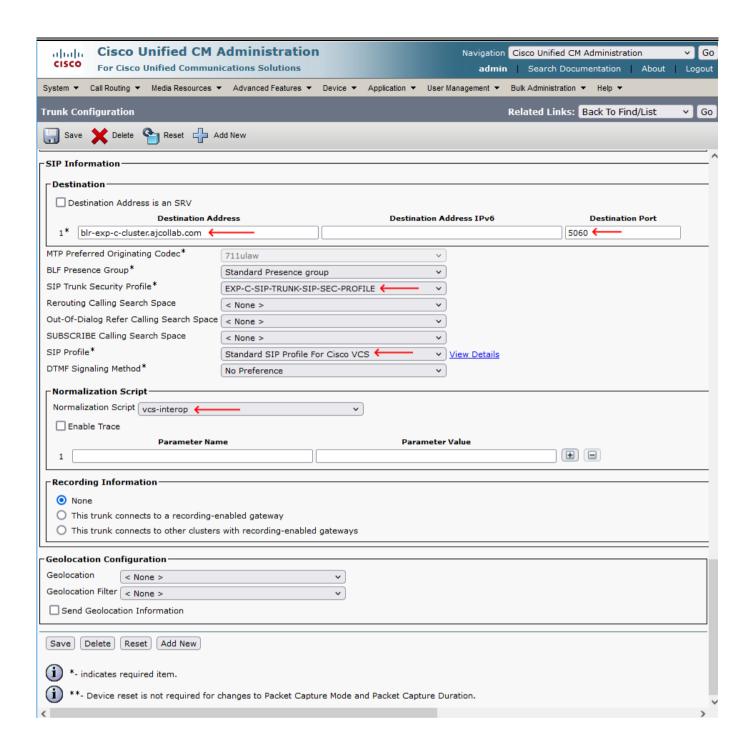


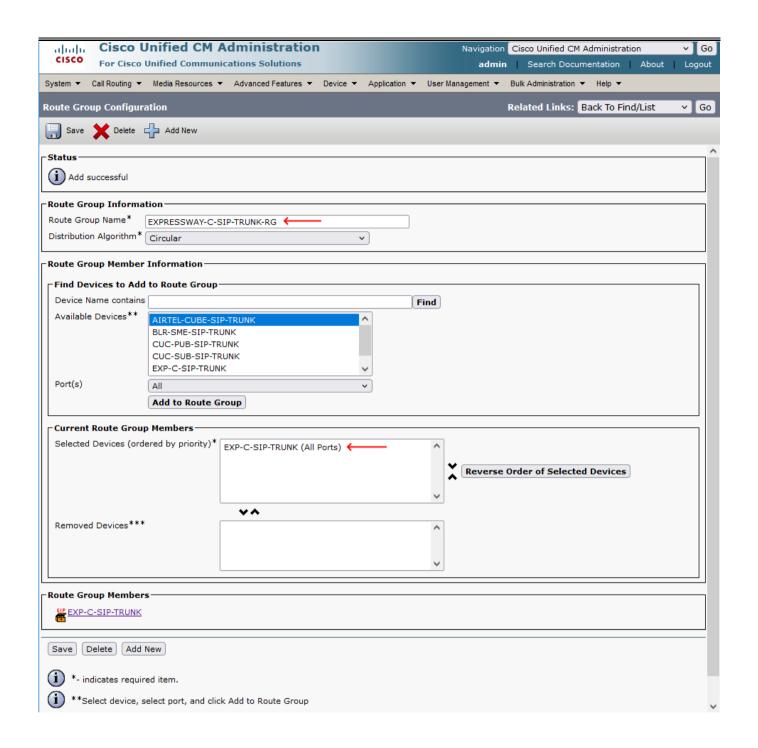


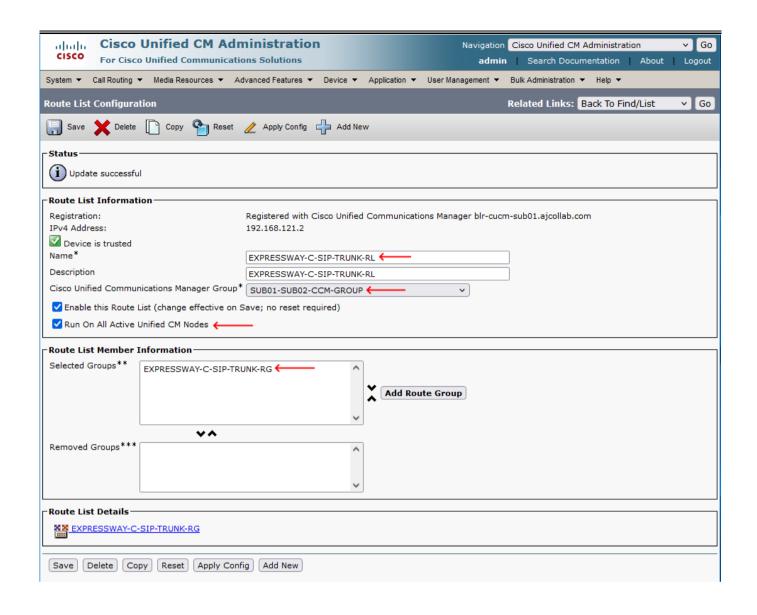


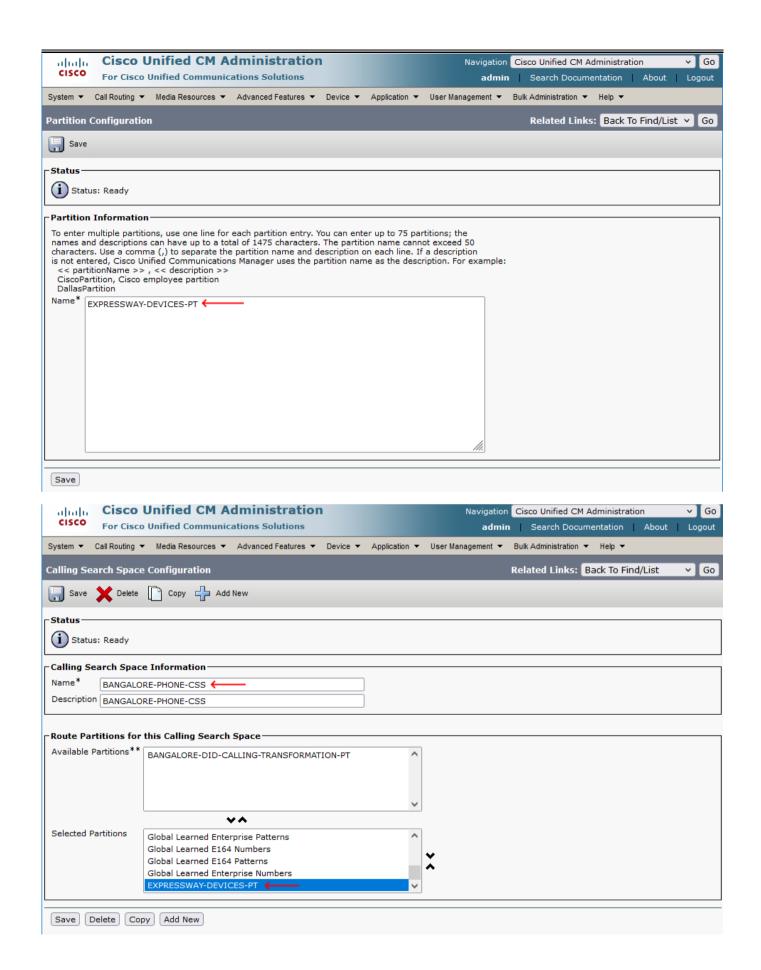


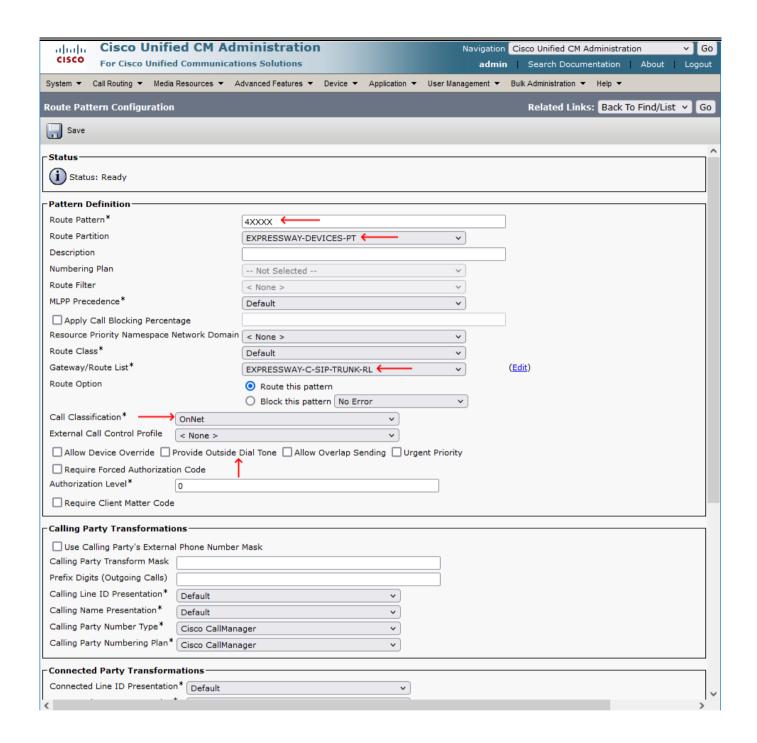




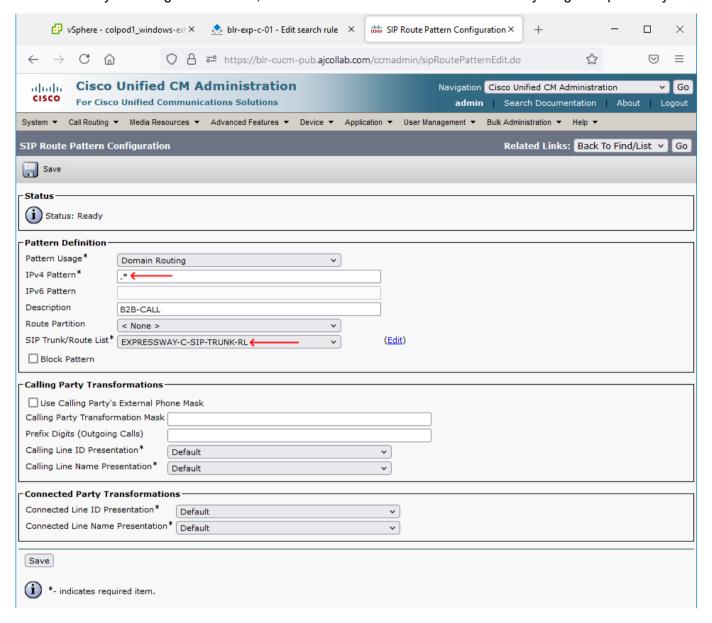


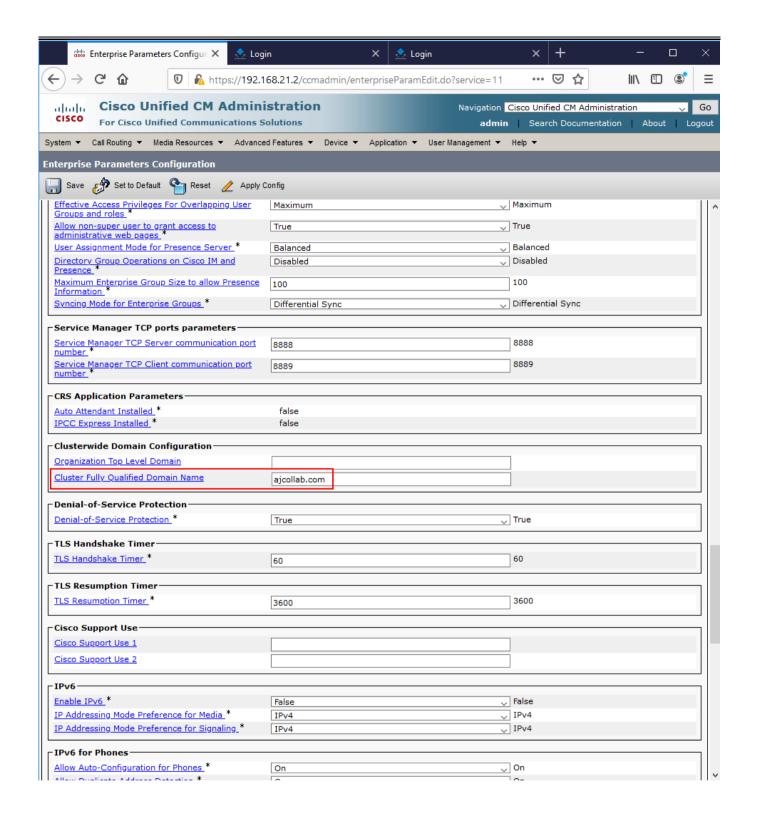




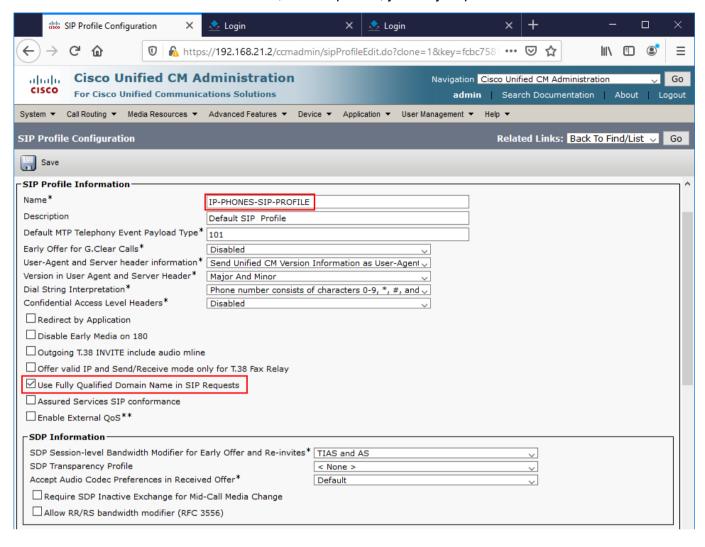


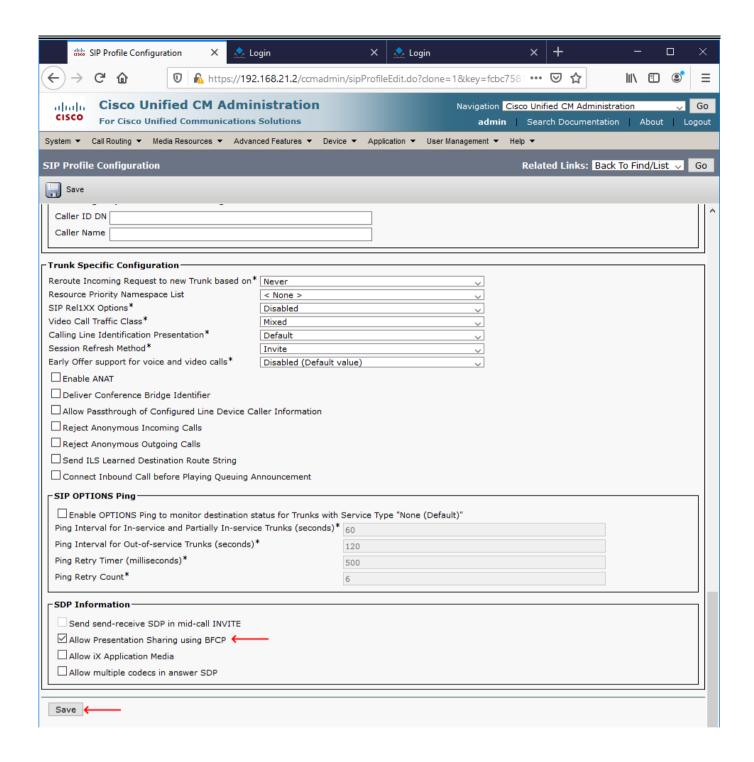
• When you configure B2B Calls, create a SIP Route Pattern that route everything to Expressway C



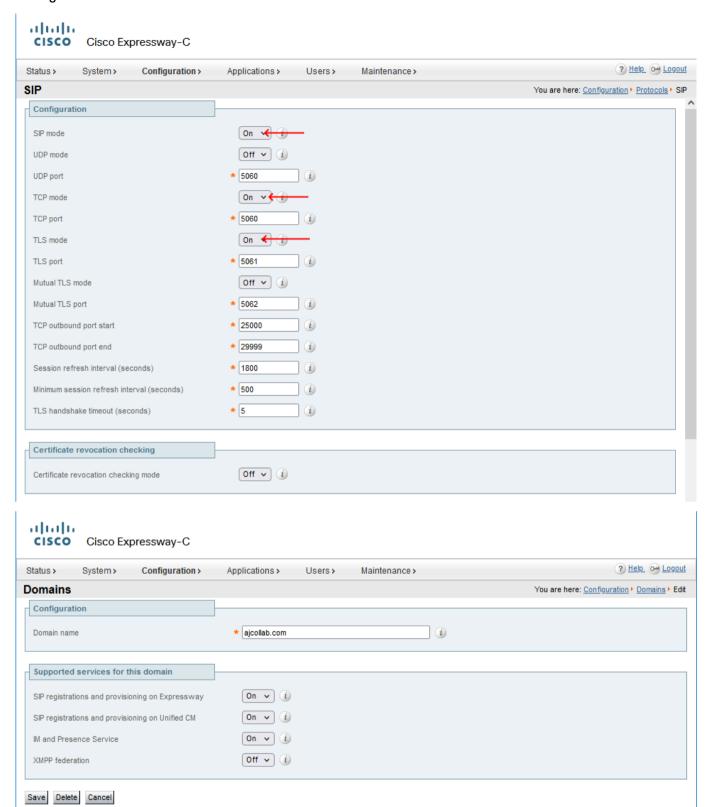


• Create a SIP Profile for IP Phones, this is optional, you may skip this

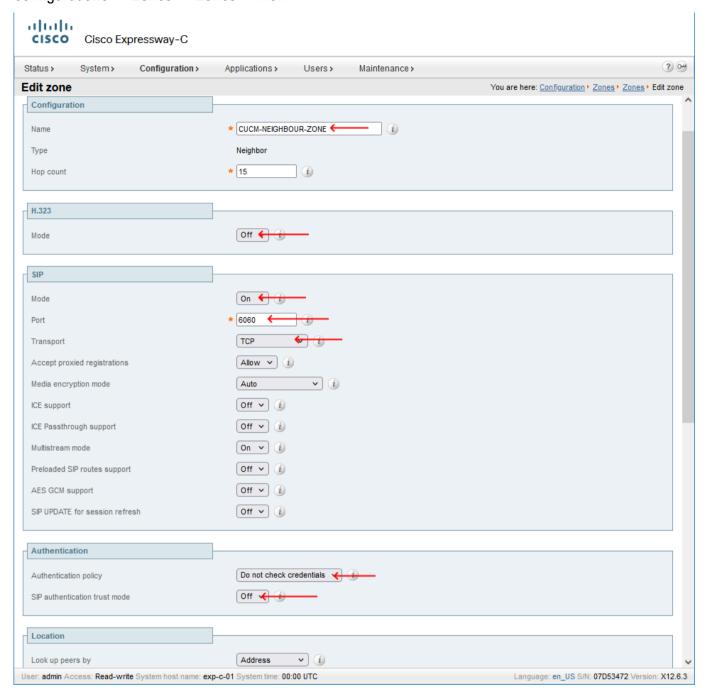


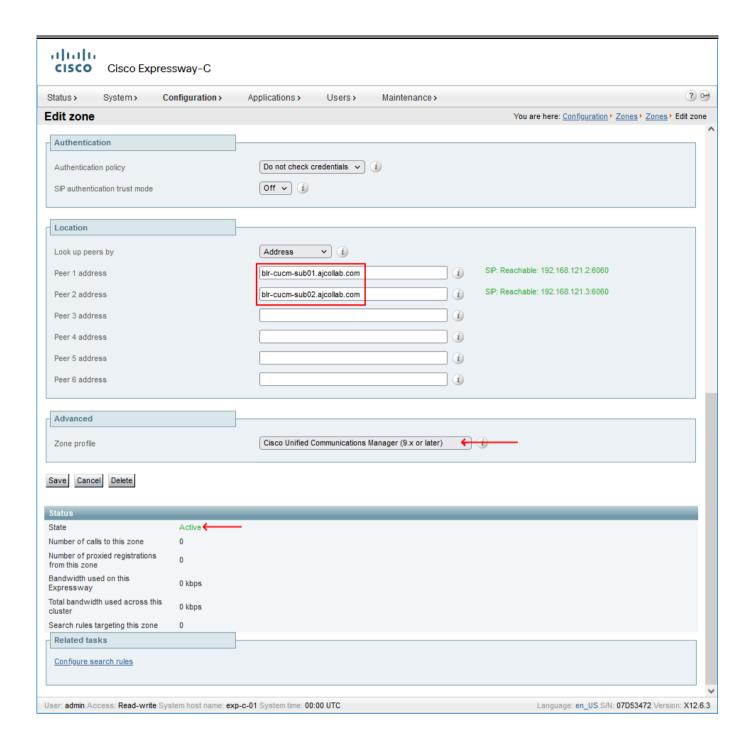


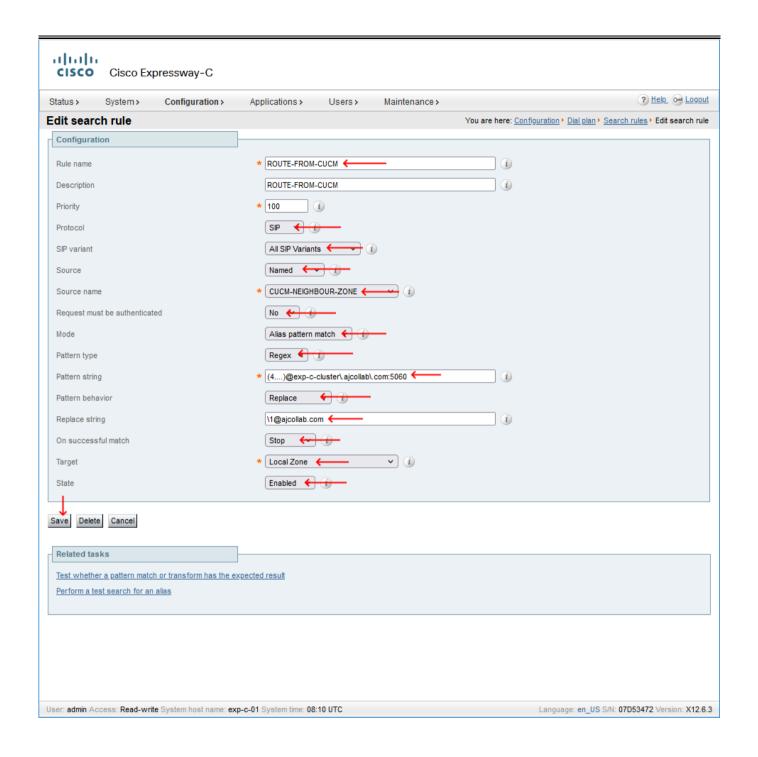
Configuration >> Protocols >> SIP

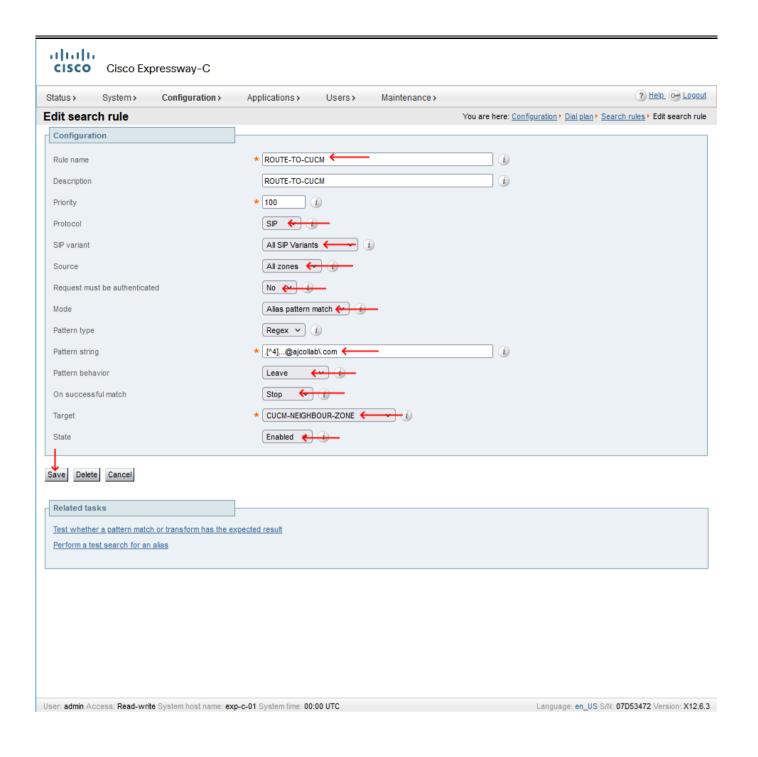


Configurations >> Zones >> Zones >> New

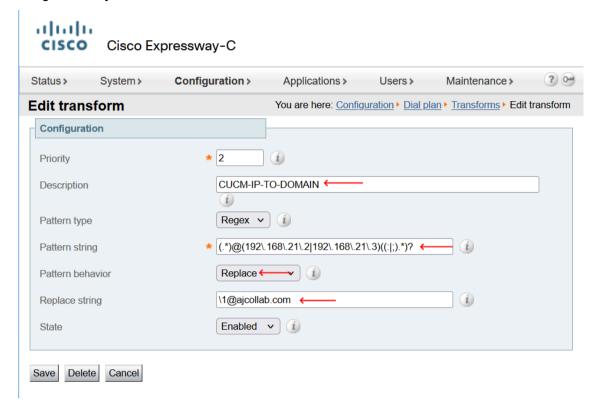




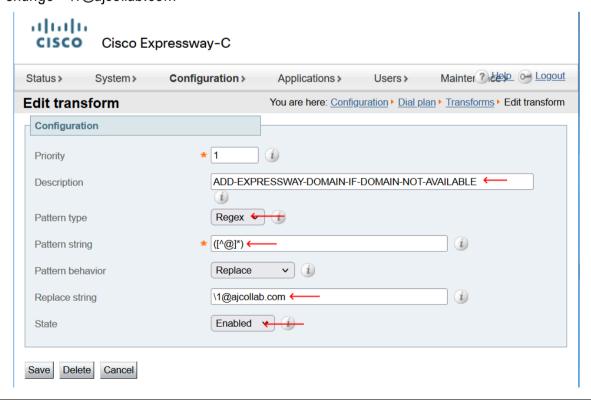


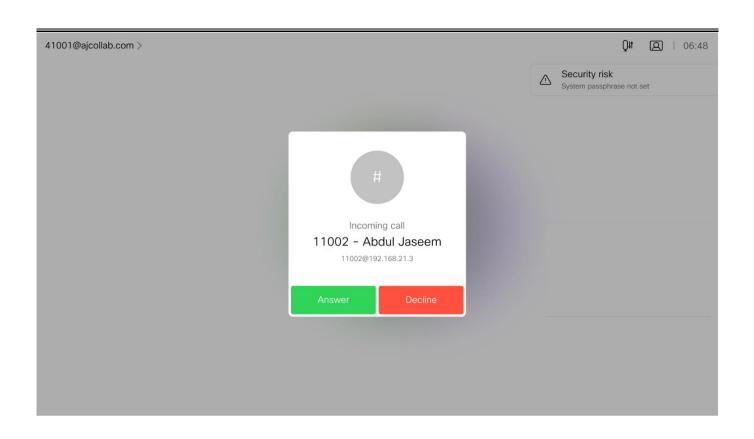


- Change CUCM IPs to @DOMAIN format
- Match = (.*)@(192\.168\.21\.2|192\.168\.21\.3)((:|;).*)?
- Change = \1@ajcollab.com



- Add @DOMAIN in every call which doesn't have '@' part
- Match = ([^@]*)
- Change = \1@ajcollab.com





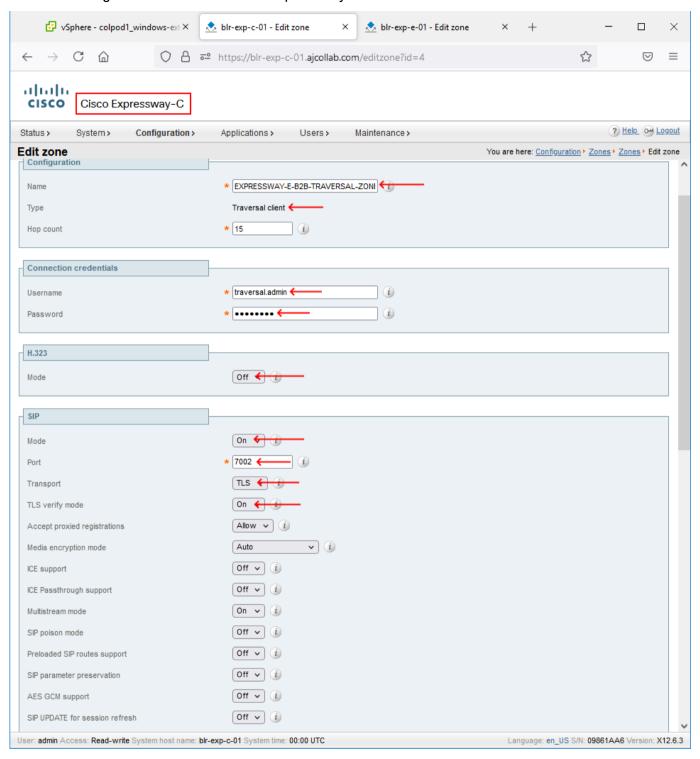
[Lab] B2B (Business to Business) Call Using Expressways

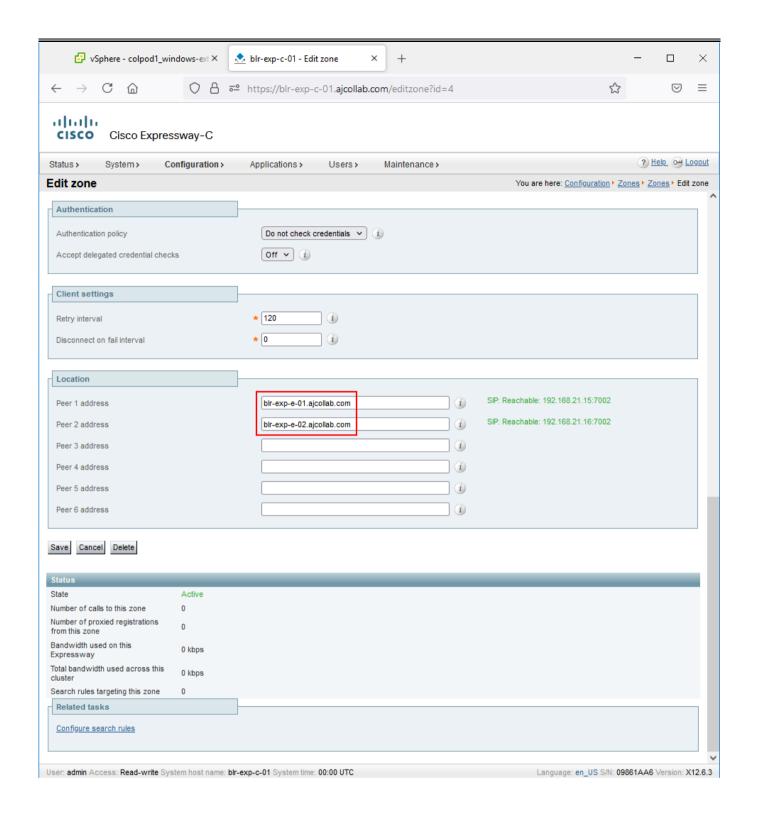


- On CUCM Create SIP Trunk from CUCM to Expressway C (Already created)
- On Expressway-C Create Neighbor Zone from Expressway-C to CUCM and Traversal Client zone from Expressway-C to Expressway E
- Expressway-E Create Traversal Server zone from Expressway-E to Expressway-C and DNS Zone to internet

Expressway C Configuration

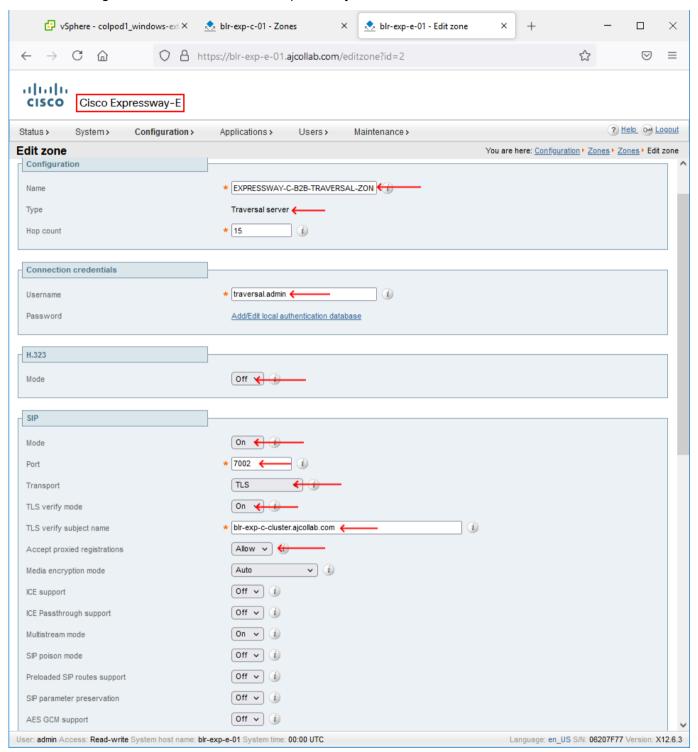
• Creating Traversal Client Zone in Expressway E

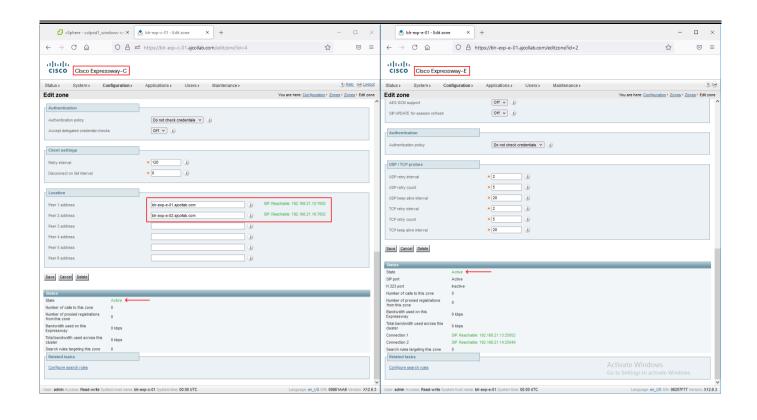




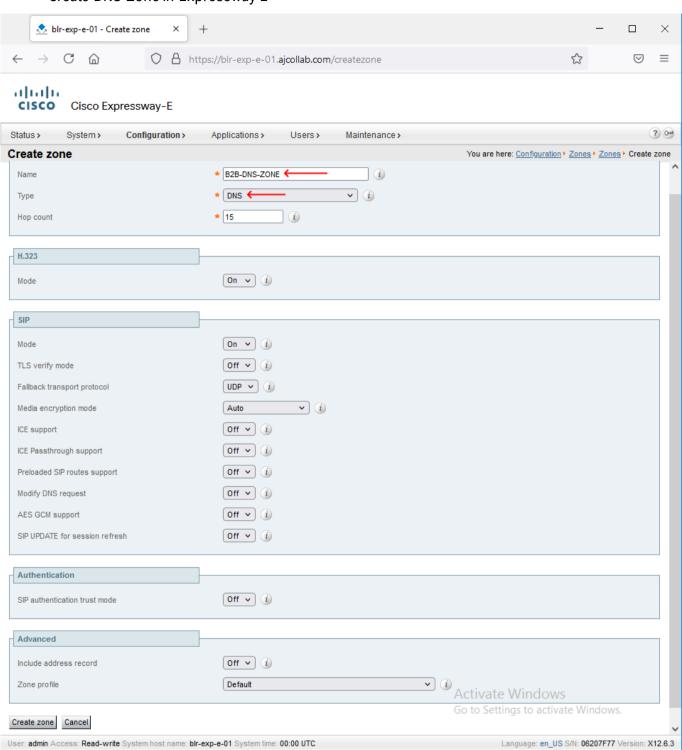
Expressway E Configuration

• Creating Traversal Server Zone in Expressway E

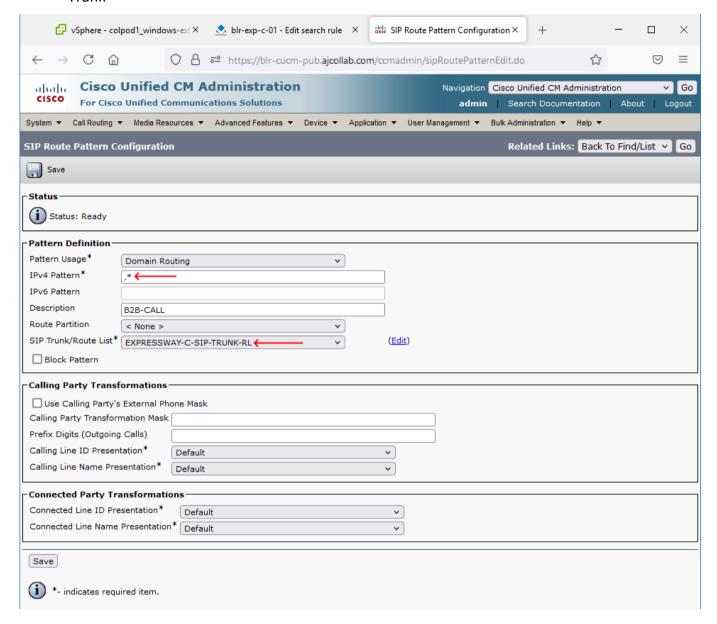




• Create DNS Zone in Expressway E

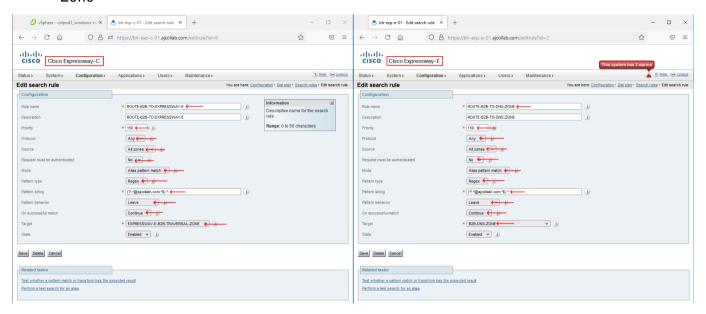


Make sure you have created B2B SIP Route Pattern in CUCM and pointed to Expressway C SIP
 Trunk



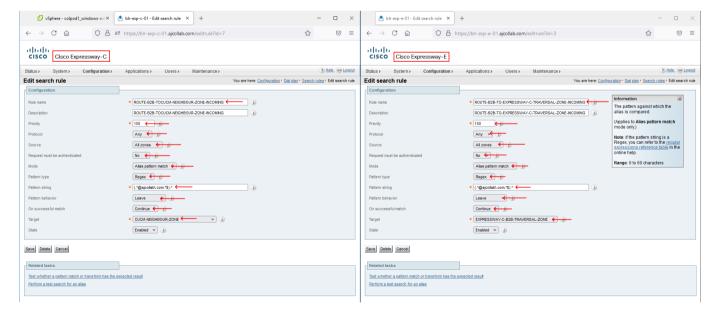
Outgoing B2B Call Search Rule on Expressway C and E

- On Expressway-C (?!.*@ajcollab\.com.*\$).* Do not match my domain and send the call Expressway-E Traversal Zone
- On Expressway-E (?!.*@ajcollab\.com.*\$).* Do not match my domain and send the call to DNS
 Zone

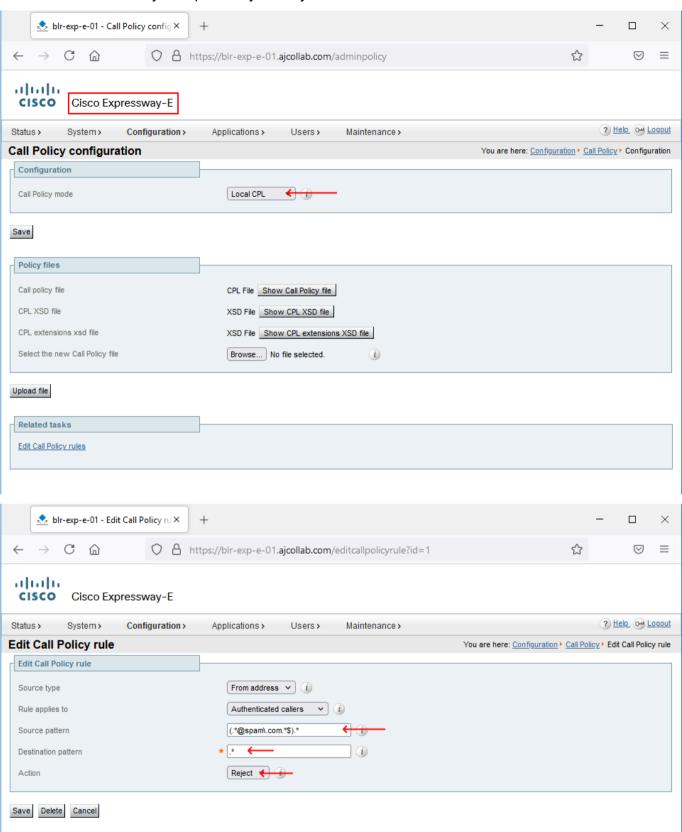


Incoming B2B Call Search Rule on Expressway C and E

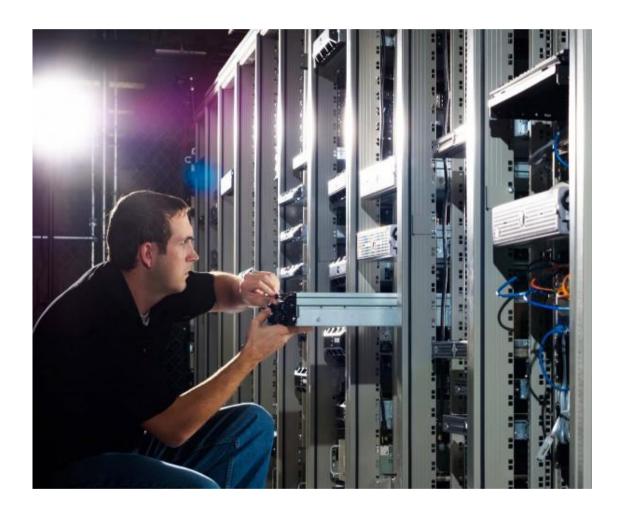
- On Expressway-E (.*@ajcollab\.com.*\$).* match my domain and send the call to Expressway-C
 Traversal Zone
- On Expressway-C (.*@ajcollab\.com.*\$).* match my domain and send the call CUCM Neighbor
 Zone



• Create Call Policy in Expressway E to reject unknown callers



Chapter 3 - Cloud Collaboration Cisco Cloud Collaboration Solution Webex Control Hub



Cisco Cloud Collaboration Solution



UCM Cloud

- It is a cloud based CUCM hosted by Cisco, interface is exactly similar to on-premise CUCM
- It is bit costly as compared to Webex, hence Cisco recommends UCM Cloud if you have 1000+ users
- The Administration interface is completely managed by customer, we can do User Management,
 Device Management, BAT, etc.
- OS Administration and Serviceability access is limited since this is a managed environment
- Selfcare portal is available to all end users

Hosted Collaboration Solution (HCS)

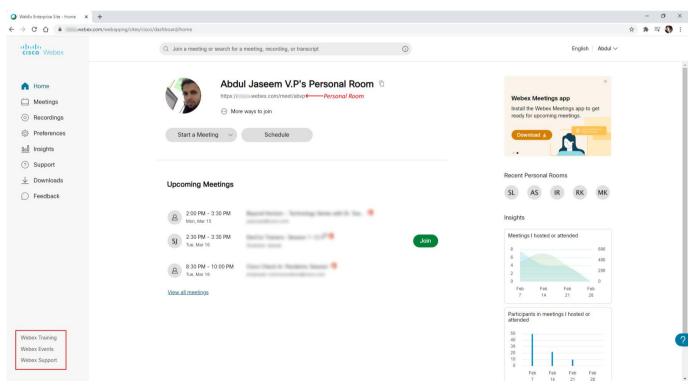
- Service Provider Hosted Collaboration Solution. CUCM is hosted by HSC Partners (Verizon, AT&T, etc.)
- Partners can offer smaller price hence it is cheaper than UCM Cloud

Cisco Webex

- Cloud based collaboration solution hosted and managed by Cisco
- Offers 3 main services Messaging, Meeting and Calling

Overview of Webex Solutions

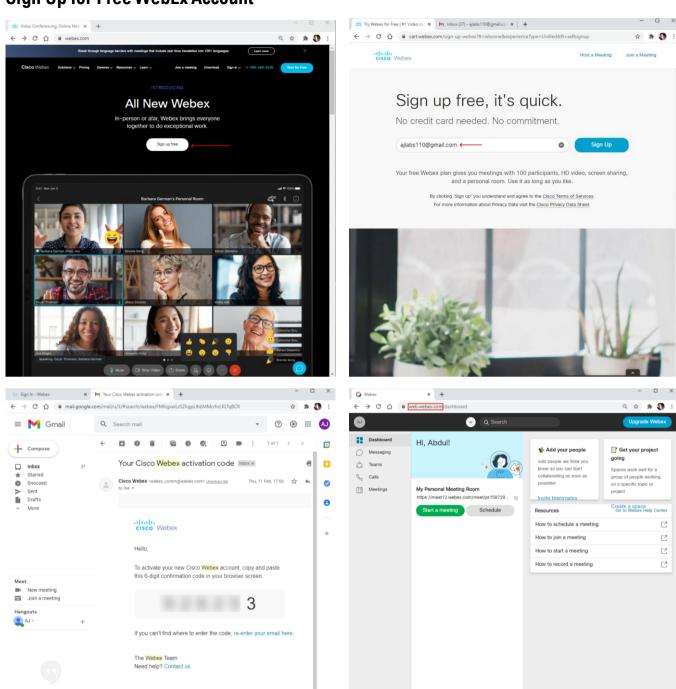
- Webex Meetings: Host video conferences with HD video, audio, and screen sharing. Designed for small to medium sized business. 1000 participants.
- Webex Personal Room: Same platform as Webex meetings, it has a static meeting ID and URL.
 Used for quick collaboration and meetings
- WebEx Events: Similar to Webex meetings but with more participants (3000 participants)
- Webex Training: Other flavor of Webex Meetings that provides impactful training experience
- Webex Support: Other flavor of Webex Meetings that provides real-time support and customer service. Kind of remote desktop management solution
- All the Webex Meeting solutions are Web Based
- Webex Teams Messaging: Connect instantly with team messaging, secure file-sharing and whiteboarding
- Webex Cloud Calling: Get a phone number and business calling system so you can make and receive calls on any device.
- Webex Contact Center: Cloud Based contact center solution



Webex Calling

- BroadCloud Calling: Powered Cisco hosted BroadSoft Cloud IP-PBX Data Center
- BroadWorks: Service Provider (Partner) hosted BroadSoft Cloud IP-PBX Date Center. This is cheaper as compared to BroadCloud
- Webex Calling powered by BroadSoft requires IP Phones registered to Control Hub to be running
 Multi Party Phone (MPP) software
- Webex Calling offers 3 PSTN options
 - 1. Cloud Connected PSTN: Direct connect to the PSTN through the cloud. Customers can choose which provide to be used. We can decide who we want to be the cloud PSTN service provider
 - 2. Local CUBE Gateway: Uses on-premise gateway to breakout to PSTN
 - 3. Bundled PSTN: We can use both of the above options to connect to PSTN
- UCM Calling: Uses on-premise CUCM to facilitate calling functionality. We can make calls from Teams just like we make call from an IP Phone registered on-premise CUCM. The client has to be in corporate network
- Hybrid Call Service: Same as UCM Calling but Teams client can be anywhere on the internet. It uses Expressway Core Connector Host

Sign Up for Free WebEx Account



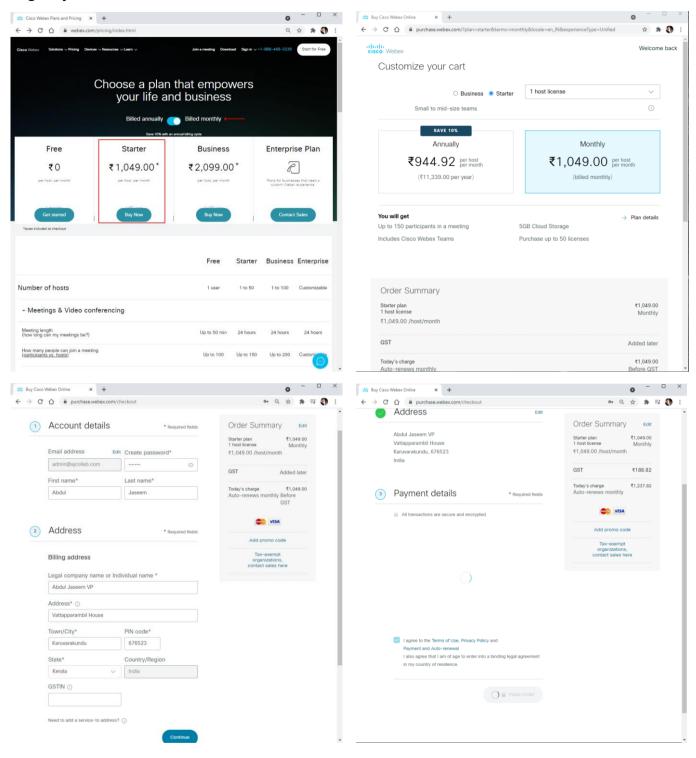
Privacy Statement Terms of Service

https://mail.google.com/mail/u/3/#inbox

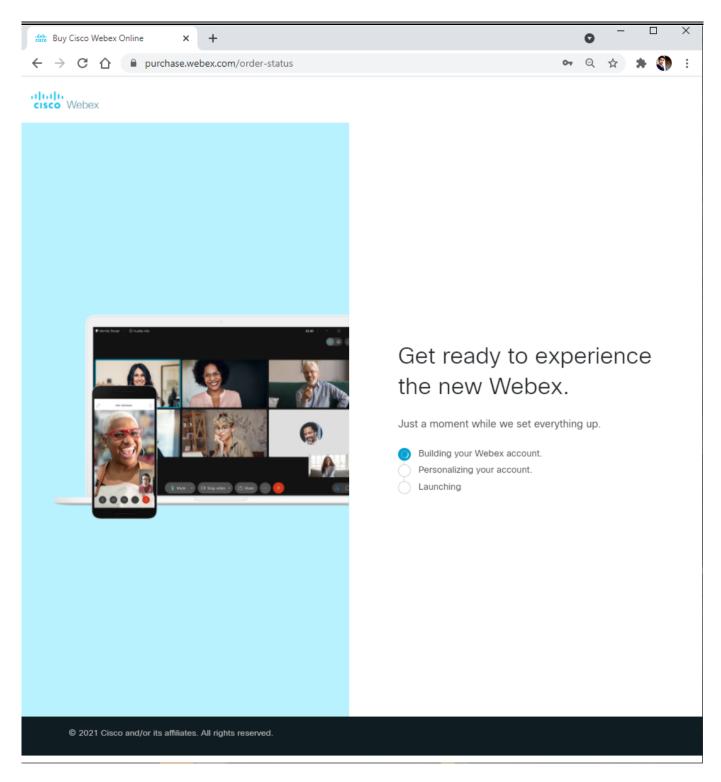
C

C

Sign Up for Paid WebEx Account



 You will receive a 6-digit confirmation code and validate it. I was unable to take the screenshot for that step

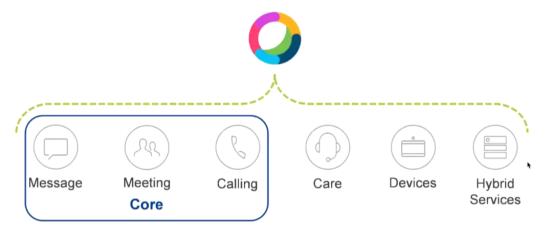


It will take quite some time to complete the setup.

Webex Teams Overview



- It is not a product, it's a Brand. Initial name was Cisco Spark and re-branded to Webex Teams
- Core Products: Messaging, Meeting and Calling
- Other Products: Care, Devices and Hybrid Services
- Messaging is the primary product that comes in every plan, meetings can be integrated to Webex
 Teams or customers can have standalone meeting site



- All these products are hosted on the Cloud and managed from a single management portal called
 Webex Control Hub
- Free account will support the core features such as Messages, Meetings and Calling, rest of the products are available only for the paid accounts

Webex Teams Infrastructure

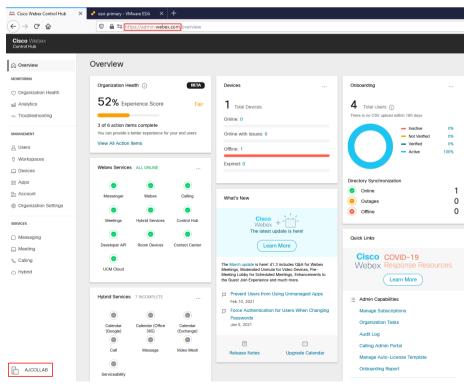


- Webex Teams built as Microservice architecture on an Open Stack framework by Cisco but many features under Teams are acquired by Cisco re-architected
- The complete Webex Teams suit (except WebEx meetings) hosted in Amazon Web Services (AWS) as of today (14/May/2021)
- WebEx Meetings are hosted in Webex Data Centers
- Webex Teams portfolio effectively uses the Auto Scaling functionality of AWS to meet the demand and load requirement
- Microservices helps to add new feature, maintain, and scale the services
- All the microservices internally communicates REST APIs via HTTP
- SIP messages are converted to HTTP APIs with the help of a server called L2SIP
- Media uses SRTP and it is end to end encrypted
- We can capture HTTP Client logs and Packets to troubleshoot issues
- Webex Teams uses Agile (continuous development and deployment) methodology
- We can build our own chat bots

WebEx Control Hub Overview

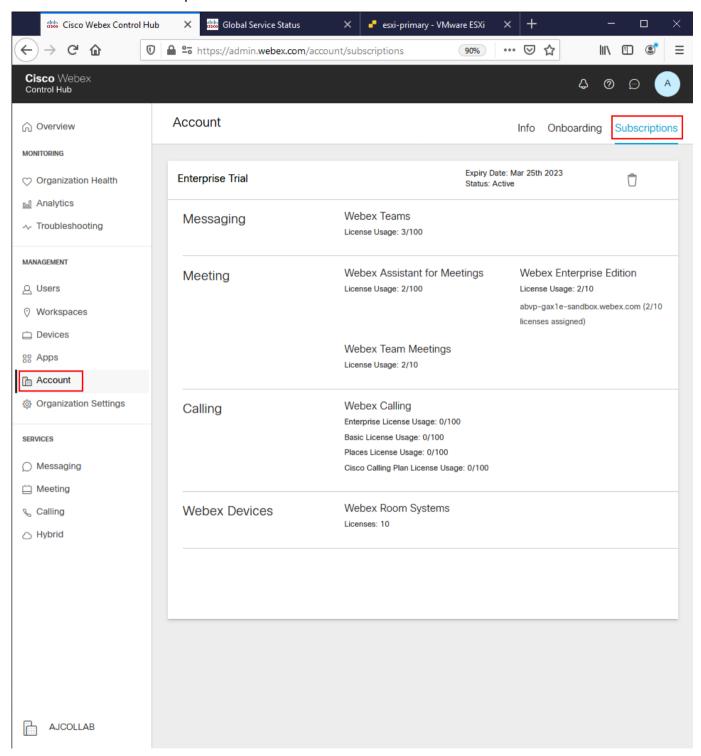


- The administrative portal that manages Webex Teams product suits
- It has several views and roles depending on who logged in (Partners used to manage customers,
 Customers used to manage users, Cisco TAC used to support everyone)
- After customer purchases the service (either from web portal or via partners), Control Hub creates an Organization
- Each organization fall under a specific geographical region, but you can have users from across the globe inside an organization
- Use Control Hub to manage your users, services, and devices. After your organization is created,
 you receive an email to get started on using the administrative functions on Control Hub

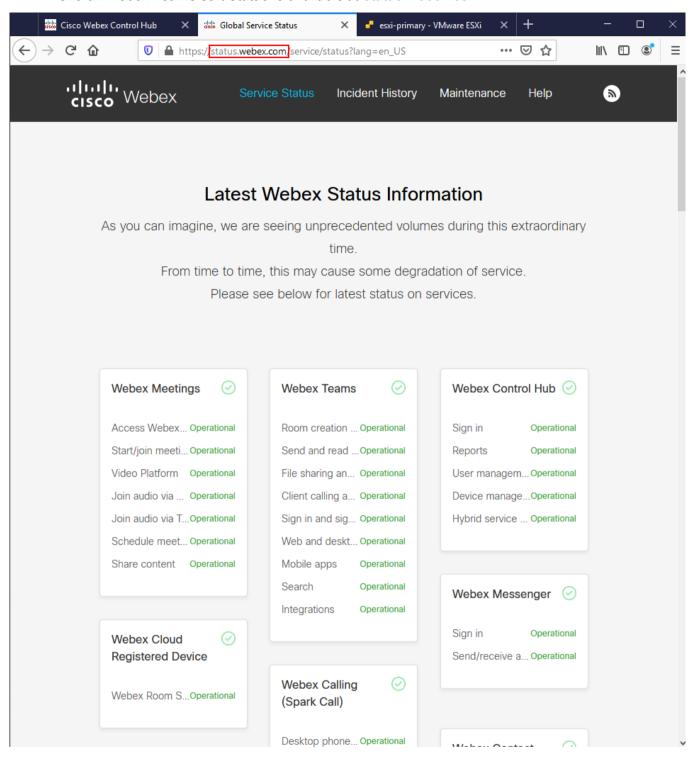


Subscription Status

Go to Accounts >> Subscriptions



Overall Webex Teams Suit Status is available at status.webex.com



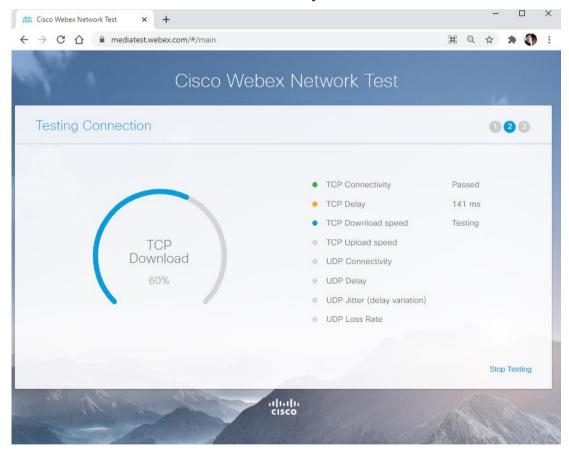
Control Hub Deployment Life Cycle

- 1. Org Creation (Partners Creates or Direct order)
- 2. Customer Admin Can Login to admin.webex.com
- 3. Customer Admin Configure Services (Directory Connect, SSO, Hybrid Services, etc.)
- 4. Customer Admin Sync / Create Users
- 5. Customer Admin Add Devices
- 6. Customer Admin creates Policies
- 7. Reporting and Diagnostics

Webex Teams Client



- Desktop Client (Windows and MAC)
- Browser Based Client (limited functionality)
- Mobile Client (Android and IOS based)
- VDI Client (Use local resources for the Audio and Video calls)
- mediatest.webex.com used to test the connectivity



Spaces

- Group chat functionality in Teams
- We can add multiple people in one Space

Teams

- Similar to Space but initially create a Team, add members to it and then create Spaces
- Team members can automatically join the Space that comes under the Teams

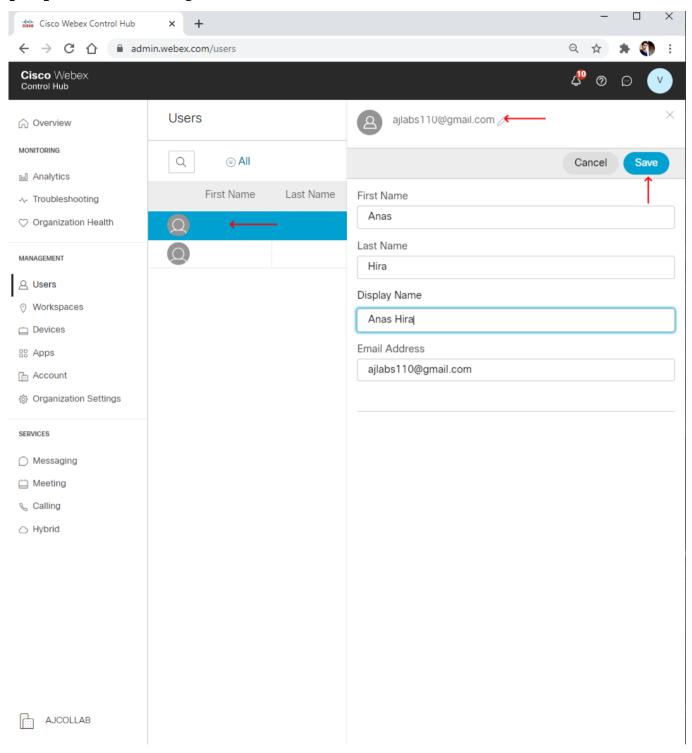
Meetings

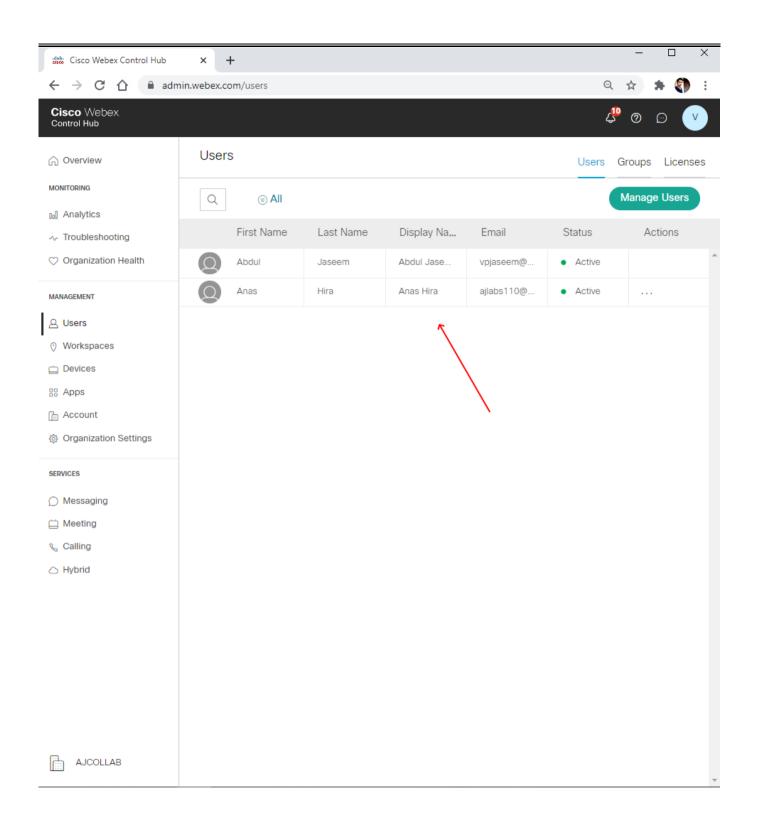
- When we have Webex Meetings, then we have option for Webex meetings otherwise it is just
 Teams meeting
- The video call option is pretty much the Teams Meeting feature
- Teams Meeting and Webex Meetings are different

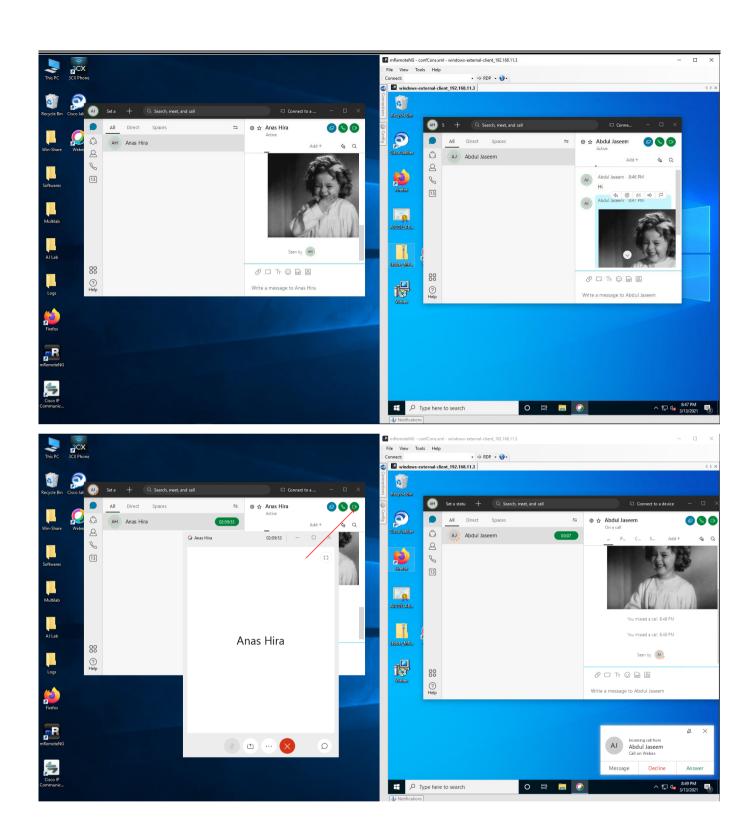
Calendar

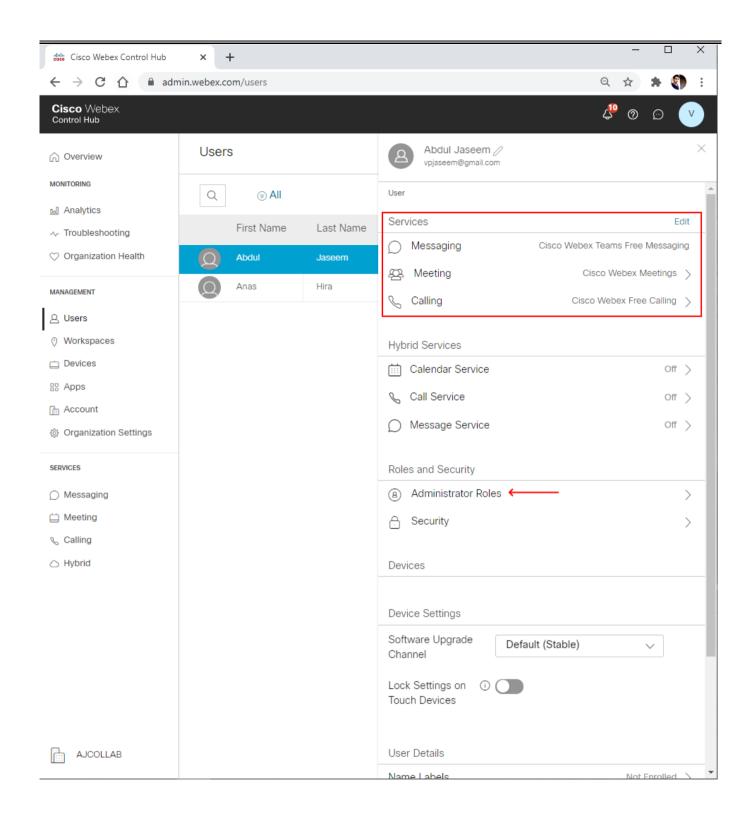
Synced with Webex Meeting Account and Outlook

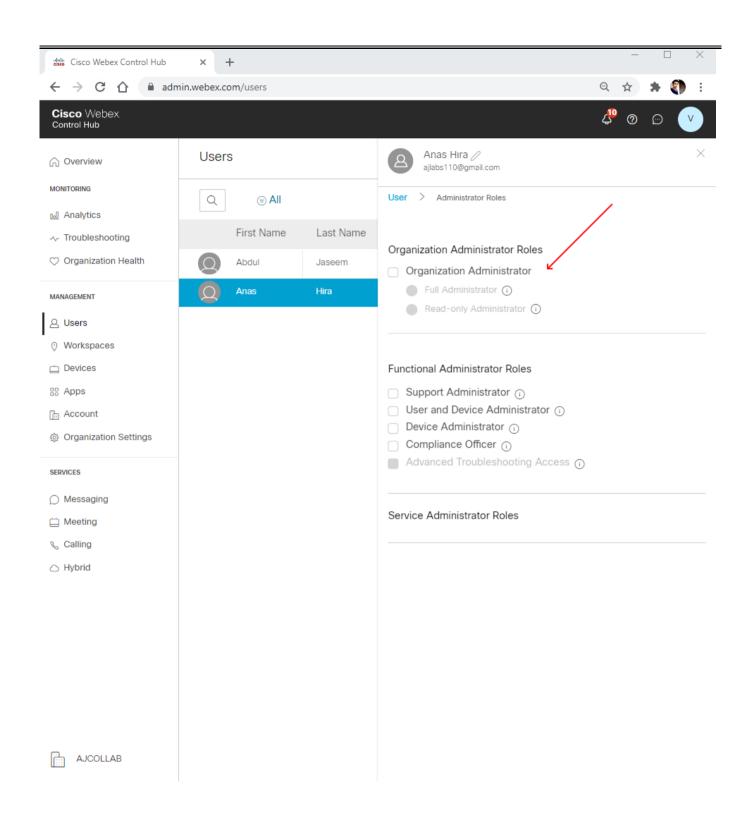
[Lab] Local User Management in Webex Control Hub

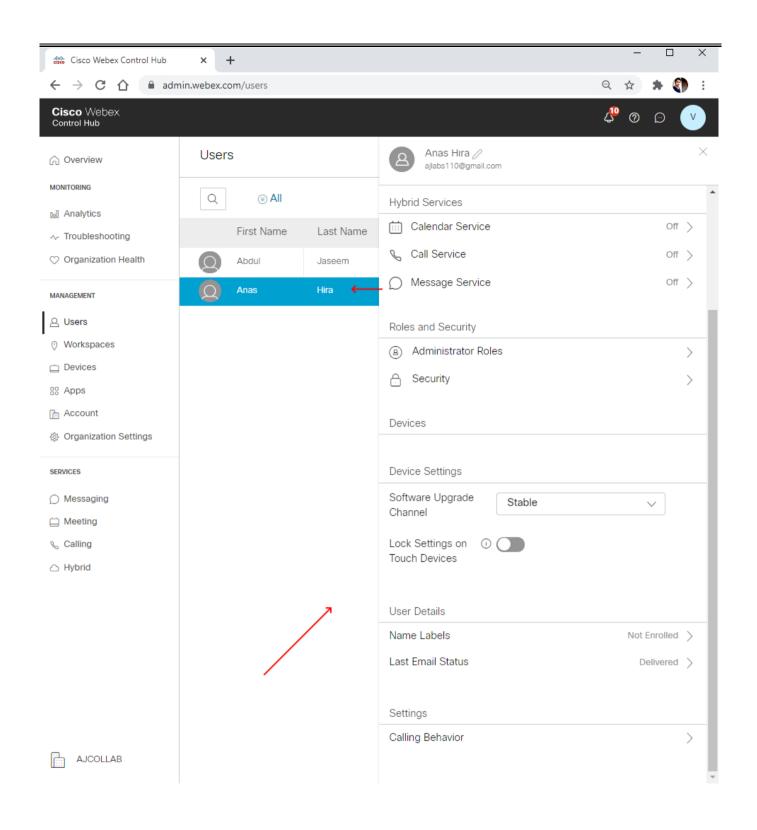






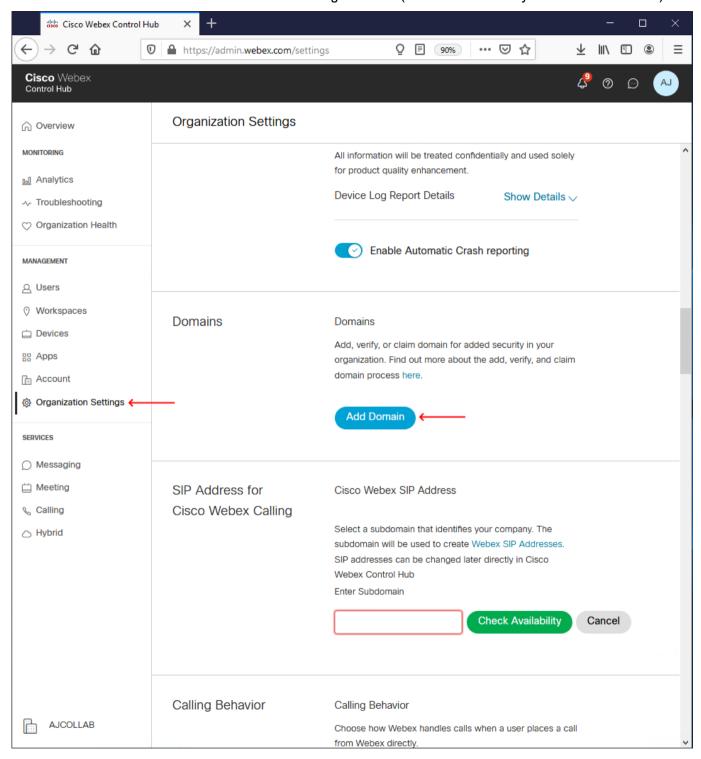


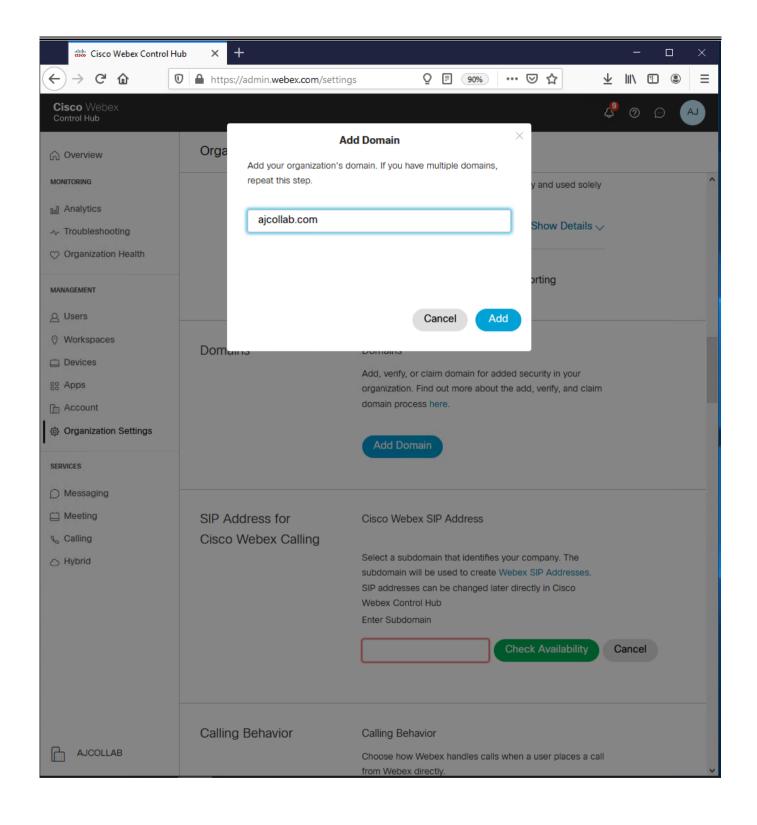


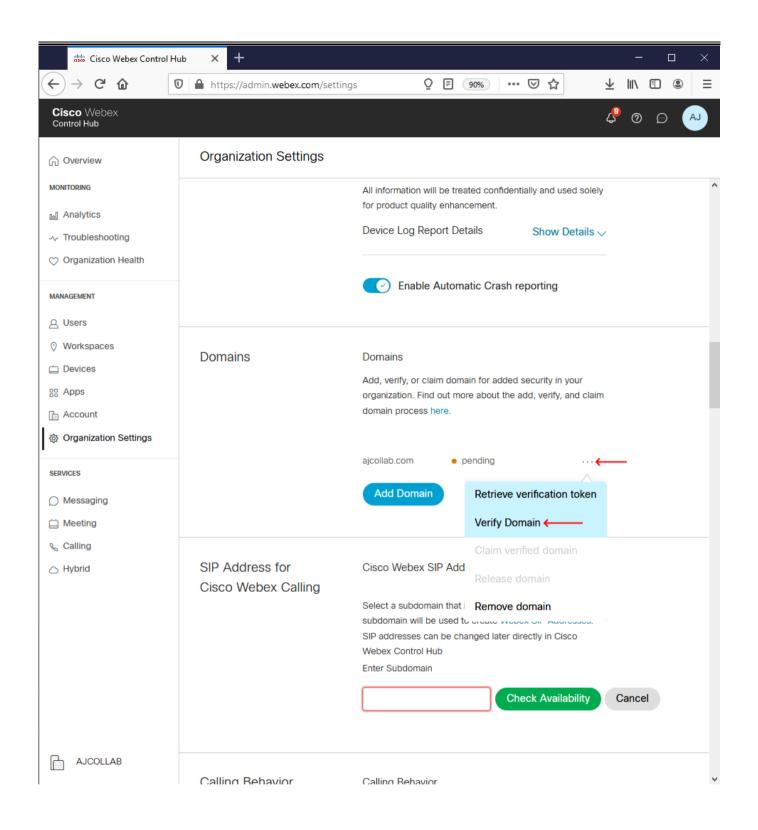


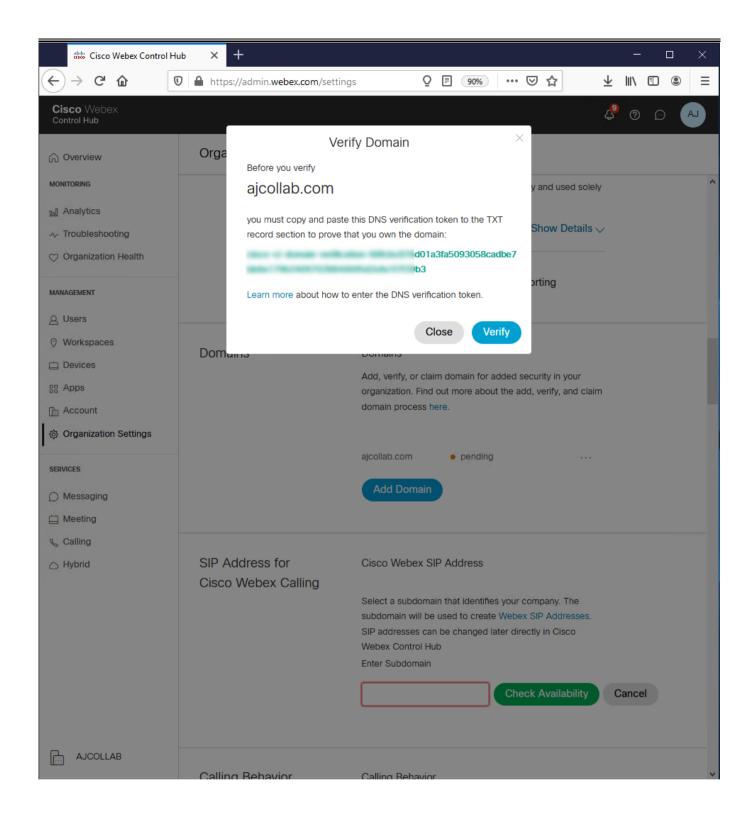
[Lab] Domain Verification

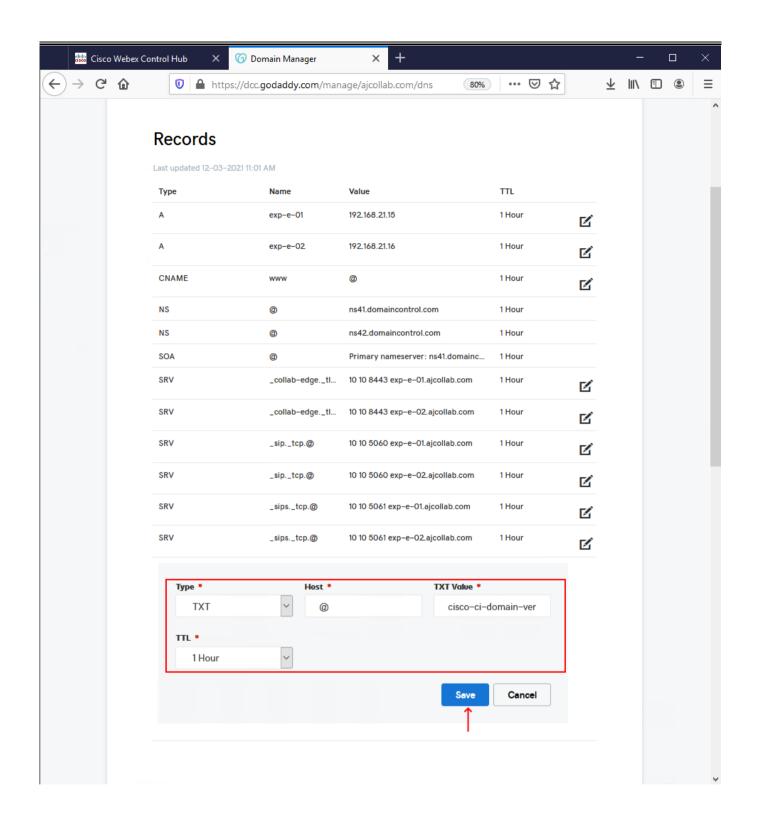
- All the users with your domain will be verified automatically, you don't need manual email verification by the user
- It allows to claim users from Consumer Organization (Free account with your domain email ID)

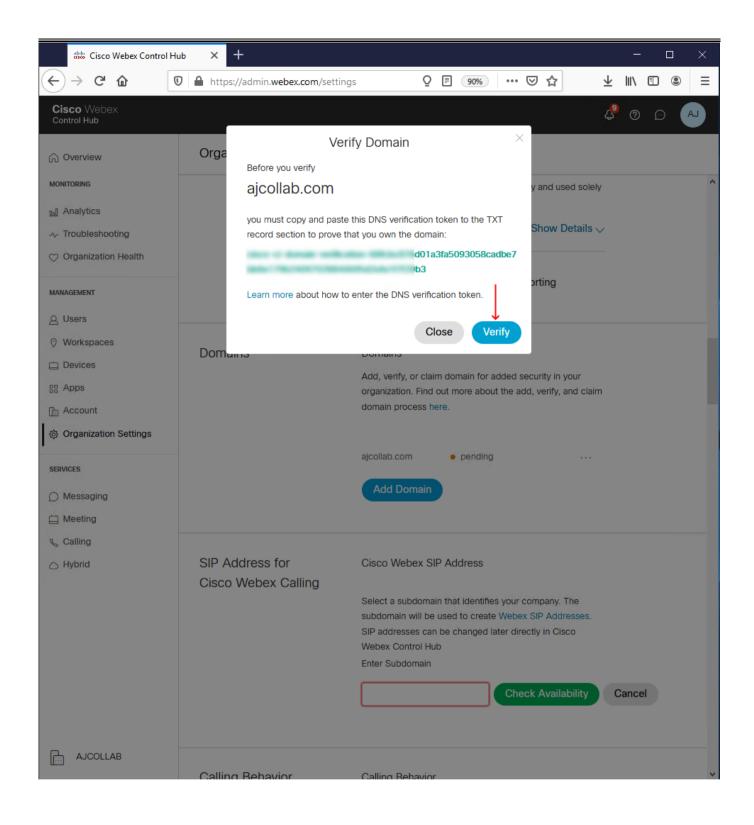


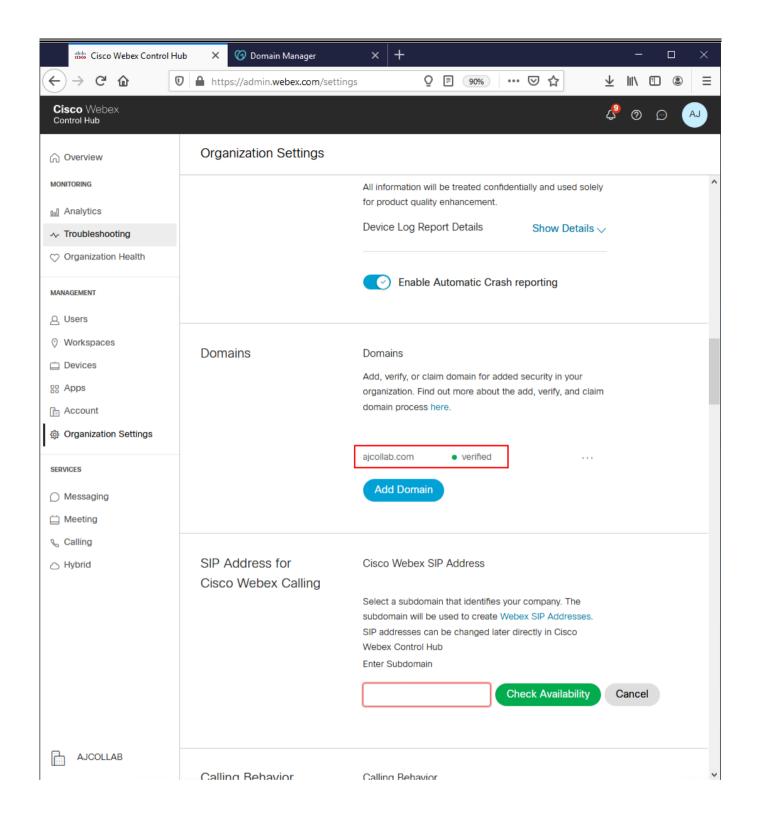


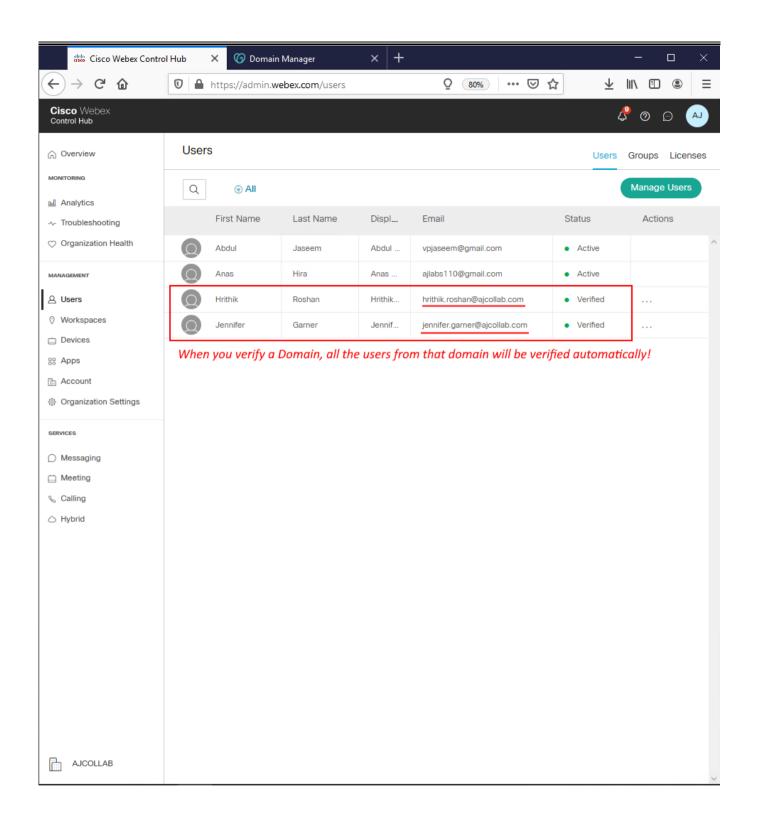






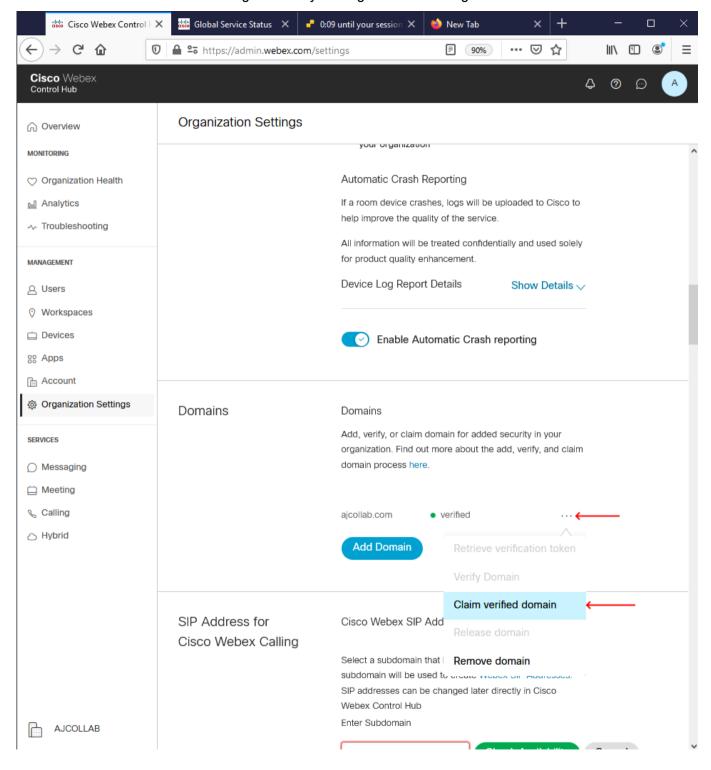






Claim the Domain

- Once you claim the domain, those users can be a part of only your organization
- They can't have any free account or can't be a part of different organization
- You must convert all existing users to your org before claiming a domain

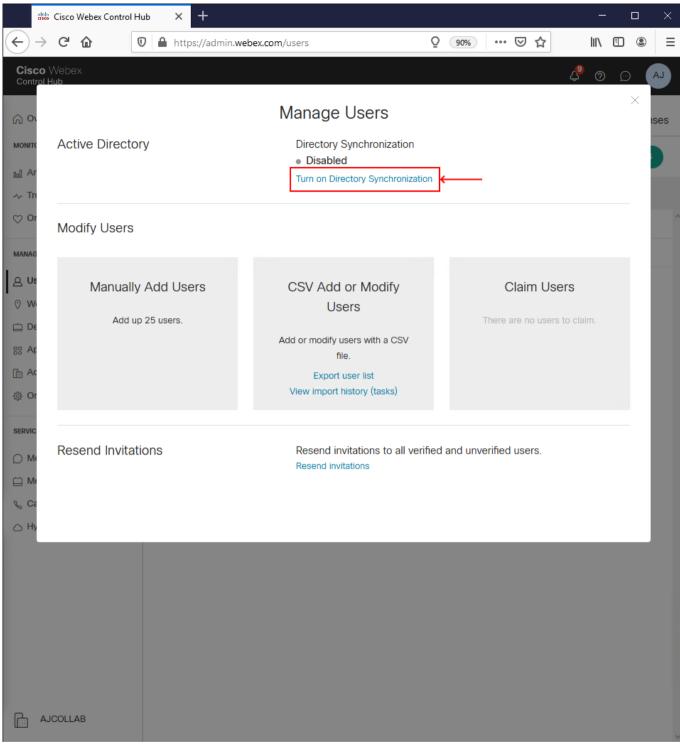


[Lab] Cisco Directory Connector

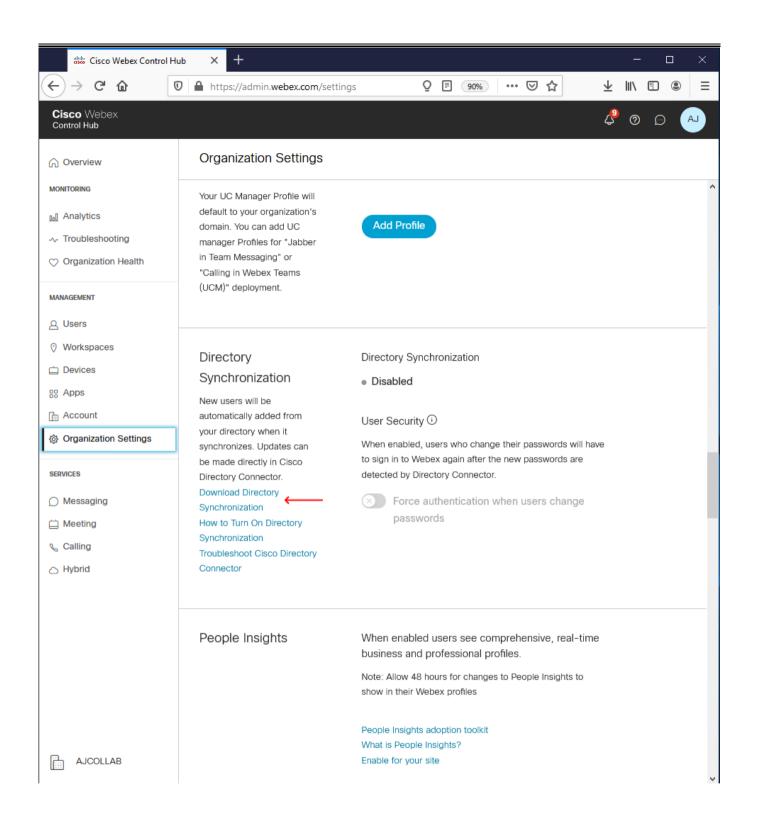


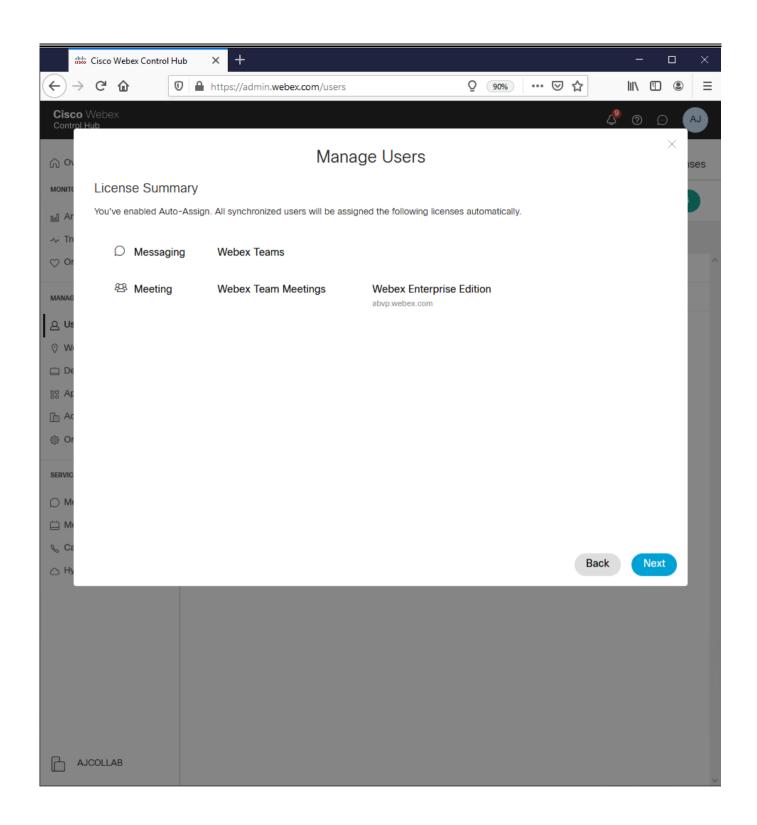
- On-Premise application for user synchronization to Cloud (like LDAP sync in CUCM but not exactly the same). It is a software that can be installed on your Windows AD
- Microsoft Active Directory becomes the primary source for user accounts in Control Hub
- We can sync entire AD with Webex Control Hub
- It is a software that can be downloaded from Control Hub and it can be installed on a Windows Server
- LDAP filters can be applied to specifically sync required users
- Directory Connecter is per domain based, if we have multiple domains, we need multiple Directory
 Connectors as well
- Directory Connector is compatible with Microsoft Azure AD as well

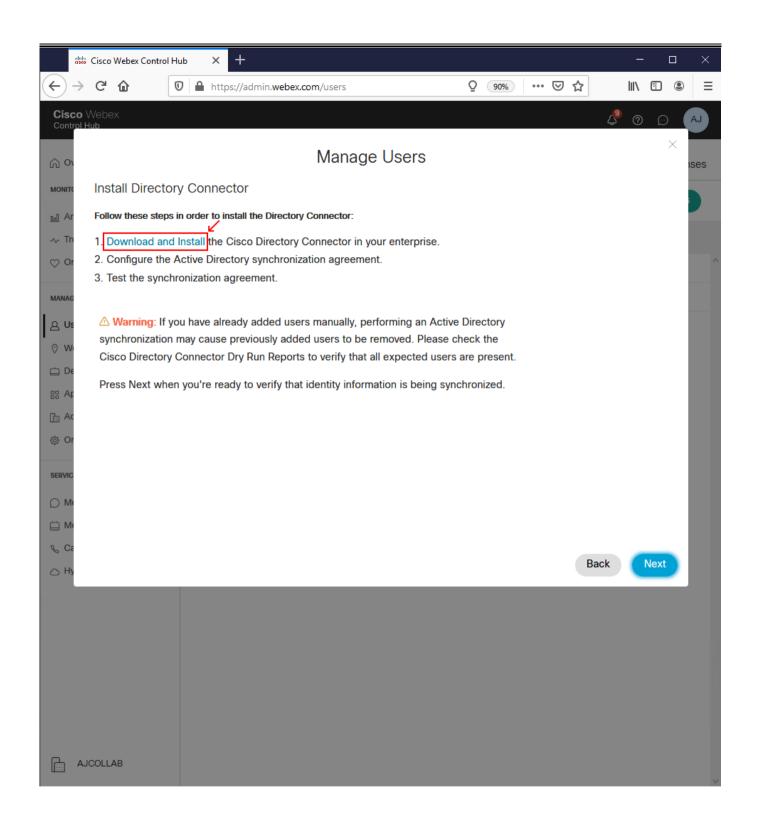
Control Hub >> Users >> Manage Users >> Turn on Directory Synchronization

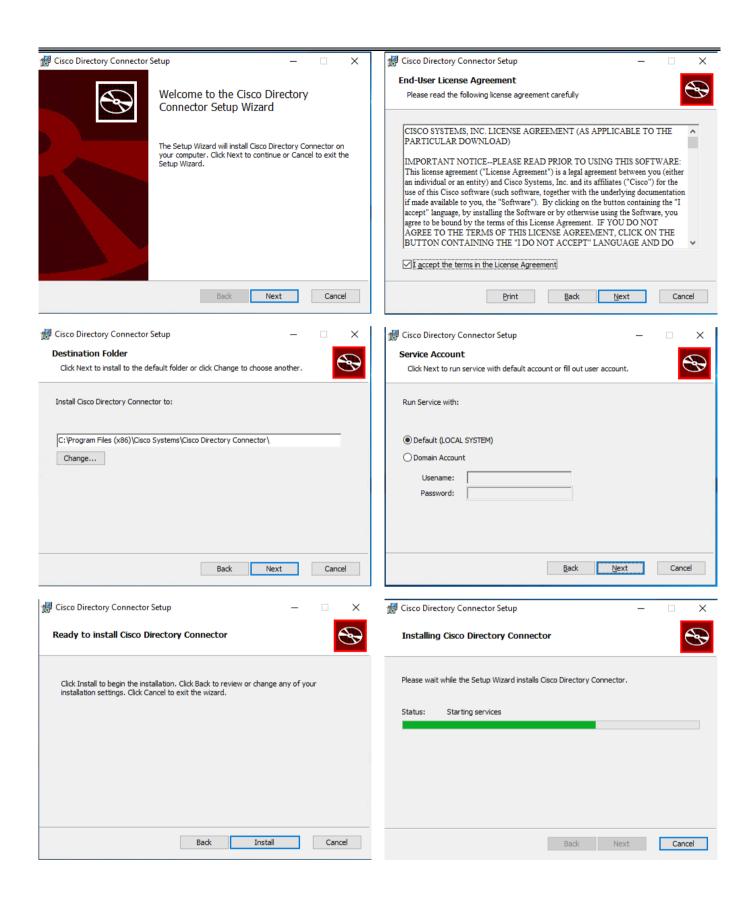


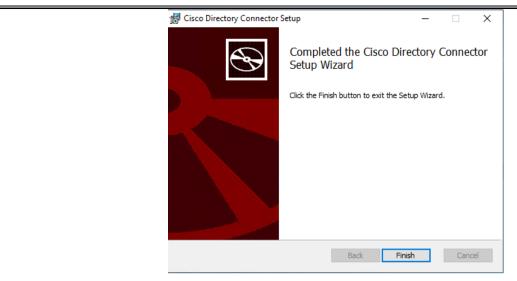
OR

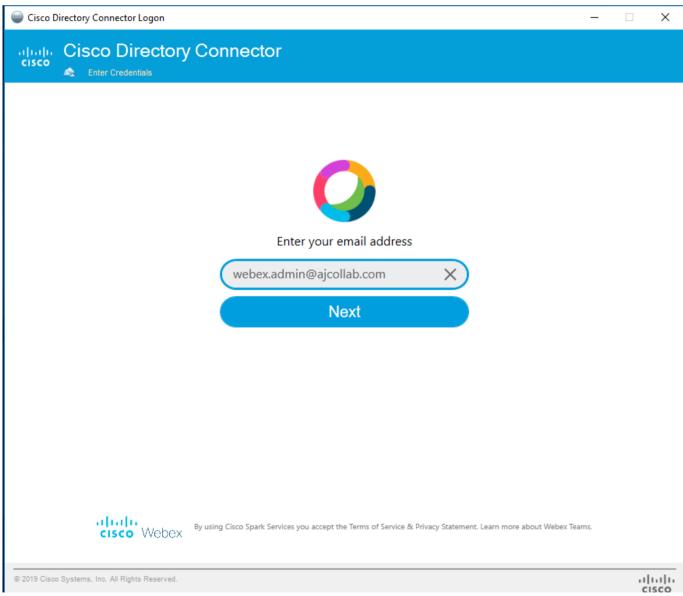


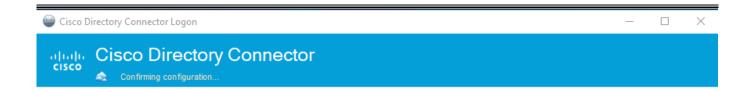


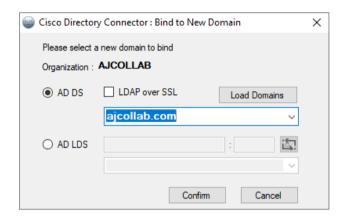




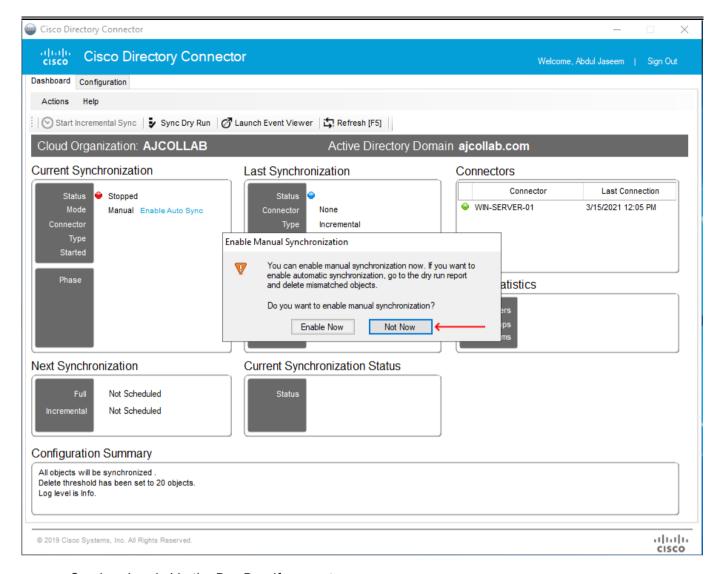




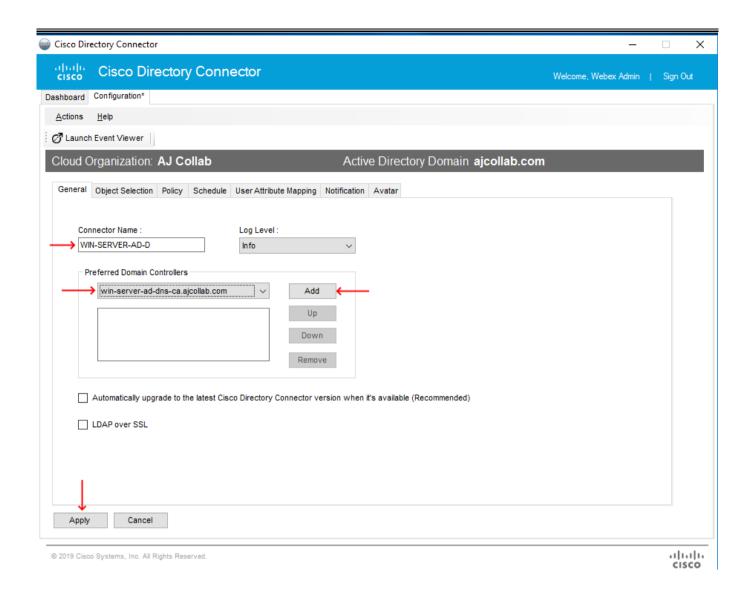


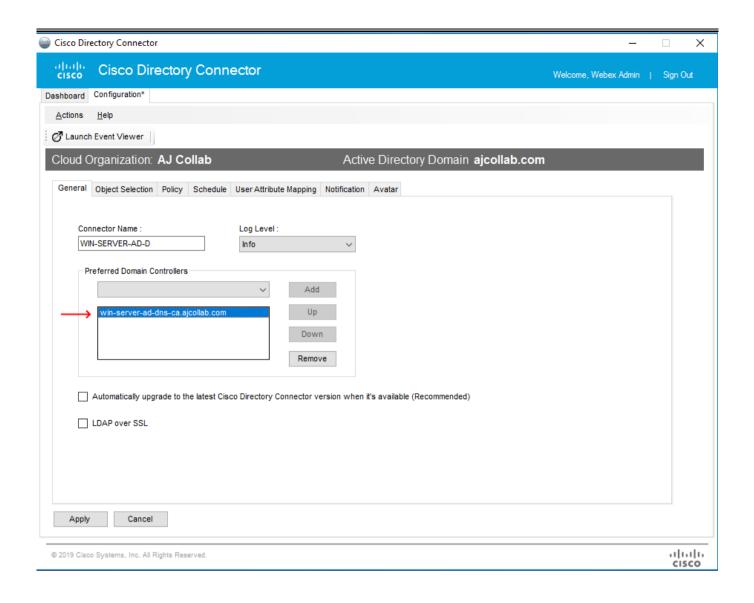


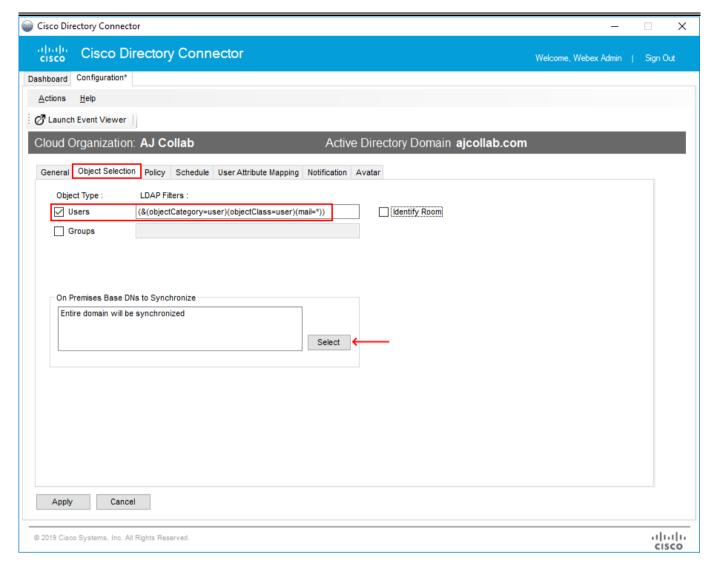
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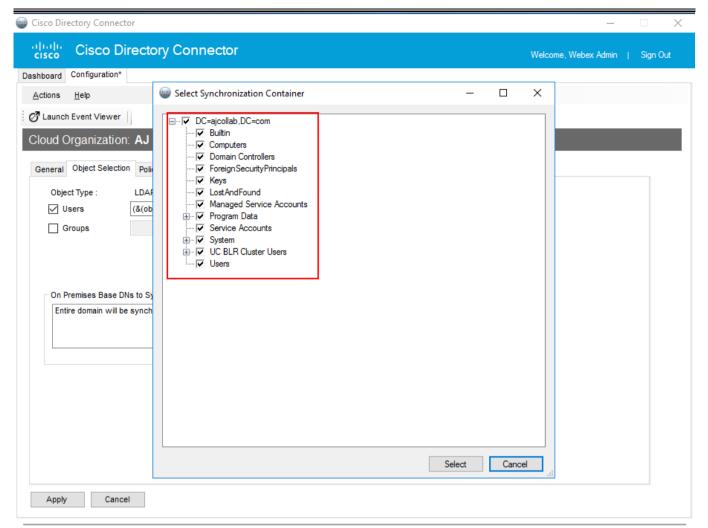
· Go ahead and skip the Dry Run if you get any pop-ups





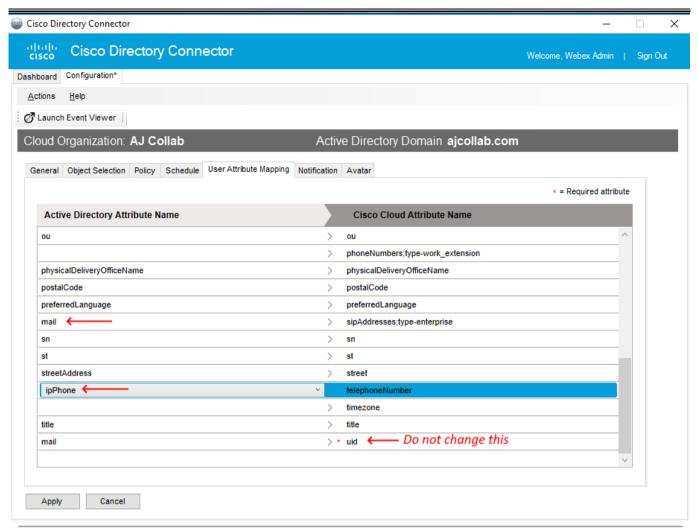


Use below custom filter, that will make sure only users with valid email ID will be synchronized
 (&(objectCategory=user)(objectClass=user)(mail=*))



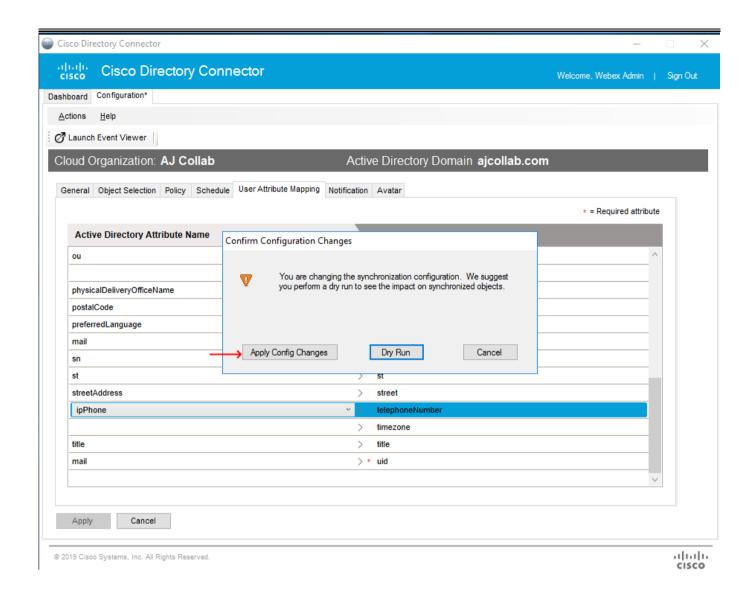
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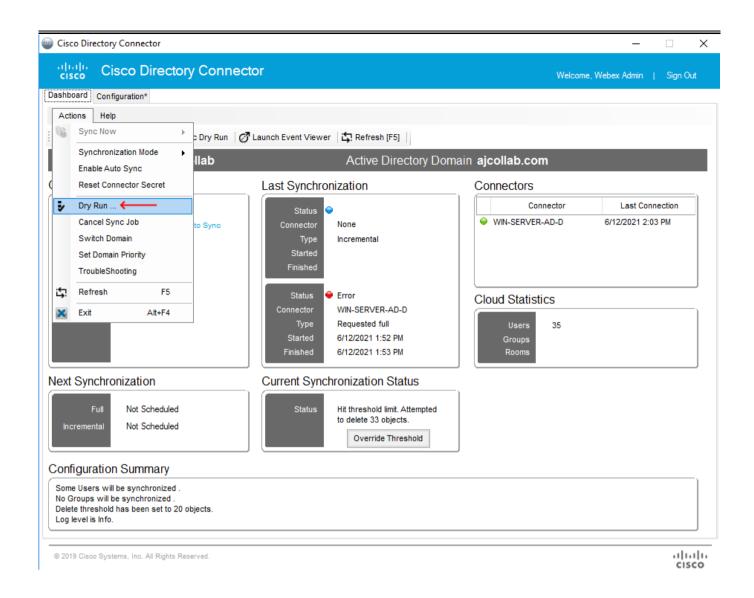
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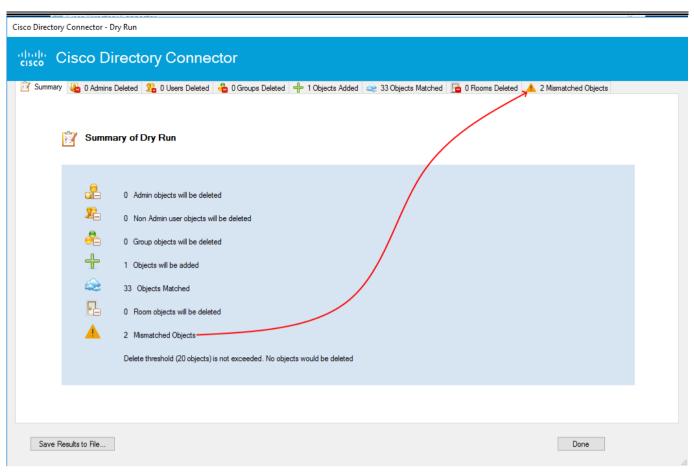


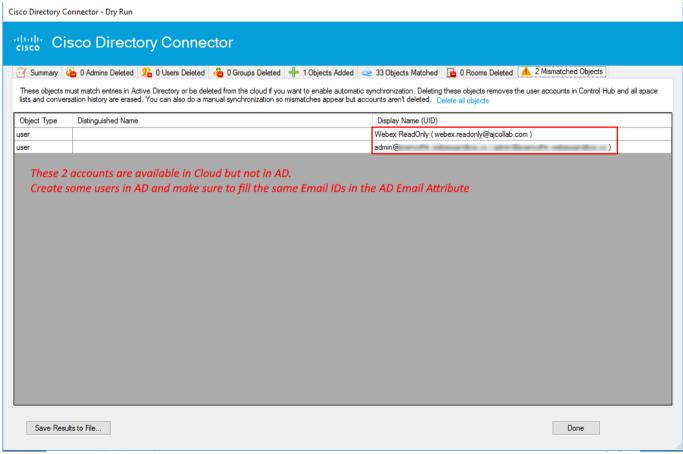
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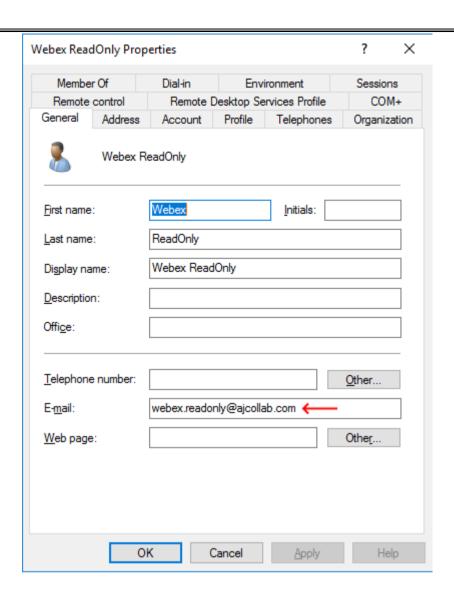
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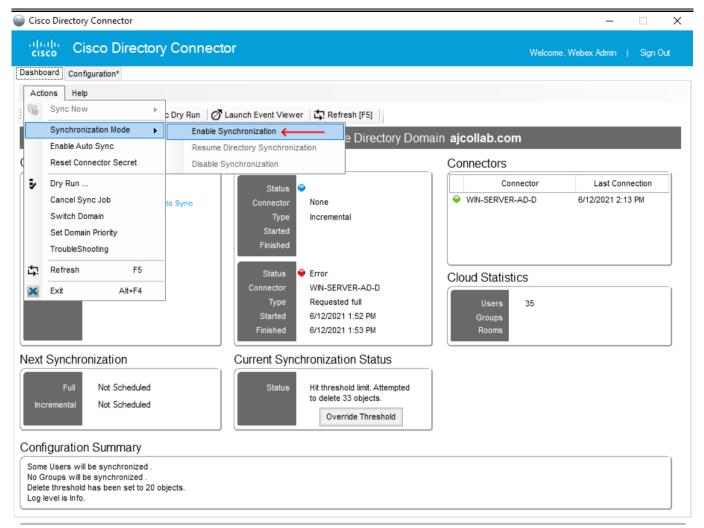


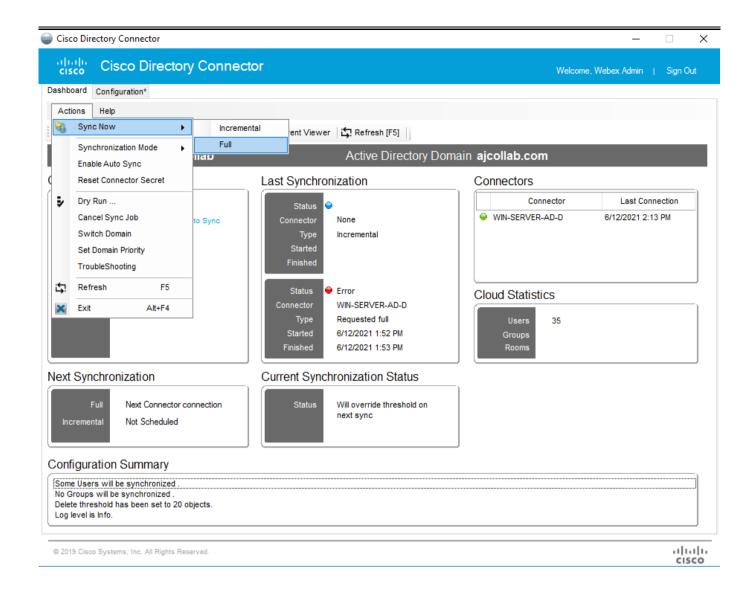


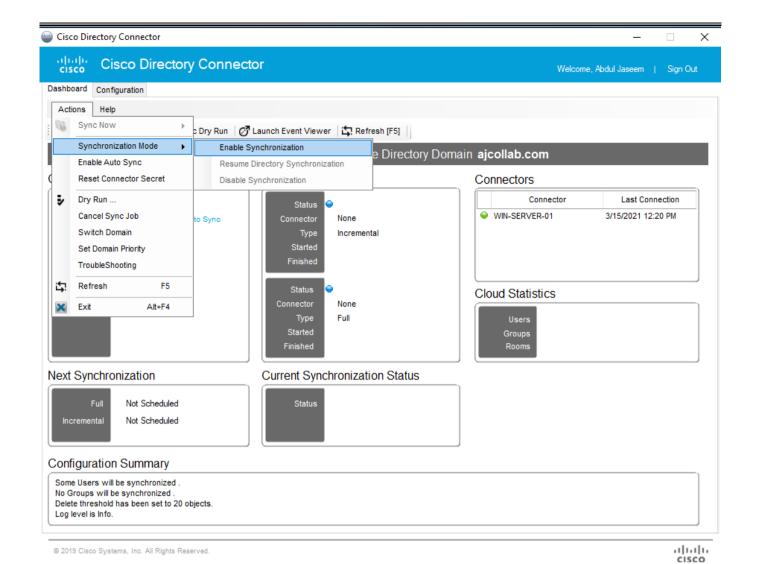




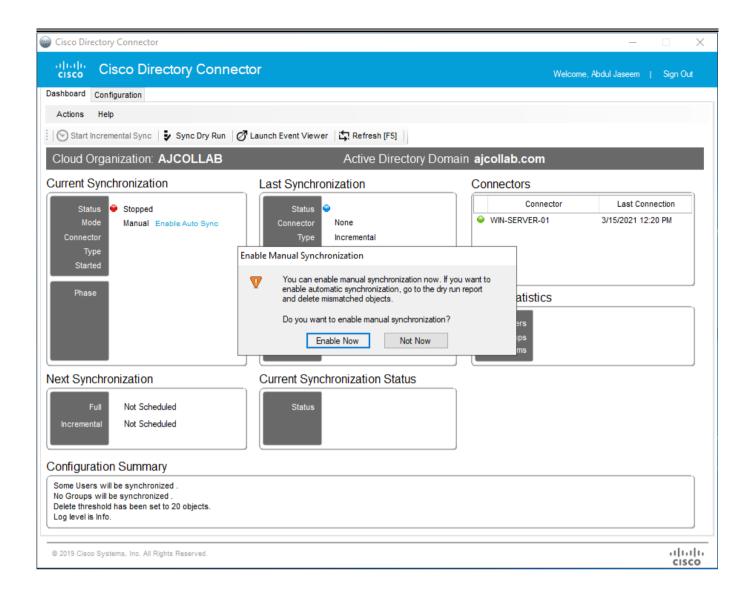


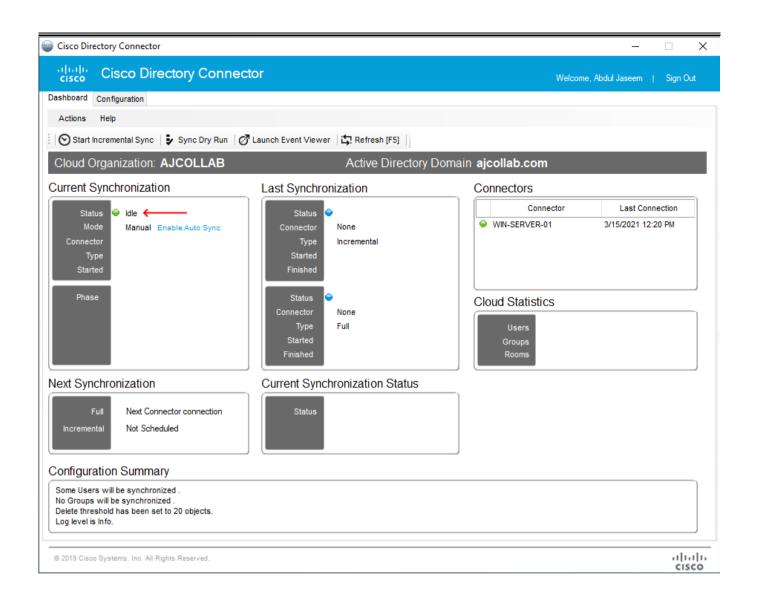


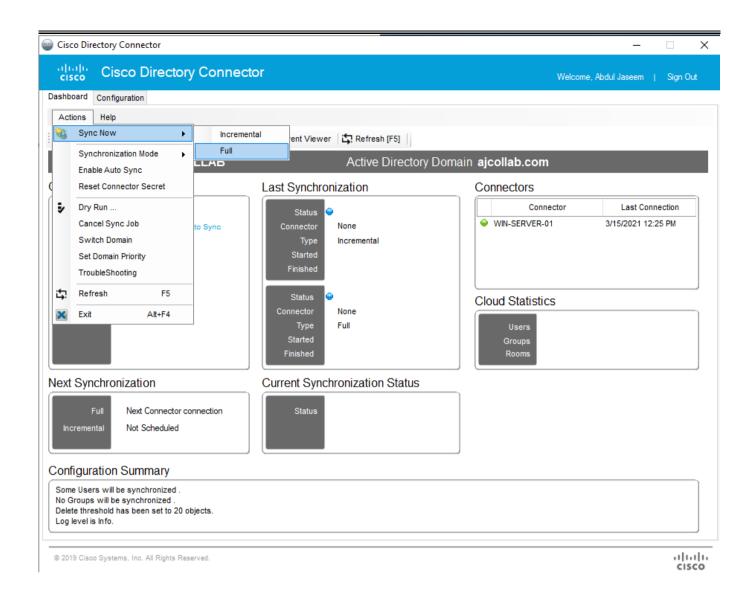


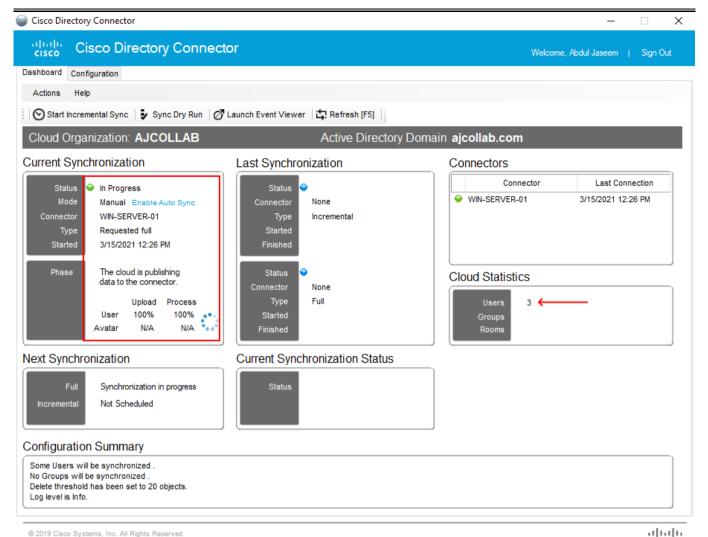


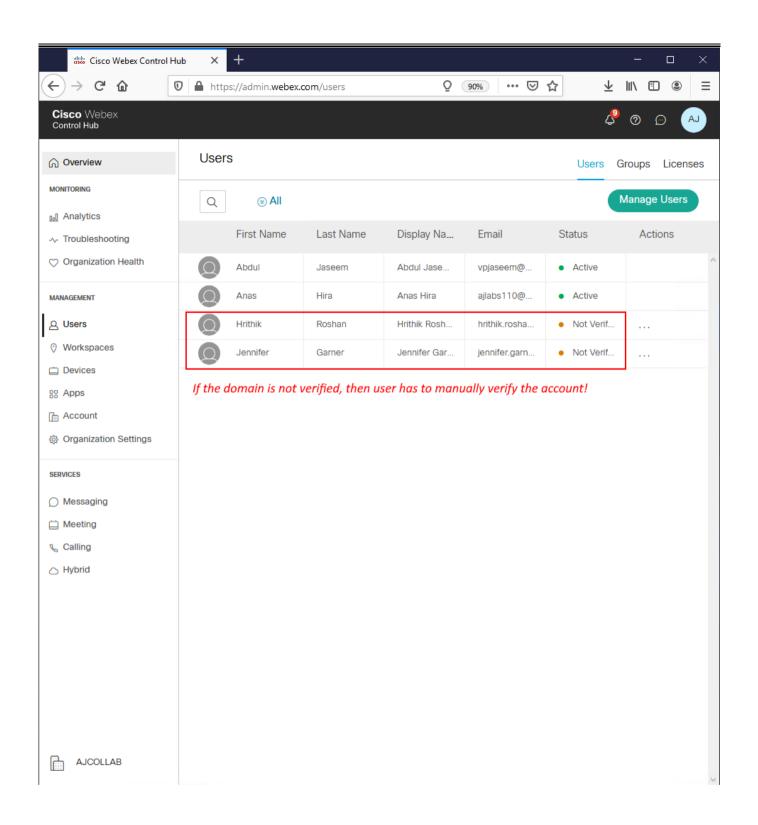
. . . .

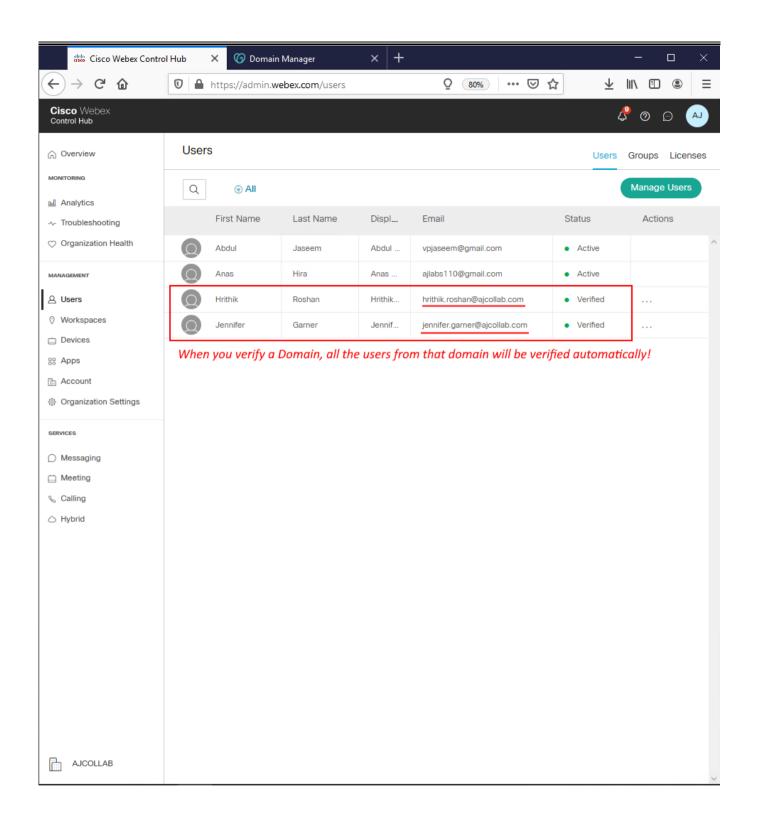


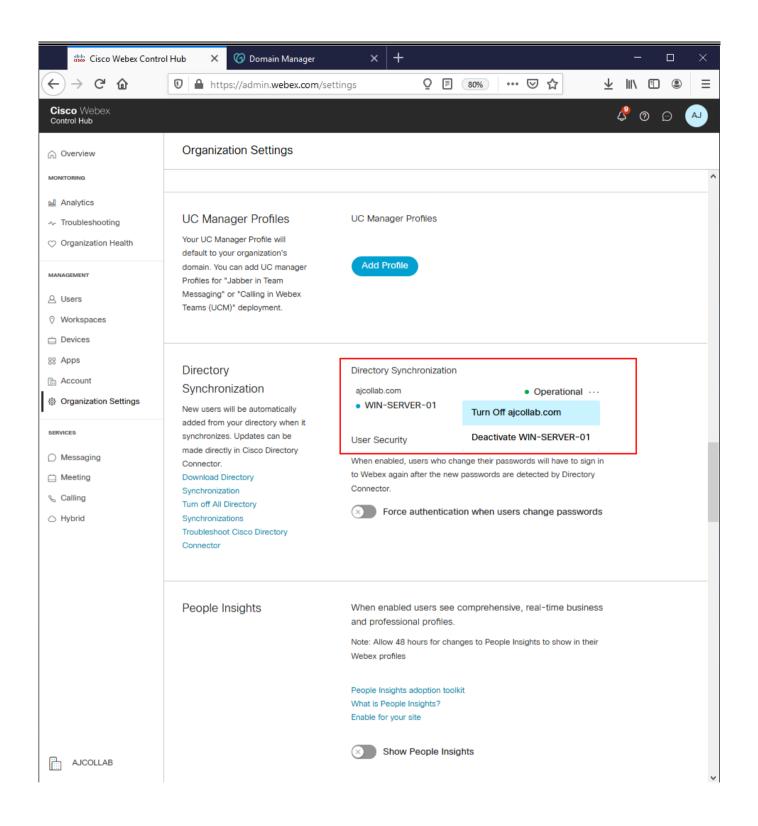




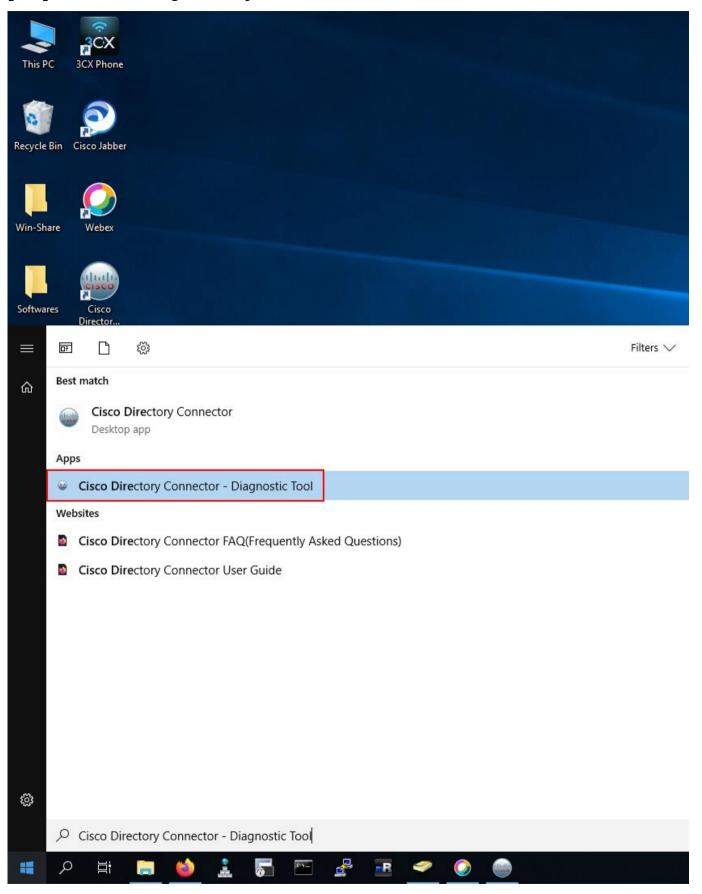


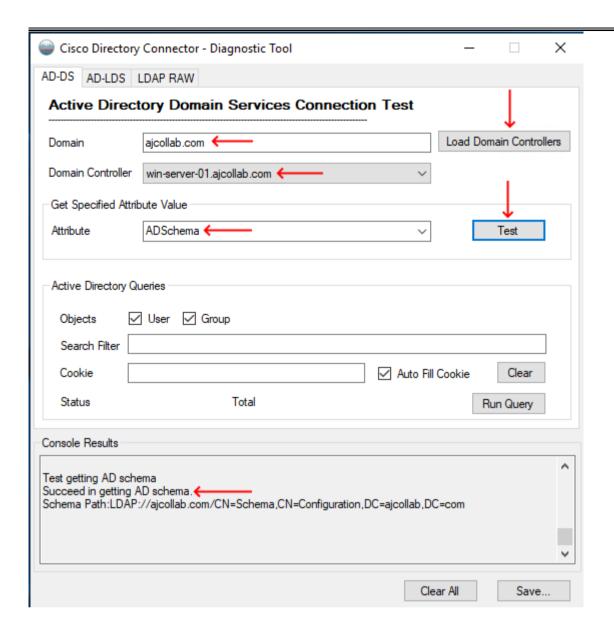






[Lab] Troubleshooting Directory Connector





Webex Teams SSO

- One set of credentials to access multiple different services
- Authentication handled by IdP (Identity Provider) server, there are 2 types of SSO
- Intra-Organizational SSO: Access resources within the organization
- Inter-Organizational SSO: Also known as federated SSO, establish trust between multiple orgs to authenticate users
- Webex Teams utilizes web federated SSO with SAML 2.0, it uses web browser to exchange identity information across HTTPS
- Identity information formatted using Security Assertion Markup Language (SAML) 2.0, XML based open standard
- Service Provider (SP): Provides the service that being utilized (application or system that user logs in to. e.g. CUCM, Webex Control Hub, etc.)
- Identity Provider (IdP): System that challenges a user for their credentials and tells the SP if the login was successful or not. Microsoft Azure and okta are cloud based IdPs Microsoft Windows ADFDS, PingiD etc. are other IdPs
- Claim: An IdP configuration that determines what information is to send to SP. Its basically taking some attributes (uid as SAM-AccountName)
- In the Webex Control Hub, the SP is Webex that is managed by Cisco and the IdP can be a
 Windows Server with ADFDS installed, that is managed by customer



Process of SSO Login

- User logs in to Service Provider (SP) web interface
- It gets redirected to IdP and submit the credentials (SAML Request)
- IdP sends information back to SP based on Claim rule and browser redirects back to the SP (SAML Response)

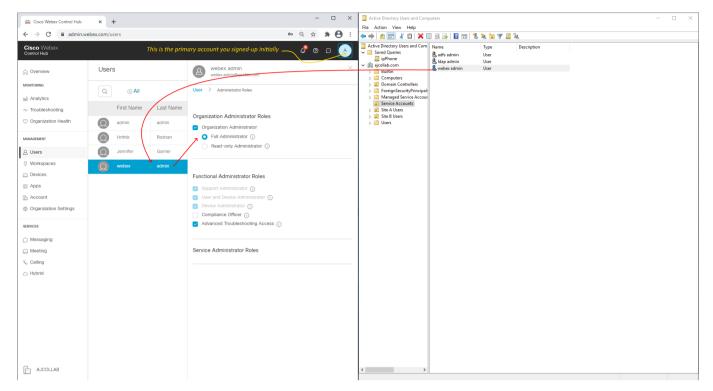
Metadata File

- SP and IdP will have its own meta data files
- Entity ID: URL of SP, this identifies which SP initiates the request
- Endpoint URL: URL of IdP
- Name ID: User attributes
- Certificates: Signing and Encryption Certificates
- Claim Rule: Defines the attributes and values that are returned in the SAML response
- Both Metadata files must be cross imported (SP to IdP and verse versa)

[Lab] Webex Control Hub SSO Configuration

Before enabling SSO, you must configure Cisco Directory Sync and make one AD synced user as 'Full Admin'

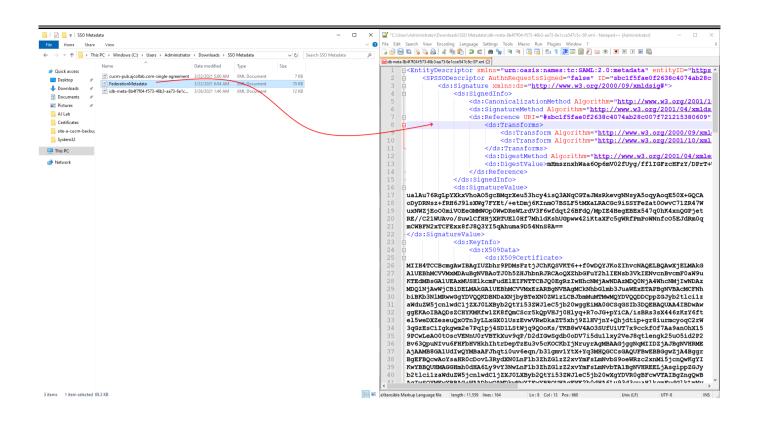
I have added a user 'webex.admin' in AD and made that user as a 'Full Admin' in Control Hub

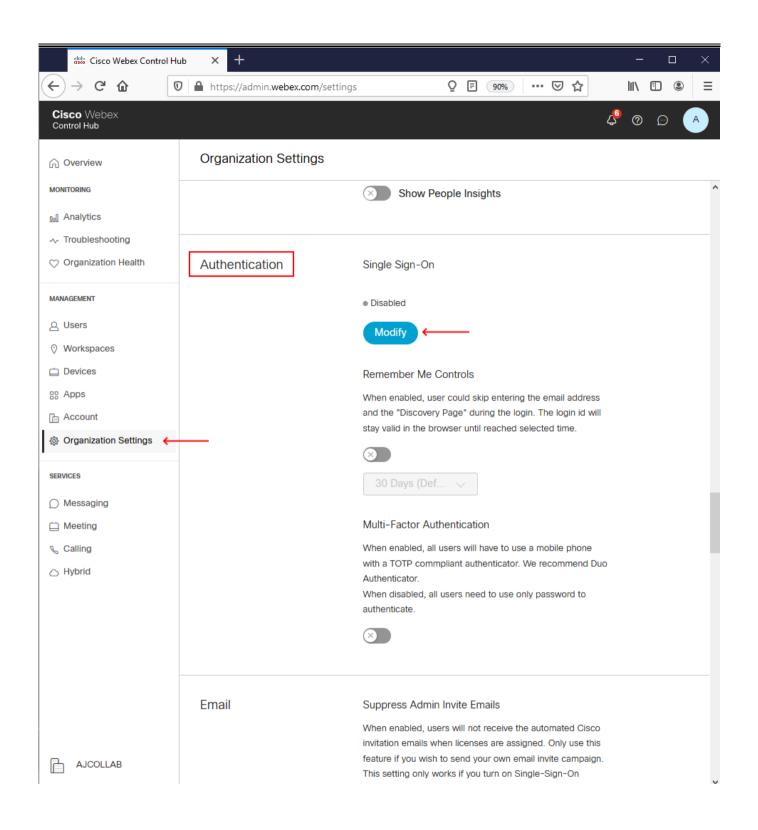


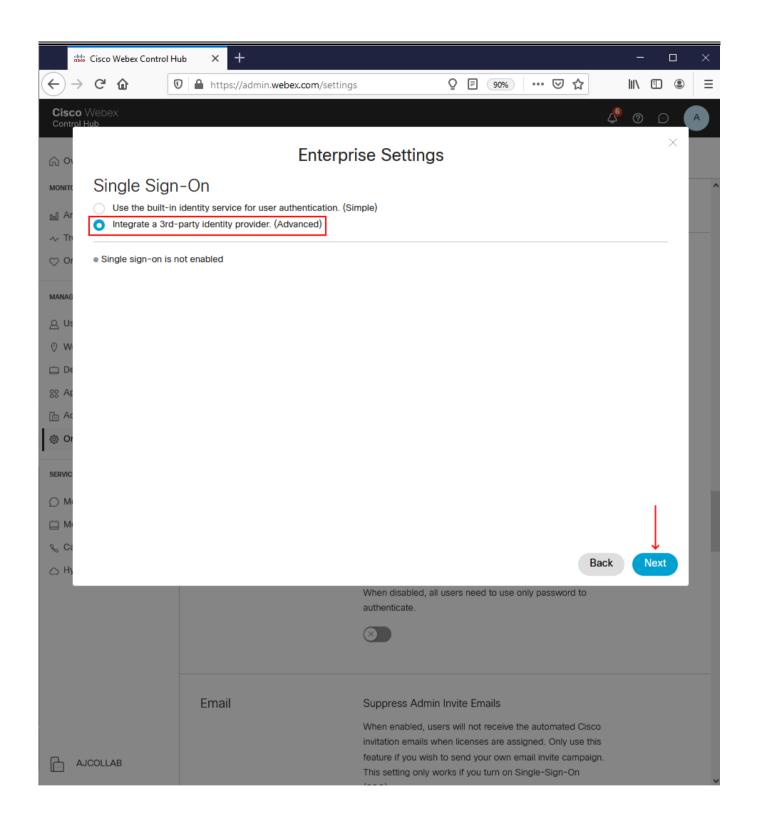
If we miss this step, after we enable SSO, we won't be able to sign in as Administrator

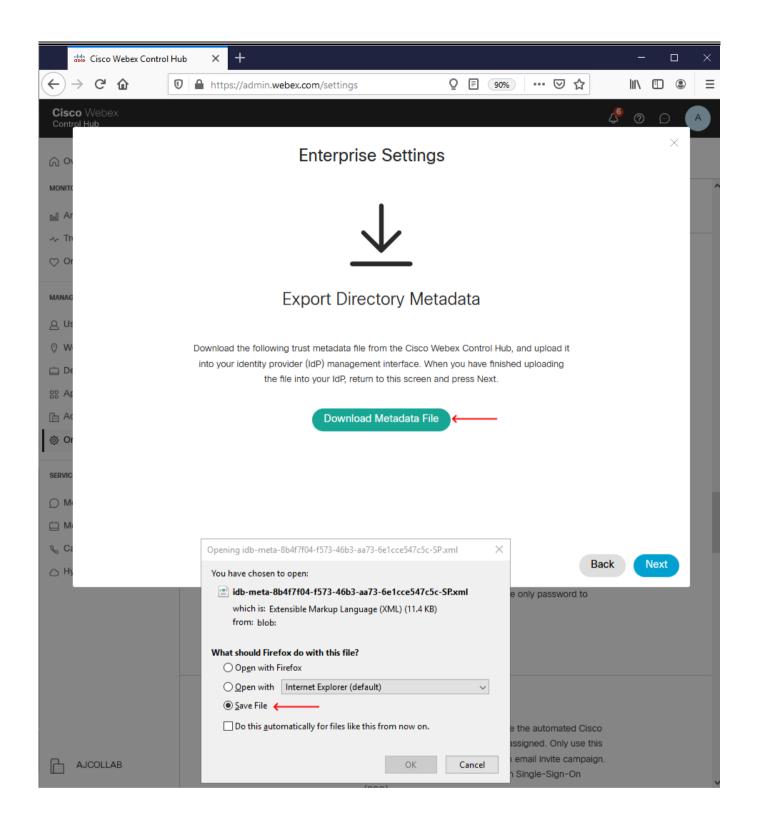
Download the ADFDS Federation Metadata from the Windows ADFDS Server

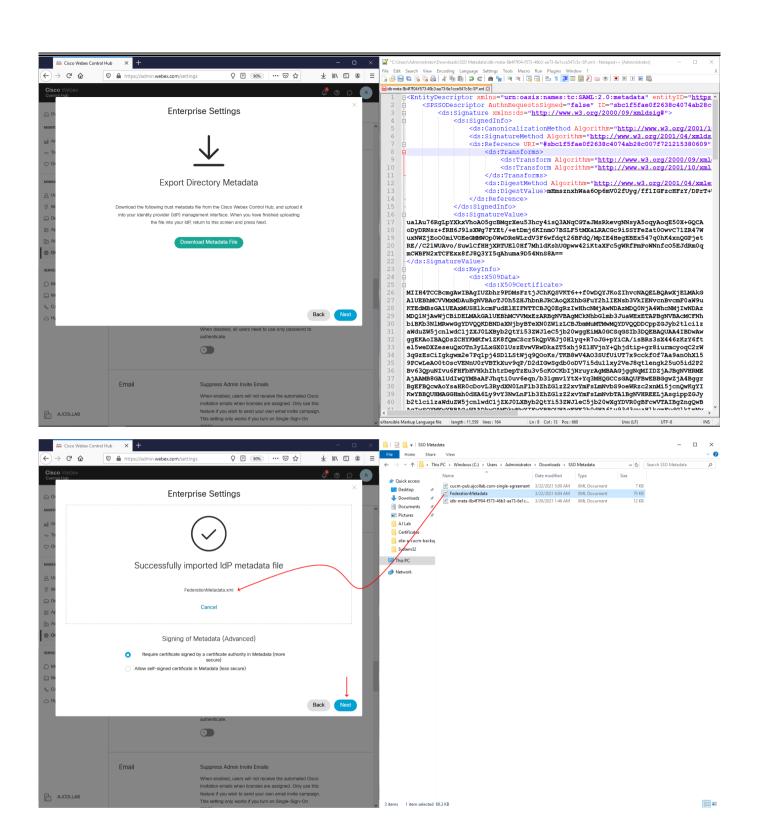
https://win-server-02-adfs.ajcollab.com/FederationMetadata/2007-06/FederationMetadata.xml

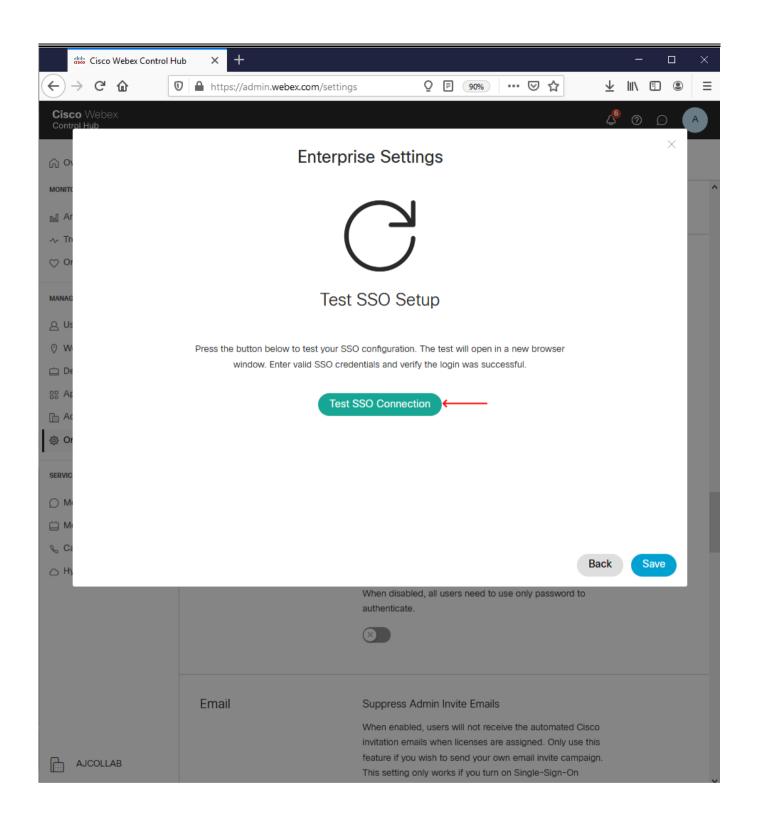


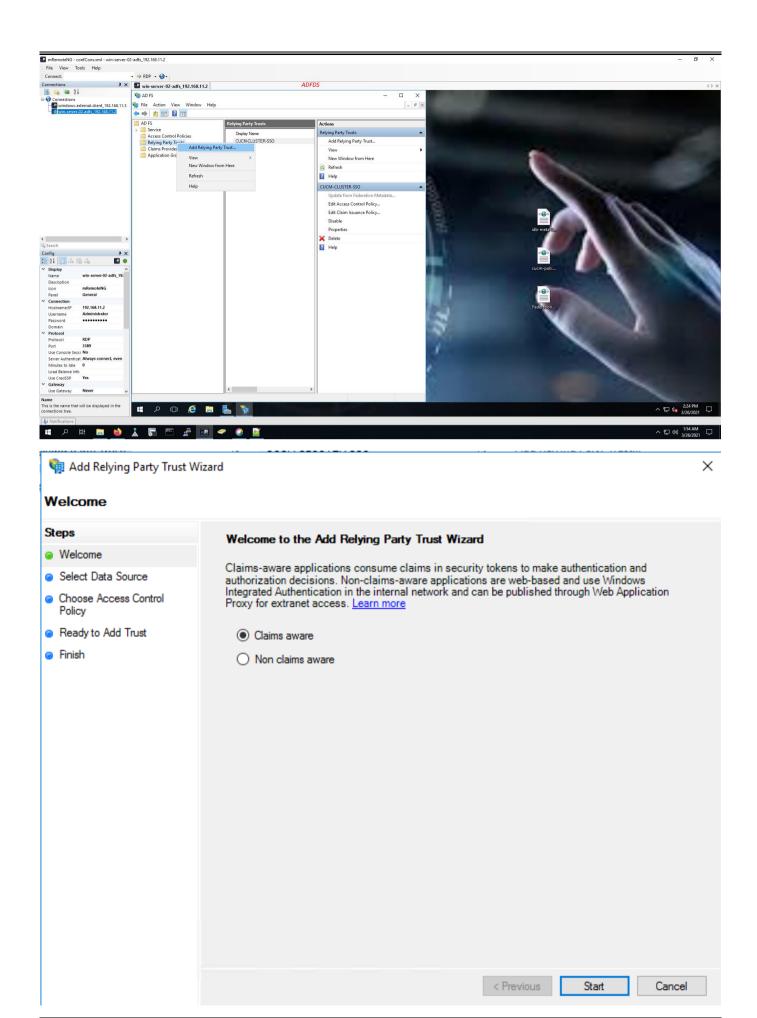


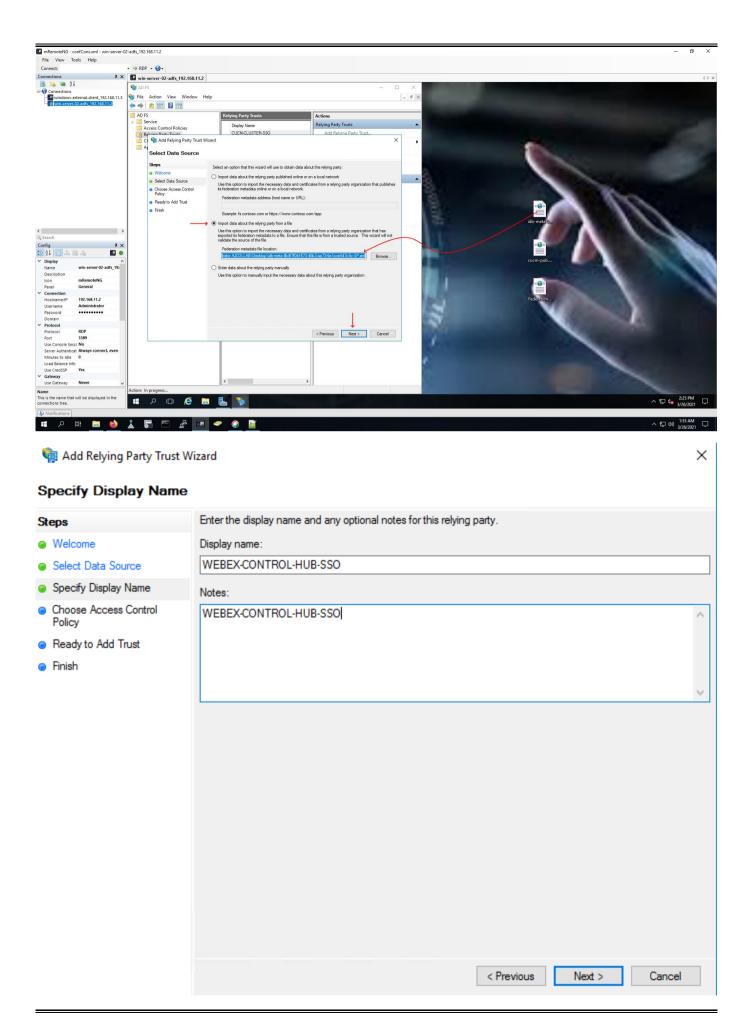


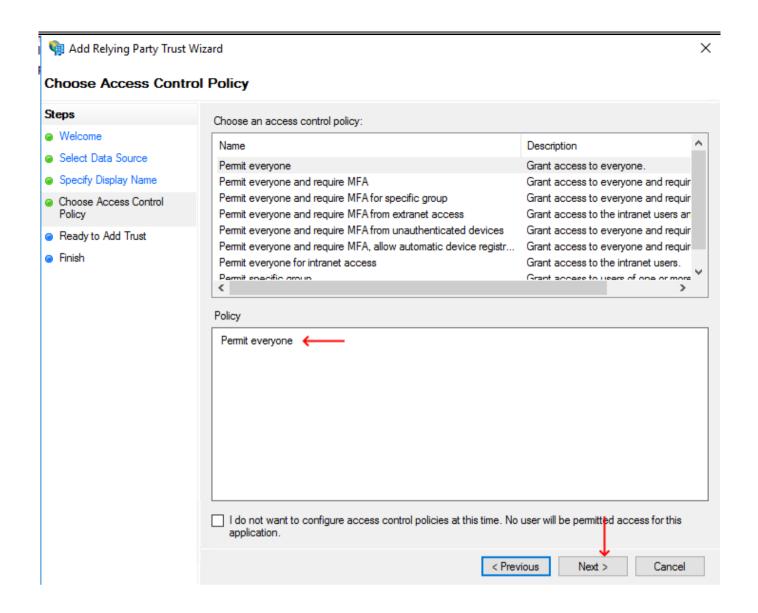


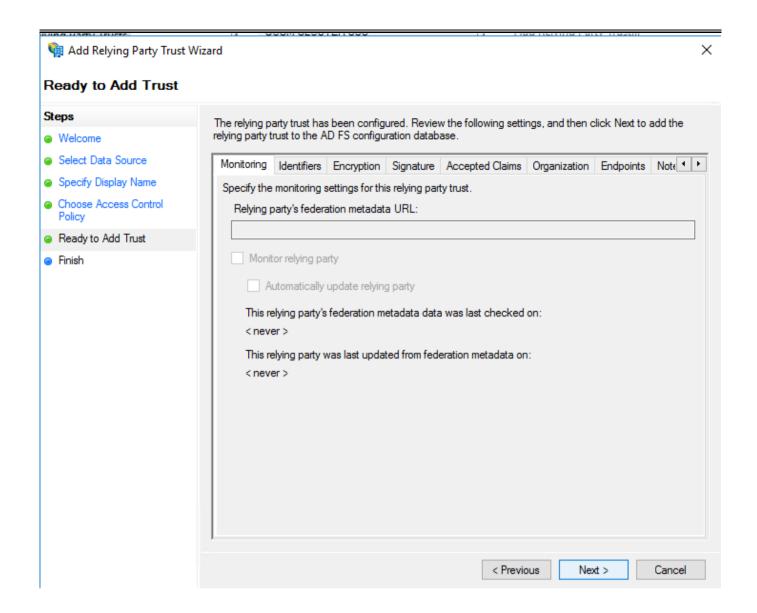








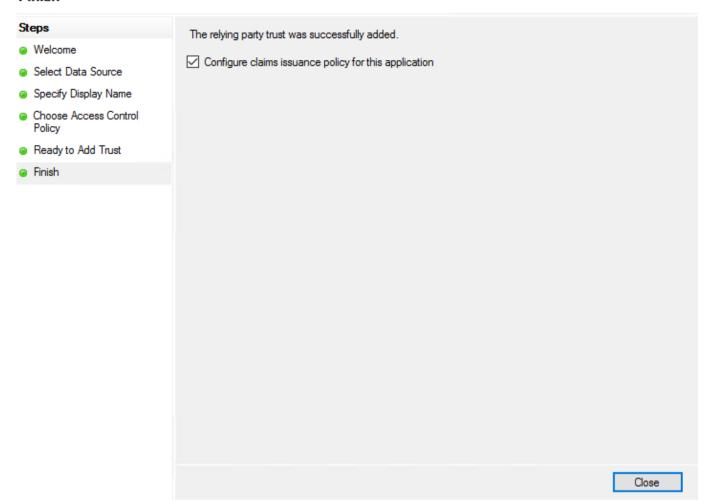


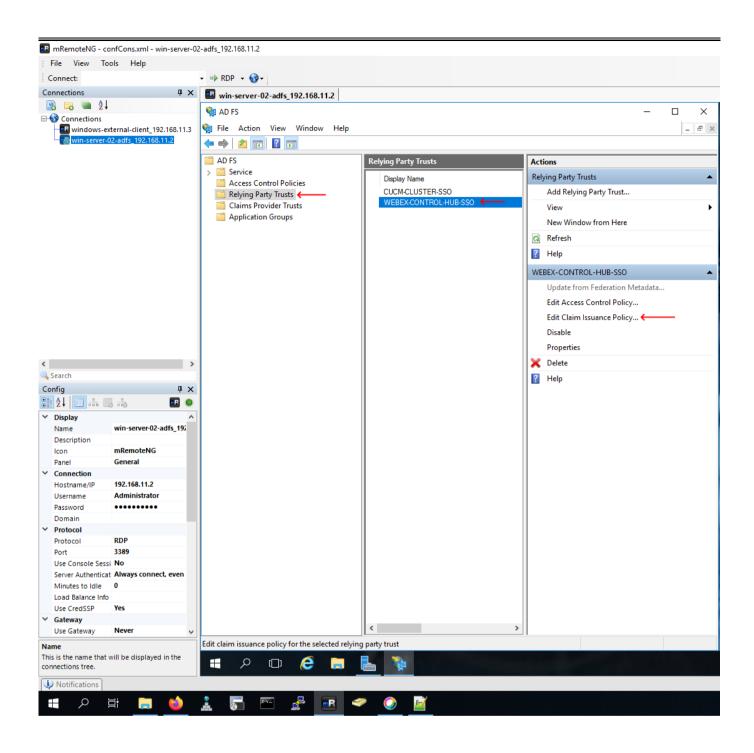


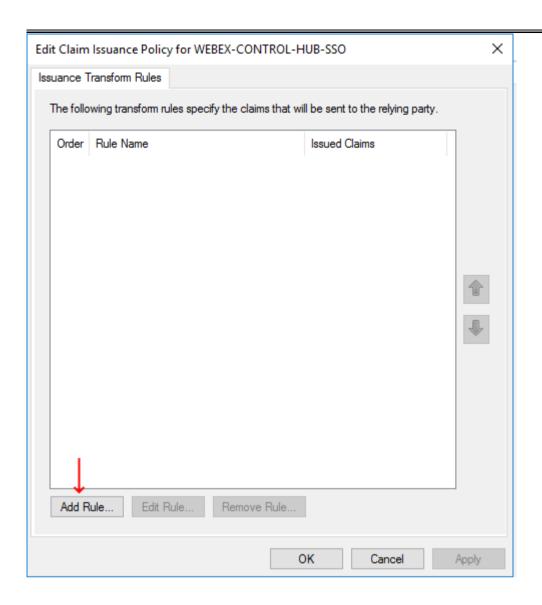


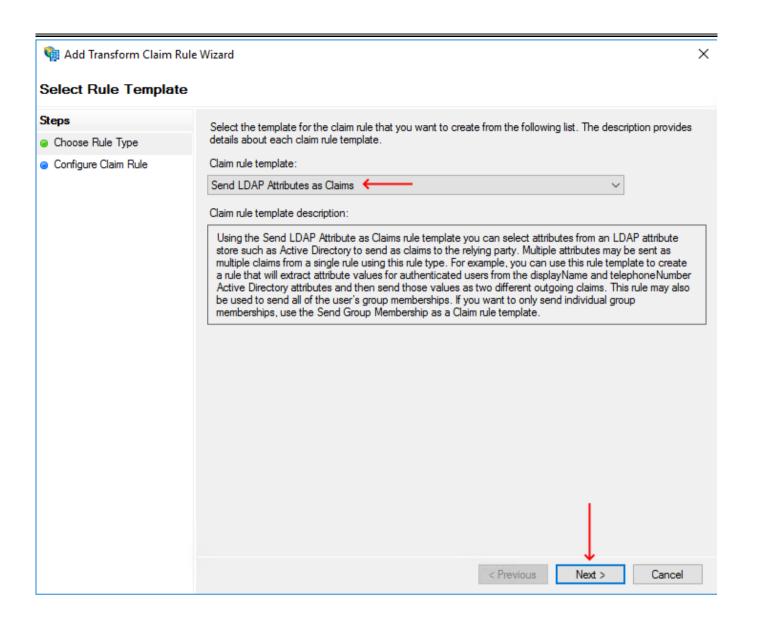
×

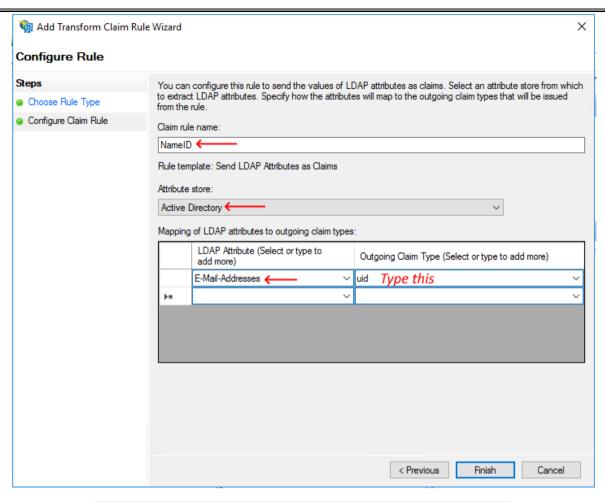
Finish

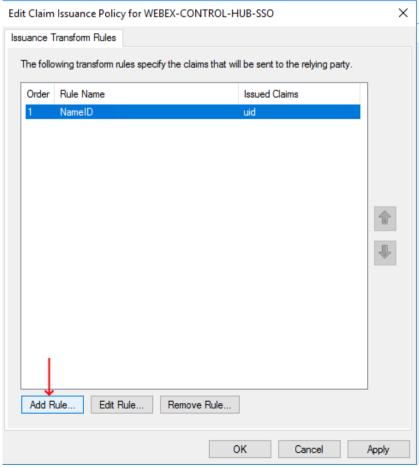




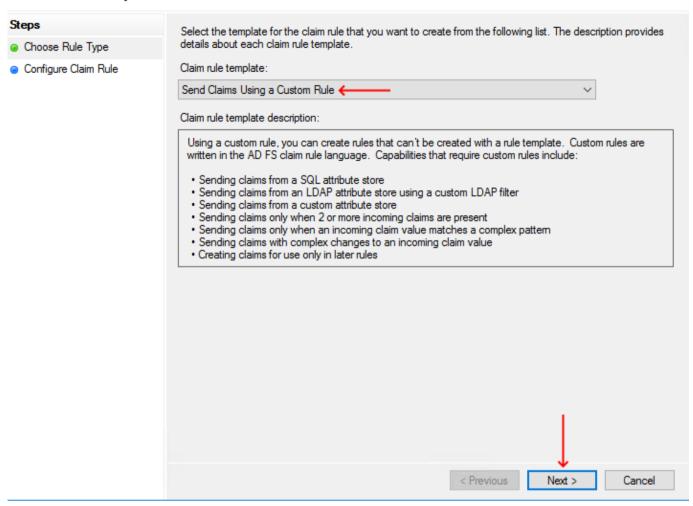


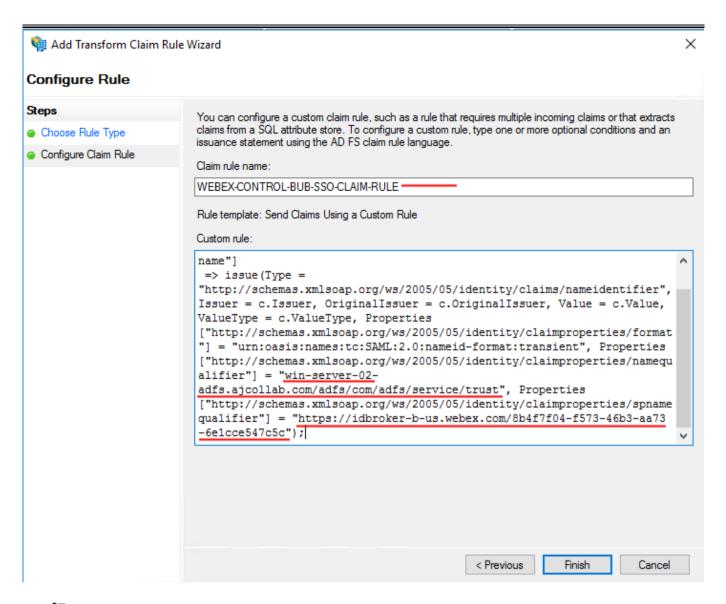






Select Rule Template





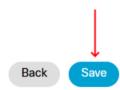
Note: The idbroker URL is available on the Webex metadata

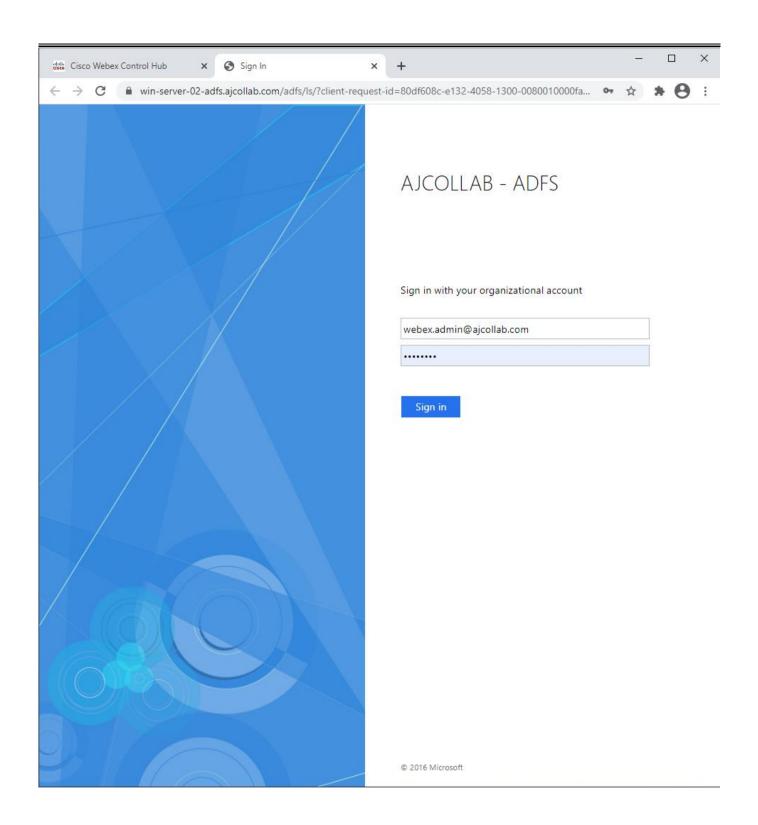
Enterprise Settings

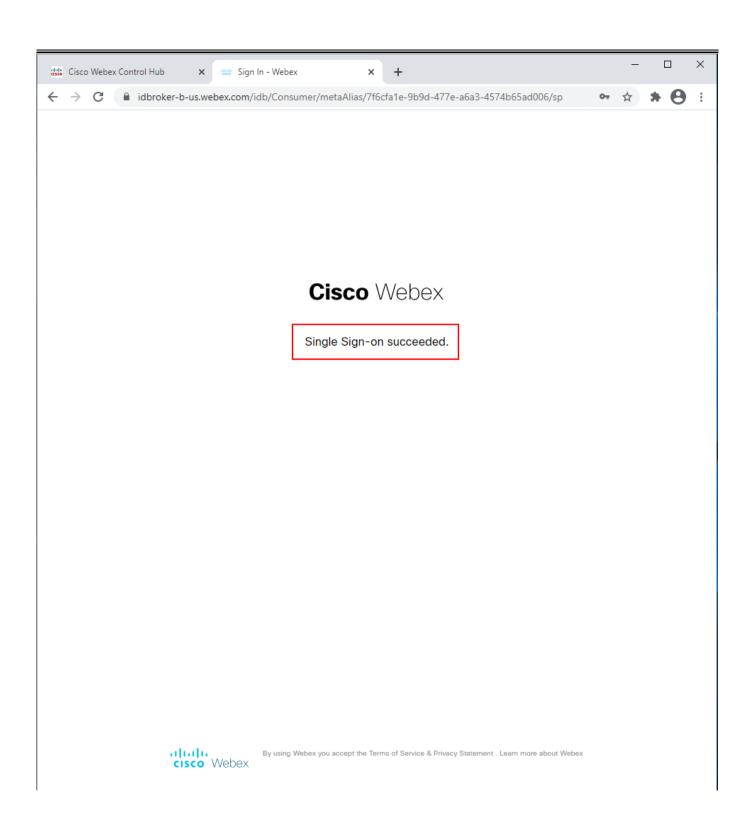


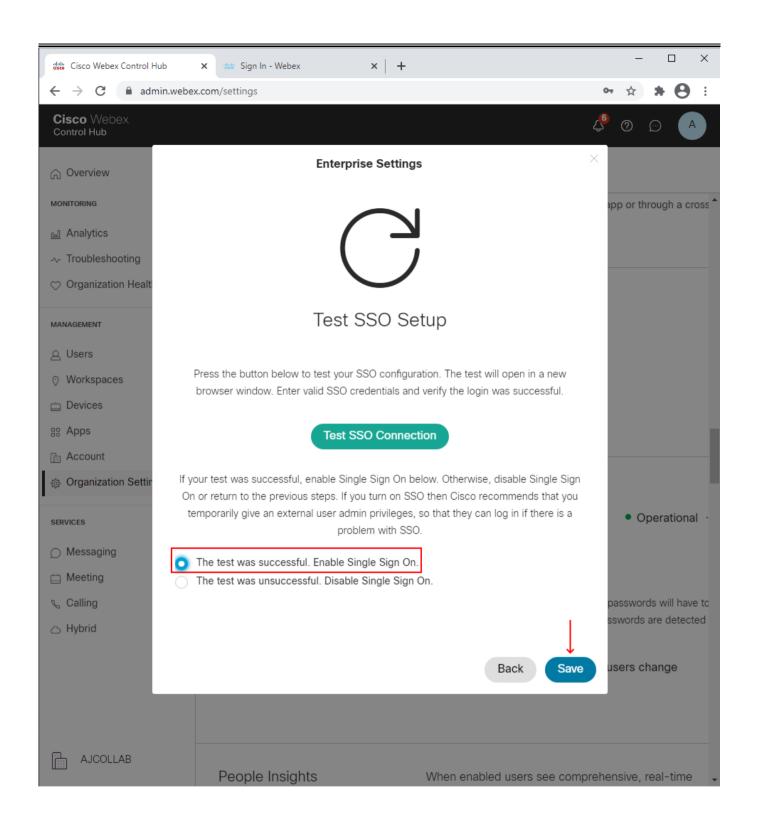
Press the button below to test your SSO configuration. The test will open in a new browser window. Enter valid SSO credentials and verify the login was successful.

Test SSO Connection ←

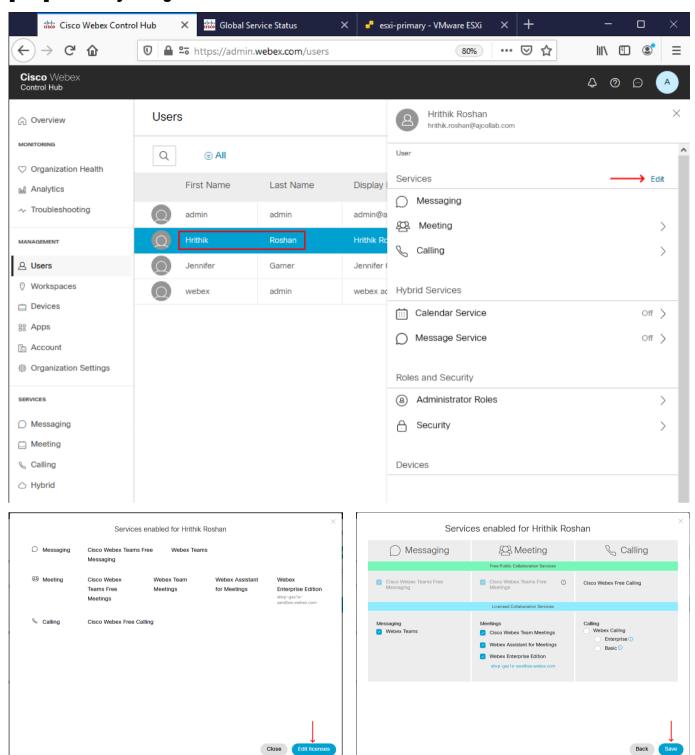








[Lab] Manually Assign License to the Users

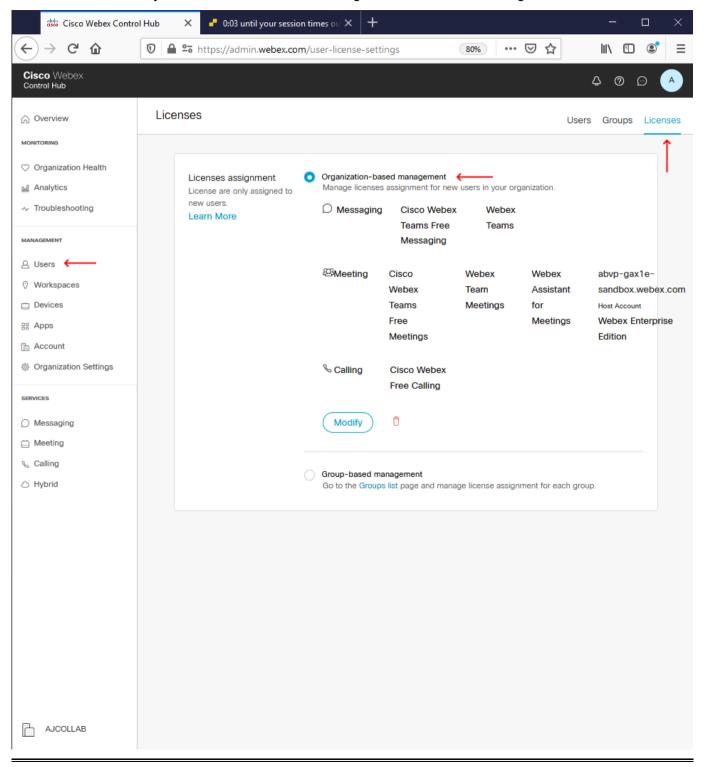


Auto License Assignment

- License Template is created and assigned to AD group
- When users Sync from a specific AD Group, those get assigned a pre-defined License Template
- We can edit / modify / disable the license template anytime

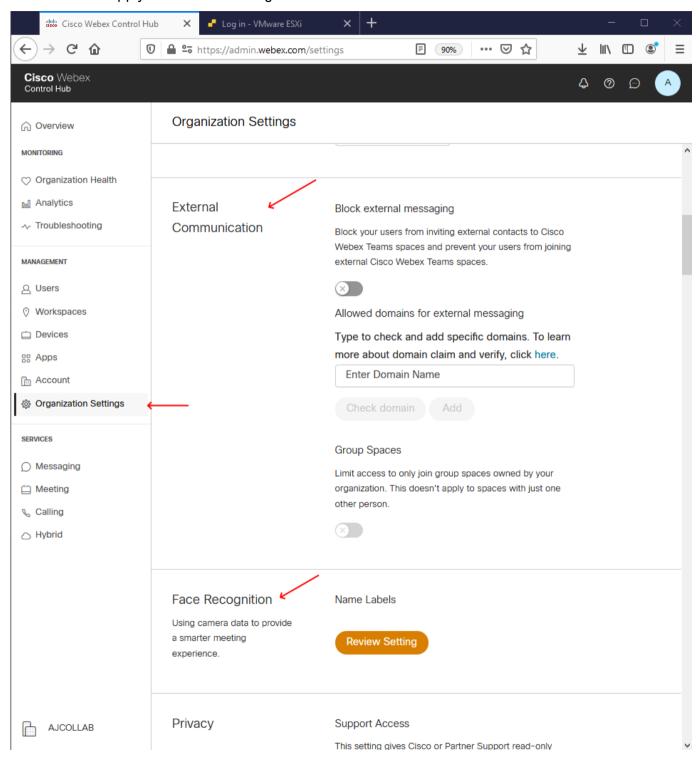
Organization Based License Assignment

- Assign license to the users automatically from at the Organization Level
- You can modify individual users once the assigned licenses from the Org

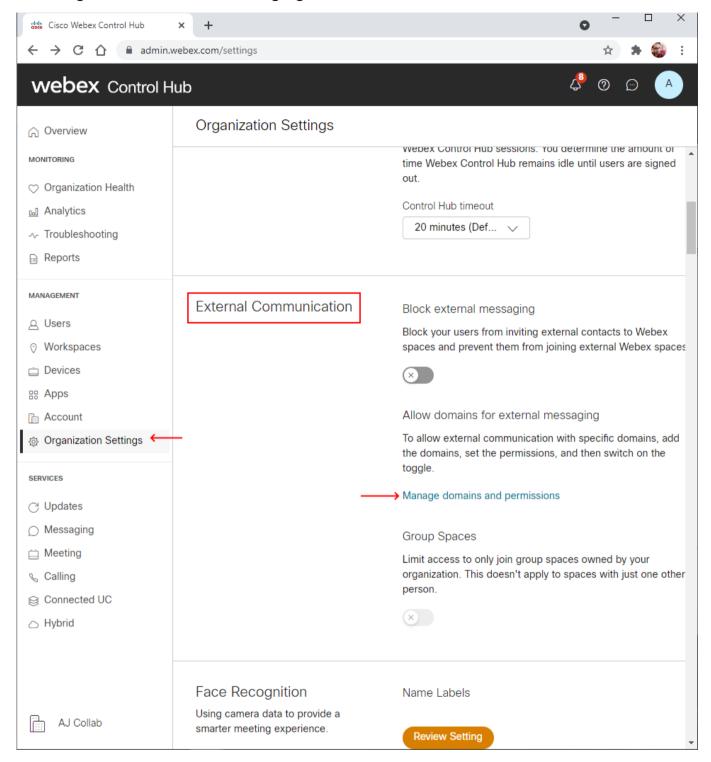


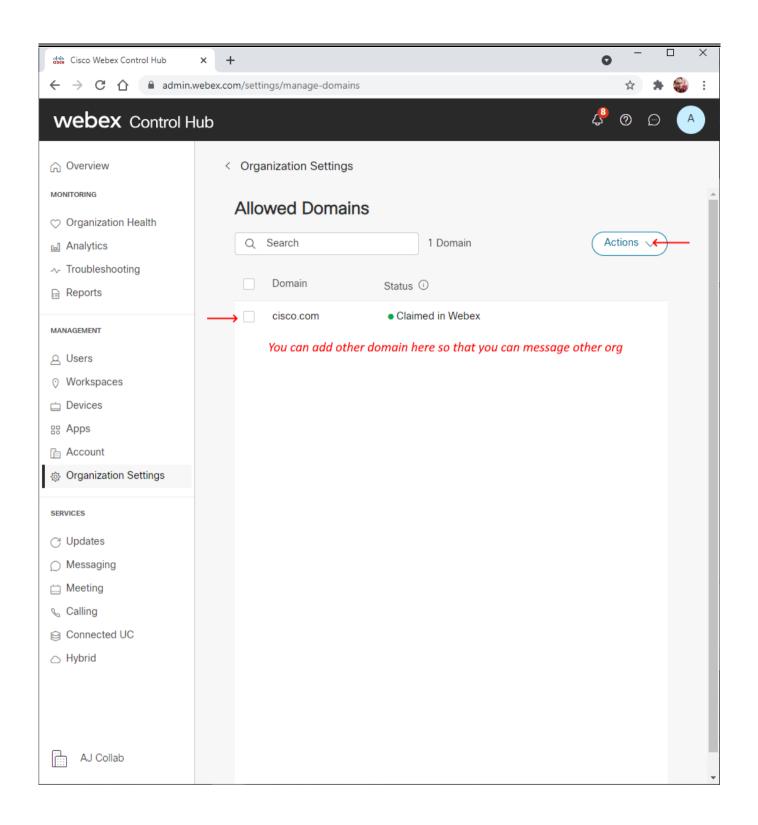
Policies and Features

- Block external communication, File Share control, Mobile PIN Lockout, Webex Assistant, Face recognition, etc.
- We can apply at User level or Org level



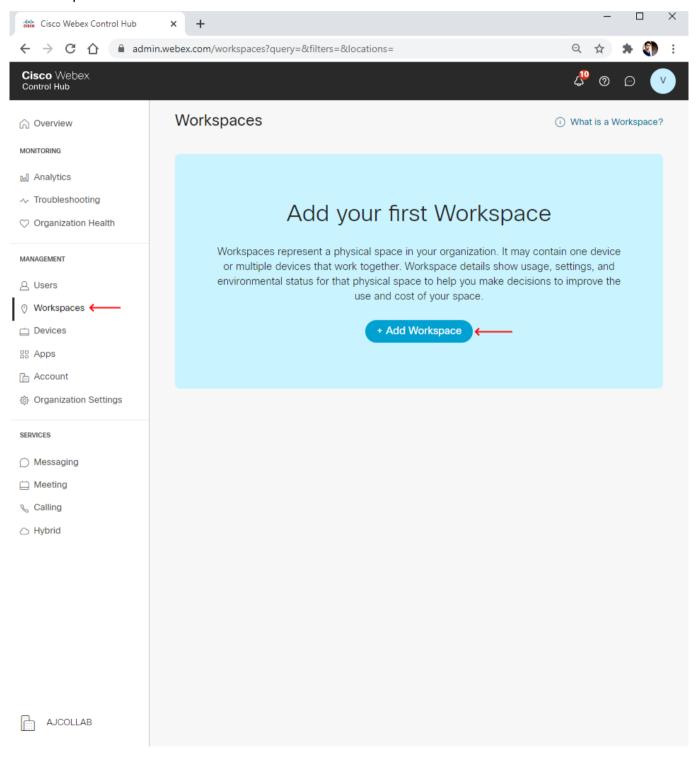
Enabling External Domain Messaging

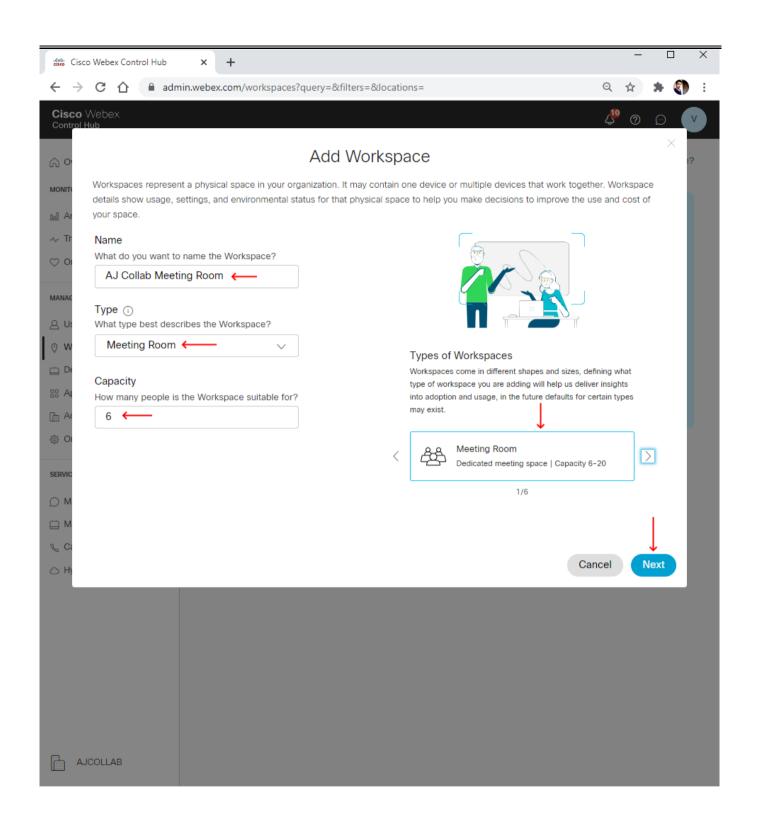


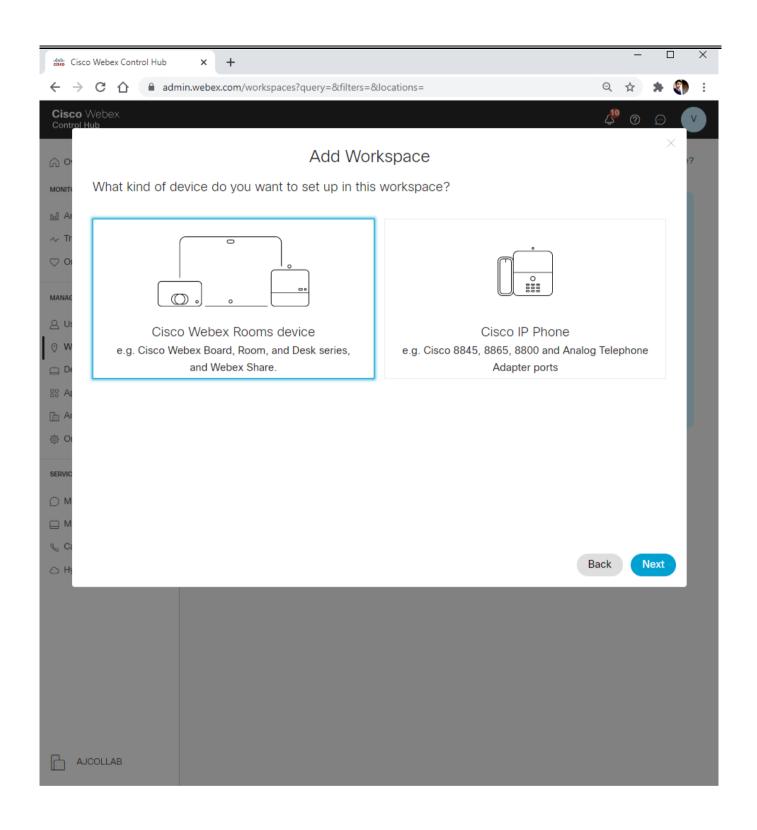


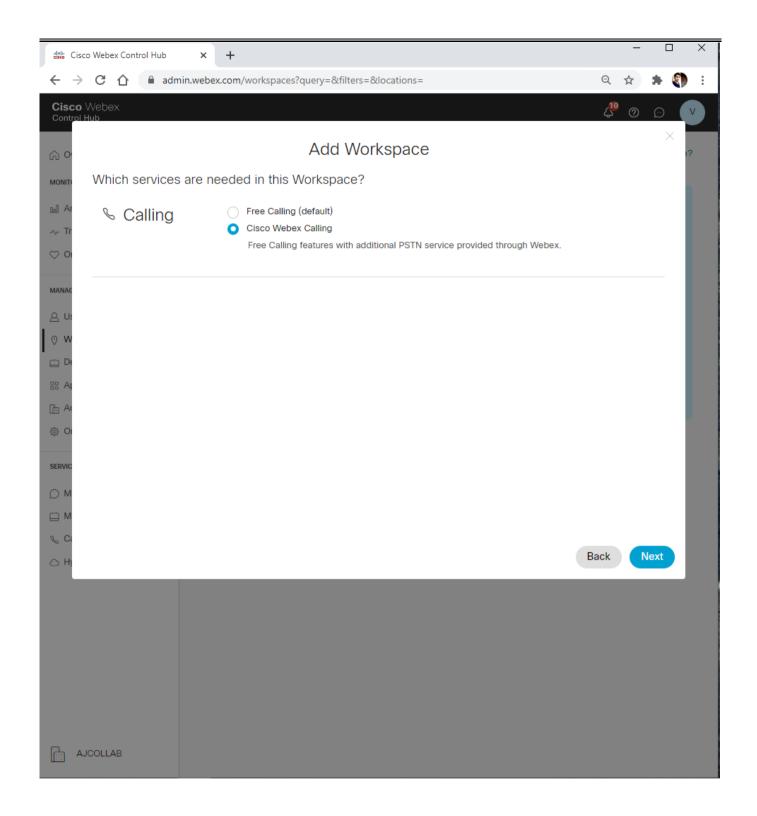
[Lab] Workspaces

- Workspaces are the physical locations of collaboration within your organization. Workspace details
 give you an at-a-glance overview of the usage, settings, and environmental status for the physical
 location
- This helps you understand the workspace conditions to make decisions that enhance the end-user experience

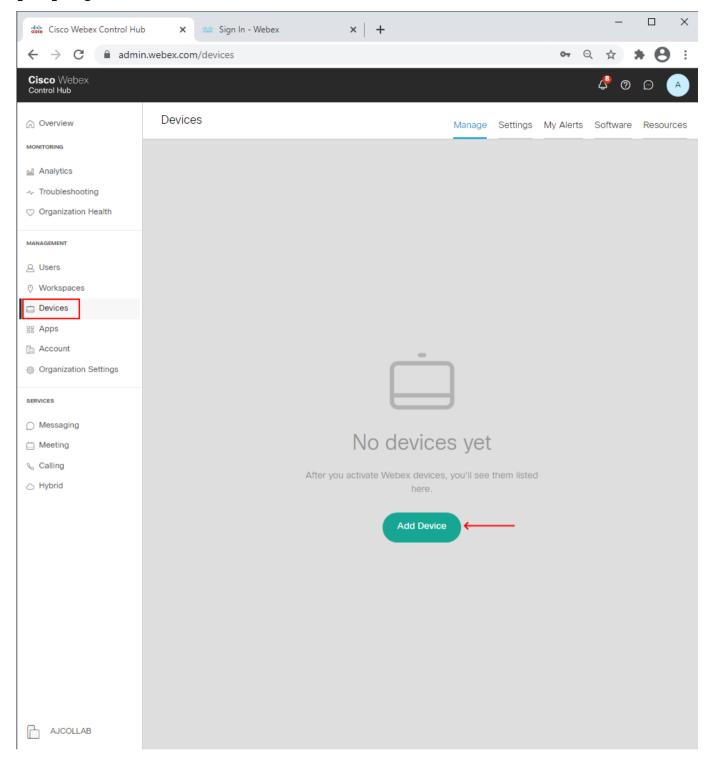


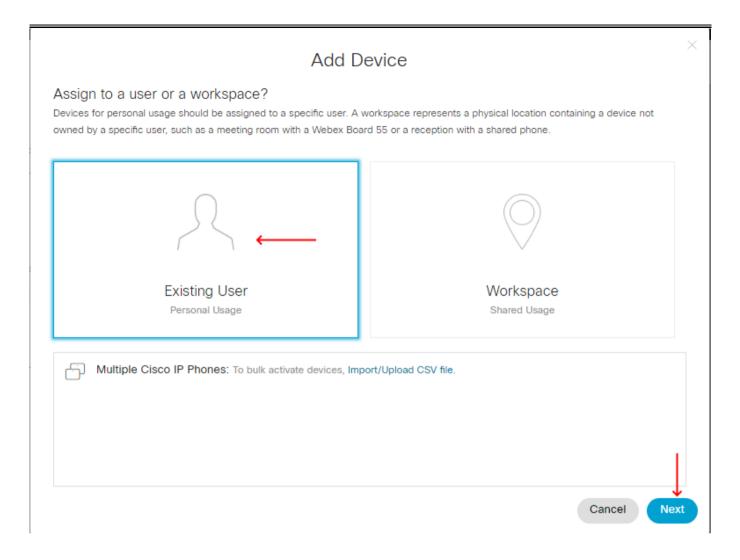






[Lab] Register Cisco DX70 in Cloud





Add Device

Which user will this device belong to?



Back

Next



Add Device

Activation Code

An Activation Code is a **one time password**. Share this code with the person setting up the device belonging to **Hrithik Roshan**, or enter or scan it yourself when prompted by the device. Once the device has been successfully activated, you will be able to find and configure it in Users or Devices.



0897-0704-2661-6729

Expires April 3, 2021 12:52 AM (PDT)

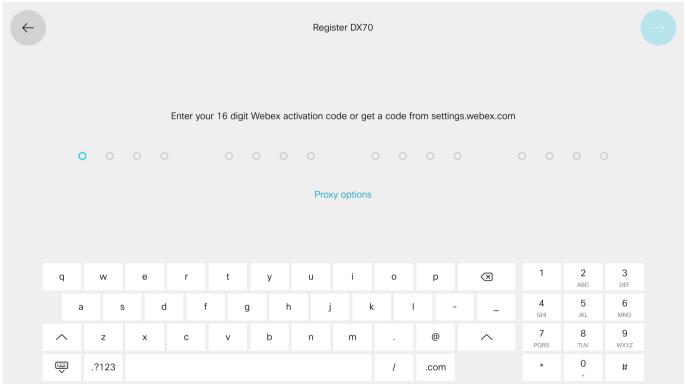




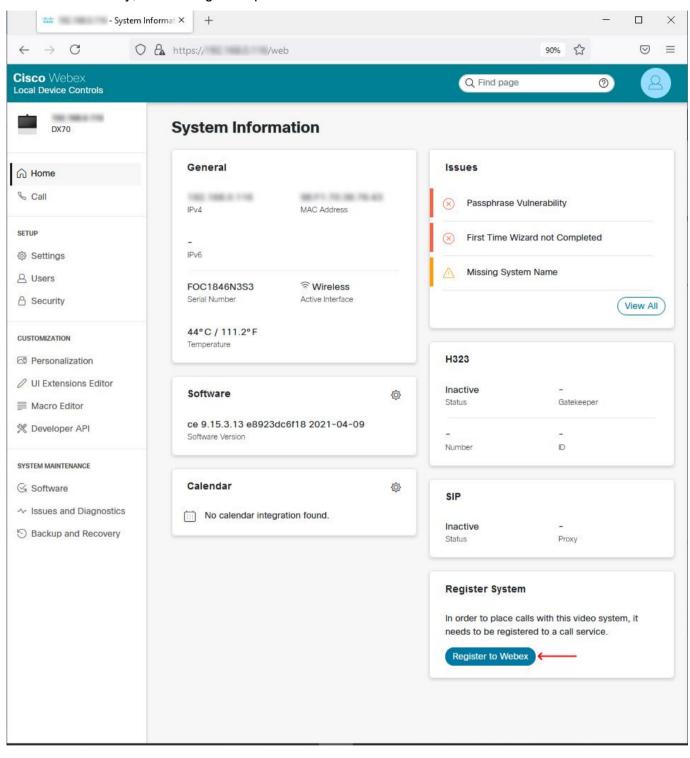


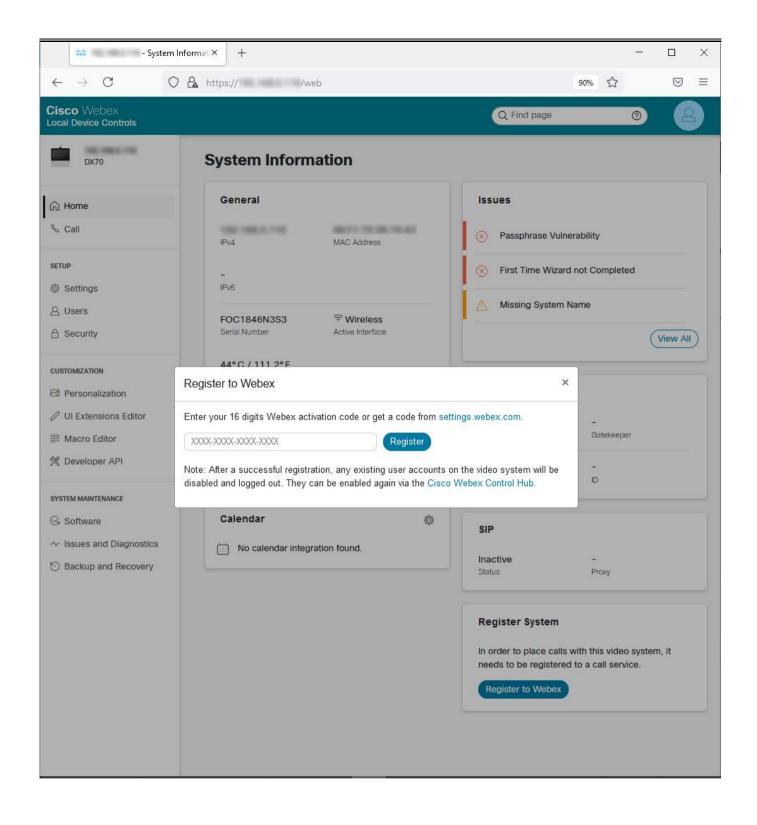
- Now restore factory the DX70
- You can enter the 16-digit code when the DX70 comes online by selecting Webex option from the device

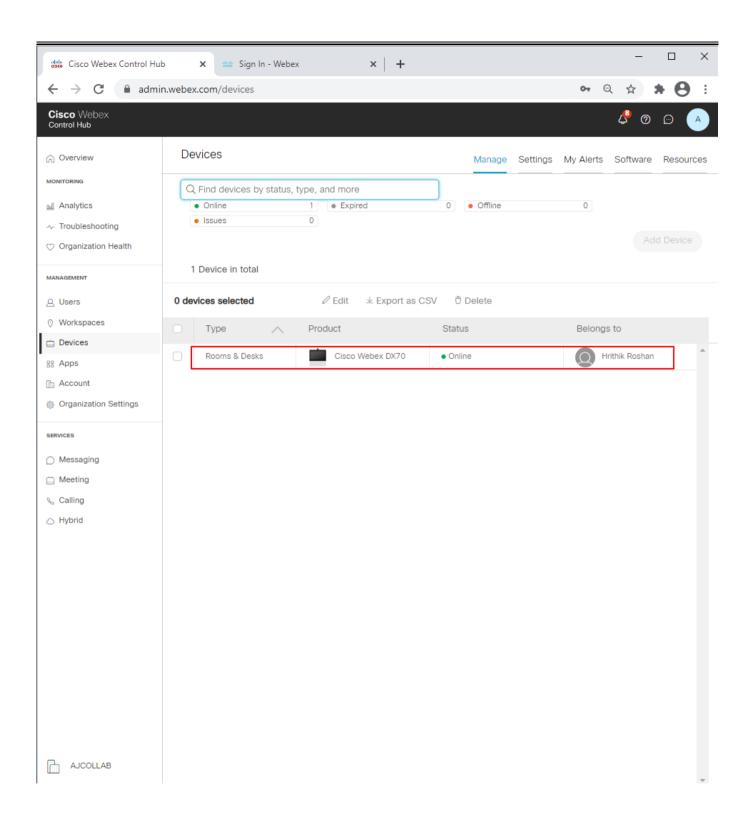


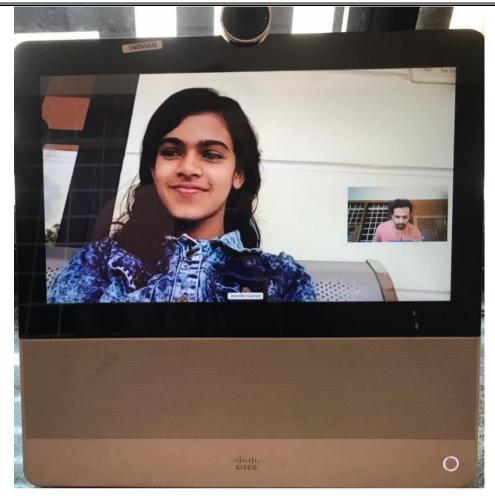


• Alternatively, use the register option from the DX70 web interface









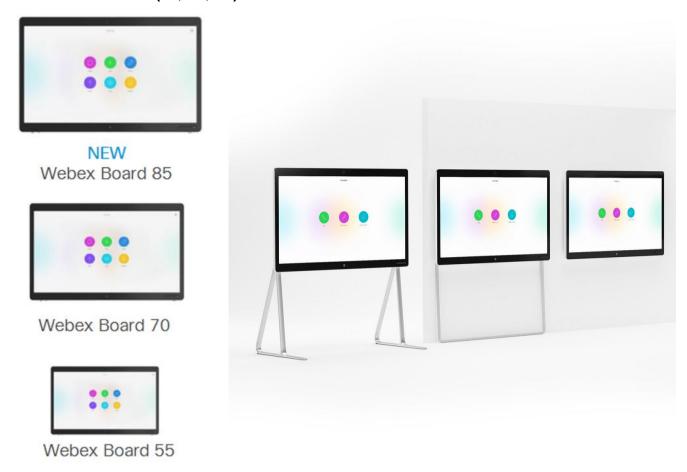
Note: Registering Cisco 8865 Phones will follow the same steps, phone must be in MPP firmware

Webex Devices



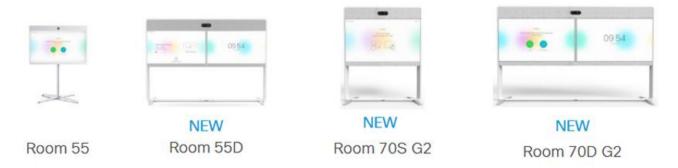
- Webex devices are broadly categorized into three such as Board devices, Room Devices and Desk devices
- Board devices used is an all-in-one touchscreen enabled and white boarding capable device
- Room devices are of 2 types: with integrated display devices called Room Series and without integrated display called Room Kit series
- Desk devices can be used as an extended screen for your Laptop and provide unified communication features

Webex Board Series (55, 70, 85)



• Watch video demonstration: Cisco Webex Board

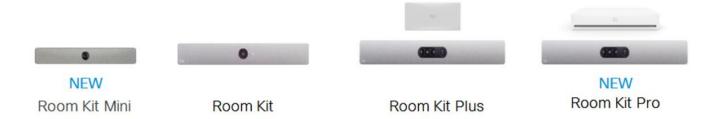
Webex Room Series



• Watch video demonstration: Cisco Webex Room Series

Webex Room Kit Series

• The display is not integrated, we can use any third party 4K display



Watch video demonstration: Webex Room Kit Mini

Cisco DX Series (DX70, DX80, Desk Pro)



Watch video demonstration: Cisco Webex Desk Pro and Cisco DX80

Cisco Telepresence SX Series (SX10, SX20, SX80)

• SX Series is now End of Sale, hence you won't see that in future



Video Demonstration: SX10

Webex Share

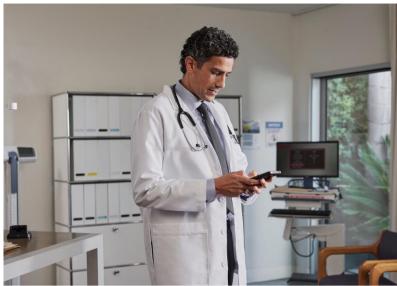
- Used to share content, screen wirelessly to any display having HDMI connectivity
- This device must be provisioned in the control hub



Watch video demonstration: Webex Share

Webex Calling

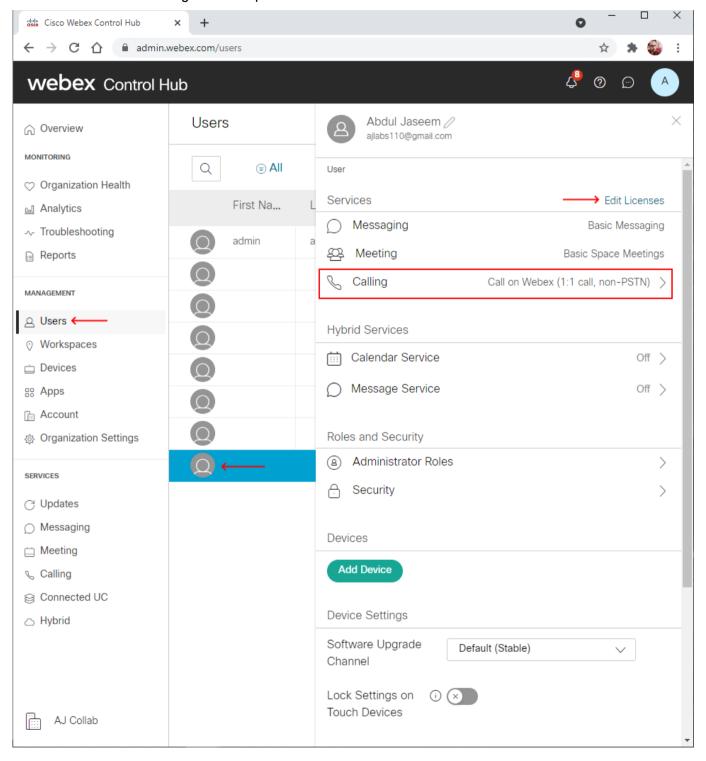


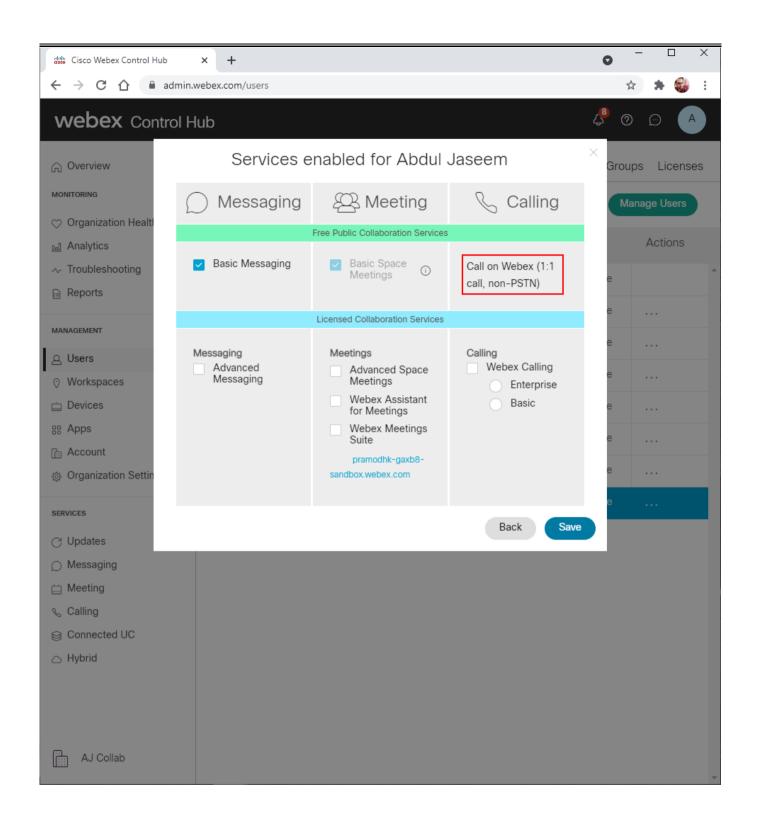


- There are basically three types of calling in Webex
- One to One (1:1) non PSTN URI calling (Default)
- Make calls directly from Webex, backed by Webex Calling (broadcloudpbx) with extension and PSTN DID support
- Unified CM Calling or UCM Calling Webex Cloud integrated with on-premises CUCM

1:1 Call Non PSTN

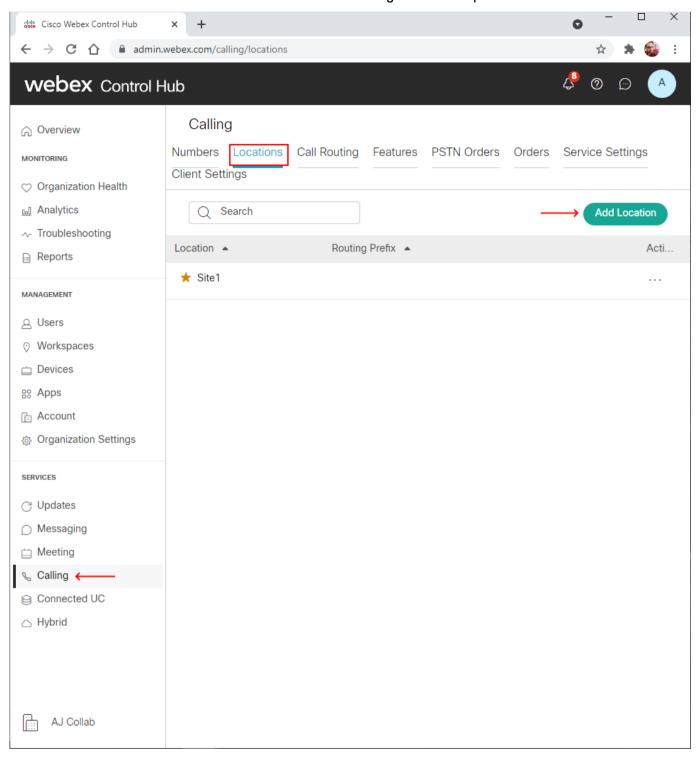
- This is the default calling option available in Webex
- Users can call using their email ID
- No additional configuration required

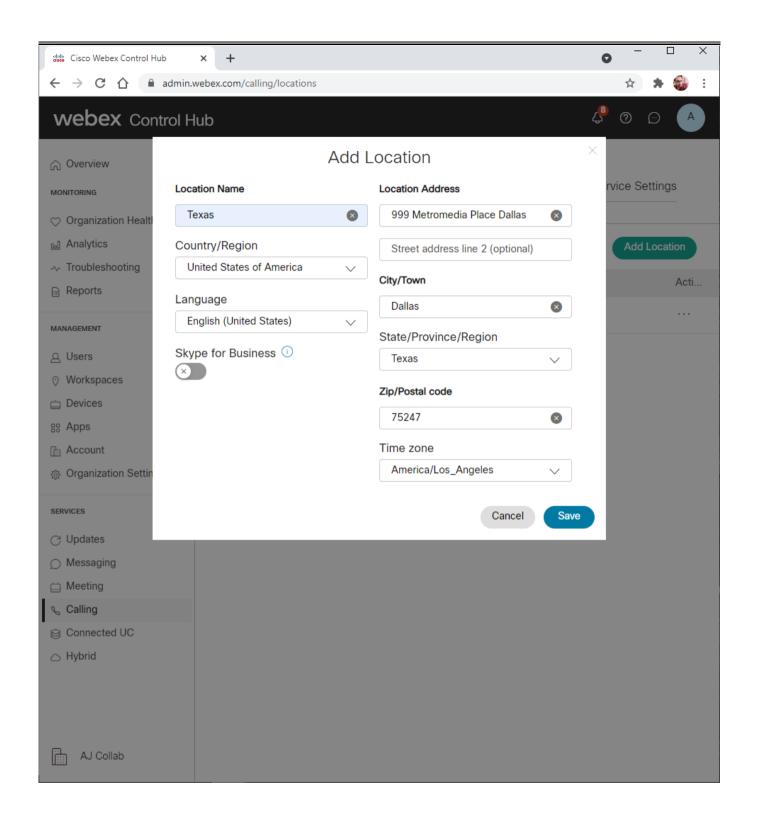


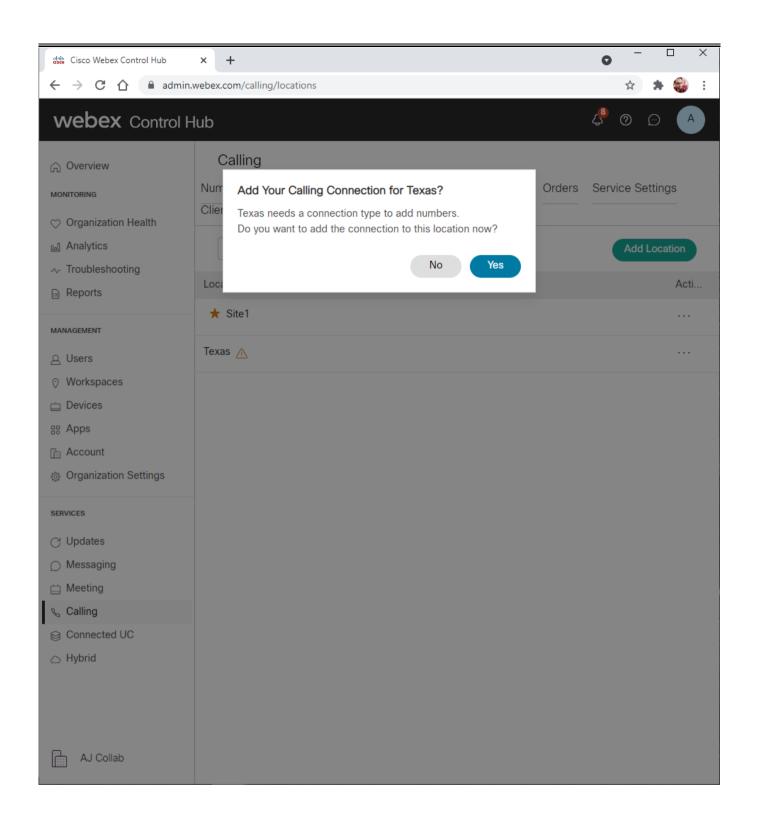


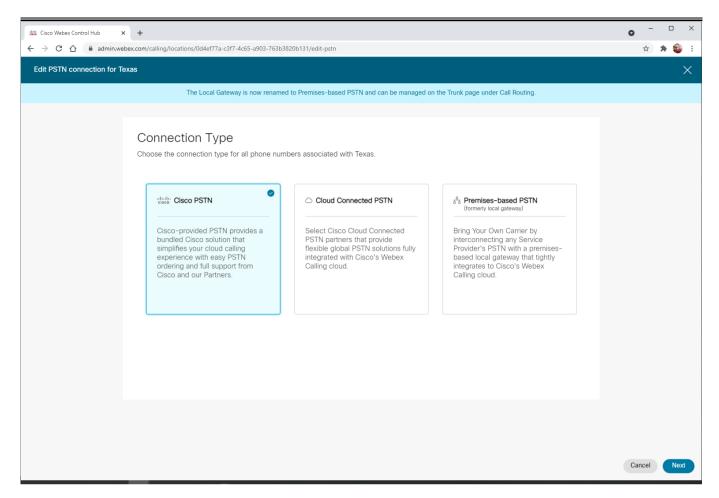
Enable location for Webex Calling powered by broadcloudpbx

We need to enable locations to enable Webex Calling broadcloudpbx



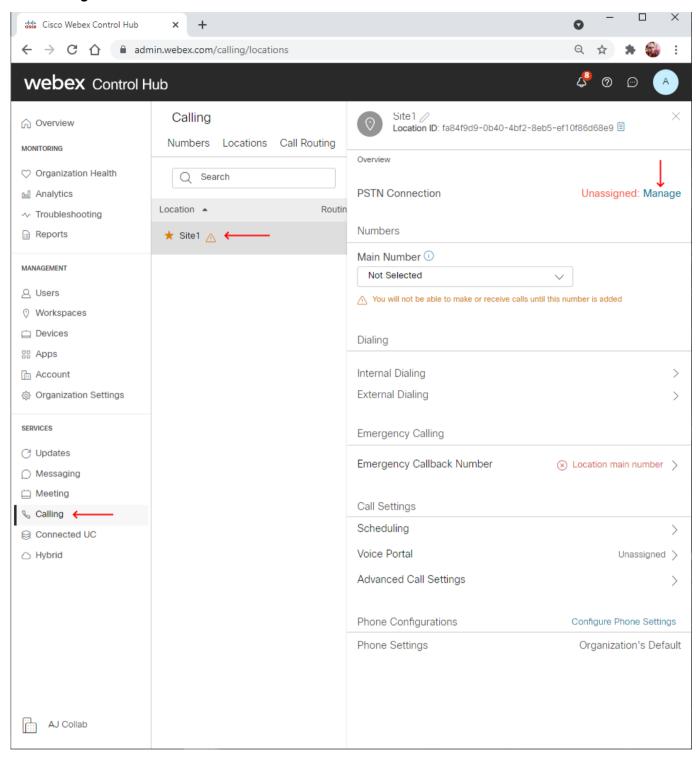


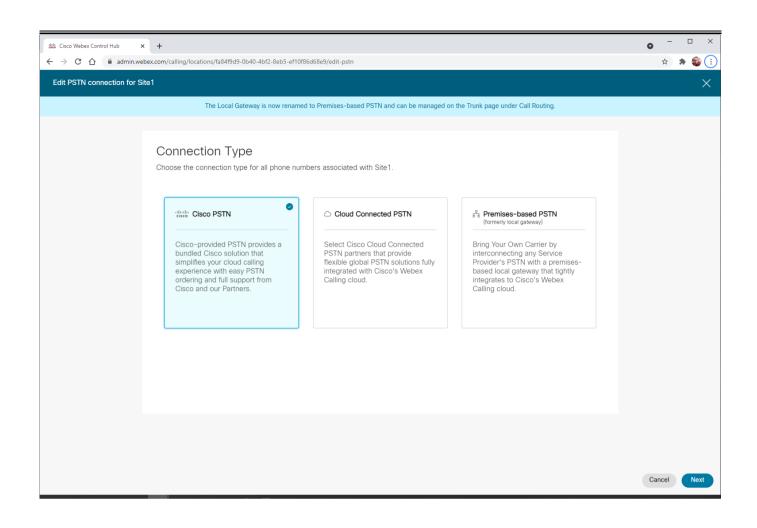


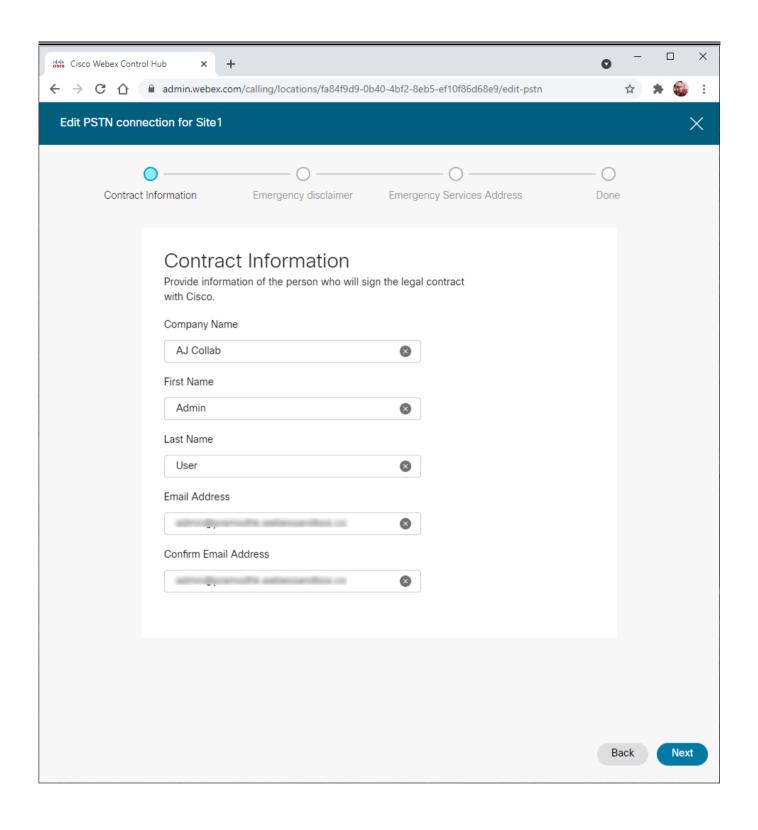


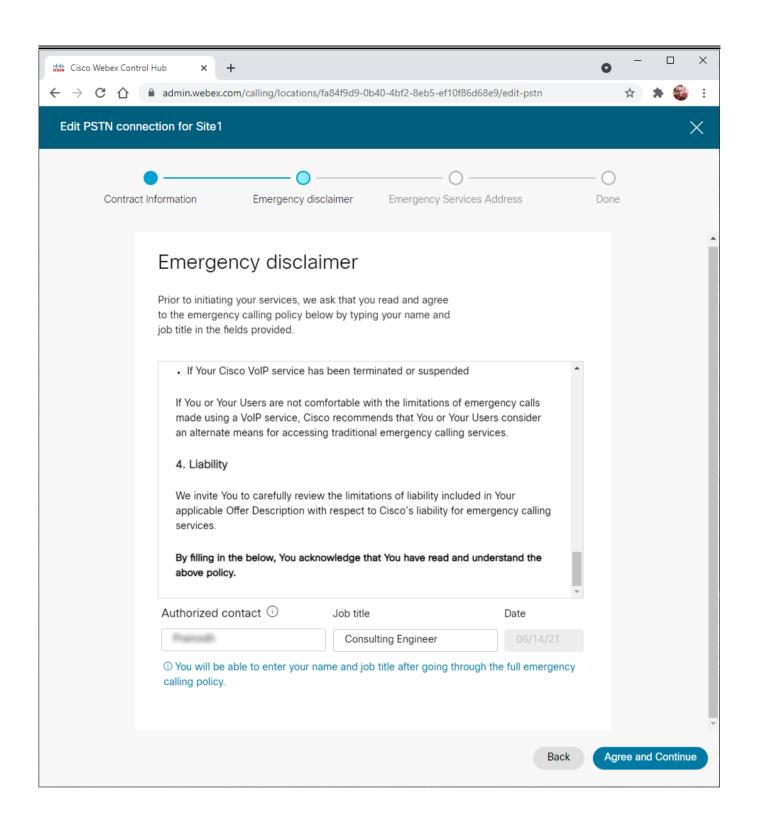
· You can select a PSTN option and proceed

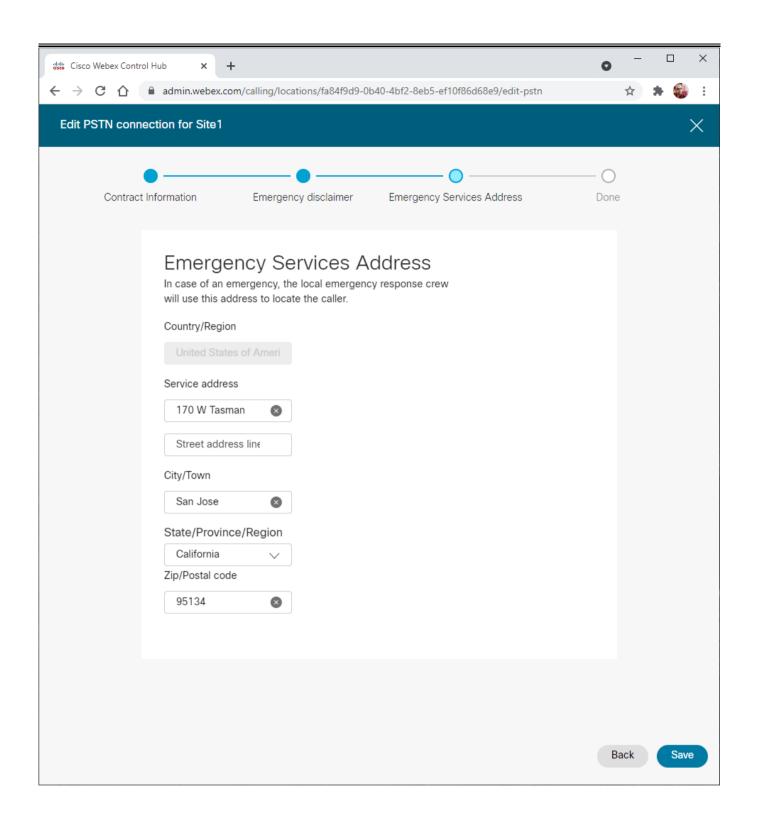
Connecting PSTN to a Site



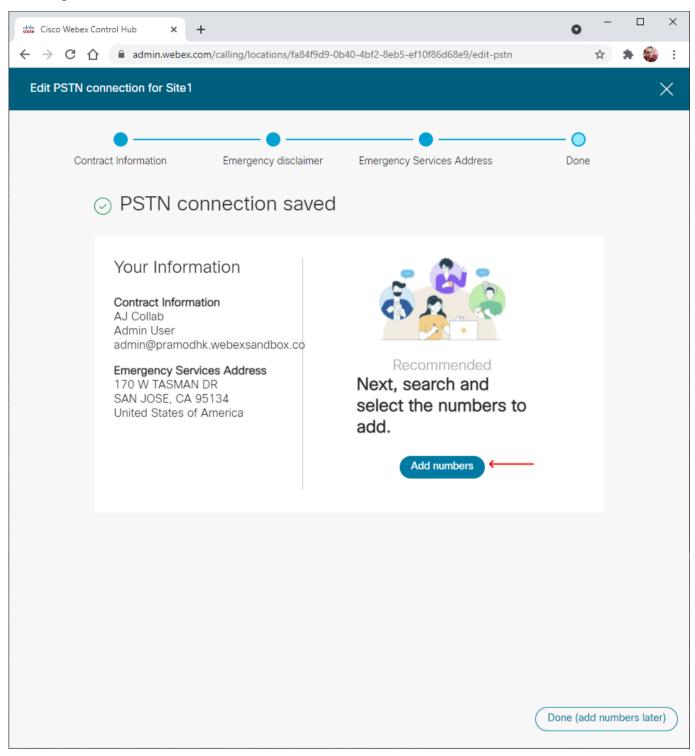


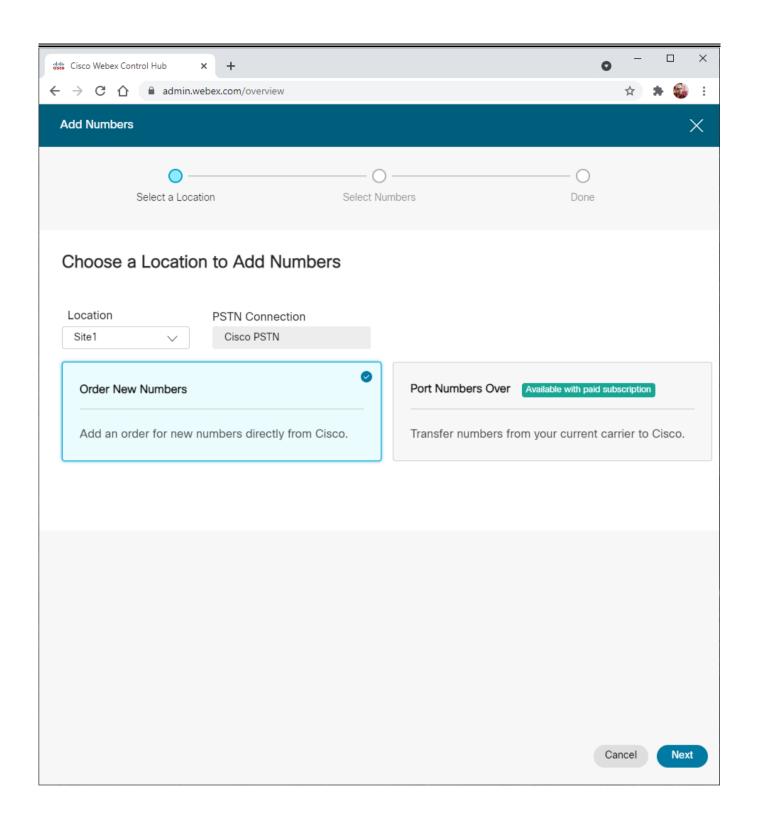


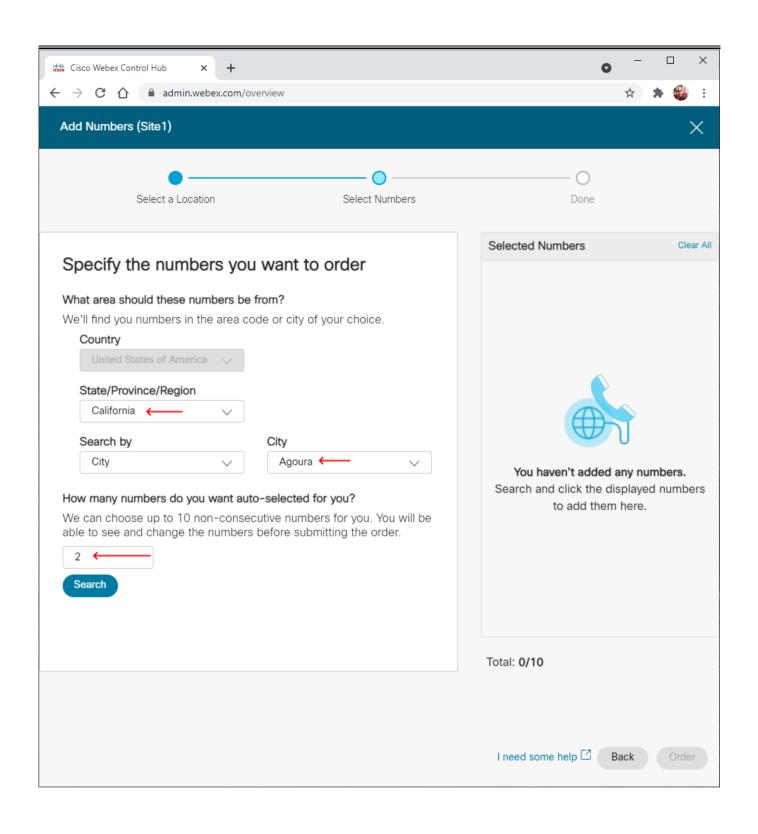


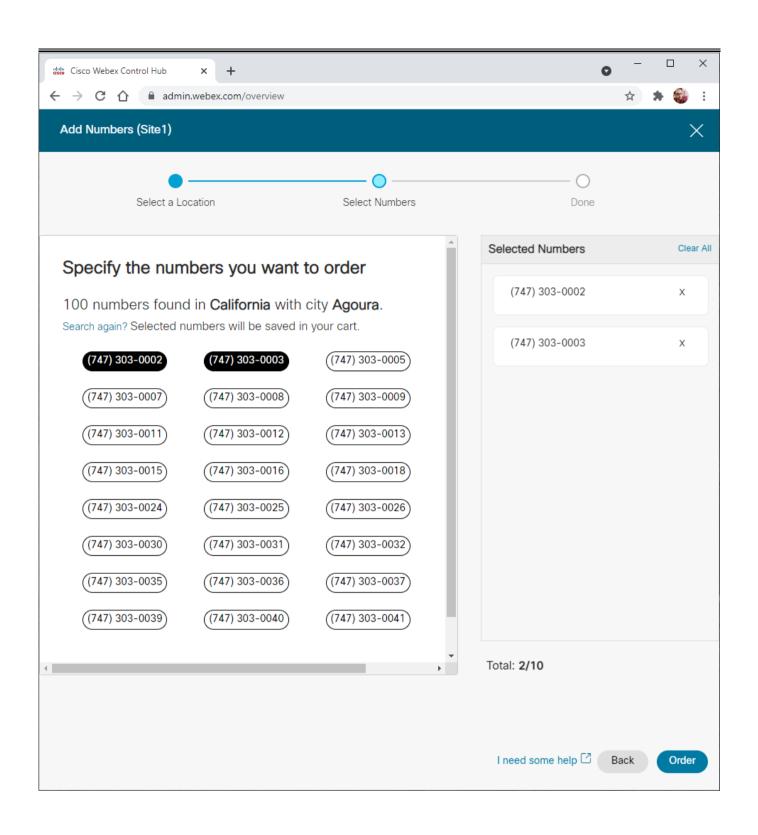


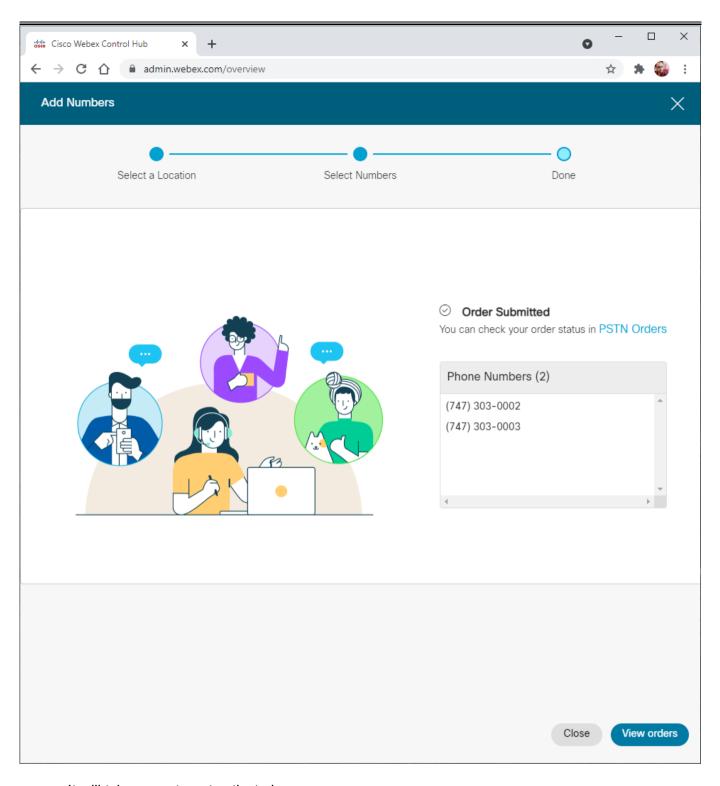
Ordering DID Number from PSTN





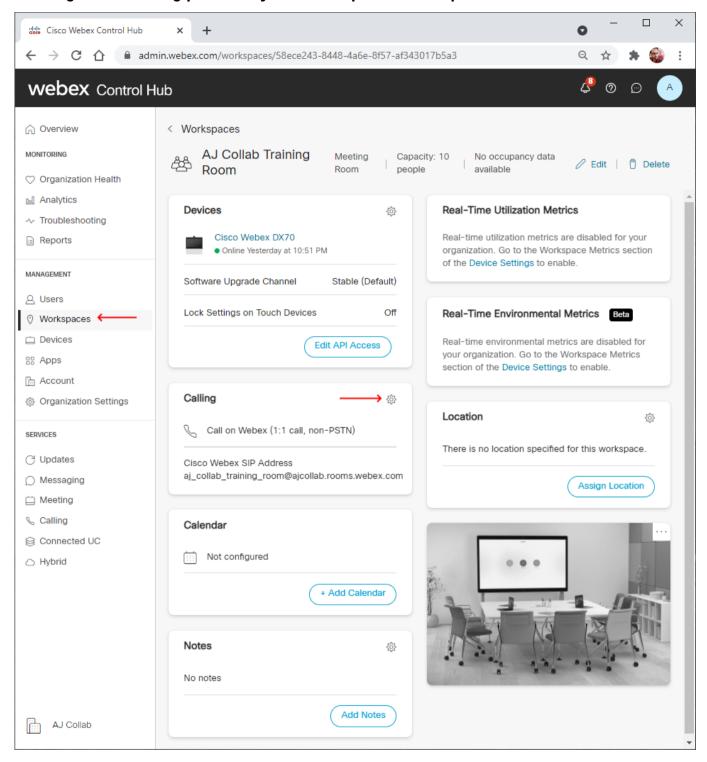


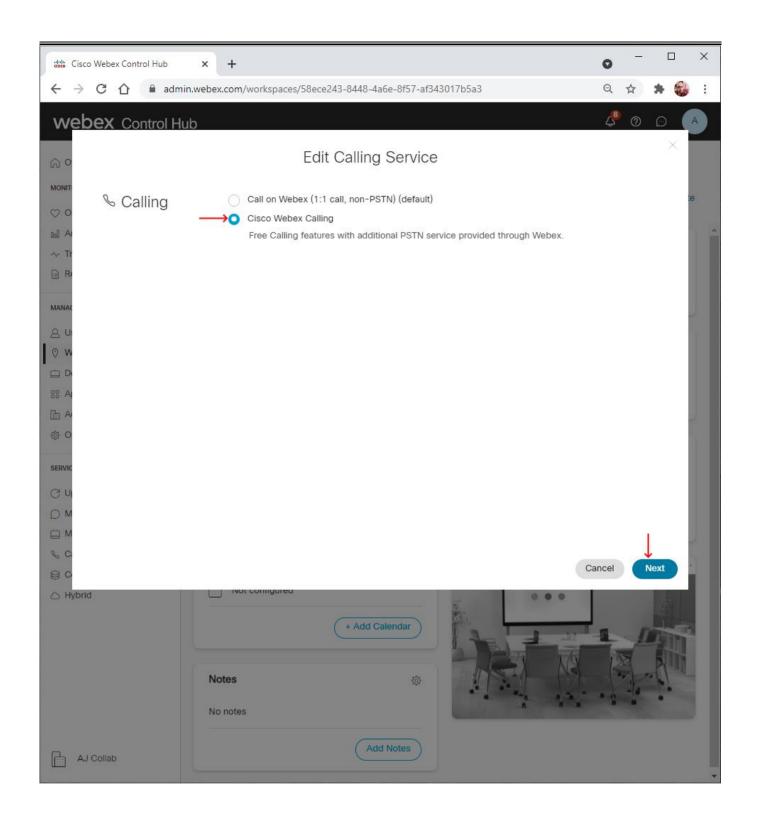


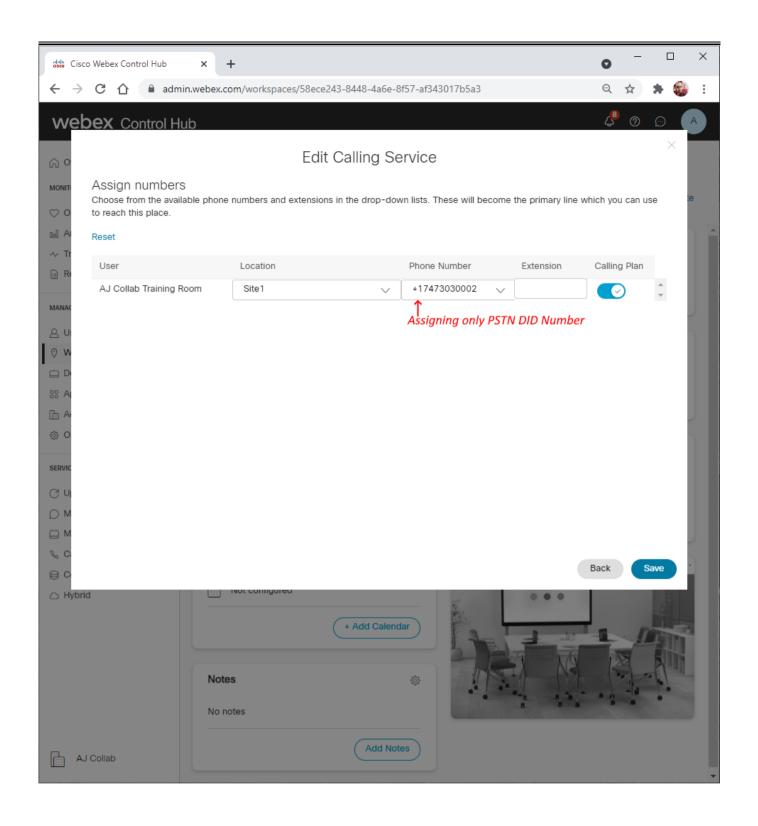


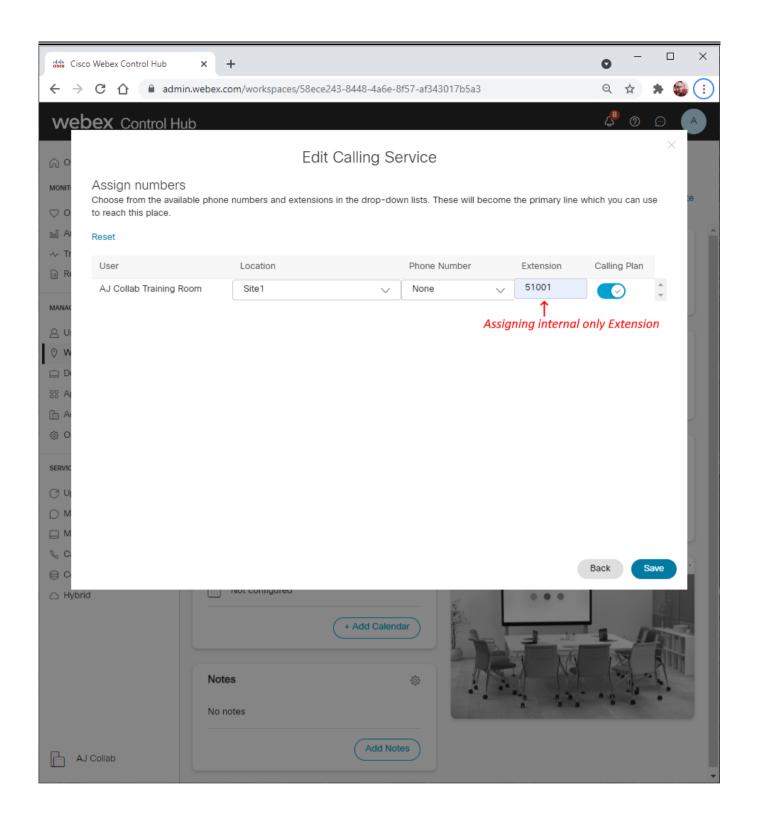
• It will take some to get activated

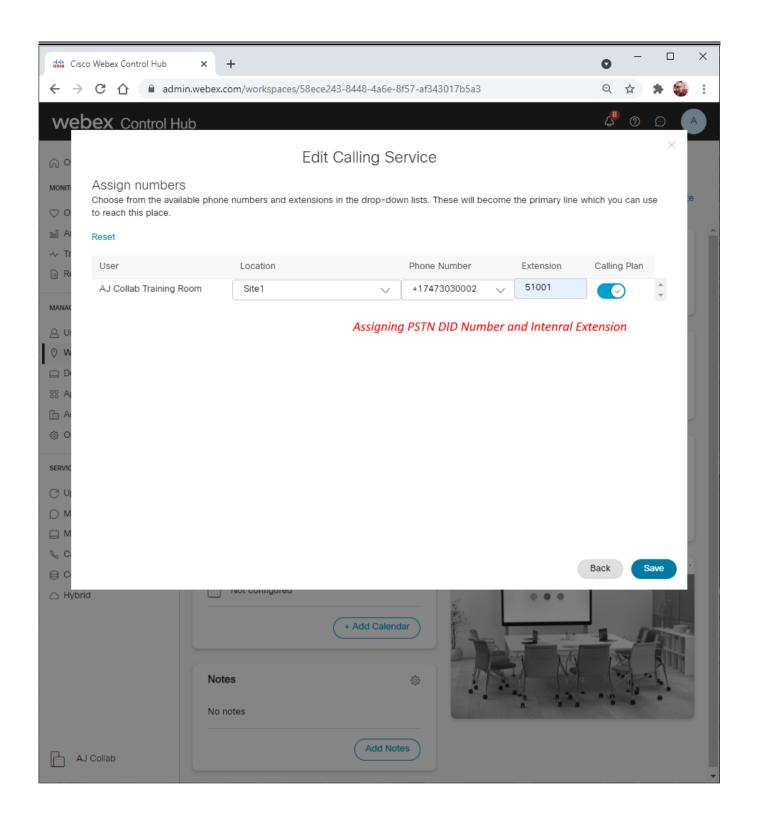
Enabling Webex Calling powered by broadcloudpbx for Workspace



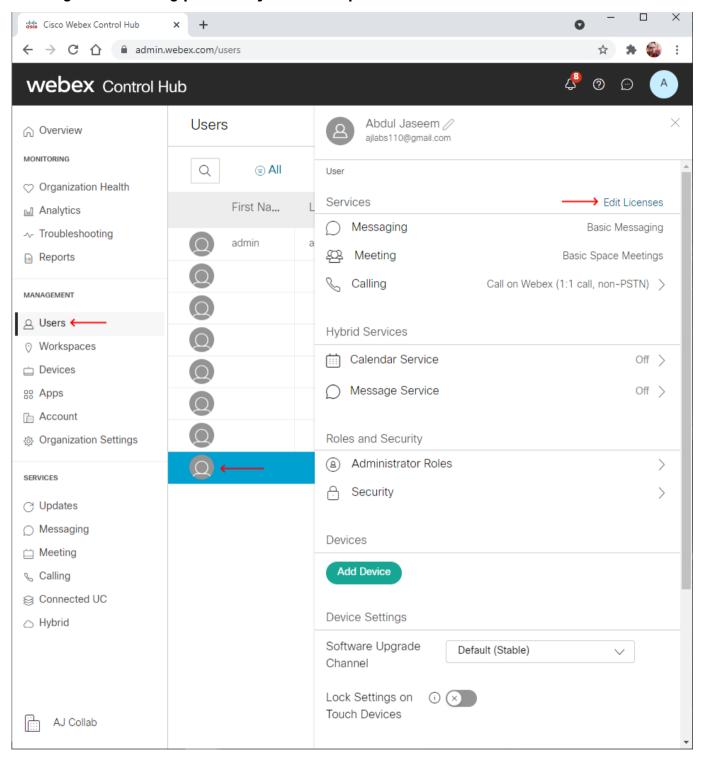


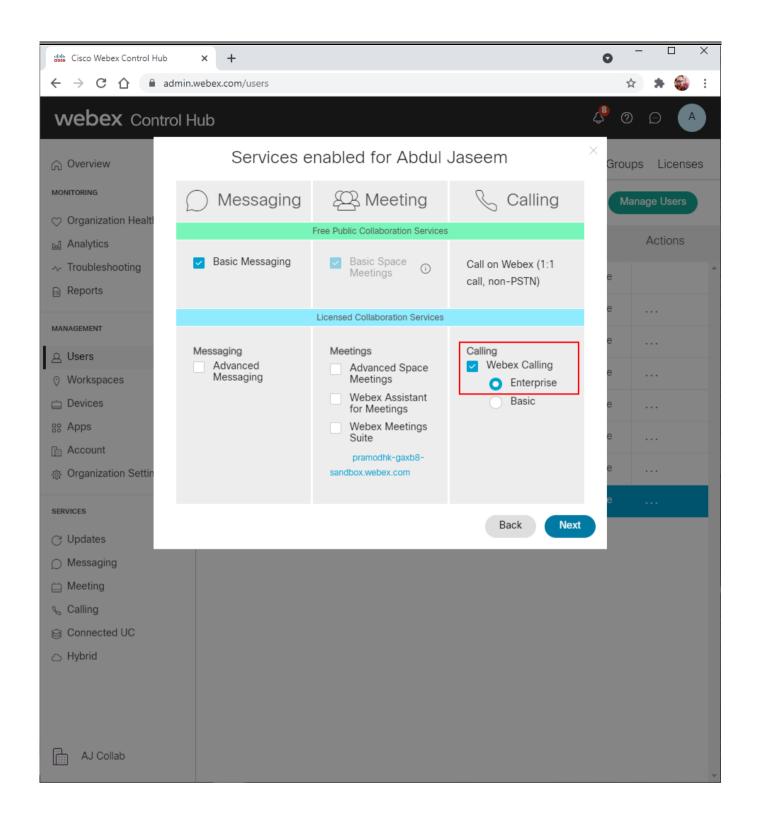


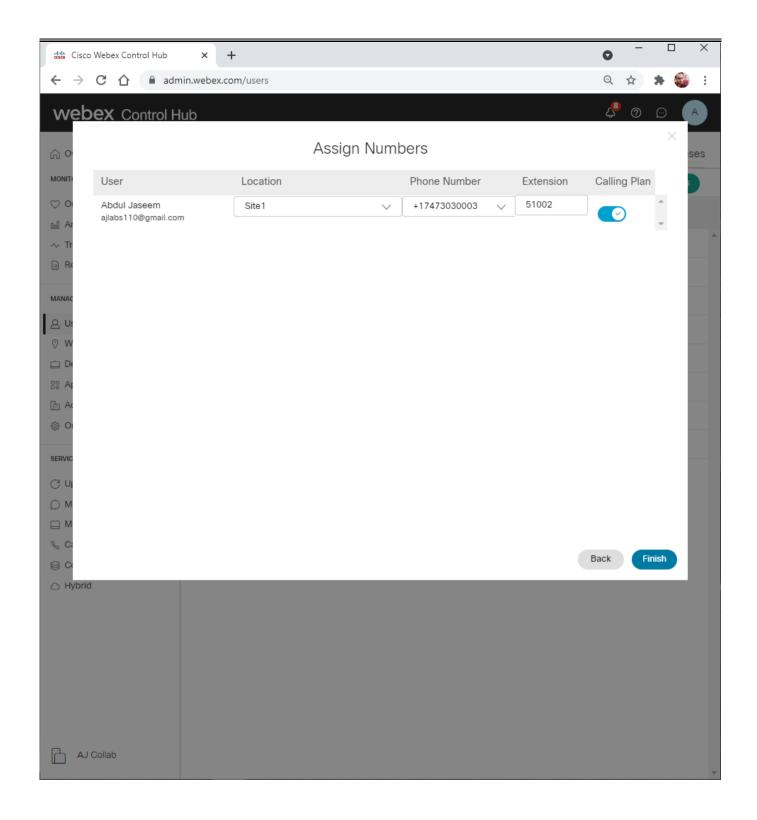




Enabling Webex Calling powered by broadcloudpbx for User





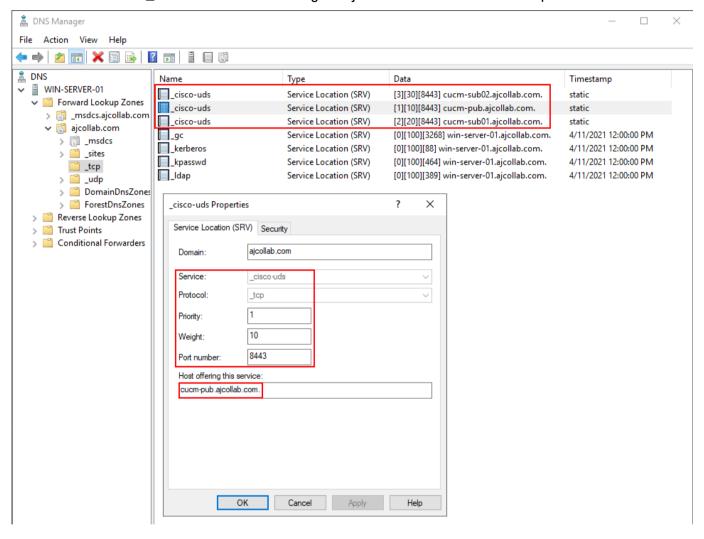


Webex Edge Solution

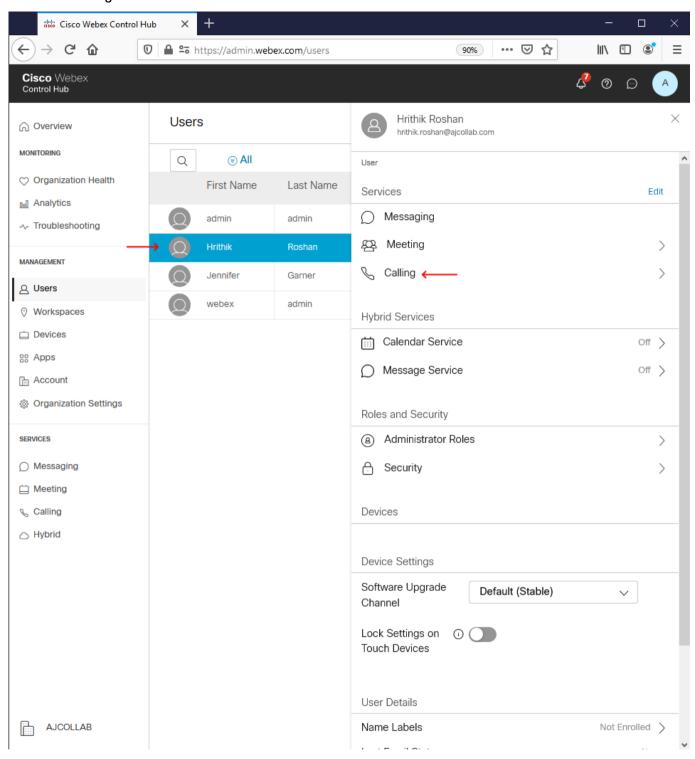
- Webex Edge Audio: Integrate on-premise CUCM and local PSTN via voice gateways in CUCM
- Webex Edge Connect: Dedicated managed QoS Enabled IP Link from on-premise to Cisco Webex Cloud through direct peering partner
- Webex Edge Video Mesh: Local instance media processing using on-premise resources

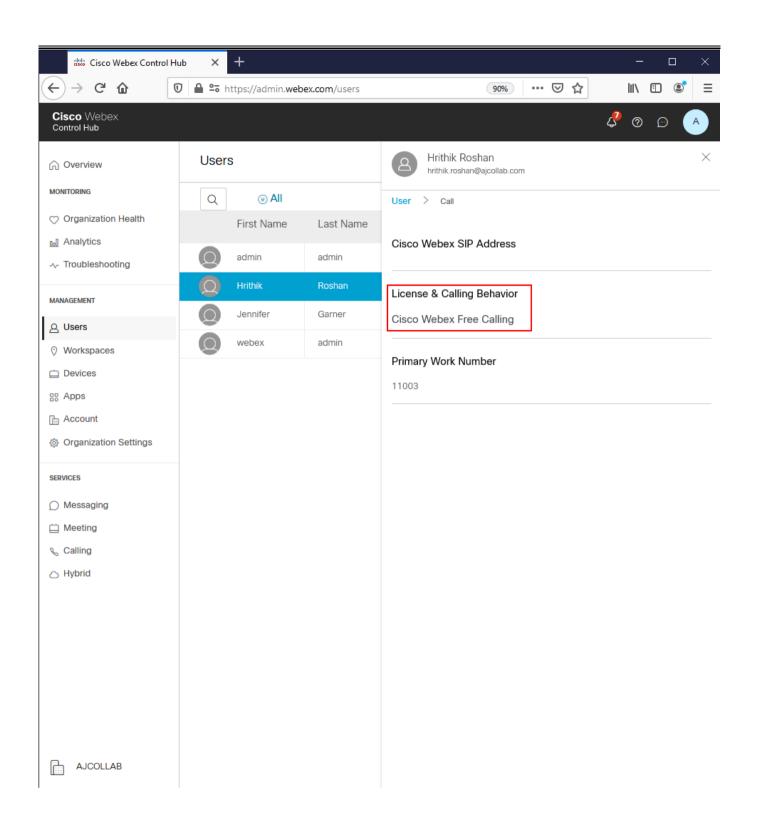
Webex UCM Calling (From Corporate Local Network)

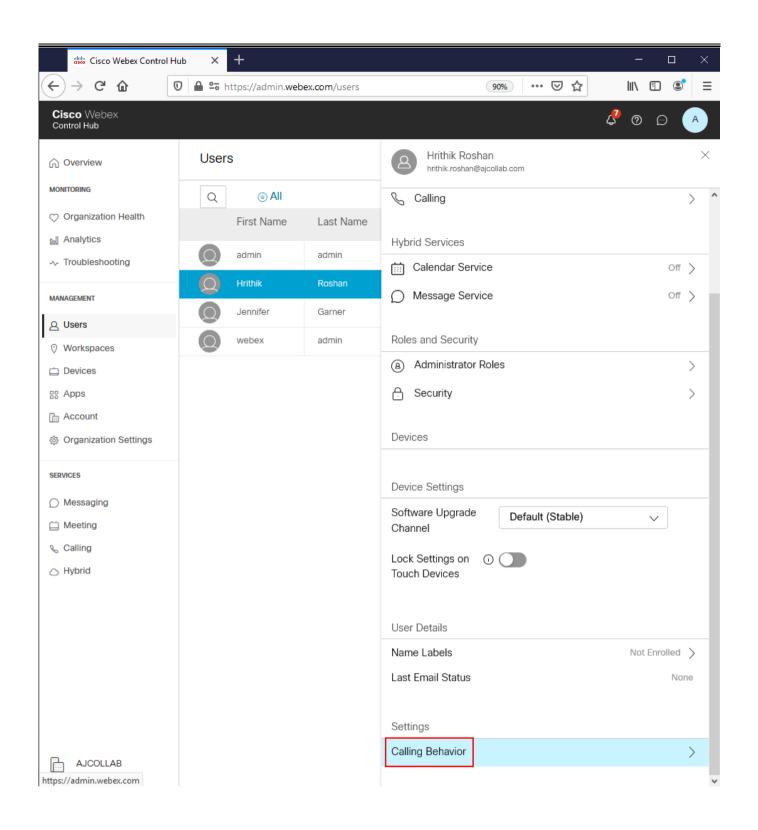
- Make Webex Team to a kind of hybrid endpoint. Some services will be served by cloud (Chat, File Sharing, etc.) and some Calling Services offered by on-premises CUCM
- The requirement of CUCM ins not going to fade away as it is a powerful call control server
- It is just like how Jabber Client gets the Phone Services via CSF
- DNS SRV for _cisco-uds must be configured just like we did for Jabber on premise SRV

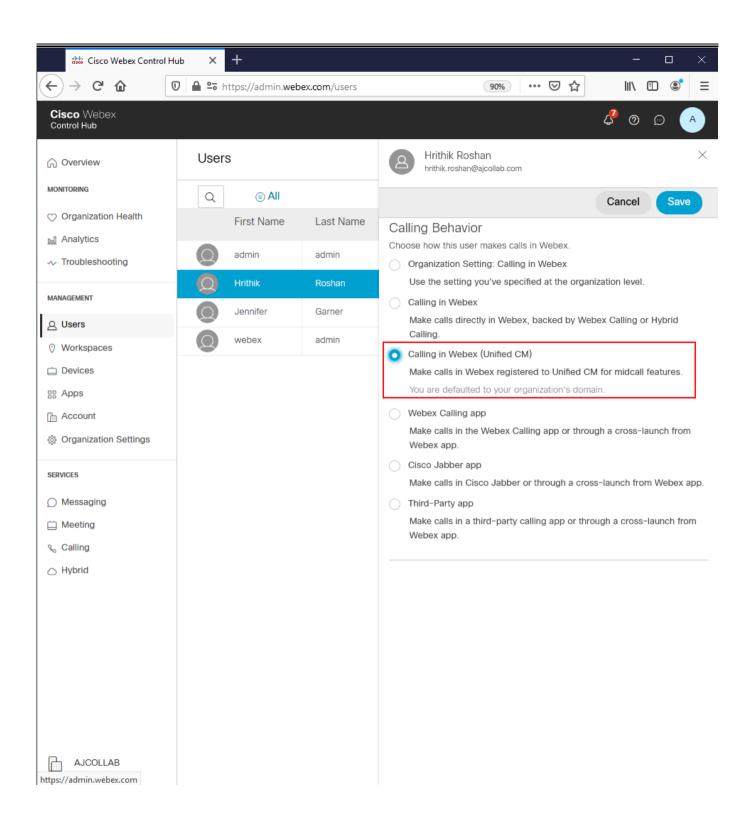


• Enabling UCM Call for individual User

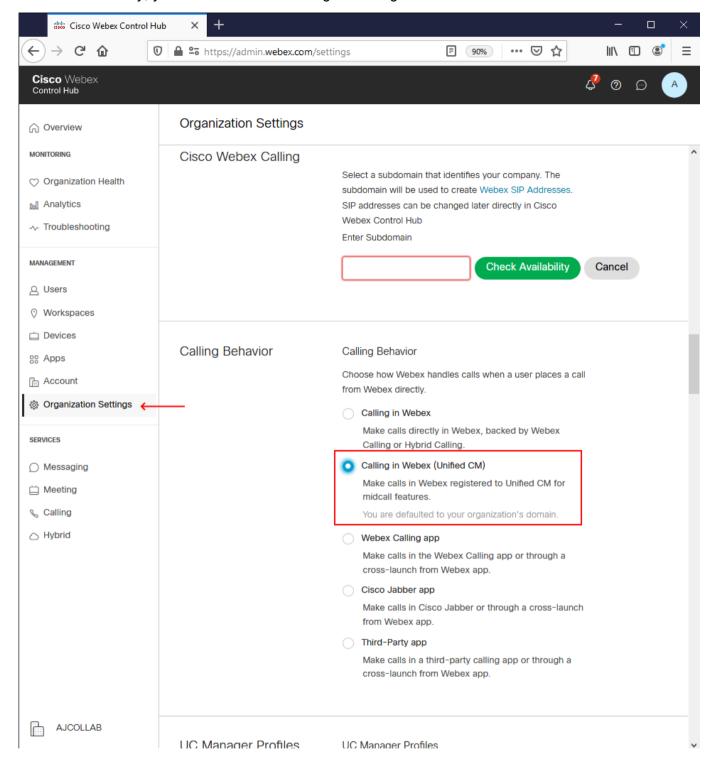


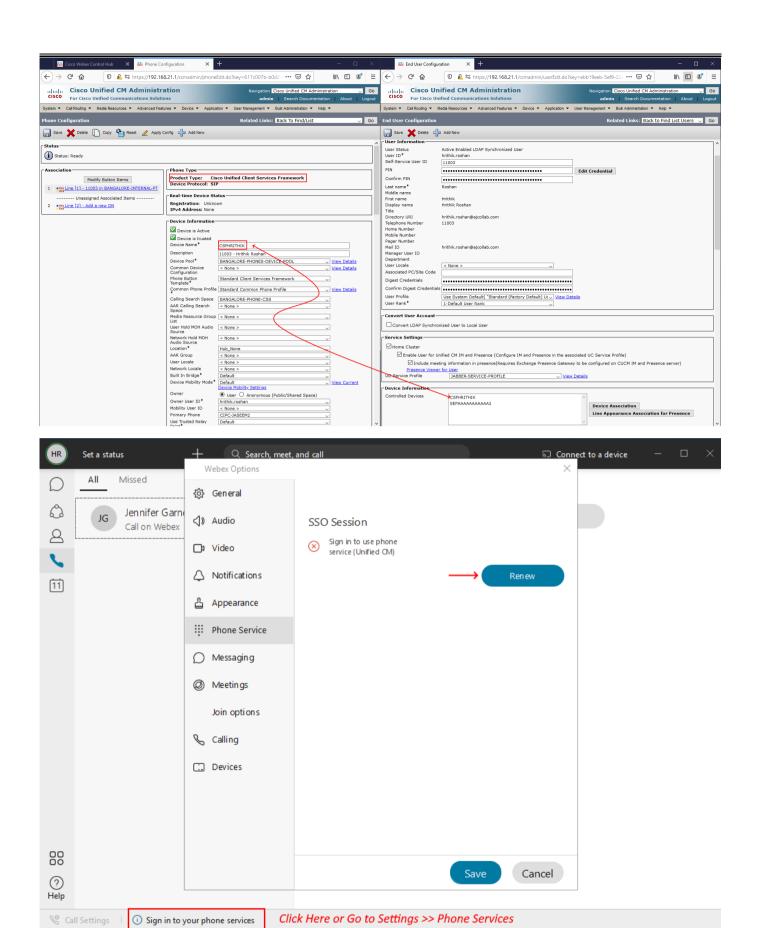


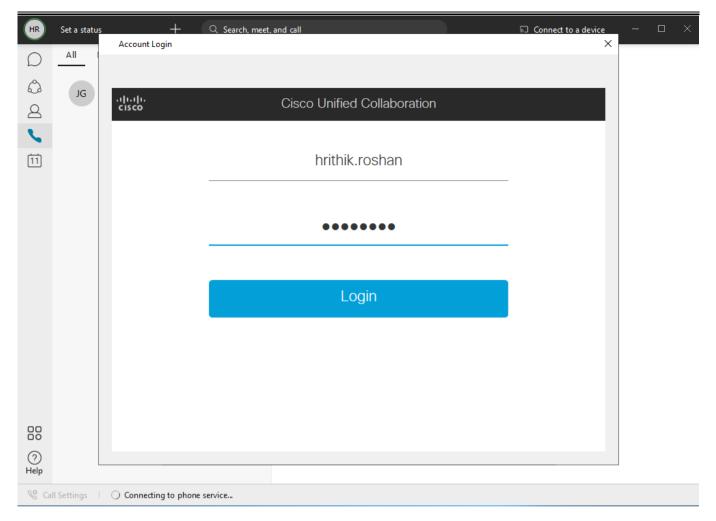




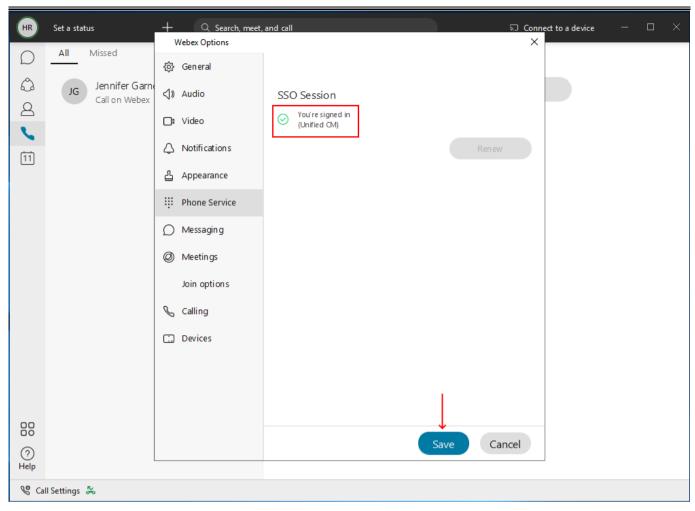
• Alternatively, you can enable UCM Calling at the Org level

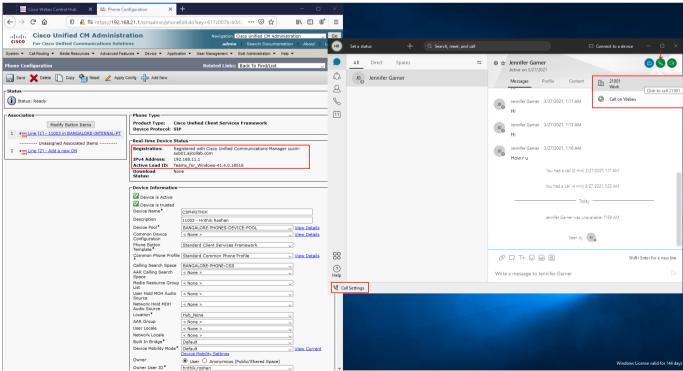


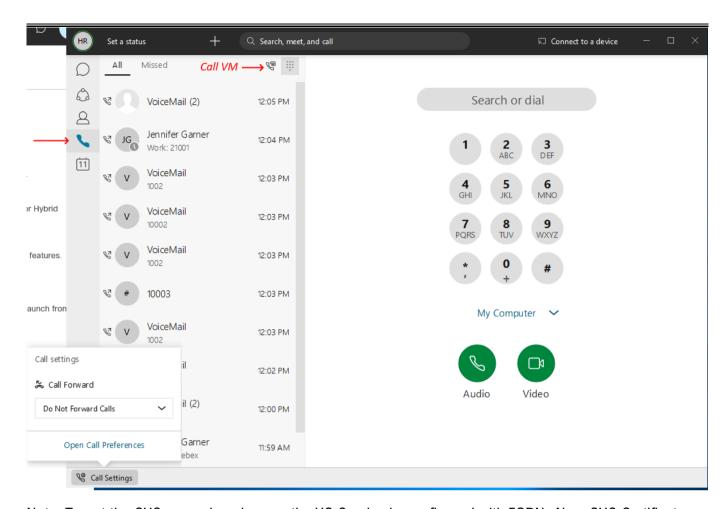




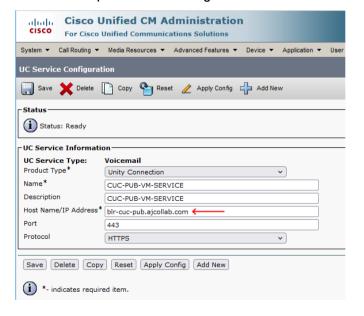
• If you have enabled SSO in Webex and CUCM, the Phone Service Login not required

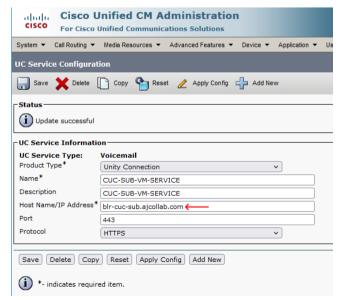


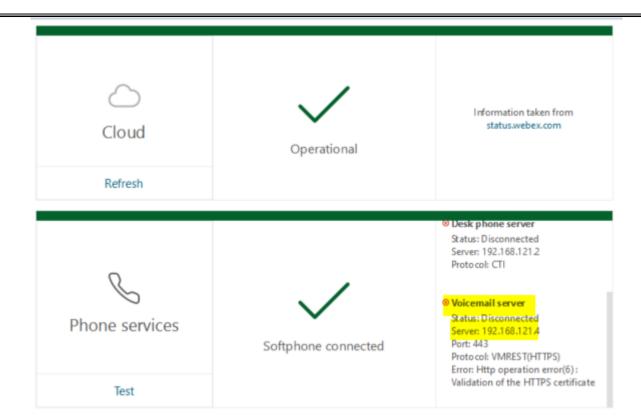


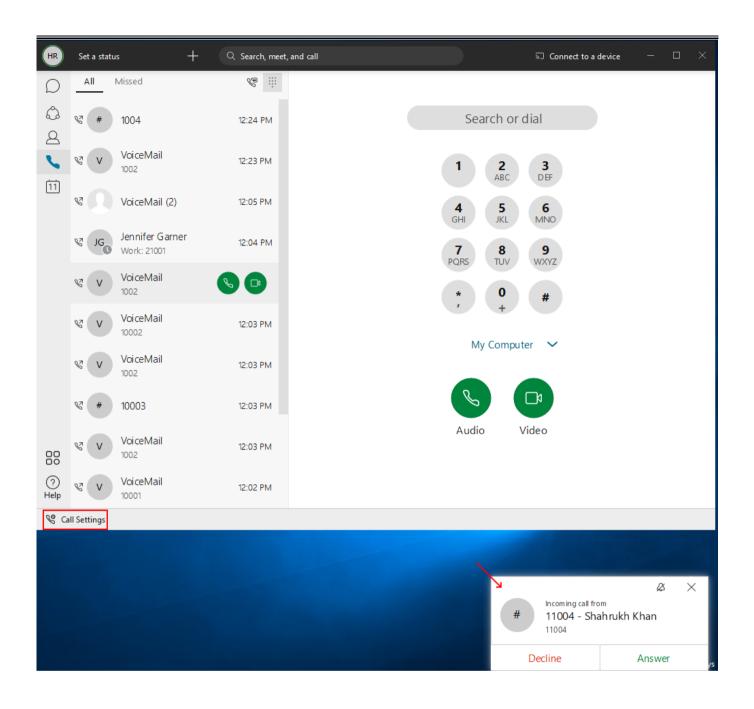


Note: To get the CUC synced, make sure the UC Service is configured with FQDN. Also, CUC Certificates has to be signed by CA and it should be trusted by the client computer. Teams client never ask for certificate expectation warning.



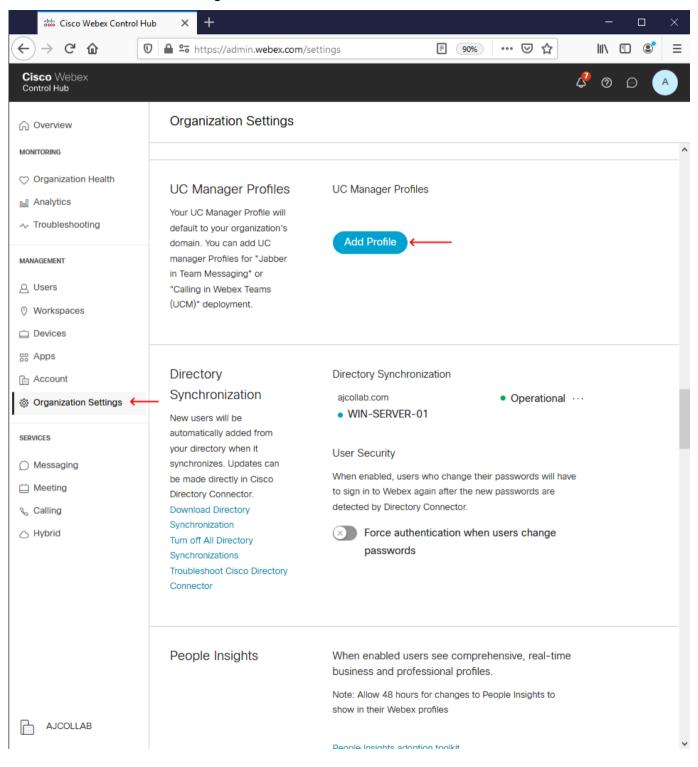


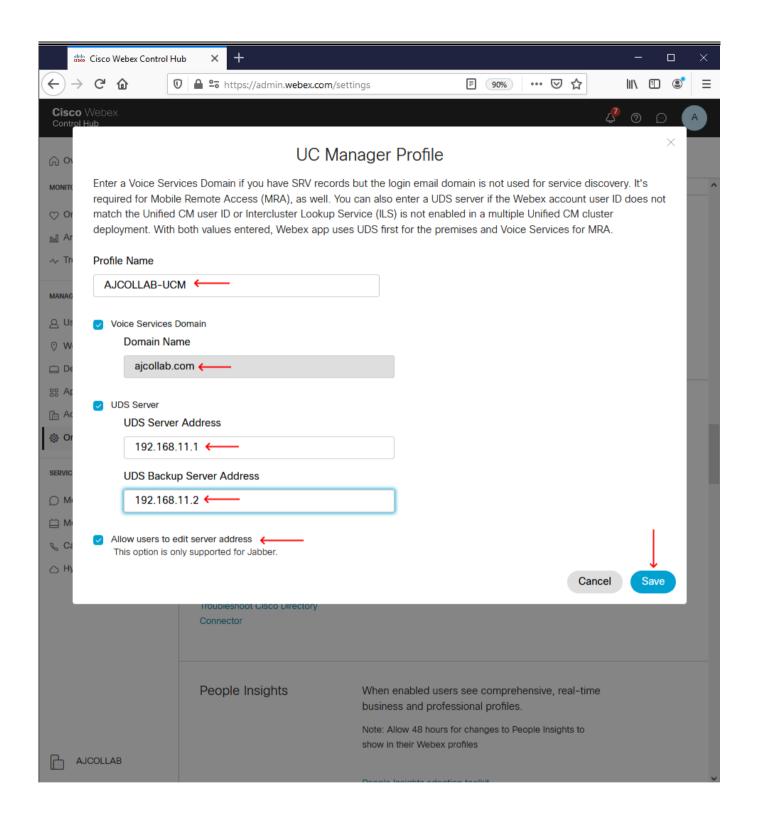


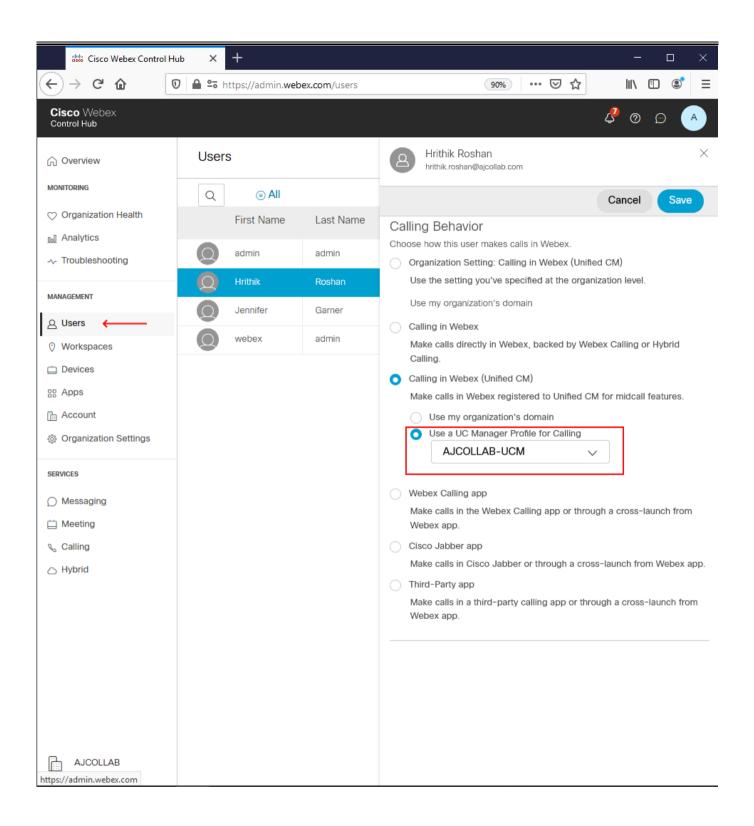


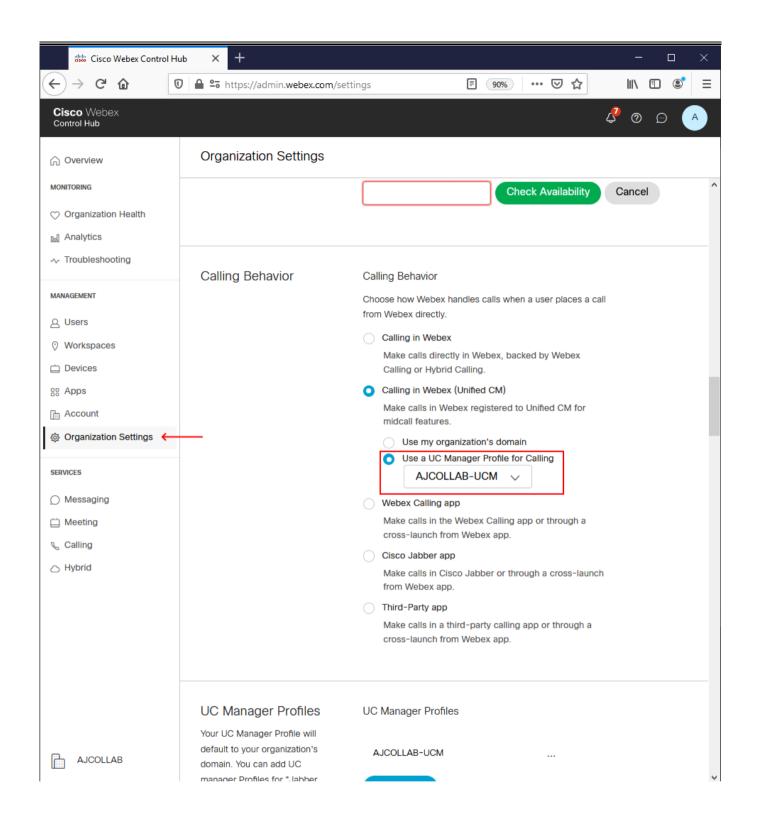
UC Manager Profile

- If your default Webex domain is not the CUCM Domain, then the service discovery fails. For
 example, if you enabled Webex teams with your personal mail ID (Gmail.com) then the service
 discovery happens for Gmail.com and you don't get anything for _cisco-uds from Gmail.com
- To overcome this issue, you can configure UC Manager Profile and hardcode the CUCM UDS
 Server details there along with CUCM voice domain



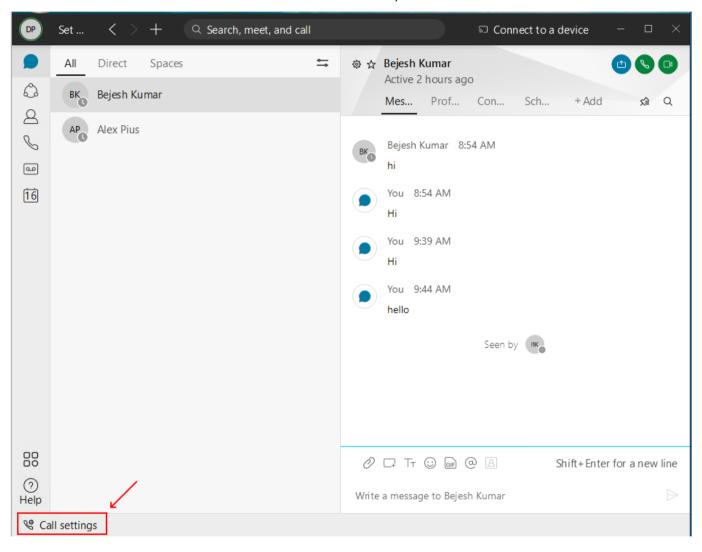


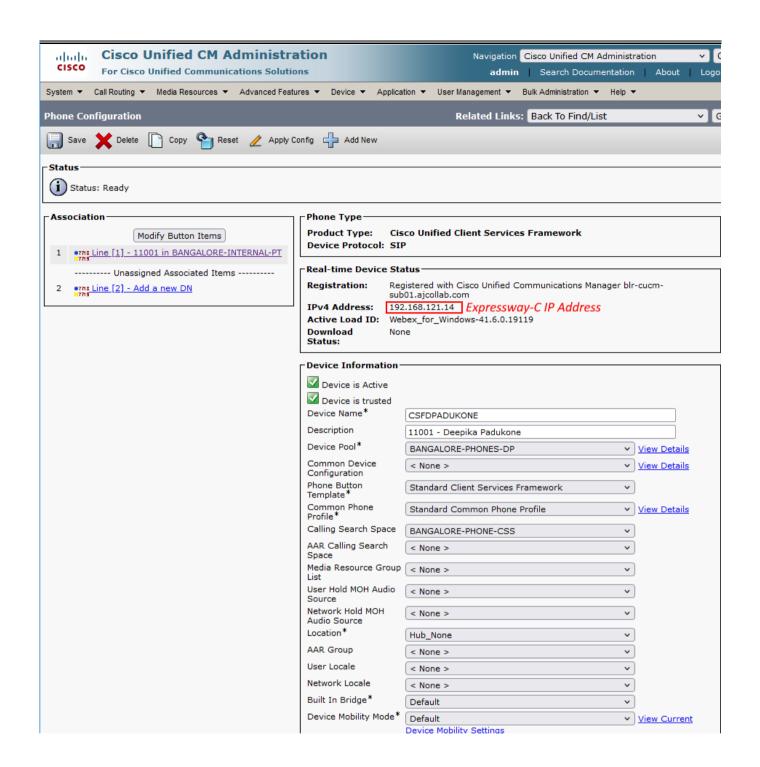




Webex UCM Calling with Expressway (MRA Solution)

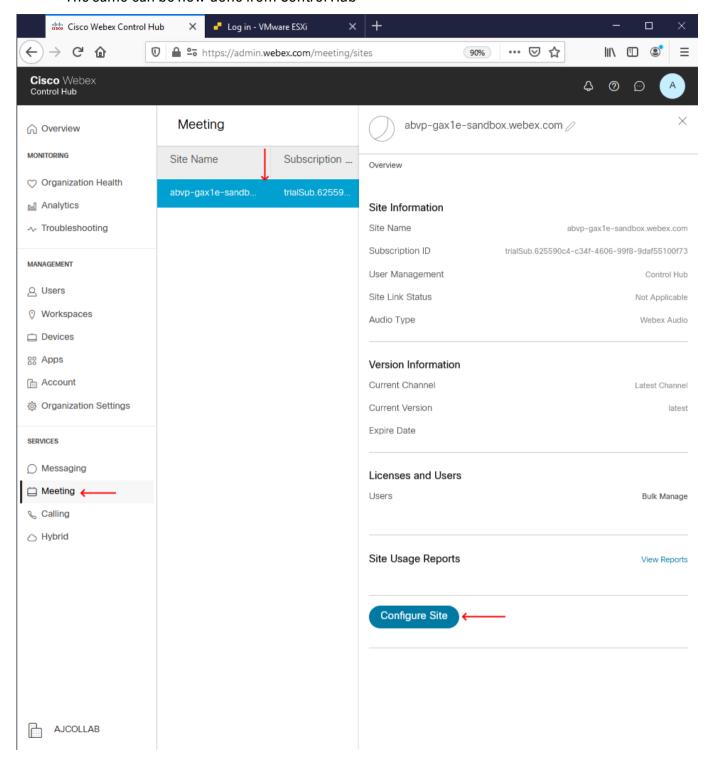
- Make sure you have the complete MRA configured as discussed in the Jabber RMA section
- Here the Teams CSF registers to CUCM just like Jabber CSF
- If you Jabber MRA configuration is correct, then there won't be any issues in Teams UCM Calling via MRA
- Make sure your Expressway E is signed by public CA (If you are in lab environment, Expressway E
 might be signed by Enterprise CA, so the Enterprise Root CA Certificate must be installed on the
 client OS where the Webex Teams software installed)

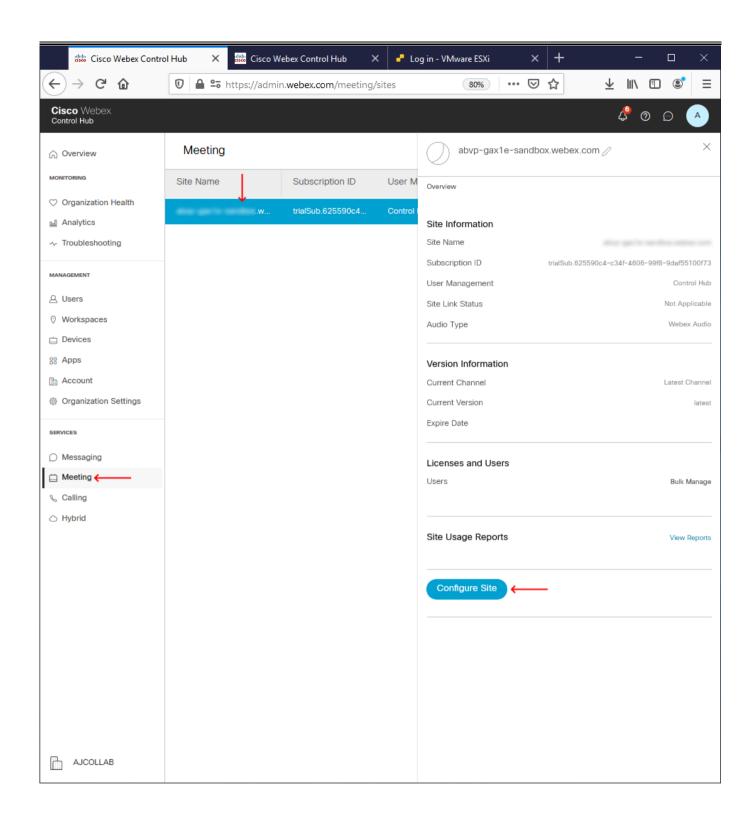


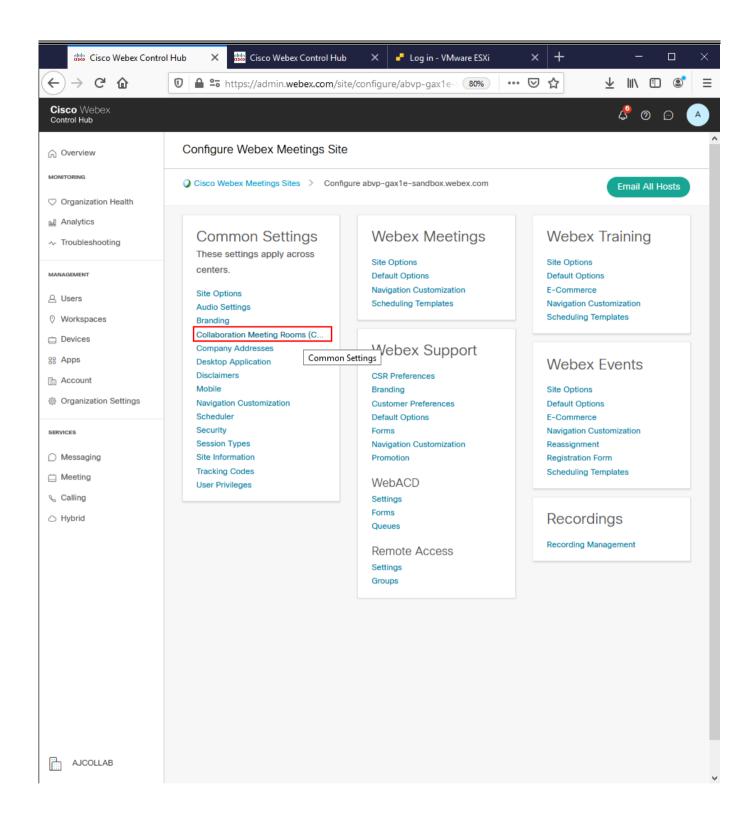


Classic Webex Meeting Site Administration

- When the Webex was initially launched, the meeting accounts are managed via Webex Site Admin page
- The same can be now done from Control Hub

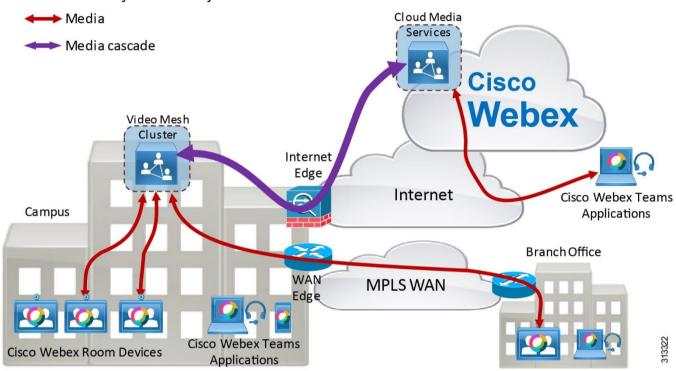






Webex Edge Video Mesh

- Solution that uses local resources for Webex Video Conferencing for on-premise attendees. Local media kept local there by saving internet bandwidth
- Extends the Cloud to on-premise, ideally 1:1 and multi-party meeting uses cloud resource always.
 Signaling and media go to and from Cloud
- Media flowing to the cloud uses huge internet bandwidth
- · It was initially known as Hybrid Media Service



- The video mesh packages the cloud meeting capabilities to an on-premise virtual machine. Video
 Mesh node can be downloaded from Webex Control Hub
- Automatic overflow to the cloud if the video mesh node is full
- Video Mesh node combines all the participants video in to one compressed media and send to the cloud over internet via cascade link
- Can be deployed as single or cluster mode

Devices that can connect to Video Mesh

Webex Registered TP Endpoints
Webex Teams App
CUCM Registered Devices (Phones, TPs, etc.)
Expressway Registered TP Endpoints

Devices that can't connect to Video Mesh

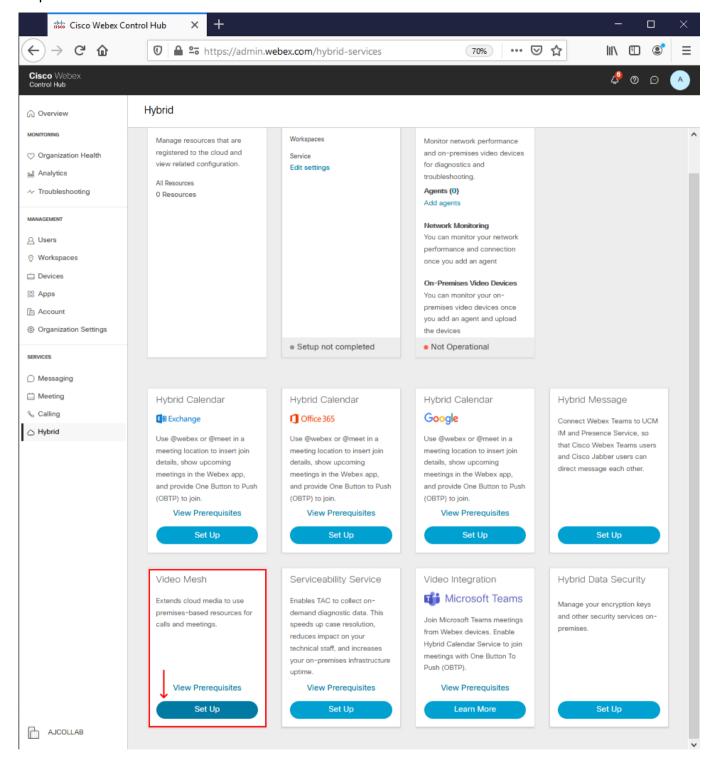
External participants from Teams App
External participants from Webex registered TP
Webex Registered UC Phone
Webex Teams browser clients (teams.webex.com)

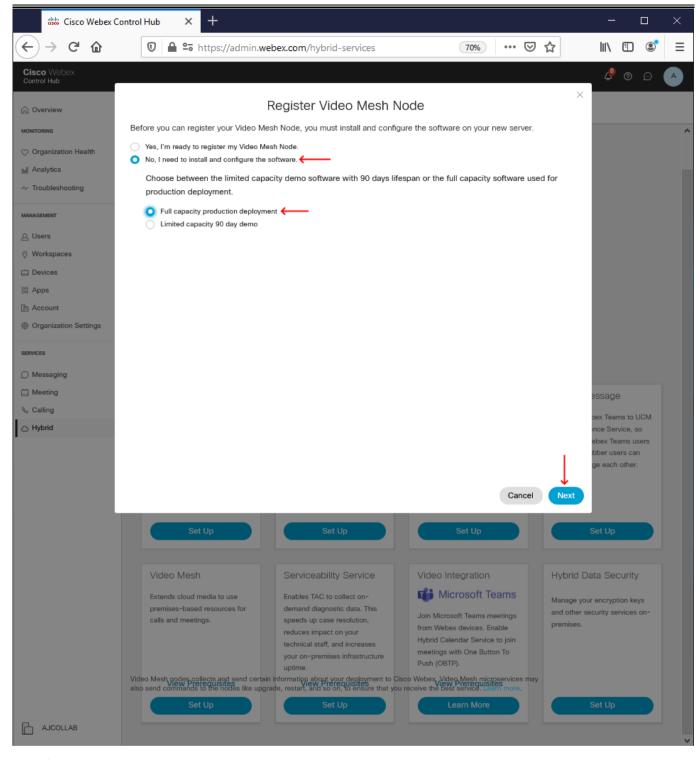
Process

- When a device registered to Control Hub, it sends a round trip time to Webex Cloud. Webex will
 identify if there is any video mesh node available and then relay that information back to the device
- Device again does a round trip time test with video mesh node and prioritize the connectivity between Webex cloud and Video mesh
- Device prefers the lowest round trip time server to get connected for meeting and conference
- CUCM registered endpoints selects the video mesh based on route pattern

[Lab] Deploying and Configuring Video Mesh Node

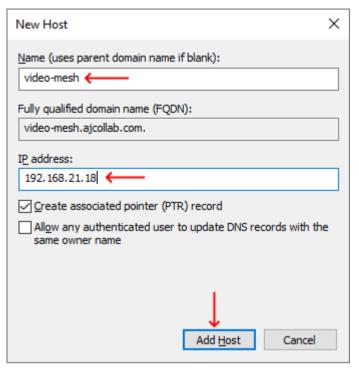
Step 1: Download the Video Mesh Node



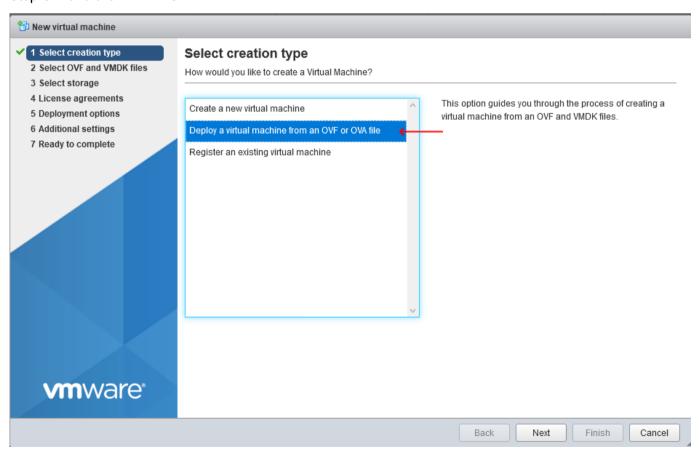


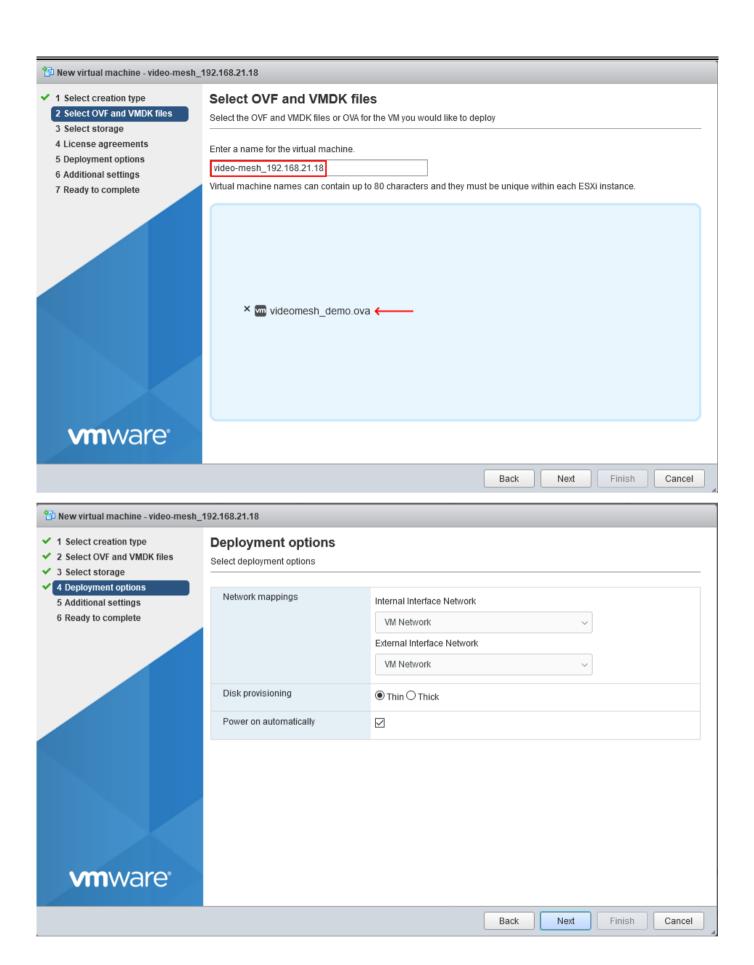
Wait for the complete the download

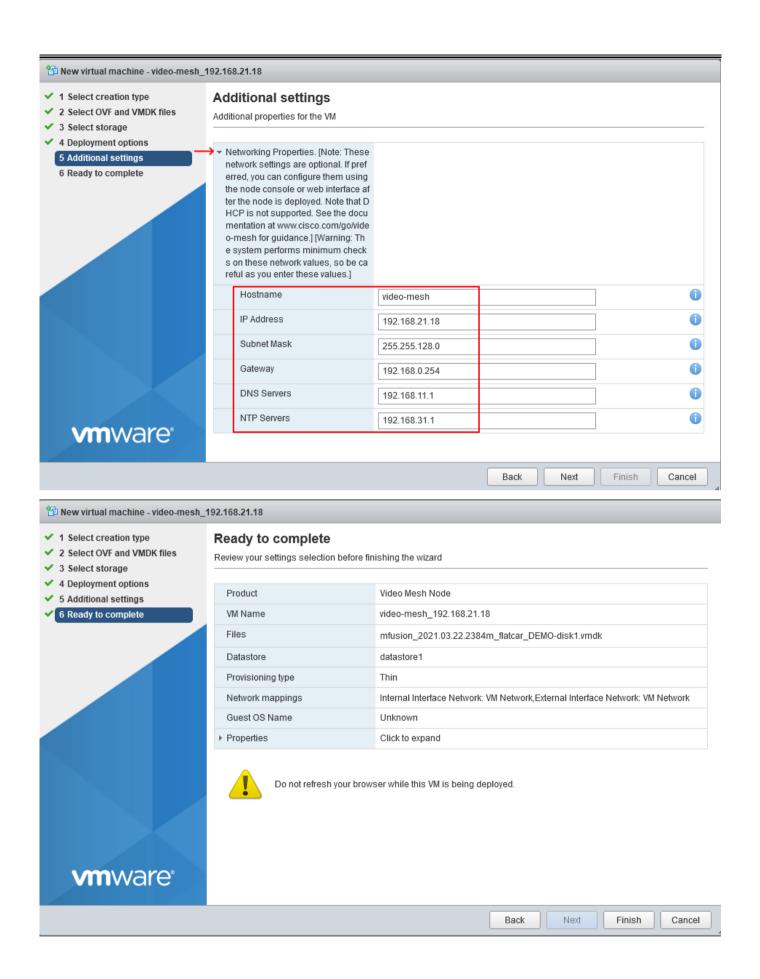
Step 2: Configure DNS A Record for the Video Mesh Node



Step 3: Build the VM in ESXi







```
video-mesh_192.168.21.18
SSH host key: SHA256:/qQ6KZ8A6Rs+URAzu+ENureqjY1KhMgAiFqEP83j/9E (RSA)
SSH host key: SHA256:747CkavSS8xVcSX39yLOef7VLX7d8Naxhm91aetpc1o (DSA)
SSH host key: SHA256:nihgZdBJnn8pIS50AI3zYs7K∪miOEWP6dbrCT08eybA (ECDSA)
SSH host key: SHA256:m3S9834TVYazUjiRWmWNrz7b3eArmR9p1fyyP3A9hTQ (ED25519)
ens192: 192.168.21.18
ens224:
video-mesh login: [    30.962393] docker0: port 3(veth59bfb74) entered blocking s
     30.9634191 docker0: port 3(veth59bfb74) entered disabled state
     30.9644671 device veth59bfb74 entered promiscuous mode
     31.1312041 SELinux: mount invalid. Same superblock, different security sett
ings for (dev mqueue, type mqueue)

31.3679891 eth0: renamed from veth56ff4fb

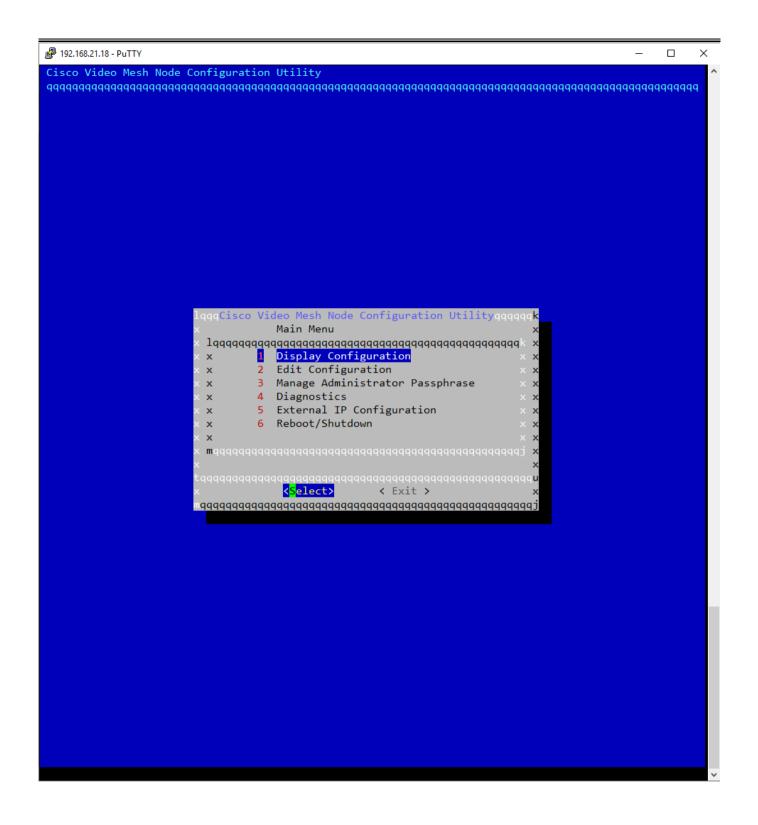
31.3798981 docker0: port 3(veth59bfb74) entered blocking state

31.3809311 docker0: port 3(veth59bfb74) entered forwarding state

37.8897111 docker0: port 4(vethe86bdb6) entered blocking state

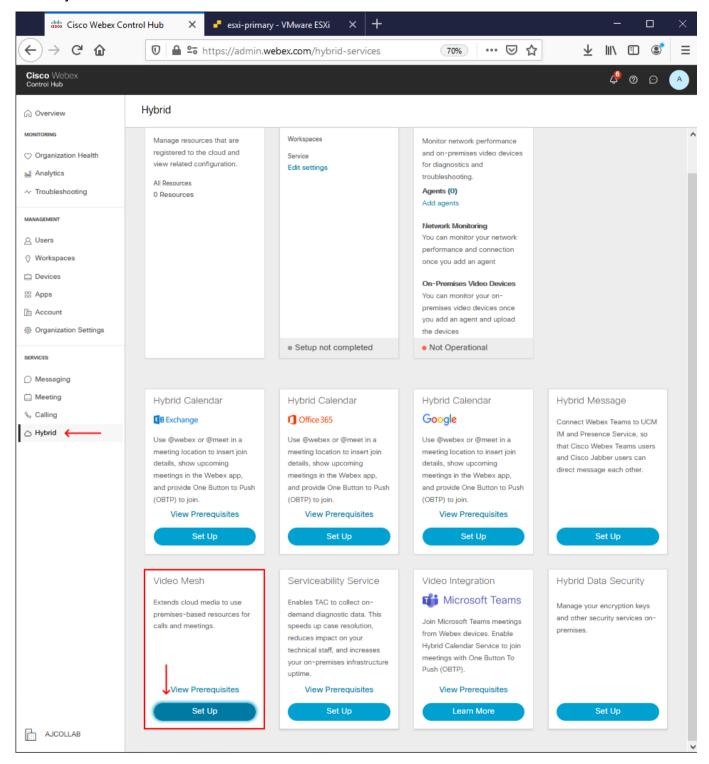
37.8910181 docker0: port 4(vethe86bdb6) entered disabled state
     37.8923431 device vethe86bdb6 entered promiscuous mode
     38.0263091 SELinux: mount invalid. Same superblock, different security sett
ngs for (de∨ mqueue, type mqueue)
     38.202139] eth0: renamed from veth28539ae
     38.2122301 docker0: port 4(vethe86bdb6) entered blocking state 38.2132351 docker0: port 4(vethe86bdb6) entered forwarding state
```

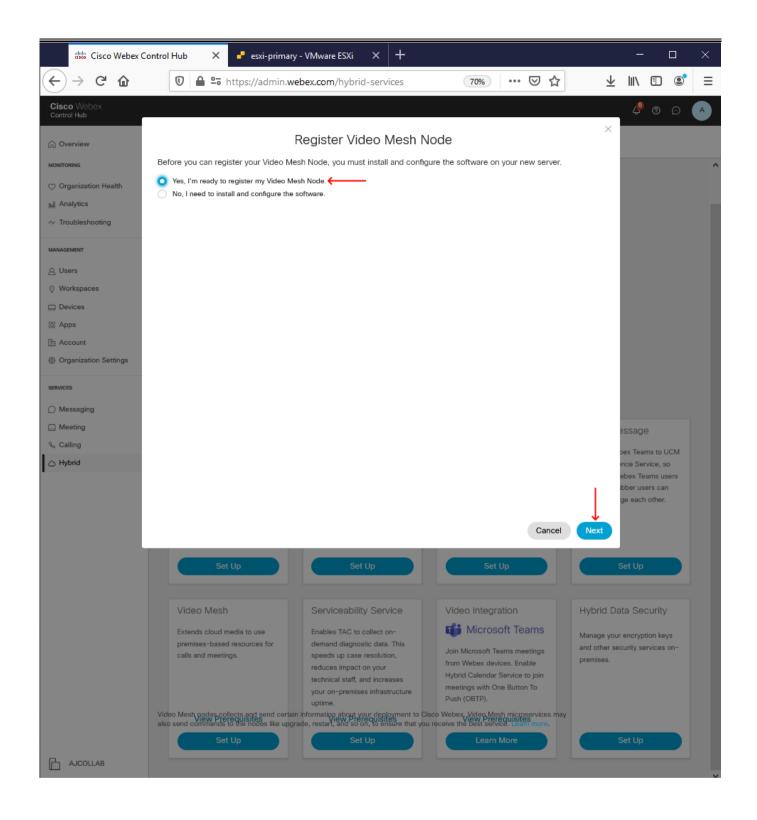
- Default username is admin and password is cisco
- When you login via SSH for the 1st time, you may challenged to reset the password

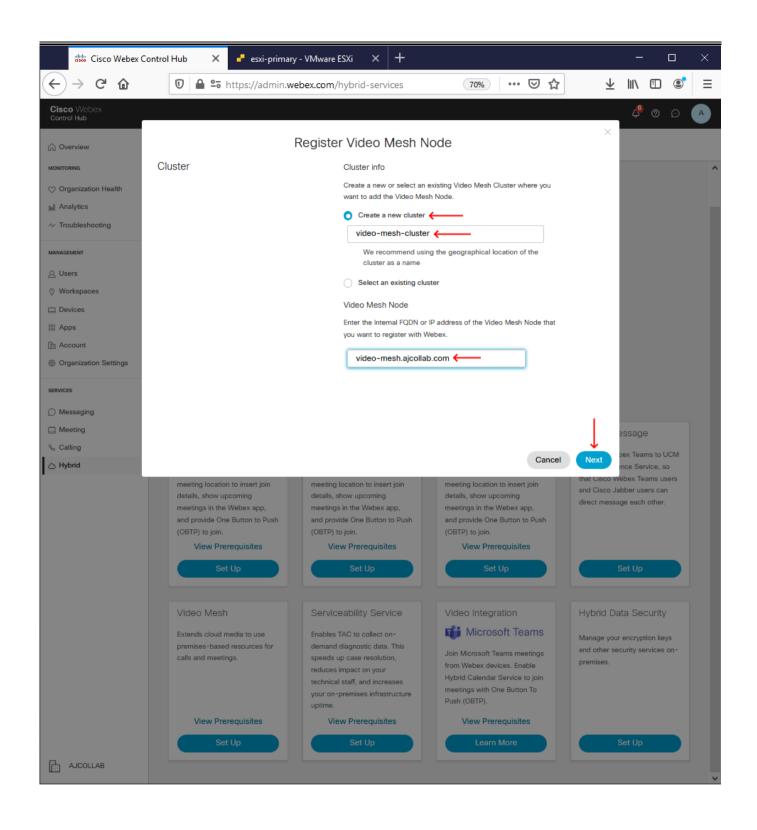


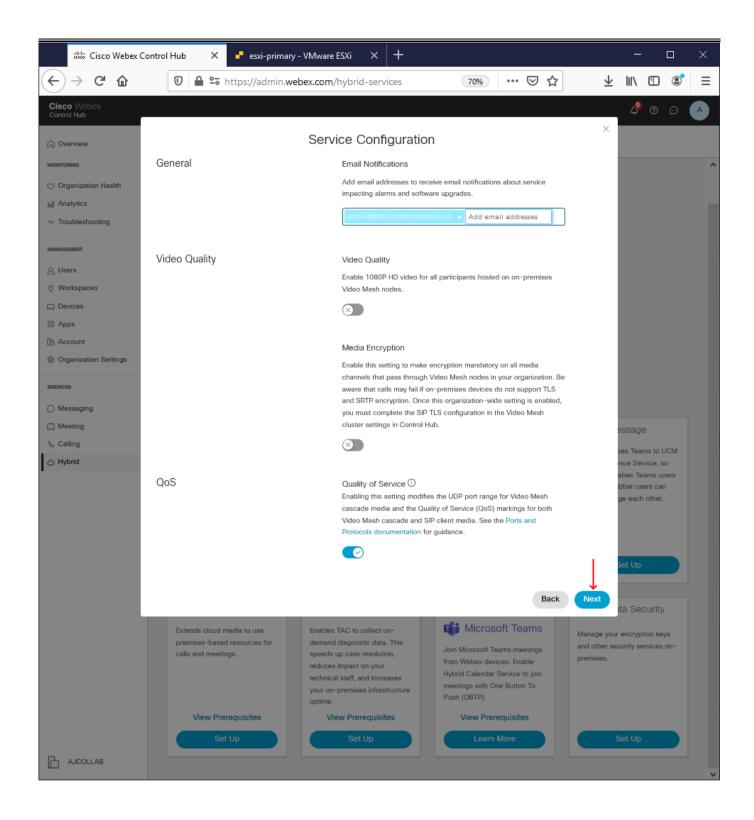
Step 4: Register Video Mesh Node to Control Hub

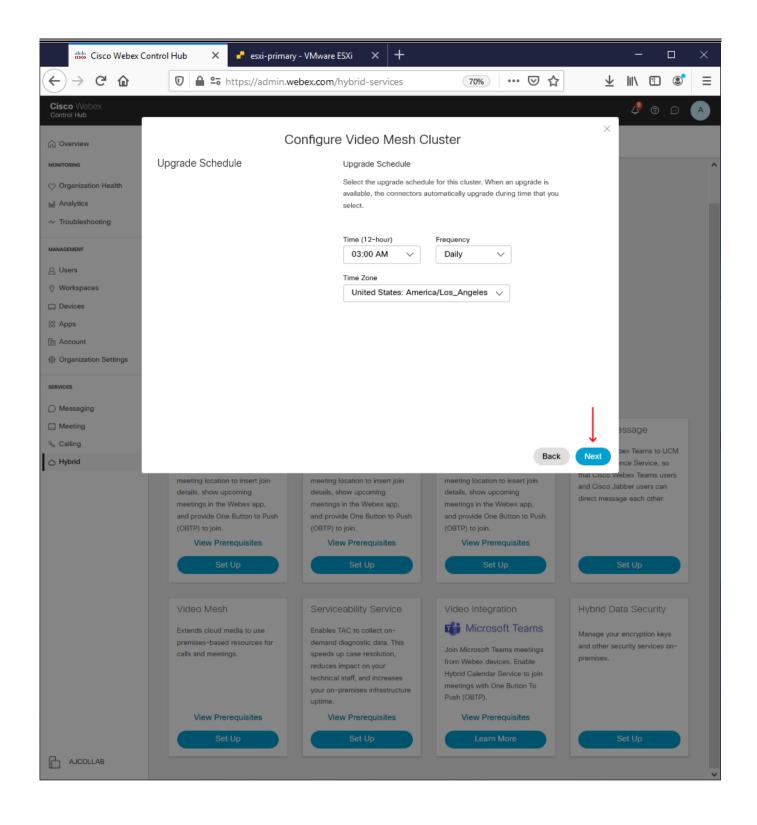
Go to Hybrid Services >> Video Mesh >>

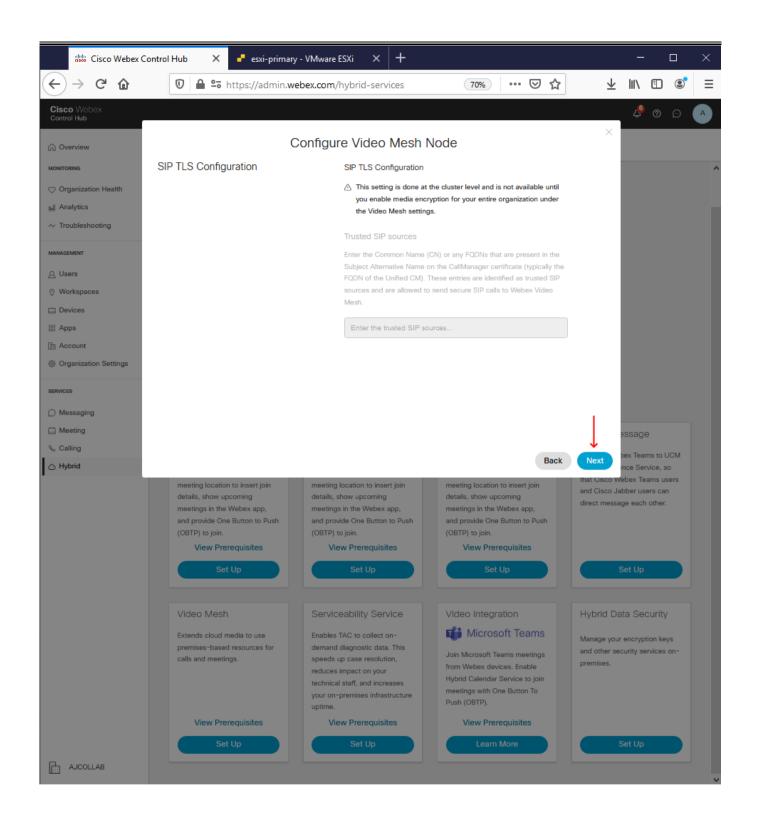


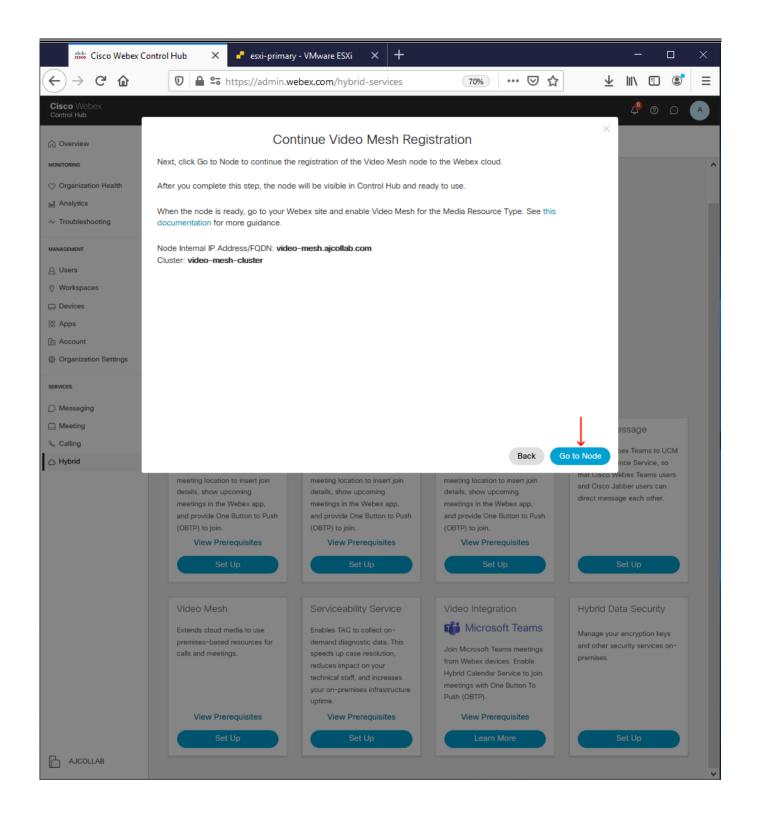


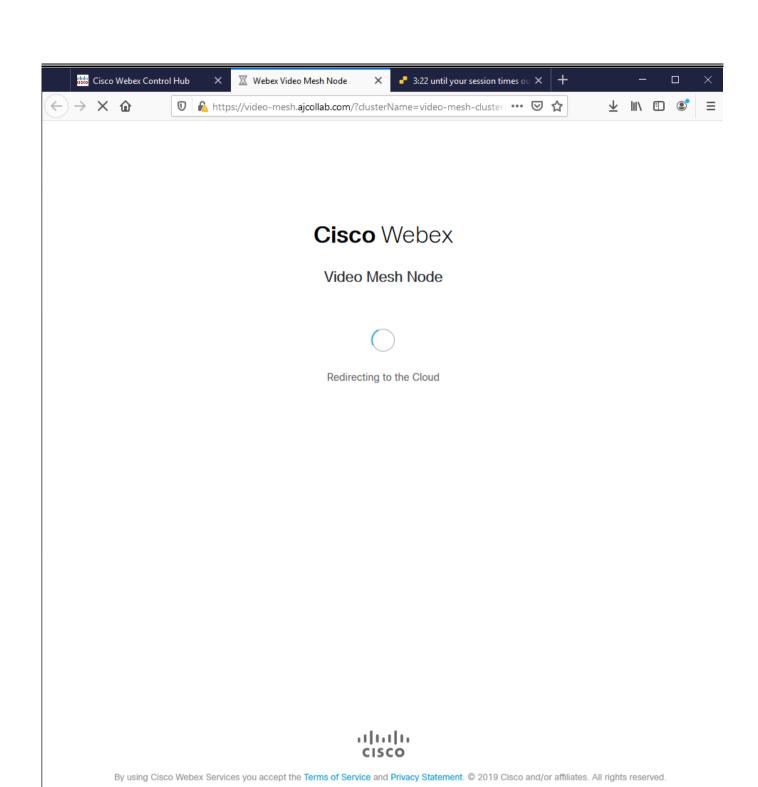




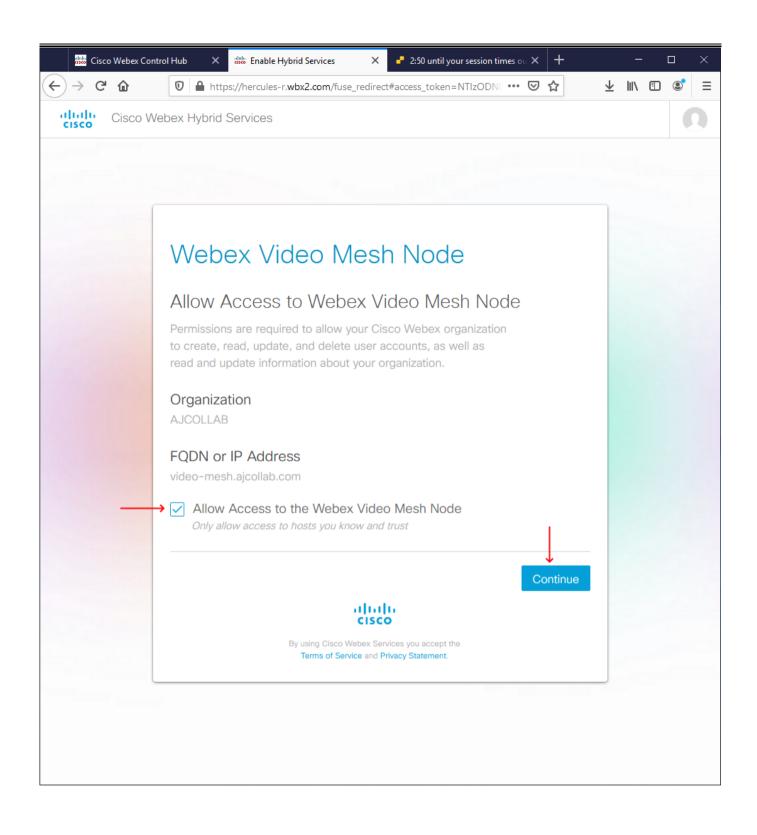


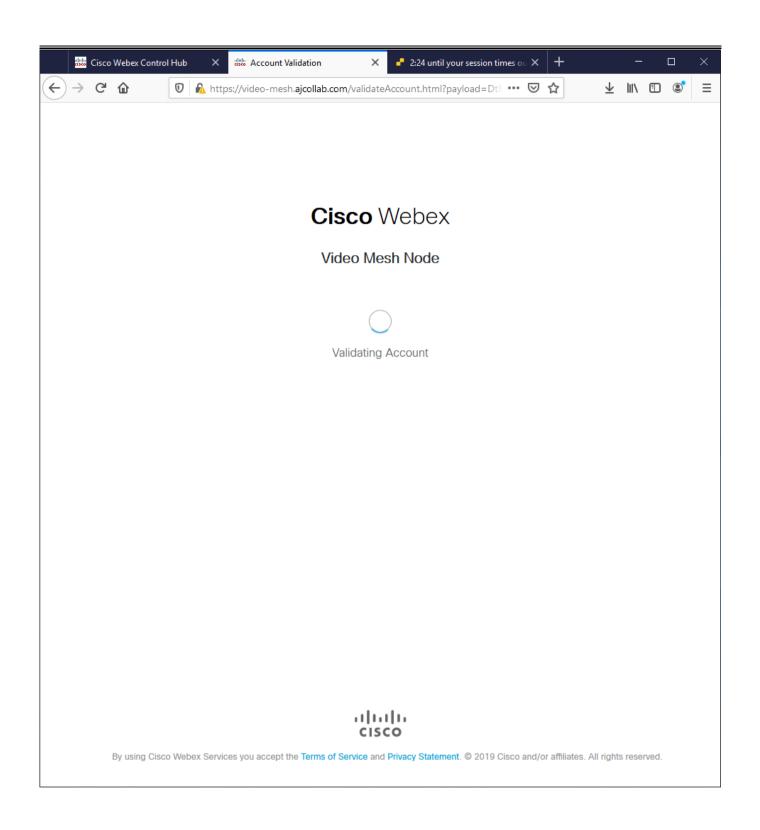


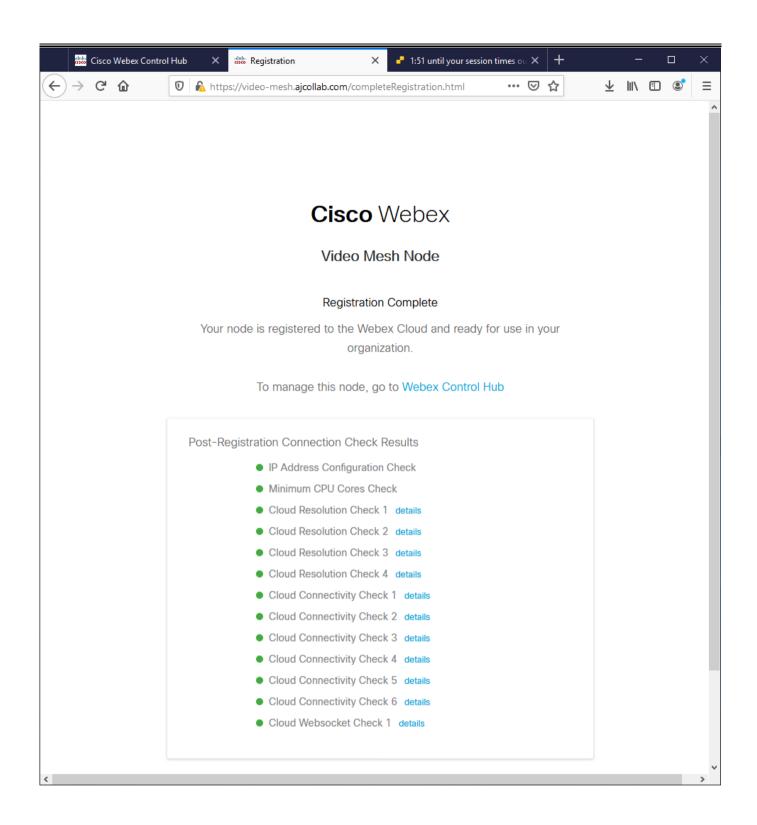




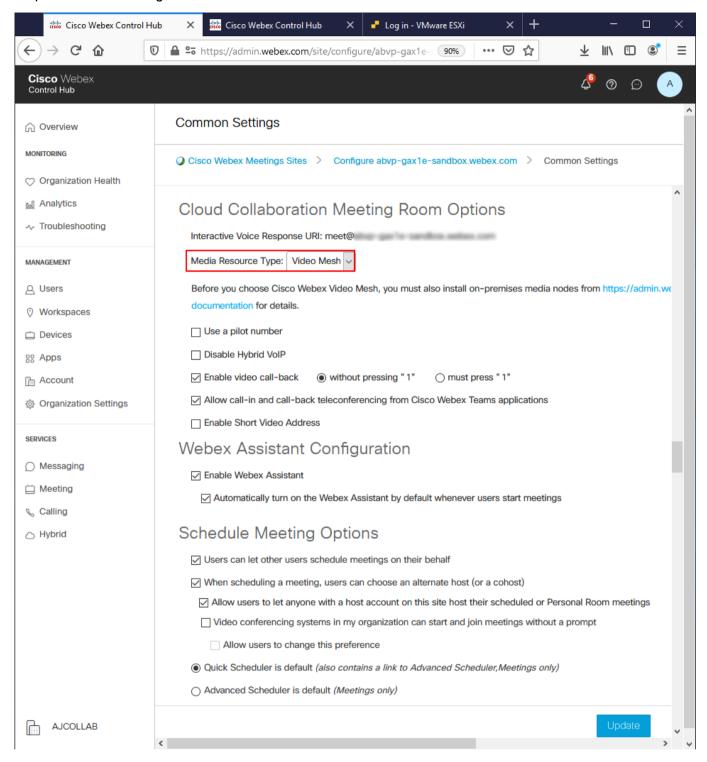
hercules-r.wbx2.com



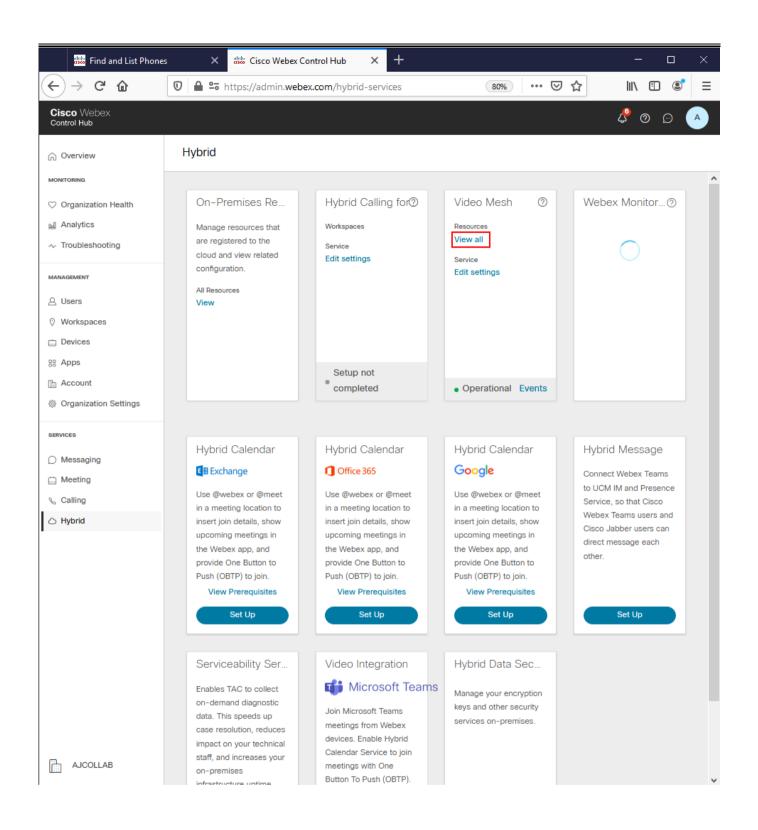


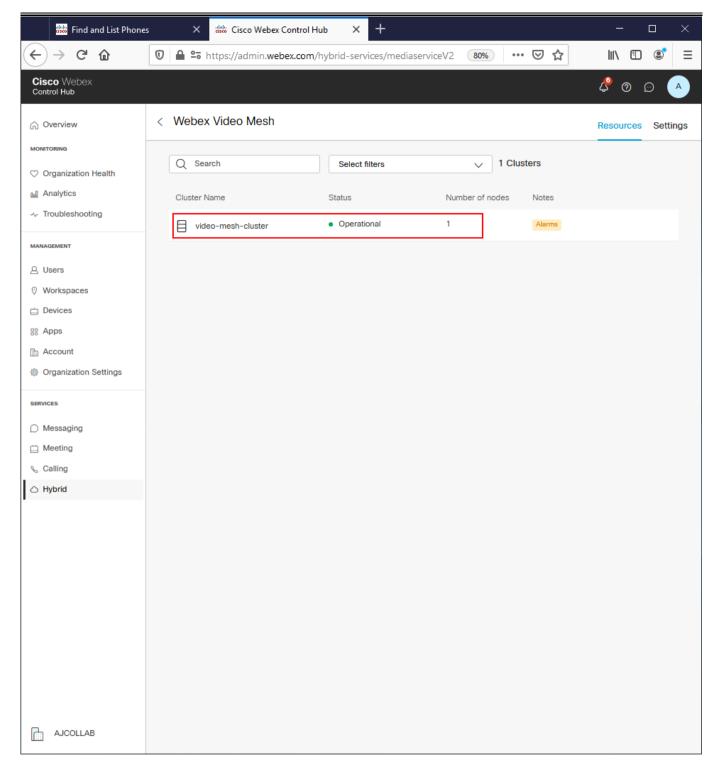


Step X: Enable meetings on Video Mesh Node

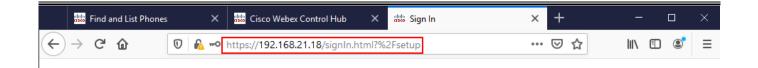


 Now Webex Registered TP Endpoint and On premise Teams Client can use Video Mesh Node but not the CUCM Registered endpoints





https://VIDEO-MESH-NODE-IP-OR-FQDN/signIn.html?%2Fsetup



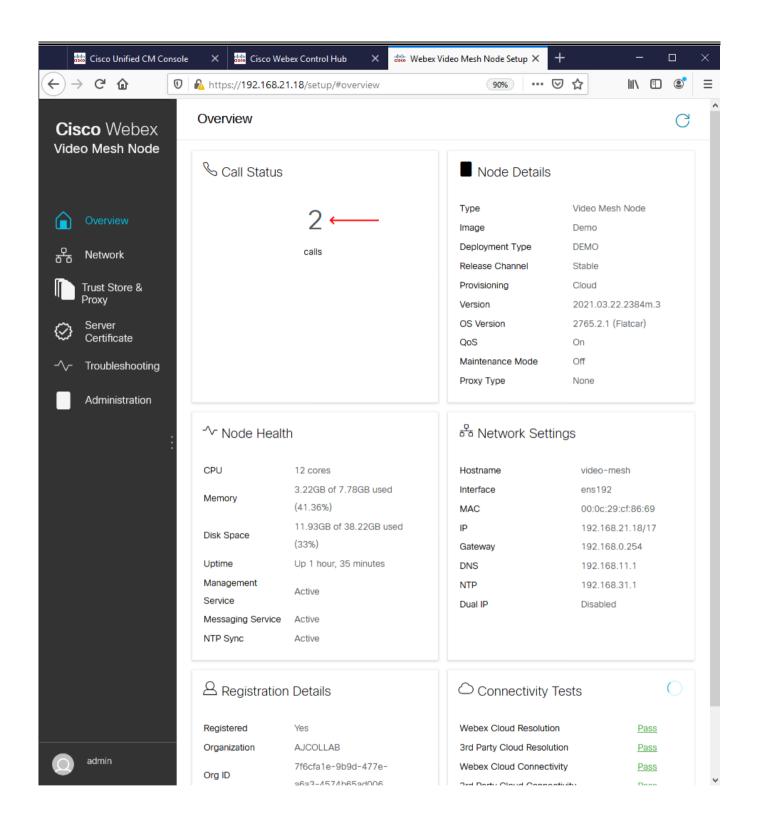
Cisco Webex

Video Mesh Node



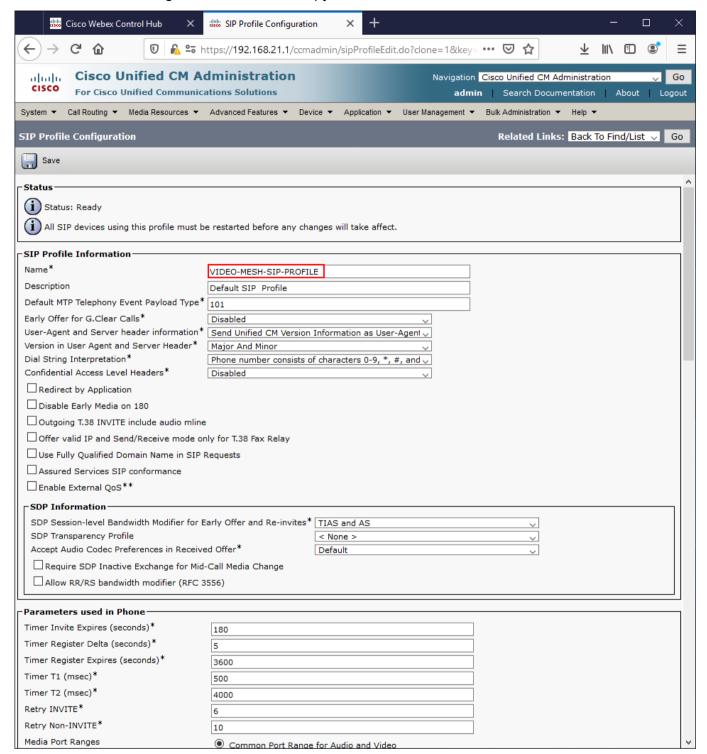
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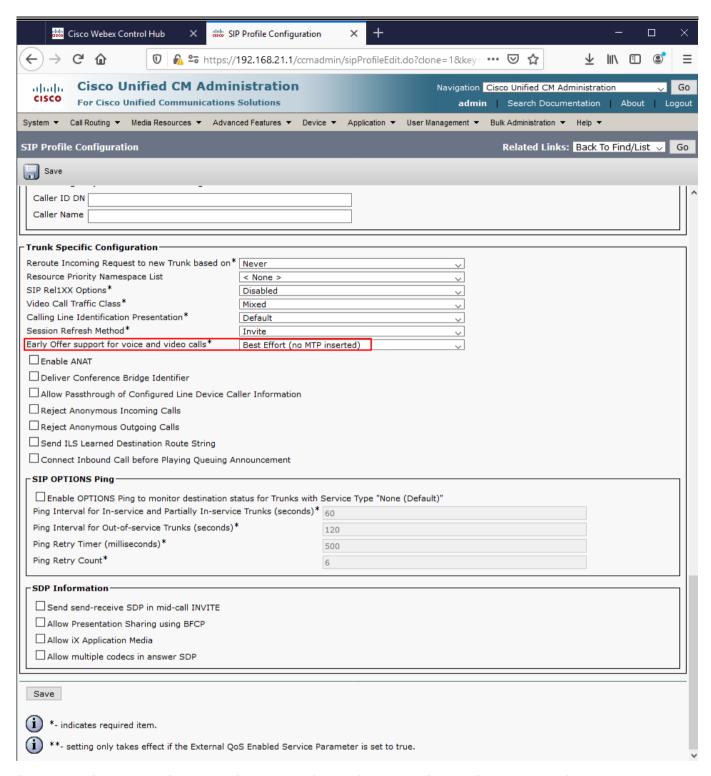
By using Cisco Webex Services you accept the Terms of Service and Privacy Statement. © 2019 Cisco and/or affiliates. All rights reserved.



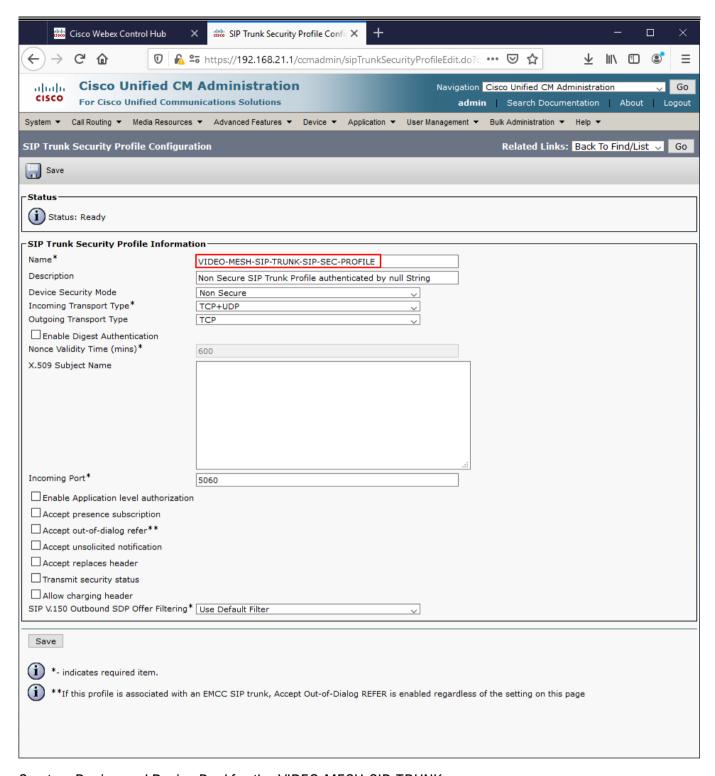
Step X: Configure CUCM for Video Mesh

Device >> Device Settings >> SIP Profile >> Copy Standard SIP Profile >>

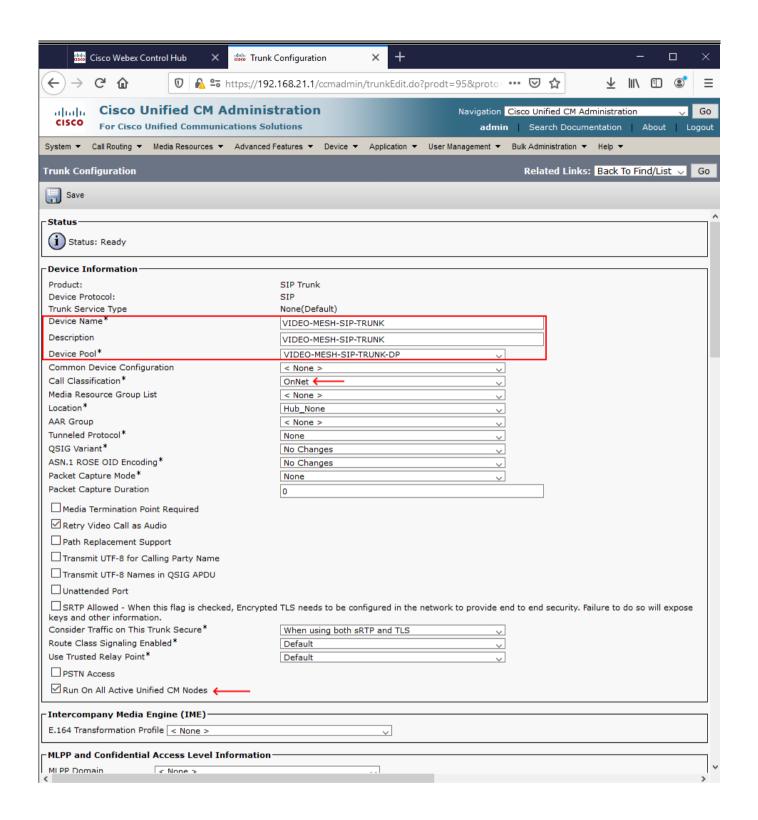


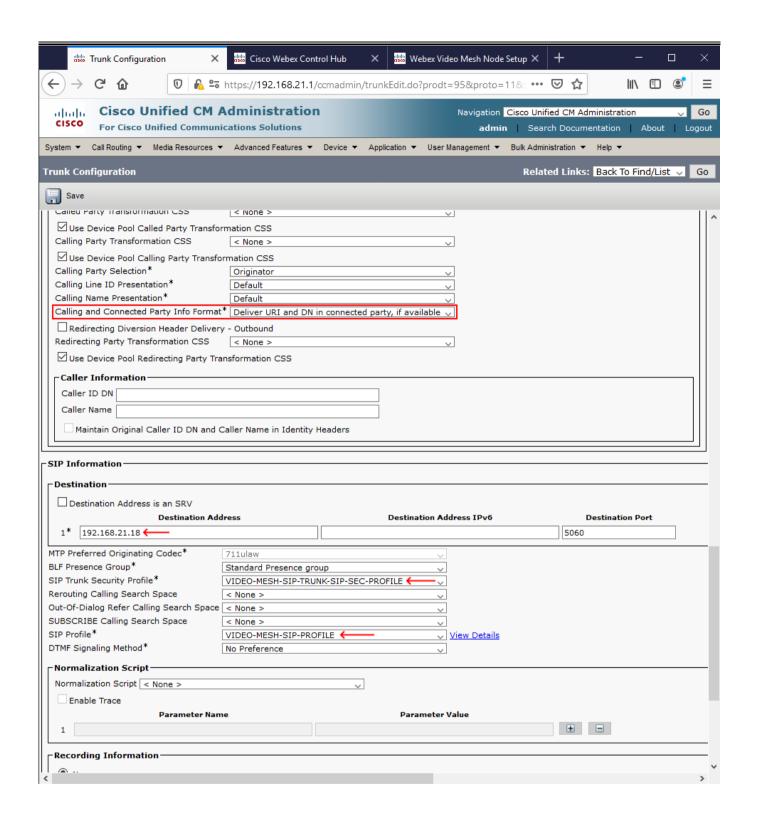


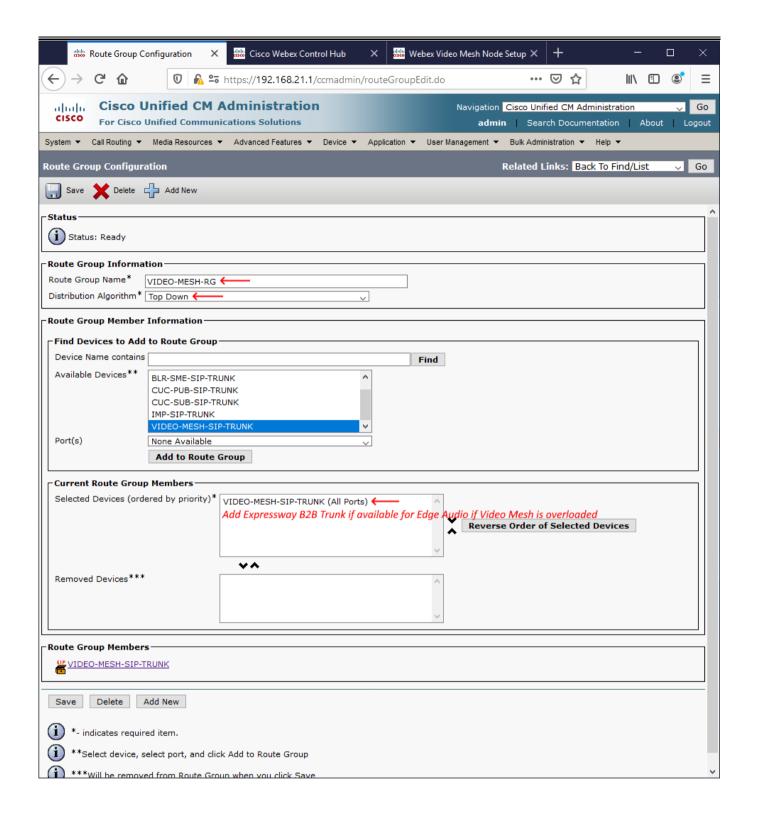
System >> Security >> SIP Trunk Security Profile >> Copy 'Non Secure SIP Trunk Profile'

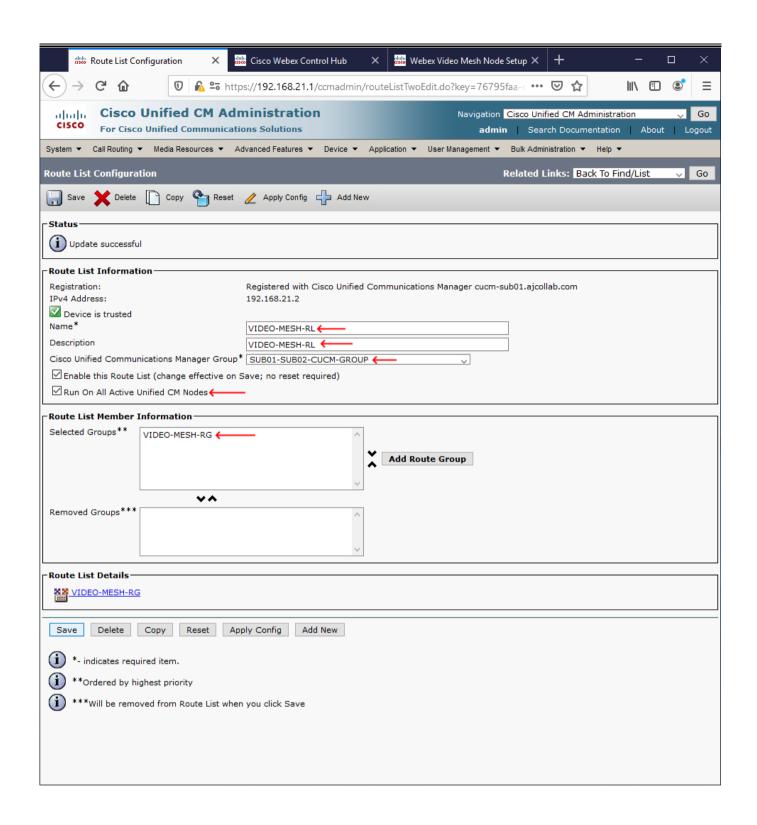


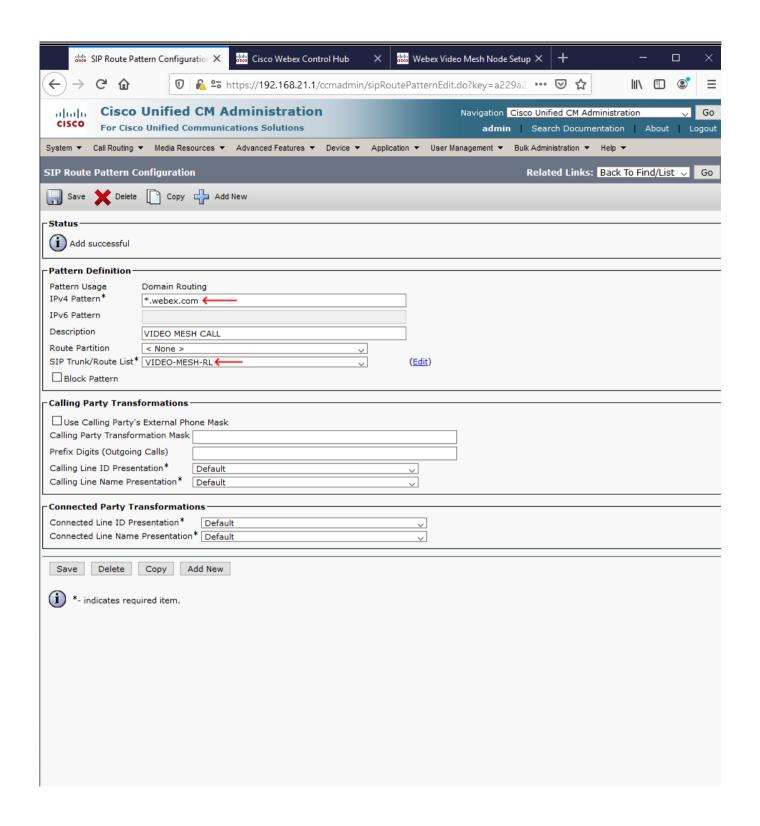
Create a Region and Device Pool for the VIDEO-MESH-SIP-TRUNK







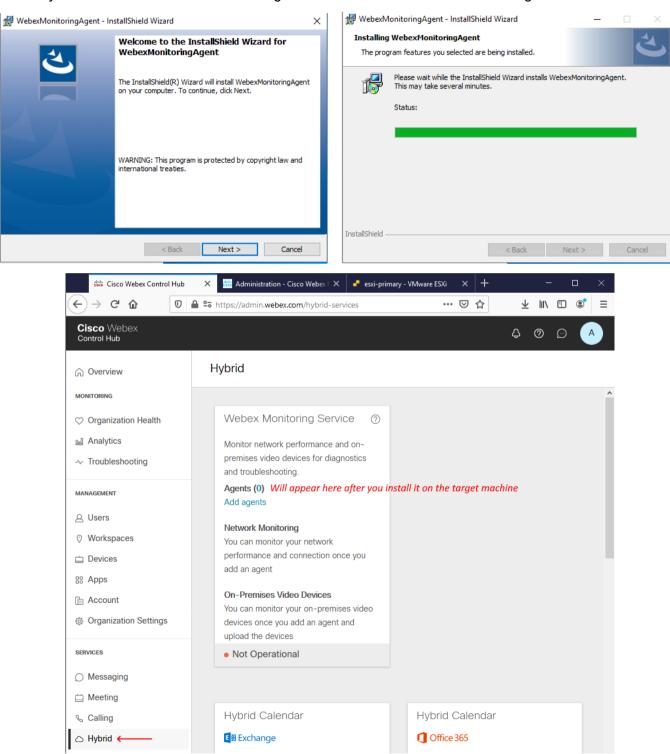




Webex Monitoring Hybrid Service

- Works on Windows Machines, Administrator installs an agent on target machine
- Agent self-register with cloud and provide service statistics and matrices of Webex service for that machine
- Agent captures the network statistics in every 15 minutes and send to Webex Cloud. Control Hub
 can retrieve those information

Go to Hybrid Services >> Webex Monitoring Service >> Download and install the agent



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About the Author



Abdul Jaseem V. P is working as a Consulting Engineer at Cisco TAC Collaboration team Bangalore, India. He has 8 plus years of experience in VoIP, Collaboration and Networking industry. He is from a South Indian state called Kerala, Malappuram district.

He is an expert in Cisco Unified Communications Manager (CUCM), Cisco Unity Connection (CUC), IM and Presence (IMP), Unified Contact Center Express (UCCX), Voice Gateways, PRI, SIP, Cisco Unified Border Element (CUBE),

Expressways, Cisco Meeting Server (CMS), Telepresence and Webex Control Hub.

He started his career as a Desktop Support Technician in a retail supermarket chain located at Dammam, Saudi Arabia. 2 years later he came back to India, worked as a Technical Trainer and then Escalation Engineer at different organizations supporting UC and Collaboration Technology.

He is CCIE Collaboration #59174 certified and holds other industry standard certifications such as vmware VCP, DevNet, AWS Solution Architect Associate and Certified Kubernetes Administrator. Good at Automation and API development, received multiple awards from Cisco for innovation and automation.

This guide is the result of his many sleepless night and personal sacrifice during 2020 COVID-19 situation. Any corrections or suggestion, feel free to contact him via LinkedIn.