CCNA Voice Prep: Cisco IP Telephony Essentials

BRKCRT-2001
Session Goals

1. Understanding Cisco CCNA voice and CCVP certification

2. To grasp the big picture behind VoIP

3. To deliver key concepts and configurations related to the current CCNA voice certification exam

Exam Tip!

4. To demonstrate on live equipment the methods used to configure a working VoIP network
Disclaimer

- This session will strictly adhere to Cisco’s rules of confidentiality

  We may not be able to address your specific question

  If you have taken the exam please refrain from asking questions from the exam

  We will be available after the session to direct you to resources to assist with specific questions or to provide clarification
Cisco Voice Certification Journey

- CCENT
- CCNA
- CCNA Voice
- CCVP
- CCIE Voice

1. CVOICE
2. CIPT1
3. CIPT2
4. QoS
5. TUC

ICND1
ICND2
IIUC

CCIE Written/Lab
Outline

- Overview of Cisco Unified Communication Solutions
- Making the Technical Transition to VoIP
- Call Flows in CUCM and CUCME
- Endpoint and End User Administration
- Telephony and Mobility Features <no time!>
- Cisco Unity Connection and Unified Presence <no time!>
Overview of Cisco Unified Communication Solutions
Challenges with Existing Voice Networks

- Teams that are not in the same physical location
- Telecommuting and mobile staff
- The need to lower costs and increase productivity
- The cost of supporting separate infrastructures for voice, video, and data
- The lack of flexibility with traditional solutions
Benefits of Cisco Unified Communications

- Enhances the way every department within your organization does business
- Be everywhere at once
- Makes everyone more efficient and productive
- Reach anyone, anywhere
- Reduces costs
- More new applications, faster
Overview of Cisco Unified Communication Solutions

- Cisco Unified Communications Manager Express
- Cisco Unified Communications Manager
- Cisco Unity Express
- Cisco Unity Connection
- Cisco Unified Presence
Call Processing

Network Infrastructure

Cisco Unified Communications Applications

Call Processing

Solution Components

Wireless Routers Switches Unified Communications Applications Cisco Unity Express for Voicemail

Add network infrastructure as required

Ability to add services: collaboration, messaging, customer contact, etc.

Mobility Services Presence Services Video Services

Call Processing Services

Presence Services Mobility Services Messaging Services Video Services

Call Processing Services

Mobility Services Presence Services Video Services

Call Processing Services

30,000 + phones

Cisco Unified Communications Manager Express

Cisco Integrated Services Routers

Cisco 7800 Series Media Convergence Servers

Cisco 7800 Series Media Convergence Servers

Cisco Unified Communications Manager Express

Cisco Unified Communications Manager Business Edition

Cisco Unified Communications Manager

Cisco Unified Communications System

Cisco 7800 Series Media Convergence Servers

Cisco Smart Business Communications System

Cisco Unified Communications Manager Express

Cisco Unified Communications Manager Business Edition
Call Processing—Cisco Smart Business Communications System

- Is composed of:
  - Cisco Unified Communications 500 for Small Business platform, which provides call processing
  - IP phones
  - Wireless access point (optional)
  - Cisco Unity Express, which provides voice mail and auto-attendant services

- Includes switching, basic VPN, and firewall capabilities

- Provides a simpler platform for easier training

- Plug-and-play

- Includes many integrated ports such as:
  - IP phone station, analog trunks, digital trunks, SIP trunks, analog stations, MOH, and expansion
Call Processing—Cisco Unified Communications Manager Express

- IP telephony solution for 240 users or less
- Part of Cisco integrated services routers
- Ideal for small business, enterprise branch office, or service provider-managed service
- Supports H.323 or SIP
- Voice mail and auto attendant with integrated Cisco Unity Express
- PBX or key switch configuration
Call Processing—CUCM Business Edition

- Key Cisco Unified Communications applications on a single server:
  - Cisco Unified Communications Manager
  - Cisco Unified Mobility
  - Cisco Unity Connection
- Effective option for medium-sized organizations with up to 500 users
- Flexible deployment options for growth
  - Single or multisite centralized configurations
- Seamless integration with entire Cisco Unified Communications system suite of products
Call Control—Cisco Unified Communications Manager

- 4.X—Windows OS
- 5.X and later—Linux Red Hat Appliance OS
- Supports up to 30,000 seats per cluster
- Signaling and device control
- Dial plan administration using a GUI
- Directory services that can be standalone or integrated with an existing directory (such as AD)
- A standardized interface to external applications for expanded functionality
VoiceMail—Cisco Unity Express

- For small deployments
  - IMAP compliant e-mail integration
  - **Browse** voice mailbox using Cisco IP Phone display
- IVR capabilities for efficient call routing and an improved customer experience
- Support for fax messages
- Easy to configure, deploy, and manage with CUCM and CUCME
- Support for many languages
VoiceMail—Cisco Unity Connection

- Integrated messaging allows receipt of voice mail using an e-mail client
- Powerful voice user interface:
  - Speech navigation for voice mail browsing
  - Speech-enabled directory dialing
  - Text-to-speech technology
- Visual display of VM on IP phone

- Personal call transfer rules:
  - Based on time of day, caller ID, and calendar
  - Forward calls to single number or series of numbers
  - Select which calls to accept in real time
- Same appliance OS as CUCM
- Scalable to 250 ports/20,000 users per server (Cisco UC 8.0)
Other Applications

- Cisco Unified Contact Center Express
  - Advanced small- to medium-sized call center functions (queuing)
- Cisco Unified Contact Center
  - Advanced medium- to large-sized call center functions
- Cisco Unified Meeting Place
  - Conferencing bridge
- Cisco Emergency Responder
  - Enhanced 911 services
- Cisco Unified Presence
  - Provides the availability status and communications capabilities of a user
- Cisco Unified Mobility
  - Gives users the ability to redirect incoming IP calls to other client devices
## Endpoints

<table>
<thead>
<tr>
<th>Commercial/Retail</th>
<th>Color Touch</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Mobility</th>
<th>Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phones: 7921G/IPC</td>
<td>Cisco Unified IP Phones: 7985G</td>
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<tr>
<td>Software: Cisco Unified Video Advantage</td>
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</table>

<table>
<thead>
<tr>
<th>Business Class</th>
<th>Conference</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phones: 7940G/7960G/7941G/7961G</td>
<td>Cisco Unified IP Phones: 7936G/7937G</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Advanced Media</th>
<th>Impress Your Friends!</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco Unified IP Phones: 7942G/7945G/7962G/7965G/7975G</td>
<td>Cisco Unified IP Phones: 8961/9951/9971</td>
</tr>
</tbody>
</table>
Making the Technical Transition to VoIP
Traditional Business Phone System

- CO Switch
- PBX
- Digital Handsets
- Local Loop
- Tie Line
- Analog or Digital Handsets
- CO Switch
- Key System
- Local Loop
- Customer Telephone
Digital Signal Processors

PSTN → DSPs → PSTN

Analog or Digital

IP Packets

Speech

IP Packets

IP
Digital Signal Processors (Cont.)

- The DSP chip performs the sampling, quantization, encoding, and optional compression step of digitization.
- It is used in both directions to convert from a traditional analog or digital voice signal to VoIP; or from VoIP to a traditional analog or digital voice signal.
- The number of simultaneous calls a chip can handle depends on the type of DSP and the codec being used.
Real-Time Transport Protocol

- Delivery services for real-time data such as voice and video
- Randomly picks even ports from UDP port range 16384–32767
- Adds the following services to UDP:
  - Payload type identification
  - Sequence numbering
  - Time stamping
RTP Control Protocol

- Can be used to monitor the quality of the VoIP stream
- Provides feedback on current network conditions
- Allows hosts that are involved in an RTP session to exchange information about monitoring and controlling the session:
  - Packet count
  - Packet delay
  - Octet count
  - Packet loss
  - Jitter (variation in delay)
- Uses a separate flow from RTP
- Paired with its RTP stream and uses the same port as the RTP stream plus one (odd-numbered port)

Double-tap to watch RTCP in action!
Packetization

- Packetization of voice is performed by DSP resources
- The DSP packages voice samples or compressed voice into IP packets
- The voice data is carried in the payload of RTP segments
- RTP is encapsulated in a UDP segment, which is encapsulated in an IP packet
Packetization—G.711 Example

G.711 20 ms of samples (160 bytes)
Packetization—G.729 Example

DSP Compression

RTP Header

20 Bytes of Voice Payload

G.729 20 ms of voice contained in packet
# Codecs—Bandwidth Implications

*G.711, G.729, and iLBC are the most Common Codecs.*

<table>
<thead>
<tr>
<th>Codec</th>
<th>G.711</th>
<th>G.726 r32</th>
<th>G.726 r24</th>
<th>G.726 r16</th>
<th>G.728</th>
<th>iLBC</th>
<th>G.729</th>
<th>G.723 r63</th>
<th>G.723 r53</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bandwidth not including overhead</td>
<td>64 kb/s</td>
<td>32 kb/s</td>
<td>24 kb/s</td>
<td>16 kb/s</td>
<td>16 kb/s</td>
<td>13.3 kb/s</td>
<td>8 kb/s</td>
<td>6.3 kb/s</td>
<td>5.3 kb/s</td>
</tr>
</tbody>
</table>
Internet Low Bitrate Codec

- Was designed for packetized communications
- Is royalty free
- Has better quality than G.729
- Has similar complexity as G.279
- Supports two fixed bit-rate frame lengths:
  - A bit rate of 13.3 kb/s with an encoding frame length of 30 ms
  - A bit rate of 15.2 kb/s with an encoding frame length of 20 ms
- Is supported only on newer Cisco Unified IP Phones:
  - IP Phone 7975G
  - IP Phone 7965G
  - IP Phone 7962G
  - IP Phone 7945G
  - IP Phone 7942G
  - IP Phone 7921G
  - IP Phone 7911G
  - IP Phone 7906G
iLBC—Packet Loss Comparison

![Packet Loss Comparison Graph]

- iLBC
- G.729A
- G.723.1

MOS vs. Packet Loss (%)
VoIP Signaling Protocols

- Signaling generates and monitors the call control information between two endpoints to:
  - Establish the connection
  - Monitor the connection
  - Release the connection

- The signaling protocol must pass supervisory, informational, and address signaling

- Signaling protocols can be peer-to-peer or client/server-based
  - Peer-to-peer allows the endpoints to contain intelligence to place calls without assistance
  - Client/server puts the endpoint under the control of a centralized intelligence point
# VoIP Signaling Protocols Comparison

<table>
<thead>
<tr>
<th>Standards Body</th>
<th>Vendor Neutrality</th>
<th>Used on Gateways</th>
<th>Used on Cisco Unified IP Phones</th>
<th>Architecture</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>ITU</td>
<td>Yes</td>
<td>No</td>
<td>Peer-to-peer</td>
</tr>
<tr>
<td>MGCP</td>
<td>IETF</td>
<td>Yes</td>
<td>Yes, limited</td>
<td>Client/server</td>
</tr>
<tr>
<td>SIP</td>
<td>IETF</td>
<td>Yes</td>
<td>Yes, Cisco Unified IP Phones and third-party phones</td>
<td>Peer-to-peer</td>
</tr>
<tr>
<td>SCCP</td>
<td>None</td>
<td>Yes, limited</td>
<td>Yes, Cisco Unified IP Phones only</td>
<td>Client/server</td>
</tr>
</tbody>
</table>
Voice VLANs

- Separates voice and data traffic
- Prevents unnecessary IP address renumbering
- Simplifies QoS configurations
- Requires two VLANs: one for data traffic and one for voice traffic
- Requires only one Ethernet cable drop for the Cisco IP phone and the PC that is plugged into the phone
- Requires two IP subnets: one for the data VLAN and one for the voice VLAN
Voice VLANs (Cont.)

- An access port can handle two VLANs:
  - Access VLAN
  - Voice VLAN

![Diagram showing tagged and untagged VLANs]
Configuring Voice VLANs

The access VLAN is used for the PC that is plugged into the IP phone.

The voice VLAN is used for voice and signaling that originates and terminates on the Cisco IP phone.

Spanning-tree PortFast mode causes spanning tree to enable the port more quickly.
Cisco SCCP IP Phone Startup Process

1. Cisco IP phone obtains power from the switch
2. Cisco IP phone loads locally stored image
3. Switch provides VLAN information to Cisco IP phone using Cisco Discovery Protocol
4. Phone sends DHCP request; receives IP information and TFTP server address
5. Cisco IP phone gets configuration from TFTP server
6. Cisco IP phone registers with Cisco Unified Communications Manager server
   Unified CM sends softkey template to SCCP phone using SCCP messages
CME and CUCM Core Administration
Review: Which Call Processing Solution?

- **Cisco Unified Communications Manager Express (CME)**
  - Supports up to 450 users (with a really big router)
  - Single router solution
  - Mostly command line based (supports router-based “GUI”)
  
- **Cisco Configuration Professional (CCP)**

- **Cisco Unified Communications Manager (CUCM)**
  - Supports virtually limitless users (max of 500 in BE)
  - Multiple server redundancy
  - Graphical, easy to use interface
Ephone and Ephone-dn Concepts

- Ephones and ephone-dns are modular Cisco IOS software constructs
- An ephone represents the configuration and setting of the physical phone
- An ephone-dn is a numeric destination that can be associated with one or more ephones
- The maximum number of supported ephones is determined by the license and hardware platform.
- An ephone can have more than one ephone-dn associated with it
- The maximum number of extensions is the same as the maximum number of ephone-dns
Ephone-dn Features

- An ephone-dn has a primary directory number assigned to it and can have an optional secondary number.
- A dn-tag is a unique value that is assigned when the ephone-dn is created.
- An ephone-dn can be single line or dual line.
  - A single line can terminate one call at a time.
  - A dual line can terminate two simultaneous calls.
- When you initially configure an ephone-dn, the system creates one or more telephony system POTS dial peers.

Primary extension number on a single-line ephone-dn that can make or receive one call at a time.

Primary and secondary extensions configured on a single-line ephone-dn in which the primary is an internal extension number and the secondary is an E.164 number.

One phone extension on a dual-line ephone-dn for ephone-dns that need call waiting, consultative transfer, and conferencing.
Basic Ephone-dn Configuration

Assigns a primary extension number to an ephone-dn

CMERouter(config)#ephone-dn 7 dual-line
CMERouter(config-ephone-dn)#number 1001

- One Virtual Voice Port
- Two Lines or Channels
- 1001
- 1001
Ephone Features

- An ephone is a software configuration of a physical phone
- It is assigned a unique phone-tag
- The physical device can be an IP phone or an analog phone attached to an ATA
- The MAC address of the IP phone or ATA is used to tie the software configuration to the hardware
- Cisco Unified Communications Manager Express automatically detects all supported Cisco Unified IP Phones
- You can associate one or more ephone-dns with an ephone
Example: Basic Ephone Configuration

MAC 000F.2470.F8F8

CMERouter(config)#ephone
CMERouter(config-ephone)#dn 7 dual-line
CMERouter(config-ephone-dn)#number 1001
CMERouter(config-ephone-dn)#exit
CMERouter(config)#ephone 1
CMERouter(config-ephone)#mac-address 000F.2470.F8F8
CMERouter(config-ephone)#button 1:7

Example:

- One Virtual Port
- MAC 000F.2470.F8F8
- ephone-dn 7: 1001
- Button 1
Example: Configuration for Multiple Ephones

CMERouter(config)#ephone-dn 10 dual-line
CMERouter(config-ephone-dn)#number 1004
CMERouter(config)#ephone-dn 11 dual-line
CMERouter(config-ephone-dn)#number 1005
CMERouter(config)#ephone-dn 12 dual-line
CMERouter(config-ephone-dn)#number 1006
CMERouter(config)#ephone-dn 13 dual-line
CMERouter(config-ephone-dn)#number 1007
CMERouter(config)#ephone 1
CMERouter(config-ephone)#mac-address 000F.2470.F8F1
CMERouter(config-ephone)#button 1:10
CMERouter(config)#ephone 2
CMERouter(config-ephone)#mac-address 000F.2470.A302
CMERouter(config-ephone)#button 1:11
CMERouter(config)#ephone 3
CMERouter(config-ephone)#mac-address 000F.2470.66F6
CMERouter(config-ephone)#button 1:12
CMERouter(config)#ephone 4
CMERouter(config-ephone)#mac-address 000F.2470.7B54
CMERouter(config-ephone)#button 1:13
Call Legs

Phone 1234 Dials a PSTN Destination

Router1
FXS 1/0/1
10.10.10.1
VoIP

Router2
10.10.10.2
PSTN

Call Leg 1: In on Router1
Call Leg 2: Out on Router1
Call Leg 3: In on Router2
Call Leg 4: Out on Router2
Dial Peers

- Dial peers are an addressable call endpoint
- They establish logical connections, or call legs, to complete an end-to-end call
- You can use dial peers inbound, outbound, or both
- Dial peers define the properties of the call leg:
  - Codec
  - QoS markings
  - VAD
  - Fax rate
- Cisco voice-enabled routers typically use two types of dial peers:
  - POTS dial peers—connect to a traditional telephony network such as FXO, FXS, E&M, BRI, PRI T1/E1, and CAS T1/E1
  - VoIP dial peers—connect over an IP network using an IP address
POTS Dial Peers

Dial peer 20 will be used to match outbound when the router receives a call setup message for 1234.

CMERouter(config)#dial-peer voice 20 pots
CMERouter(config-dialpeer)#destination-pattern 1234
CMERouter(config-dialpeer)#port 1/0/1
VoIP Dial Peers

Dial peer 20 matches inbound

Dial peer 30 matches outbound

Phone 1234 dials 2010

CMERouter1

FXS 1/0/1

Lo0 - 10.10.10.1

CMERouter1(config)#dial-peer voice 20 pots
CMERouter1(config-dialpeer)#destination-pattern 1234
CMERouter1(config-dialpeer)#port 1/0/1
CMERouter1(config)#dial-peer voice 30 voip
CMERouter1(config-dialpeer)#destination-pattern 2...
CMERouter1(config-dialpeer)#session target ipv4:10.10.10.2
Destination Pattern Options

- Common destination pattern wildcards:
  - Plus (+)
    - Preceding digit occurs one or more times
  - Asterisk (*) and pound sign (#)
    - Not valid wildcards; are DTMF tones
  - Comma (,)
    - Inserts a one-second pause
  - Period (.)
    - Specifies any one wildcard digit
  - Square brackets ([ ])
    - Indicates a range of digits within the brackets
  - T
    - Indicates a variable-length pattern
Matching Outbound Dial Peers

Destination pattern is matched based on longest number match

Example 1: Dialed number 555-1234 will match dial peer 4
Example 2: Dialed number 555-1235 will match dial peer 3
Example 3: Dialed number 555-2000 will match dial peer 2
Example 4: Dialed number 551-1234 will match dial peer 1
The **preference** command defines the order of preference when multiple dial peers have an equally good destination pattern defined.

- Preference values can be set from 0 to 10.
- Lower values are more preferred.
- The default preference on dial peers is 0.
Demo: Setting up Phones in CME

From Zero to Hero

- Step 1: Core router configuration
- Step 2: Voice VLAN / DHCP Pool
- Step 3: Core CME Configuration
- Step 4: Ephones / Ephone-DN
- Step 5: Outside Calling Route Plan
# CUCM: Endpoint Configuration Methods

<table>
<thead>
<tr>
<th>Method for Adding IP Phones</th>
<th>Advantages</th>
<th>Disadvantages</th>
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</thead>
<tbody>
<tr>
<td>Autoregistration</td>
<td>▪ Devices automatically added</td>
<td>▪ Default Settings, random DN</td>
</tr>
<tr>
<td></td>
<td>▪ Default Settings, random DN</td>
<td>▪ Modifications needed</td>
</tr>
<tr>
<td>Unified CM BAT</td>
<td>▪ Bulk add</td>
<td>▪ MAC addresses required in BAT files</td>
</tr>
<tr>
<td>Unified CM Auto-Register Phone Tool</td>
<td>▪ Very scalable</td>
<td>▪ Cisco CRS required</td>
</tr>
<tr>
<td></td>
<td>▪ MAC addresses not required</td>
<td>▪ Complex configuration</td>
</tr>
<tr>
<td>Manual Configuration</td>
<td>▪ Simple</td>
<td>▪ MAC addresses required</td>
</tr>
<tr>
<td></td>
<td></td>
<td>▪ Time-consuming</td>
</tr>
</tbody>
</table>
Jeremy’s Favorite: Autoregistration

- Supported by all Cisco IP phones
- Existing endpoints are not affected
- Automatically adds Cisco IP phones not found in database (based on MAC addresses)
- Phones assigned the next available directory number of the configured range
- Cisco Unified Communications Manager BAT can be used to make bulk changes after autoregistration
CUCM Autoregistration Configuration

Enter Directory Number Range for Autoregistration

Enable Autoregistration
Manual IP Phone Configuration
Step 1: Adding the IP Phone

Select phone type, e.g. 7960
Select phone protocol (SIP or SCCP)
**Manual IP Phone Configuration**

**Step 2: Phone Configuration**

- **Required parameters:**
  - MAC Address
  - (Device Pool)
  - (Phone Button Template)
  - (Common Phone Profile)
  - (Location)
  - (Built-In Bridge)
  - (Privacy)
  - (Device Mobility Mode)
  - Device Security Profile

- () = parameters with default values
Manual IP Phone Configuration
Step 3: Directory Number Configuration

- Required parameters:
  - Directory Number
  - Presence Group
  - Auto Answer
  - Visual Message
  - Waiting Indicator
  - Policy
  - Ring Setting (Phone Idle)
  - Maximum Number of Calls
  - Busy Trigger

- () = parameters with default values
User Management Options

- One-by-one manual configuration using Cisco Unified Communications Manager Administration
- Bulk configuration using Cisco Unified Communications Manager BAT
- LDAP integration (for end users only):
  - LDAP synchronization
    For user provisioning
    Personal and organizational user data are managed in LDAP
  - LDAP authentication
    For user authentication
    Passwords managed in LDAP
# Cisco Unified CM End-User Data Location

<table>
<thead>
<tr>
<th></th>
<th>No LDAP Integration</th>
<th>LDAP Synchronization</th>
<th>LDAP Authentication</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Personal and...</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Organizational Settings:</td>
<td>Local</td>
<td>LDAP (replicated to local)</td>
<td>LDAP (replicated to local) or Local</td>
</tr>
<tr>
<td>User ID</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>First, Middle, and Last Name</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Manager User ID and Department</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone Number and Mail ID</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Password</strong></td>
<td>Local</td>
<td>Local</td>
<td>LDAP</td>
</tr>
<tr>
<td><strong>Cisco Unified CM Settings:</strong></td>
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<tr>
<td>PIN and Digest Credentials</td>
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<td></td>
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<tr>
<td>Groups and Roles</td>
<td>Local</td>
<td>Local</td>
<td>Local</td>
</tr>
<tr>
<td>Associated PCs</td>
<td></td>
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<tr>
<td>Controlled Devices</td>
<td></td>
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<td></td>
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<tr>
<td>Extension Mobility Profile and CAPF Presence Group and Mobility</td>
<td></td>
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</tr>
</tbody>
</table>
User Search Bases

- Two synchronization agreements are used
  - One synchronization agreement specifies User Search Base 1 and imports users jsmith, jdoe, and jbloggs
  - The second synchronization agreement specifies User Search Base 2 and imports users jjones, bfoo, and tbrown

No Synchronization Agreement for Service Accounts

CCM Dir Mgr Is not Imported
External Calling in CUCM (time permitting)
Endpoint Dialing

- **On-Net Dialing**: calls that originate and terminate on the same telephony network (e.g., internal IP phone to IP phone calls within the same cluster)

- **Off-Net Dialing**: calls that originate from a telephony network and terminate on a different telephony network (e.g., IP phone to PSTN calls)
Dial-Plan Configuration in Cisco Unified Communications Manager

Route Pattern:
- Matches dialed number for external calls
- Performs digit manipulation (optional)
- Points to a route list for routing

Route List:
- First level of path selection
- Performs digit manipulation
- Points to prioritized route group(s)

Route Group:
- Second level of path selection
- Points to the actual device(s)

Devices:
- Gateways (H.323, MGCP)
- Trunks (SIP, H.323)
## Route Pattern: Commonly Used Wildcards

<table>
<thead>
<tr>
<th>Wildcard</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>x</td>
<td>Single digit (0–9, *, #)</td>
</tr>
<tr>
<td>@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>!</td>
<td>One or more digits (0–9)</td>
</tr>
<tr>
<td>[x-y]</td>
<td>Generic range notation</td>
</tr>
<tr>
<td>[^x-y]</td>
<td>Exclusion range notation</td>
</tr>
<tr>
<td>.</td>
<td>Terminates access code</td>
</tr>
<tr>
<td>#</td>
<td>Terminates interdigit timeout</td>
</tr>
<tr>
<td>&lt;wildcard&gt;?</td>
<td>Matches zero or more occurrences of any digit that matches the previous wildcard</td>
</tr>
<tr>
<td>&lt;wildcard&gt;+</td>
<td>Matches one or more occurrences of any digit that matches the previous wildcard</td>
</tr>
</tbody>
</table>
# Route Pattern Examples

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>1234</td>
<td>Matches 1234</td>
</tr>
<tr>
<td>1*1x</td>
<td>Matches numbers from 1<em>10 to 1</em>19</td>
</tr>
<tr>
<td>12xx</td>
<td>Matches numbers from 1200 to 1299</td>
</tr>
<tr>
<td>13[25-8]6</td>
<td>Matches 1326, 1356, 1366, 1376, 1386</td>
</tr>
<tr>
<td>13[^3-9]6</td>
<td>Matches 1306, 1316, 1326, 13*6, 13#6</td>
</tr>
<tr>
<td>13!#</td>
<td>Matches any number that begins with 13, is followed by one or more digits, and ends with #; 135# and 13579# are example matches</td>
</tr>
</tbody>
</table>
Partitions and Calling Search Spaces

- A partition is a group of numbers with same reachability
  
  Any dialable patterns can be part of a partition (directory numbers, route patterns, translation patterns, voice-mail ports, Meet-Me conference numbers, etc.)

- Calling search space is a list of partitions and includes the partitions that are accessible by this CSS
  
  A device can call only those numbers located in the partitions that are part of its calling search space

  Assigned to any entity that can generate a call routing request, including phones, phone lines, gateways, and applications
Partition <None> and CSS <None>

- Before partitions and CSS are configured, all entities that can have a partition (i.e., called entities such as directory numbers, route patterns, etc.) reside in partition <None>, and all entities that can have a CSS (calling entities such as phones or trunks) are assigned with CSS <None>

- Entities that are in partition <None> are always accessible (regardless whether the calling entity has a CSS or not)

- Entities that have CSS <None> assigned can only access entities that are in partition <None>
Partition and CSS Scenario

- **Your requirements:**
  1. Lobby phone can call some internal numbers (no manager) and local PSTN
  2. Employees can call all internal numbers and local PSTN
  3. Manager can call all numbers

Route Patterns:
- Local: 9.[2-9]XXXXXXXXXX
- INTL: 9.011!
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