CCNA Voice

Guide Version 1.0

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CCNA VOICE Guide 1.0 (43 Pages)

CCNA VOICE (640-461) Syllabus

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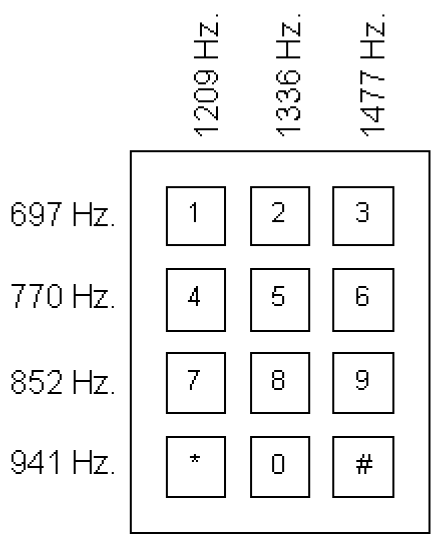
Traditional Voice System

- Traditional telephone network has been placed in 1900s.
- An analog phone uses properties of electricity used to convey audio information over cables.
- When you speak in to an analog phone, the sound that come out of your mouth are converted to a form of electricity.
- For a voice call to occur, certain informations must be passed between the telephony device and the initiator such as on-hook, off-hook, availability, dialing, Ringing etc. These are referred as SIGNALING.

3 Type of Analog Signaling

- For a voice call to occur, certain informations must be passed between the telephony device and the initiator such as on-hook, off-hook, availability, dialing, Ringing etc. These are referred as SIGNALING.
  1. ***Supervisory Signaling***: Involves the detection of the change of status of a loop or trunk. (ON HOOK, OFF HOOK, RING)
  2. ***Address Signaling***: Involves passing dialed digits (PULSE, DTMF)



  3. ***Informational Signaling***: Provide audible tone to user, which indicates certain conditions (Dial tone, Busy, Ring Back, Congestion, Conformation)

FXO, FXS Signaling Methods

FXO & FXS indicates pass signaling via two methods.

Loop Start (PSTN)

- When handset picked up (off-hook) connects the 48V circuit that draws current from the device (Router, PSTN, etc.) & indicates the change in status
- Router or PSTN provide dial-tone to the end device
- User dial the number and destination Rings
- Destination off hook, (Router, PSTN) bridge 2 calls each other
- (Glare issue: when user dial & other used off-hook same time)

Ground Start (PBX)

- It requires ground voltage detection to create loop.
- Used in trunk lines

(Note: Loop start & Ground start are less important in VoIP since these are too old technologies, we have to configure only what matches the other side)

Analog Connectivity Issues

- Distance limitation (Signal strength low, need repeater, amplify noise)
- Wiring limitation (1 Call per line)

Digital Connections and Signaling

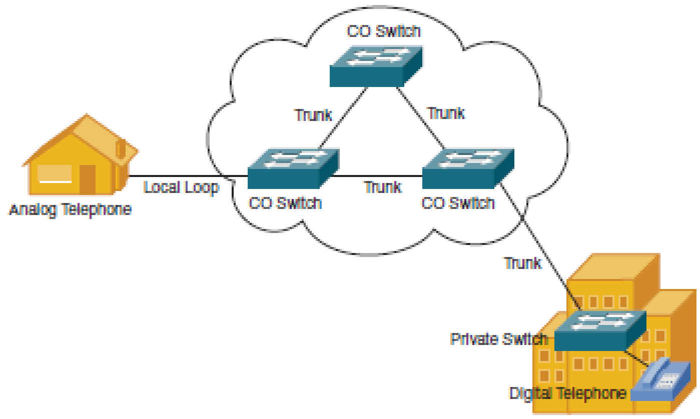
- It uses TDM to transmit voice traffic.
- T1 (24 channels) and E1 (32 Channels) digital circuit
- CAS, CCS, ISDN
- (Refer CVOICE Guide)

Understanding PSTN

- When phone system was originally created, individual phones were wired together to allow communication. If you wanted to connect more than one person you need multiple cables and ports. PSTN eliminates this difficulty.
- Public Switched Telephone Network, PSTN is a huge voice network having dial plan to all over the world.
- PSTN is a network of networks, which connects multiple telecom providers together in to a massive worldwide network.
- All the telecom service provides in the world communicate together via a common protocol called SS7 (Signaling System 7)

- When user makes a call, the first CO performs an SS7 lookup to locate the destination, once found SS7 responsible for call routing through the voice network

Components of PSTN



- ***Analog Telephone***: Convert audio in to electrical signal. Able to connect directly to PSTN
- ***Local Loop***: Link between customer premises and telecom service provider
- ***CO Switch***: Central Office Switch provides services to the local loop, includes Signaling, Digit Collection, Call routing, Call setup & tear down
- ***Trunk***: Provide connection between CO Switches or Private switches (PBX)
- ***Private Switch (PBX)***: Allows business to operate miniature PSTN inside the company. Cost saving for nearby calls.
- ***Digital Telephone***: Typically connects to PBX systems. Convert audio in to binary values.

(Note: Many believe that PSTN will eventually be absorbed to Internet)

Understanding PBX

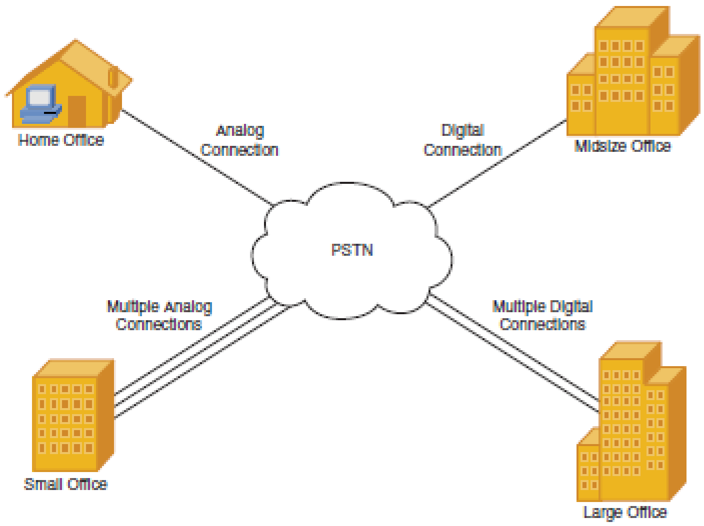
- Many organizations have thousands of phones internally, if they purchase direct PSTN connections the cost would be huge! PBX (Private Branch Exchange) eliminates this issue. PBX allows to internally manage in-house telephones.
- It gives flexibility to internal users to make calls without using PSTN resources (free internal extension to extension call)
- External calls handled by Trunk lines.

PBX Components

- ***Line Card***: Provide connection between telephone and PBX

- ***Trunk Card***: Provide connection from PBX to PSTN or to other PBX
- ***Control Complex***: Intelligence behind PBX system. Call Routing, setup etc.

PSTN Connections



--PSTN Numbering Plan (Country Code, Area Code, Site code, Subscriber Number)

Why VoIP?

- Convert analog voice signal to digital and then placing the binary values as data packet with IP Addressing headers and finally sending over data network.
- Reduced cost of communication: VoIP calls are absolutely free of cost (internet or WAN charges are applicable)
- Reduced cost of cabling: Single Ethernet cable carries voice & Data
- Take your Phone with you (Mobility)
- IP Soft Phone
- Feature-rich communication: Voice, Data & Video combined together (E.g.: IVR)
- Open Standard: PBX are basically proprietary locked. VoIP uses ICT/IP standard

Process of Converting Analog Signals to Digital Signal

1. Sampling

Sampling is the process of cutting an audio stream

Human ear : 20 – 20,000 Hz

Human Speech : 200 – 9,000 Hz

Telephone Channel : 300 – 3,400 Hz (Enough to convey mood of the caller)

Nyquist Theorum : 300 – 4,000 Hz (2x4000 = 8000 Samples per sec)

2. Quantization

Assigning values to sampled values.

A sample is a numeric value consumes a single byte (8 Bit) of information.

3. PCM (Pulse Code Modulation)

Convert the quantized value to digital value of 8 bits

8 Bit can represent 0 – 255, even though Quantization limited to +127 to -127

1st bit used to represent the polarity

a Law (Everywhere) : 1 = +ve; 0 = -ve

µ Law (US, Japan, Canada) : 0 = +ve; 1 = -ve

8000 Samples x 8 bits = 64000 = 64Kbps (Band Width Standard Voice Call)

4. Compress (Optional)

Send a single sound sample and tell the remote device to continue playing that for certain time. It reduces the bandwidth.

Codecs based on compression technique

Common compression: G.729 = 8 Kbps

CODEC (COder-DECoder / COMpression-DECompression)

An audio codec is a device or computer program capable of coding or decoding a digital data stream of audio. The way our voice is coded to packet

CODEC	Sampling	B/W	MOS	Payload	Sample Size (ms)	
G.711	8000 Hz	64Kbps	4.3	240/160	30/20	Including TCP/UDP header total ≈ 80Kbps
G.722	16000 Hz	64Kbps	4.5			Wideband Codec
G.729	8000 Hz	8Kbps	3.92	30/20	30/20	More DSP used, high complexity codec
G.729A	8000 Hz	8Kbps	3.7			Less DSP requirement
iLBC	8000 Hz	15.2Kbps	4.1	50/38	30/20	Industry standard, Open source

G.726 : 32Kbps MOS = 3.85

G.728 : 16Kbps MOS = 3.61

Default VoIP Sample Size = 20mSec

DSPs (Digital Signal Processors)

Basic function of a router to route traffic and it is not a processor intensive task. VoIP is processor demanding task, here comes DSPs to offload the processing responsibility of voice related task from the processor of the router.

A DSP chip performs all Sampling, Encoding and compressing functions.

VIC + DSPs = Processing Voice to Packets

Cisco bundles these chips to PVDMs (Packet Voice DSP Modules)

PVDM2-8	Provides 0.5 (Half) DSP chip
PVDM2-16	Provides 1 DSP chip
PVDM2-32	Provides 2 DSP chip
PVDM2-48	Provides 3 DSP chip
PVDM2-64	Provides 4 DSP chip

DSP consumption is based on the Codecs

Packetizing an Audio (Audio Payload)

1 Sec Audio → 8000 Samples →

1 Sec Audio → 20mSec Packets → 50 Packets Per Sec

8000/50 = 160 Samples in a Packet

Such a packet is called Audio Payload

RTP (Real Time Transport Protocol)

- Protocol to send the codec (Protocol of voice) carry Audio Payload between devices
- RTP operates at the transport layer OSI model.
- RTP is one way stream only. In a 2 way conversation, there will be dual RTP stream between the devices.
- RTP Port Numbers: Random even UDP ports from 16,384 to 32,767

Audio Payload	UDP	**RTP**	IP	Data Link

Payload Type	Sequence Number	Time Stamp

1. Payload Type: Specify Audio or Video communication
2. Sequence number: Allows remote device to put the packet back in the order
3. Time Stamp: Sampling Size 20msec or 30mses

RTCP (Real Time Transport Control Protocol)

- At the time RTP starts, RTCP also engages. Its primary job is statistics reporting which includes Packet Count, Packet Delay, Packet Loss, Jitter (delay variations)
- RTCP Port Numbers: Random odd UDP ports from 16,384 to 32,767
- RTCP packets will be exchanged between every 5 seconds

--CUCM Cluster

--CUCM Call Processing

Understanding Components of Cisco Unified Communication (CME, CUCM-BE, CUCM)

Cisco Call Manager Solutions/ Voice Management Products (IP PBX)

1.CME (450 Limited end devices, no redundancy, router hang, CLI Based also GUI-CCP)

2.CUCMBE (1000 devices, Software in as server platform, No redundancy)

3.CUCM (30,000 devices, Dedicated Call Processing solution)

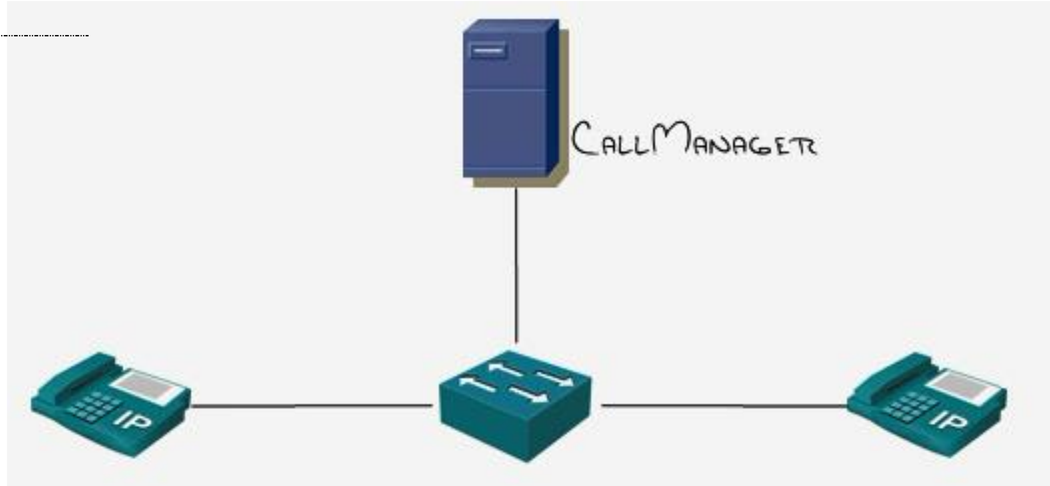
Understanding CUCM (Cisco Unified Communication Manager)

CUCM Version History

Selsius developed call Manager later Cisco purchased → Cisco Call Manager V2.4

<table>
<tr><th>Version</th><th>Platform</th><th>Hardware</th><th></th></tr>
<tr><td>V2.4</td><td>Windows NT</td><td>Any Hardware</td><td rowspan="3">Windows</td></tr>
<tr><td>V3.X</td><td>Windows 2000</td><td>MCS Servers only (Cisco Approved Hardware IBM/HP)</td></tr>
<tr><td>V4.X</td><td>Windows 2000</td><td>MCS</td></tr>
<tr><td>V5.X</td><td>Package installation on Linux</td><td>MCS</td><td rowspan="2">Linux</td></tr>
<tr><td>V6.X</td><td>Linux Based OS</td><td>MCS</td></tr>
<tr><td>V7.X</td><td>Virtual Machine Appliance on VMWare ESXi</td><td>MCS</td><td rowspan="2">Virtual Machine</td></tr>
<tr><td>V8.X</td><td>Virtual Machine Appliance on VMWare ESXi</td><td>Cisco Unified Computing System (UCS)</td></tr>
</table>

Features of CUCM



- Mind of Voice network
- Full support for Audio/ Video telephony
- Built in disaster recovery system
- Directory support (IBM IDS to Windows ADDS)
- Server based cluster deployment
- Device Control
- Dial plan & Permisions
- Phone feature and control
- 20 Server per cluster (Publisher, Subscribers (more), TFTP, DHCP, DNS, Application (Voice Mail), Media resources)

Functions of CUCM

Call Processing

Signaling end devices

Phone feature Administration

Directory Services

Link to external applications

CUCM Hardware Requirements

CUCM 7.X	CUCM 8.X	CUCM 9.X	CUCM 10.X	CUCM 11
2 GHz Processor	2.4 GHz Processor			
2 GB RAM	6 GB RAM			
72 GB Hard Disk	72 GB Hard Disk			

CUCM Installation

1. Setting Virtual Machine (Virtual Hardware)

Create New Virtual Machine → Typical Next → Select ISO → Credentials (Linux Root) → VM Name, Location → 80 GB → Next → Customize Hardware (RAM 2GB, Bridge) → Finish

2. Installing CUCM

Media Check No → Agree → OK → Install → Proceed → Upgrade No → Windows data Import No → Continue → Time Zone (Asia/ Culcutta) → Automatic Negotiation of NIC Yes → Change MTU Size NO → Use DHCP NO (Host Name, IP, Mask, Gateway) → DNS NO → Credential (Platform Administration) → Certificate Info → 1st Node YES → NTP (Real time YES/ LAB NO) → Clock Info → Database Access Password (For DB replication, Communication between nodes) → SMTP NO → Web Page Credentials → Platform Configuration Complete OK

CUCM CLI (10%)

CLI is accessed via SSH protocol using terminal programs (Putty, Secure CRT, Tera Term, etc.)

1. admin:show network eth0 - for getting the IP of CUCM
2. admin:set network ip eth0 142.100.69.10 255.255.0.0 - Change IP
3. admin:set network gateway 142.100.69.11 - Change Gateway
4. admin:set network status eth0 up - Enable network Adapter
5. admin:set network status eth0 down - Disable network Adapter
6. admin:utils network ping 142.100.0.1 - Ping
7. admin:utils network traceroute - Tracert
8. admin:set network domain TEST.COM - Set Domain Name
9. admin:set network hostname TEST_CUCM - Set Host name for CUCM
10. admin:set network dns primary 142.100.69.11 - Set Primary DNS
11. admin:set network dns secondary 142.100.69.12 - Set Secondary DNS
12. admin:utils service restart Cisco Tomcat - Restart web Services
13. admin:utils service list - List all Services
14. admin:utils system restart - Reboot CUCM
15. admin:utils system shutdown - Shutdown CUCM

CUCM Web Administration (90%)

Support IE, Firefox. Doesn’t Support Google Chrome

Go to browser https://[IP ADDRESS_OF_CUCM]/ccmadmin

5 Consoles in CUCM

1. Cisco Unified CM Administration: Core/ Day to day Administration
2. Cisco Unified Service Ability: For Services, Issues Debug
3. Cisco Unified OS Administration: Like an OS (Change IP, Ping, etc)
4. Cisco Unified Disaster Recovery: Backup & restore DB
5. Cisco Unified Reporting: Report, Status report, Call Duration, etc

1. CUCM Admin Interface

This is the main administration of CUCM

System Menu:

Includes tasks for the configuration of Subscribers, CM Group, Device Mobility Group, Device Pool, Region, Location, SRST, Service Parameter, Enterprise Parameter

Call Routing Menu:

Includes tasks to define call routing system, Call hunting, Class of Control, Features (Intercom, Call Park, Call Pickup, etc.)

Media Resources:

For configuring MOH, Annunciator, Media Termination Point, Transcoder etc.

Advanced Feature:

For configuring Voice Mail Integration, Extension Mobility, VPN Features etc.

Device Menu:

For configuring Gateways, Gate Keeper, Trunks, IP Phones, Remote Destination, Device Settings, Phone button and Soft key templates etc.

Application Menu:

CUCM Assistant configuration wizard and Plugin downloads

User Management Menu:

For configuring Application User, End User, User Group and Roles

Bulk Administration Menu:

Perform bulk operation for repetitive tasks

Help Menu:

Access to local searchable help files

2. CUCM Service Ability Interface

For activating and deactivating some basic services in CUCM and troubleshooting

Alarm Menu:

Option for system monitoring and performance, health monitoring

Trace Menu:

Configuring and troubleshooting trace settings

Tools Menu:

Access to CDR Analysis, Service Activation, Control Centre Feature Service, System logs

SNMP Menu:

Configuring SNMP (Simple network Management Protocol) and its authentication

Help Menu:

Access to local searchable help files

3. CUCM OS Administration Page

Here administrator can monitor and interact to the Linux Based OS such as

- Monitor hardware resource (CPU, Disk Speed) utilization
- Check upgrade software version
- Configure NTP (Network Time Protocols) server IP
- Manage server security, Digital Certificate
- TAC Remote Assistance account
- PIG to other devices

4. Disaster Recovery System Interface

Provides DRS backup and restore capability

5. CUCM Reporting Interface

Method to access system reports, logs, issue logs.

Understanding Cisco IP Phones

IP Phone Front Panel

Display	Standard display for IP Phone
MWI	Message Waiting Indicator: Ringing alert, MSG alert (Blink)
Line Buttons	To assign DNs or Speed dial, Special functions etc. Line 1 must be a DN always (Right side to display)
Soft Keys	To assign special features (Redial, Transfer, Hold, etc.)
Selector Button	To navigate through different options
Dial Pad	0 to 9 and *, # numbers
Directory Button	To access cooperate directory
Message Button	To access messages
Service Button	To access services offered by the company
Settings Button	Check IP, Reset, Factory restore, etc.
Help (Info) Button	Help topics
Volume UP/ Down	To adjust volume
Headset Button	To enable headset

Mute Button	To Mute
Speaker	To activate speaker phone

IP Phone Back Panel

Auxiliary Port	(RS232) To connect expansion module
10/100 SW	To connect IP Phone to the switch (Network)
10/100 PC	To connect co-located PC (IP Phone acts as a mini switch)
Power Port	48V DC Power (If there is no PoE)
Headset Port	To connect Cisco Headset (RJ11)
Handset Port	To connect receiver handset

Types of IP Phones

7905	Single Line button, No display, No Auxiliary & PC Port
7906	Single Line button, Small display, No Auxiliary & PC Port
7910, 7911, 7912	Single Phone button, Small display, PC Port, No Auxiliary Port
7920, 7921, 7922	Single Phone Button, IP Based Mobile phones (Wi-Fi)
7935, 7936	1 Line buttons, Auxiliary & PC port, For Conference
7940, 7941, 7942, 7945	2 Line buttons, Auxiliary & PC port, Big Screen
7960, 7961, 7962, 7965	6 Line buttons, Auxiliary & PC port, Big Screen, 7965 is Color
7970, 7971, 7972	8 Line buttons, Auxiliary & PC port, Big Screen, Color
7985, 7986	8 Phone Button, Auxiliary & PC port, Big Screen, Color, Support video
Exp Modules 7914, 7915, 7916	These are not phones, but Expansion modules. 14 Line buttons, Uses Auxiliary port to interface. We can add 2 exp module to an IP Phone
8800	Smart Phones
8900	Smart Phones, Video Support, Inbuilt camera
9900	Smart Phones, Video Support, Inbuilt camera

Protocols & Services used by an IP Phone

IP Phones requires set of Protocols and services

- NTP (Network Time Protocol)
- CDP (Cisco Discovery Protocol)
- DHCP (Dynamic Host Configuration Protocol)
- TFTP (Trivial File Transfer Protocol)
- PoE (Power Over Ethernet)
- DNS (Domain Name System)

NTP (Network Time Protocol):

IP standard provides network based time synchronization. It ensure same time on all devices based on time different time zones. Time synchronization is important for many functions to operate

CDP (Cisco Discovery Protocol):

Is a cisco proprietary L2 protocol provides network mapping information for directly connected Cisco devices

DHCP (Dynamic Host Configuration Protocol):

DHCP is faster, easier, widely accepted method to distribute IP information to the clients. It provide following information to the IP Phones

- IP Address
- Subnet Mask
- Default Gateway
- DNS Server
- TFTP Server IP

DHCP can be provided by an existing DHCP server or Local router or CUCM itself

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. Server uses Port 67 and client uses port 68.

DORA Process in DHCP

Discover: Broadcast: Can somebody give me an address?

Sender 0.0.0.0 UDP 68, Receiver 255.255.255.255 UDP 67

Trying to discover whether there is any DHCP Server

Offer: Broadcast: I can give you 192.168.1.2, Mask, G/W, DNS, Lease Duration

Sender 192.168.1.1 UDP 67, Receiver 255.255.255.255 UDP 68

Request: Broadcast: I want to accept the offer, allow me to use 192.168.1.2 for 3600 seconds. Client takes the address offered by the Server pool

Sender 0.0.0.0:68, Receiver 255.255.255.255:67

ACK: Unicast to confirm the acceptance of provided IP info

PoE (Power Over Ethernet):

Provides DC power over Ethernet cabling. Helps less cabling efforts and cost saving (Wiring, No Power supply needed)

DNS (Domain Name System)

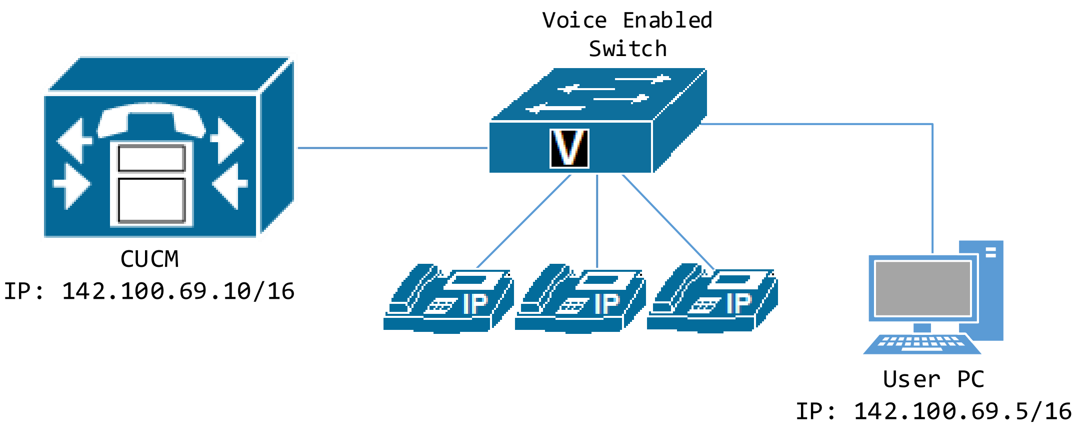
DNS provides IP to Domain mapping and vice versa. It is not critical to IP Phones. ADNS server must be external to the CUCM Cluster; DNS is not a service that CUCM can offer.

---Other VoIP Devices

1. ATA
2. VG224: 2 Ethernet Ports + 24 FXS Ports
3. VG248:
4. Cisco VT Advantage
5. Cisco VT Camera

IP Phone Registration in CUCM

IP Phone registration (Manual)



Step 1: Service Activation:

Cisco Unified Services Ability → Tools → Service Activation

- Cisco CallManager
- Cisco Tftp
- Cisco DHCP Monitor Service

→ Save

Go to Tools → Control Center – Feature Services check the service activation status. It *should be 0 days always*. We can Stop, Start, Refresh those services from here.

Step 2: Configure CUCM DHCP Server (if there is no DHCP Server)

Cisco DHCP Monitor Service must be enabled

Cisco Unified CM Administration → System → DHCP → DHCP Server

Find → Add New

Host Server (Select) → Save

System → DHCP → DHCP Subnet

Find → Add New

DHCP Server*	<Select_SERVER_NAME>
Subnet IPv4 Address*	<142.100.0.0>
Primary Start IPv4 Address*	<142.100.0.50>
Primary End IPv4 Address*	<142.100.0.100>
Primary Router IPv4 Address	<GATEWAY:142.100.69.254>
IPv4 Subnet Mask*	<255.255.0.0>
Primary TFTP Server IPv4 Address (Option 150)	<142.100.69.10>

→ Save

IP Helper Address for DHCP Broadcast

Router(config)#interface ethernet 1/0

Router(config-if)#ip helper-address 142.100.0.254

Step 3: IP Phone Registration (Manual)

Go to Device → Phone → Find → Add New

Phone Type* (Select type) → Next → Select the device protocol: SCCP → Next

Device Information	
Device Name*	<MAC_ADDRESS(SEP005056C00001)>
Description	<Cisco IP Communicator>
Device Pool*	<SELECT_POOL_OR_DEFAULT>
Phone Button Template*	<Standard CIPC SCCP_OR_SELECT>
Softkey Template	<SELECT_FROM_LIST>
Common Phone Profile*	<Standard Common Phone Profile>
Protocol Specific Information	
Device Security Profile*	<Standard SCCP Non-Secure Profile>

→ Save

Association Information

Line [1] - Add a new DN → Directory Number Information

Directory Number* <1000> → Save

Alert Name : Called party Name when the phone is Ring back. If Phone A calling Phone B, "Calling NAME B" is displayed on Phone A when alert name configured on Phone B.

Display Name : Internal caller ID, appear on the remote phone

Go to Device → Phone to see registered phones

IP Phone Registration Automatic

Step 1 & 2 is same

Remove Unassigned DN from Call Routing → Route Plan Report → Unassigned DN → Find → Select All → Delete

Cisco Unified CM Administration → System → Cisco Unified CM → Find

Select Server

Under Auto-registration Information

Starting Directory Number* <1000>

Ending Directory Number* <1010>

Remove check mark from

Auto-registration Disabled on this Cisco Unified Communications Manager

→ Save

Go to Device → Phone to see registered phones

- Cisco Unified CM Group → Check → Auto-registration Cisco Unified Communications Manager Group (All the auto reg phones comes under this group)
- Enterprise Parameters → Auto Registration Phone Protocol* → SCCP
- Device → Device Settings → Device Defaults (We can change the 'Device Pool' and 'Phone Button Template' for Auto reg phones)

IP Phone Registration (BAT) (Check after Device Pool & Button Template)

BAT is a fast way to add, remove, modify database entries for almost every part of CUCM Database.

1. Service Activation

Cisco Unified Serviceability → Tools → Service Activation → Cisco Bulk Provisioning Service

2. Creating Phone Template

Cisco Unified CM Administration → Bulk Administration → Phone Template → Find → Add New → Phone Type → <Cisco 7960> → Next → Select the device protocol: SCCP → Next →

Template Name*	<7960_SCCP_FINANCE_DEPARTMENT_PHONES>
Device Pool*	<BANGALORE_DP>
Phone Button Template*	<Standard 7960 SCCP>
Common Phone Profile*	<Standard Common Phone Profile>
Device Security Profile*	<Cisco 7960 - Standard SCCP Non-Secure Profile>

→ Save

Click Line [1]

Line Template Name* <LINE_7960_SCCP_FINANCE_DEPARTMENT_PHONES>

→ Save

Related Links → Configure Device (7960_SCCP_FINANCE_DEPARTMENT_PHONES) → Go

→ Save

3. Creating Phone File Format

Bulk Administration → Phone → Phone File Format → Create File Format → Find →

Add New (or copy 'Simple Phone Format' then edit)

Format Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONE_FILE_FORMAT>

Device Fields: Device Name, Description

Line Fields: Directory Number

IP Phone Service Maximums: Maximum Number of Lines: 1

→ Save

4. View File Format

Bulk Administration → Phone → Phone File Format → Add File Format →

Format File Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONE_FILE_FORMAT>

Then click (View File Format) link, copy the content to a notepad file.

DEVICE NAME,DESCRIPTION,DIRECTORY NUMBER 1

Sample: ***(7960_SCCP_FINANCE_DEPARTMENT_PHONES_BAT.txt)***

```
DEVICE NAME,DESCRIPTION,DIRECTORY NUMBER 1
02004C4F4F51,SALES1,1001
02004C4F4F52,SALES2,1002
02004C4F4F53,SALES3,1003
02004C4F4F54.SALES4.1004
```

5. Upload File .txt file

Bulk Administration → Upload/Download Files → Add New →

File: → Browse → 7960_SCCP_FINANCE_DEPARTMENT_PHONES_BAT.txt

Select The Target* <Phones>

Select Transaction Type <Insert Phones - Specific Details>

(Optional 'Overwrite File if it exists.**')

→ Save

6. Validate Phones

Bulk Administration → Phones → Validate Phones →

Validate Phones Specific Details

File Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONES_BAT.txt>

Phone Template Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONES>

Job Description <7960_SCCP_FINANCE_DEPARTMENT_PHONE_VALIDATE>

Bulk Administration → Job Scheduler → Find → Click Job ID → Log File Name and check the status

7. Insert Phones

Bulk Administration → Phones → Insert Phones →

File Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONES_BAT.txt>

Phone Template Name* <7960_SCCP_FINANCE_DEPARTMENT_PHONES>

(Optional: Override Options)

Job Information

Job Description <7960_SCCP_FINANCE_DEPARTMENT_PHONES_INSERTING>

Run Immediately <Check>

→ Submit

Bulk Administration → Job Scheduler → Find → Click Job ID → Log File Name and check the status

Go to Device → Phones to see registered Phones.

Phone Button Template

Defines the behavior of Phone Buttons. Two buttons are assigned for DN by default, but we can customize our own.

Device → Device Settings → Phone Button Template → Find → Copy Standard 7960 SCCP → Button Template Name* <Type 1 Standard 7960 SCCP> → Save →

Button Information

Button	Feature	Label
1	Line**	MY NUMBER
2	Speed Dial	<SPEED DIAL>
3	Privacy	<Privacy>
4	Service URL	<Service URL>
5	Speed Dial BLF	<Speed Dial BLF>
6	Call Park BLF	<Call Park BLF>

→ Save

Device → Phone, Select phone by clicking in the device Name

Device Information

Phone Button Template* <Type 1 Standard 7960 SCCP> - The one we created!

→ Save

New Button template has been applied; we can apply this template while registering IP Phones also.

Configure Template for Particular Phone

Device → Phone, Select phone by clicking in the Device Name again

Association Information

Select buttons

2. Speed Dial Settings

Speed Dial Settings

	Number	*Label*	*ASCII Label*
1	1003	MANAGER	MANAGER

→ Save → Close

Abbreviated Dial Settings

	Number	Label	ASCII Label
2	1004	USER1	USER1

AbbrDial softkey is used to dial Abbreviated speed dial

4. Add a new SURL

5. Add a new BLF SD (Busy Lamp Field / Speed Dial)

Destination	*Directory Number*	*Label*	*Label ASCII*
1001	<None>	MANAGER	MANAGER

→ Save → Close

6. Add a new BLF Directed Call Park (Busy Lamp Field/Directed Call Park Button Settings)

Directory Number:Partition	*Label*	*Label ASCII*
	PARK1	PARK1

→ Save → Close

Softkey Template

Controls what Softkey button functions are available to the user (Redial, Hold, Conference, Transfer, Park, Mobility etc.)

Device → Device Settings → Softkey Template → Find

Copy 'Standard User'

Name* <SPECIAL_USER>

Description <FOR_SPECIAL_USERS>

Related Links: Configure Softkey Layout → GO

Select a call state to configure (ON HOOK, OFF HOOK, ON HOLD, CONNECTED, etc.) and assign Required Softkeys.

→ Save

Device Pool

It provides a set of common configuration to a group of devices. Typically we are creating one devoice pool per an area (but we can create more if needed).

System → Device Pool to see the existing device pool

The components of device pool is given below.

1. Cisco Unified CM Group

It defines a top-down ordered list of redundant subscriber servers to which the phone can register. It includes maximum of 3 subscriber server plus optional SRST reference. The goal of CUCM Group is to provide server redundancy for IP Phones.

SUB-SERVER1	Primary Call Manager
SUB-SERVER2	Secondary Call Manager
SUB-SERVER3	Tertiary Call Manager

Phone initially register with Primary Call Manager if it goes down, IP Phone contact Secondary Call Manager and so on.

Configuration

System → Cisco Unified CM Group → Find → Add New

Put Name to the Group

Cisco Unified Communications Manager Group Members

Available Cisco Unified Communications Managers	SUB-SERVER1 SUB-SERVER2 SUB-SERVER3 SUB-SERVER4
	▼▼▼▼▼▼ (Select & Pull-down)
Selected Cisco Unified Communications Managers (CUCM Group 1)	<SELECTED_SUB-SERVER1> <SELECTED_SUB-SERVER2> <SELECTED_SUB-SERVER3>
Selected Cisco Unified Communications Managers (CUCM Group 2)	<SELECTED_SUB-SERVER2> <SELECTED_SUB-SERVER1> <SELECTED_SUB-SERVER3>

NB: If we check ***Auto-registration Cisco Unified Communications Manager Group***, all the Auto Reg phones comes under this group

2. Date & Time Group

If the devices are not in NTP server's time zone, the displayed time will be wrong. 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 zone.

Configuration

System → Date and Time Group → Find → Add New

Group Name*	BANGALORE_TIME_1
Time Zone*	Indian Standard Time - (GMT+5:30)
Separator*	/ (Slash)
Date Format*	D/M/Y
Time Format*	12-hour

3. Region:

It used to set bit rate for a call. Same region uses High Bit Rate Codec and region to other region uses low bit rate codec. (Same region call: G.711, Region to Other region: G.729)

Configuration

System → Region → Find

Add New

Name* **BANGALORE_REGION** → Save

Modify Relationship to other Regions

BANGALORE_REGION G.711

→ Save

Add New

Name* **CHENNAI_REGION** → Save

Modify Relationship to other Regions

BANGALORE_REGION G.729

CHENNAI_REGION G.711

→ Save

Add New

Name* **DELHI_REGION** → Save

Modify Relationship to other Regions

BANGALORE_REGION G.729

CHENNAI_REGION G.729

DELHI_REGION G.711

→ Save

4. Location:

To set Bandwidth, defines maximum amount of band width used by calls from a location. Each call is tracked and bandwidth used is deducted from the total of that location. If bandwidth of the particular location exceeds the call may dropped or routed to PSTN.

Configuration

System → Location → Find → Add New

5. SRST Reference

System → SRST → Find → Add New

Name* <BANGALORE_GATEWAY_SRST>

Port* 2000

IP Address* <142.100.69.254>

→ Save

6. MGRL

7. Auto Registration CSS (Auto Call to Admin)

8. LRG

Configure Device Pool

System → Device Pool

Device Pool Settings

Device Pool Name* BANGALORE_DP

Cisco Unified Communications Manager Group* BANGALORE_GROUP1

Roaming Sensitive Settings

Date/Time Group* <BANGALORE_TIME_1>

Region* <BANGALORE_REGION>

SRST Reference* <BANGALORE_GATEWAY_SRST>

→ Save

Applying Device Pool to Phones

Device → Phone → Phone Configuration page → Device Pool <BANGALORE_DP>

→ Save

IP Phone Boot up Process

1. Obtain power

Sources

- Cisco PoE (10W)
- IEE802.3AF (15.4W)
- Mid-span Power
- Electrical Power

Process

From CDP message switch can be able to analyze how much power needed for an IP Phone Cisco switch send FLP (Fast Line Pulse 147KHz), phone receive and send back FLP. Then switch provide required power. 3rd party phone will take full power.

2. Run Boot Strap Loader

IP Phone runs a bootstrap loader and expand the firmware image that is stored inside the flash memory of phone. This will initiate the software and hardware of IP Phone.

3. Obtain VLAN information

Phone send CDP request message to get VLAN (VOICE) information. If voice VLAN information not configured, there will be a delay in this stage.

To avoid the delay use #switchport voice vlan untagged

4. Obtain IP Address

IP Address can be obtained from DHCP server.

- IP Address
- Subnet Mask
- Gateway IP
- TFTP Server IP

If DHCP server in a remote location #ip-helper address <IP> command is used to change DHCP broad cast to Unicast.

5. Download Configuration File

Phone contacts TFTP server and request its configuration file. Each phone has customized configuration created by CUCM and uploaded to when administrator creates and modifies the phone.

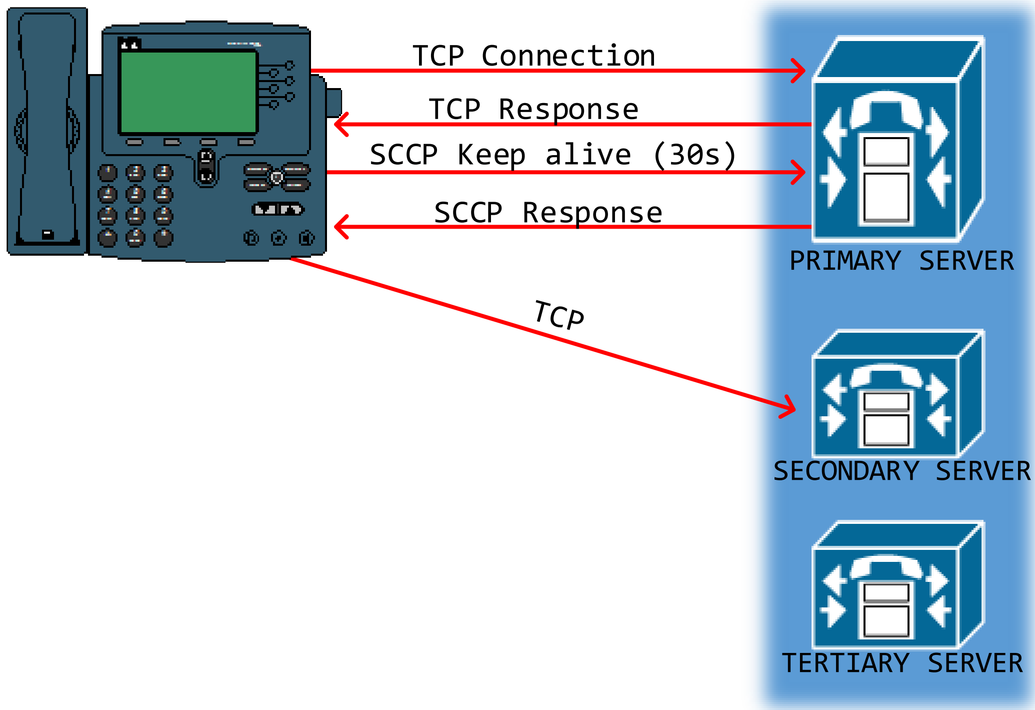
For SCCP Phones : **SEP<MAC-ADDRESS>.cnf.xml**

For SIP Phones : **SIP<MAC-ADDRESS>.cnf.xml**

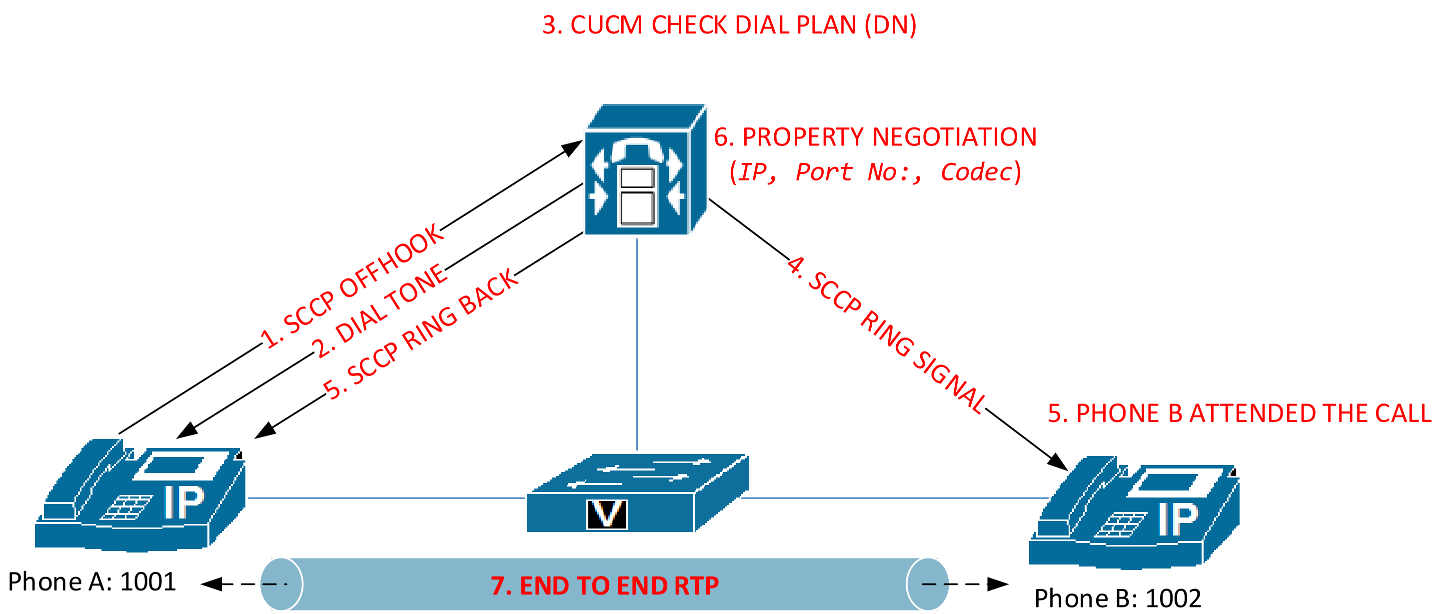
If above file is not present **xmldefault.cnf.xml** will be downloaded (For auto reg also)

6. Registering with CUCM

Phone register with primary CUCM listed in the configuration file. CUCM server will send other configurations such as DNs, Softkeys, etc. via SCCP messages in the last stage of registration process.



CUCM Call Flow



Key Protocols

- SCCP (Signaling Protocols)
- SIP (Signaling Protocols)
- RTP

Call Flow

1. 'A' sends SCCP OFF HOOK signal to CUCM
2. CUCM sends DIAL TONE to 'A'

3. ‘A’ dials DIRECTORY NUMBER (DN) [Digit by Digit (SCCP) or En block (SIP)]
4. CUCM check DIAL PLAN (DN, Manipulation, Call Privilege, Call Coverage)
5. When match found CUCM send SCCP RING signal to ‘B’ and SCCP RING BACK signal to ‘A’
6. ‘B’ attended the call
7. Property negotiation (IP, Port No:, Codec)
8. Establish end to end RTP media streaming + RTCP

Maximum Number of Calls and Busy Trigger

Maximum Number of Calls: = Active calls + outgoing calls

Busy Trigger: = Active Calls + Incoming Calls

Maximum Number of Calls >= Busy Trigger

User Accounts

Why would we create End users?

- Next evolution of phone system (Like Skype)
- Soft Phones requires user logins (Call center setup)
- SIP Phone need user account
- User can manage their own phones (Reducing administrator load - Speed Dial, Change Language)
- Extension mobility (Like Roaming Profile in Windows)
- Tracking (All international Phone calls report per user)

1. End User

End users are typically people who type User Name & Password in a login screen (usually web page) to access some features or control.

- Associated with actual person
- For individual interactive login
- Used to assign user features and admin rights
- Create locally & Can be integrated with LDAP (Light Weight Directory Access Protocol- Microsoft, Netscape, SUN, IPLANET)

2. Application User

Application user is typically applications which send authentication information to read or write something on the system.

- Associated with an Application
- For non-interactive logins
- Used for application authorization
- Create locally, unable to integrate with LDAP

Structure of User Accounts

User accounts in a CUCM server contains 2 sections (General & CCM Data)

General Data

- User ID/ First/ Last/ Middle Name
- Department
- Phone Number/ Mail ID
- Password

CCM Data

- PIN Number
- CUCM Administration Privileges
- Associated IP Phones/ DN
- Feature Access

End User Configuration

1. Manual Local data (No LDAP)
2. BAT
3. LDAP Sync: General info managed from LDAP server, Password managed from CUCM
4. LDAP Authentication: Password managed by LDAP

1. End User Manual Configuration Local Data

Cisco Unified CM Administration → User Management → End User → Add New

User ID	:	Jaseem
Password	:	Ciscovoice123
PIN	:	123 (Extension Mobility)
Last Name	:	vp
Associated Devices	:	<Select Phones> → Save

User Management → User Group → Standard CCM End Users → Add → Find → Add Selected

http://[IP_of_CUCM]/ccmuser

Now user can login to user page and manage (limited access) the associated phones.

User Options → Device →

Line Settings, Speed dial, User Settings, Password, PIN, etc.

Managing User Groups, Roles, Privileges (Sub Administrators)

1. User assigned to Group
2. Group assigned to Roles

3. Roles assigned to Privileges

1. Creating Groups

User Management → User Group → Add New

Name : Read_Only_Administrator_Group

Add End User to the Group → Select User → Add Selected → Save

2. Creating Roles

User Management → Roles → Find → Copy/ Add New

Role Information: Application →

Cisco Call Manager Administration

Cisco Call Manager Service Ability, etc. → Next

Name : Read_Only_Administrator_Roles

3. Select Privileges according to our requirement

4. Assign Roles

User Management → User Group → Roles (!) → Assign Role to Group → Select Roles (Read_Only_Administrator_Role)

Standard CCM Admin Users (To get the Web Page Access)

→ Add Selected → Save

2. End User BAT Configuration

Step 1. Service Activation

Cisco AXL Web Service

Cisco Bulk Provisioning Service

Cisco TAPS Service

Step 2. Create Template

Bulk Administration → User → User Template → Find → Add New

User Template Name* : Sales_Users_Template

Check : Default Password to user ID

User Group : Standard CCM End User

→ Save

Step 3. Create CSV File (Use .xlt file to generate .txt CSV file)

Fill the text file with required info

Step 4. Upload File

Bulk Administration → Upload/ Download → Add New

Browse,

Select The Target* : Users

Transaction Type : Insert Users

Step 5. Insert Users

Bulk Administration → Users → Insert Users

LDAP

LDAP allows organizations to create single, centralized directory information store. LDAP holds informations about User Account, Passwords, Privileges, etc. Informations in LDAP available to other applications & softwares hence no need to maintain separate directories. It simplifies user administration because only need to store user information in a single place.

LDAP Sync: some user data is maintained in LDAP Server & replicated to CUCM. General data obtained from LDAP will be read only. But some user attributes can be edited from CUCM. User password is local to the CUCM only.

LDAP Sync: General info managed from LDAP server, Password managed from CUCM

LDAP Authentication: It redirects the password authentication from CUCM to LDAP System. Centralized password system. It problem in LDAP authentication, when LDAP server fails nobody can login.

LDAP Mechanism

1. All existing User account in CUCM database are deactivated (not deleted)
2. Accounts whose CUCM User ID exactly match with user ID in LDAP are reactivated & its attributes updated from LDAP to CUCM
3. LDAP accounts are replicated to CUCM database
4. CUCM deactivated accounts will be deleted after 24 hours

3. Adding End User by LDAP (LDAP Sync)

We are discussing LDAP integration using Windows Active Directory. Windows ADDS perform full sync of all records every time, leads server busy and affect performance, hence schedule to off hours. Here Password managed by CUCM

Windows Server 2012 Side Configuration

1. Installing Windows ADDS

i. Server Manager → Add Roles and Features → Server Roles

- Active Directory Domain Services
- Active Directory Lightweight Directory Services

Install → Restart Server

ii. Promote Windows server 2012 to a Domain Controller (Configure ADDS)

Action Center flag → Promote this server to a domain controller

Add a new forest → Domain name <smart.com> → Next

Directory Service Restore Mode (DSRM) <********Ciscovoice123>

→ Next → Next → Next → Next → Next → Install

2. ADDS Configurations

Server Manager → Tools → Active Directory Users and Computers

Right click on the domain name <SMART.COM> → New → Organizational Unit

Give a name <CUCM Users> → OK

Expand the OU and create users ***(Last Name is used for SYNC)***

CUCM Side Configuration

Cisco Unified Serviceability → Tools → Service Activation → Cisco DirSync → Save

Cisco Unified CM Administration → System → LDAP → LDAP System

Check Enable Synchronizing from LDAP Server

LDAP Server Type <Microsoft Active Directory>

LDAP Attribute for User ID <sAMAccountName> → Save

System → LDAP → LDAP Directory → Find → Add New

!!!** Existing End Users found in the corporate directory will be deleted**!!

LDAP Directory Information

LDAP Configuration Name* <CUCMLDAP1>

LDAP Manager Distinguished Name* <administrator@smart.com>

LDAP Password* <Ciscovoice123>

Confirm Password* <Ciscovoice123>

LDAP User Search Base* <ou= CUCM Users, dc=smart, dc=com>

LDAP Directory Synchronization Schedule

Perform Sync Just Once <Check>

LDAP Server Information

Host Name or IP Address for Server	LDAP Port
142.100.128.0	389

→ Save → Perform Full Sync Now

Check status from User Management → End User → Find

4. CUCM LDAP Authentication

LDAP Authentication: Password managed by LDAP (Password not replicating, entire Authentication redirected to LDAP server)

Cisco Unified CM Administration → System → LDAP → LDAP Authentication

Check Use LDAP Authentication for End Users

LDAP Manager Distinguished Name <administrator@smart.com>

LDAP Password <Ciscovoice123>

Confirm Password <Ciscovoice123>

LDAP User Search Base <ou= CUCM Users, dc=smart, dc=com>

Host Name or IP Address for Server	LDAP Port
142.100.128.0	389

→ Save

CUCM User Phone Telephony Features

CUCM supports wide range of features.

Some basic annunciator will can be activated using

Cisco Unified CM Administration → Tools → Service Activation → Cisco IP Voice Media Streaming App → Save

It is also called Call Coverage Features, that refers several feature and mechanism that all incoming calls are answered

1. Shared Line

Same Directory Number for many phones, all those phones will ring same time.

Cisco Unified CM Administration → Device → Phone → <Select Phone> → Line [2] - Add a new DN

→ Directory Number* : 1111 → Save

NB: All the phones will be ringing simultaneously, when connected to one, others will be in 'Remote in Use' mode. When the connected phone goes to 'Hold' others can 'Unhold or Resume' the call (Less Privacy)

To overcome the issue, enable privacy in service parameter

Cisco Unified CM Administration → System → Service Parameters

Server : <Select Server>

Service : Cisco Call Manager (Active)

Enforce Privacy Setting on Held Calls : True

2. Conference on Shared Line (Barge & Privacy)

If 2 phones have shared line configured & one of the hone is using that line, the second phone can force a three-way conference by using Barge feature. The conference is hosted on first phones 'built in Conference Bridge'. It doesn't use CUCM resources for conferencing. Phone itself have some DSPS (Built in Bridge)

Create a Phone Button Template with 'Privacy Button' and apply to the shared line phones.

Enable 'Barge Soft Key' while 'Remote in Use' state, or use 'Standard Feature Soft key template'

(Single Phone)

Device → Phone → <Select Phone>

Built In Bridge : On

Privacy : Off

Or (Cluster wide)

Cisco Unified CM Administration → System → Service Parameters

Server : <Select Server>

Service : Cisco Call Manager (Active)

Builtin Bridge Enable : On

Privacy Setting : False

3. Join Conference

This uses CUCM resources to perform conference.

Enable 'Join' soft key while connected state

Answer call → put on hold → Answer next call → Put on hold → Join button

3 call on hold press Join Button

4. Call Back

Monitor the availability of another user when he come back.

Add soft key 'Call Back (CallBack)' template while 'on hook & ring out' state.

Dial a Number + CallBack

5. Call Forward

All or some calls to be forwarded to a destination number by the User or Administrator either at the phone itself or at CUCM User/Admin pages. During call forward, the CSS in device & Line are ignored, hence Call Forward CSS must be configured.

1. Forward All from the Phone

Set Call forward All (CFwdALL) all soft key while 'On Hook' state

2. Conditional Call Forward

Go to line page → Call Forward and Call Pickup Settings

Forward All

Forward Busy Internal

Forward Busy External

Forward No Answer Internal

Forward No Answer External

Forward No Coverage Internal

Forward No Coverage External

Forward on CTI Failure

Forward Unregistered Internal

Forward Unregistered External

No Answer Ring Duration (seconds) : 10

Create End User & Associate a phone, then he can also configure Forwards.

6. Transfer

Transferring is a kind of forwarding done after speaking between two parties.

1. Consulting Transfer

There will be a consulting between 2 parties before transferring. We have to press the Transfer Softkey 2 times, one to start consulting and again to transfer call.

Also user can Press Transfer Softkey + DN + Transfer Softkey without consulting.

Enable 'Transfer (Trnsfer)' soft key while Connected state

2. Direct Transfer

Call 1 : Hold

Call 2 : Active

Select Held call the Press DirTrfr

DirTrfr : Call 1 & Call 2 get connected

Enable 'Direct Transfer (DirTrfr)' soft key while Connected state

7. Call Park

Allows user to temporarily attach a call to a Park Slot number. It is a special kind of holding.

Call Routing → Call Park →

Call Park Number/Range : 222X

Cisco Unified Communications Manager : CUCM-SUB01

Enable ‘Park’ soft key while connected state

System → Service Parameters →

Call Park Display Timer* : 10 (Shows message “Call Park At 222X”)

Call Park Reversion Timer* : 60 (Ring back the initial phone)

8. Direct Call Park

Call Routing → Direct Call Park →

Number : 333X

Reversion Number : 1005

Retrieval Prefix : *

→ Save

Use ‘Transfer (Trnsfer)’ and transfer to 333X (eg: 3330). To retrieve call dial *3330 from any phone. If not answered 1005 will ring.

9. Call Pickup

Ability to pick someone else ringing phone.

Define a soft key template while ‘on hook’ state consisting following keys

Pick Up (PickUp)

Group Pick Up (GPickUp)

Other Pickup (oPickup)

<u>Create Pickup Groups</u>

<u>Eg. Group 1:</u>

Call Routing → Call Pickup Group

Call Pickup Group Name : SALES_GROUP

Call Pickup Group Number : 1999

Description : SALES_GROUP_PICKUP_NUMBER

→ Save

<u>Eg. Group 2:</u>

Call Routing → Call Pickup Group

Call Pickup Group Name : PURCHASE_GROUP

Call Pickup Group Number : 2999

Description : PURCHASE_GROUP_PICKUP_NUMBER

→ Save

<u>Associate Phones to Pickup Group</u>

Device → Phones 1 → Line → Call Forward and Call Pickup Settings →
Call Pickup Group : SALES_GROUP

Device → Phones 2 → Line → Call Forward and Call Pickup Settings →
Call Pickup Group : SALES_GROUP

Device → Phones 3 → Line → Call Forward and Call Pickup Settings →
Call Pickup Group : PURCHASE_GROUP

Device → Phones 4 → Line → Call Forward and Call Pickup Settings →
Call Pickup Group : PURCHASE_GROUP

1. Auto Pickup

Phones in same group can be picked up with a single soft key. When Phone 1 is ringing, hit 'PickUp' soft key and 'Answer'

To get single button answer

System → Service Parameters →

Auto Call Pickup Enabled* : True

2. Group Pickup

Ringing Phones in a group can be picked up from other group. When Phone 3 or Phone 4 ringing, Phone 1 or 2 can pick up by hitting 'GPickUp' + GROUP_NUMBER (eg: 2999)

3. Direct Pickup

'GPickUp' soft key followed by Directory number in same group. Or associate all the groups.

4. Other Pickup

Call Routing → Call Pickup Group → Select a Pickup Group from 'Available Call Pickup Groups' and add to 'Selected Call Pickup Groups' → Save

Do the same for other Pickup groups that is associate all Groups

Now just press 'oPickup' soft key to pick other group calls

10. Do not disturb

Add soft key 'Toggle Do Not Disturb (DND)' while on hook

When we press 'DND', under CUCM phone the 'Do Not Disturb' filed will be checked.

DND Option : Ringer off

DND Incoming Call Alert : Flash only/ Beep only/ Disable

11. Services

Future topic

SIP Phone Registration

1. Creating SIP End User

Create an END USER

Name : sipuser

and Add user to Group →

- Standard CCM End Users
- Standard CTI Enabled

→ Add Selected → Save

2. Registration

Device → Phone → Add New →

Phone Type* : Third-party SIP Device (Basic) → Next

MAC Address* : AAAABBBBCCCC <DUMMY_MAC_ADDRESS>

Description : SIP_PHONE1

Device Pool* : Default

Owner User ID* : sipuser

Phone Button Template : Third-party SIP Device (Basic)

Device Security Profile* : Third-party SIP Device Basic - Standard SIP Non-Secure Profile

SIP Profile* : Standard SIP Profile

Digest User : sipuser <SELECT_END_USER>

→ Save

Line [1] - Add a new DN → Directory Number : 1111 → Save

3. Login to SIP Phone (3CX Soft Phone)

Settings → Accounts → New

Account Name : <OPTIONAL>

Caller ID : <OPTIONAL>

Extension : 1111

ID : sipuser <END_USER_ID>

Password : 123

Specify IP Address of SIP Server

I am in the office- local IP : 142.100.64.11 <IP_OF_CUCM>

Associate end user to sip phone

DNA (Dialed Number Analyzer):

Cisco Call Manager Dialed Number Analyzer enables you to enter in calling and called numbers and receive a report that describes the following:

- How CallManager intends to route the call
- How CallManager changes the calling party number
- How CallManager changes the called party number

Service Activation

Cisco Unified Crevice Ability → Tools → Service Activation (PUBLISHER) → Cisco Dialed Number Analyzer → Save

Access DNA Page

Cisco Unified Crevice Ability → Tools → Dialed Number Analyzer

or https://<CUCM IP ADDRESS>/dna

Perform Analysis

Analysis → Phones → Select Phone → Enter Dialed Digit → Analyze

Analysis → Trunks → Select Trunk → Calling Party Number, Called Party Number → Analyze

CUCM Reporting

Cisco Unified Reporting tool pulls informations from a range of sources related to troubleshooting and maintaining & system analysis of CUCM.

Cisco Unified Reporting → Go or

https://<IP ADDRESS OF CUCM>/cucreports

CAR (Call Detailed Record Analysis & Reporting)

- Every call that CUCM processes can be logged. It contain information about Call, Voice Clarity, etc. These logs are called CDR (Call Detailed Record) & 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.

To activate CAR,

Cisco Unified Crevice Ability → Tools → Service Activation (PUBLISHER) → Cisco CAR Web Service → Save

Then go to https://<IP ADDRESS OF CUCM>/car

If external billing server is to be used, activate Cisco SOAP CDRonDemand Service must be enabled. (SOAP – Simple Object Access Protocol)

CDR (Call Detailed Record)

It contain Call Starting time, ending time, Call Duration, etc.

After activating CAR Service, you should configure CDR to load data.

CM Administration Page → System → Service Parameter → Select Server →

CDR Enabled Flag*: True: It determines whether CDRs will be generated. This must be set on all servers. Default is false

CDR Log Calls with Zero Duration Flag: True: CUCM records failed calls or calls which duration less than 1 second.

Display FAC in CDR: Controls whether FAC used to make a call will be included in CDR.

CMR (Call Management Record):

Packet sent, Received, Packet loss, Jitter

CM Administration Page → System → Service Parameter → Select Server →Call Diagnostics Enabled: Enabled Regardless of CDR Enabled Flag

Exporting CDR & CMR

Generating CDR Reports

The following reports are available

- Bills: Individual, Department
- Top N: By Charge, By duration, By Number of Calls
- Manager Call Usage, Assistant call usage
- IP Phone Services: Number of users subscribed to the services, and utilization percentage of each

Cisco Unified RTMT (Real Time Monitoring Tool)

RTMT allows Administrator to collect, view, interpret, and monitor various counters, trace files, and log files generated by CUCM.

It is a software installed on Administrators work station. It is available to download from CUCM plugins menu itself.

Cisco Unified CM Administration → Applications → Plugins → Find → Cisco Unified Real-Time Monitoring Tool – Windows → Download

Capabilities of RTMT

- Monitor system health
- Generate e-mail alerts for objects that fall below or exceed defined threshold value
- Collect, view different trace files
- View Syslog messages
- Configure & Monitor Performance counters
- Gateway Activity (H.323, MGCP, FXO/FXS, T1/E1, SIP/IC Trunks) informations

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

Click IP Phone IP Address to get web page

User can do this from Security settings of the Phone

Dependency Record

Step 1 Choose System > Enterprise Parameters.

Step 2 Scroll to the CCMAdmin Parameters area of the window.

Step 3 From the Enable Dependency Records drop-down list box, choose False.

A dialog box displays with a message about dependency records. Read the information carefully before clicking OK.

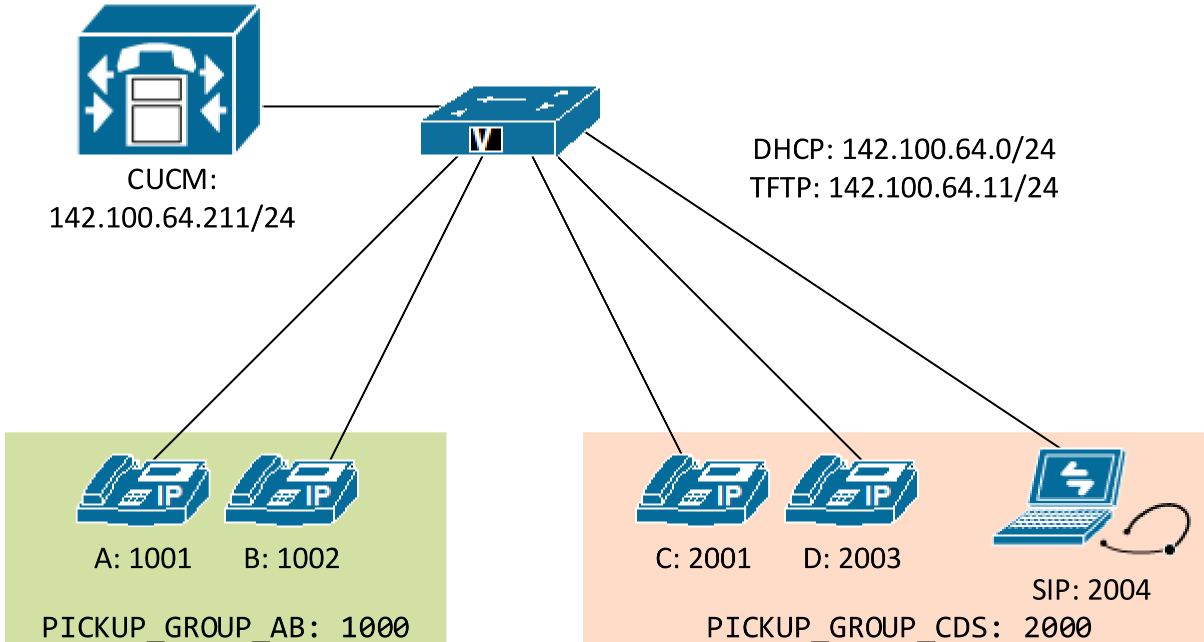
Step 4 Click OK.

The field displays True.

Step 5 Click Update.

Step 6 Close the browser that you are using; then, reopen the browser. This makes the parameter take affect for the entire system.

CCNA VOICE REAL LAB



1. Configure DHCP Server & POOL (142.100.64.X/24)
2. Register Phone A & B Manually and assign numbers given in the scenario
3. Register Phone C & D Automatically and specify the Auto registration range
4. Configure a SIP Phone by adding another end user (sipuser)
5. Configure shared line (5001) in A, B, C and D
6. Enable Barge in B only, assign privacy button also
7. Configure SPEED DIAL BUSY LABP FIELD to 2004 SIP phone by adding Phone button template.
8. Configure 2 device pool (US, INDIA) and put A, B in US-DP and C, D in INDIA-DP & set codecs
9. Configure Call Forward No Answer (20 Sec) to SIP Phone 2004
10. Configure a Park Slot 111X and Directed Call Park at 222X with * Prefix
11. Assign A, B in to a PICKUP_GROUP_AB with group number 1000
12. Assign C, D and SIP Phone in to a PICKUP_GROUP_CDS with group number 2000
13. Configure all kind of PICKUP options
14. Add a read only Administrator (readadmin)
15. Enable call back feature in all Phones
16. Configure 2 end user (user1, user2). Associate A, B to User1 & C, D to User2