What You Make Possible

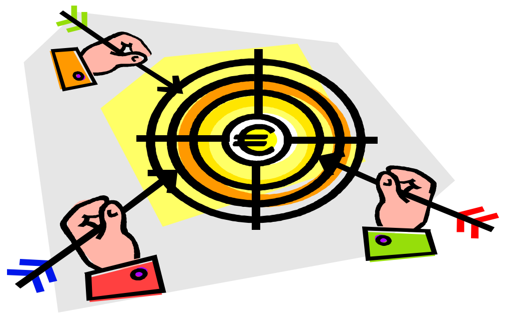
TOMORROW
starts here.

CCNA Voice: Real World Overview and Certification Preparation

BRKCRT-9103

TOMORROW starts here.

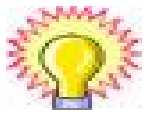
Session Goals



1. Understanding Cisco CCNA Voice and CCNP Voice certification

2. To grasp the big picture behind VoIP; takes away the fear

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 live!

Cisco Voice Certification Journey

CCENT	ICND1
CCNA	ICND2
CCNA Voice	ICOMM
CCNP Voice	1. CVOICE 2. CIPT1 3. CIPT2 4. TVOICE 5. CAPPS
CCIE Voice	CCIE Written/Lab

Outline

- Overview of Cisco Unified Communication Solutions
- Making the Technical Transition to VoIP
- CME and CUCM Core Administration
- External Dialing in CUCM and CUCME

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 organisation does business
- Makes everyone more efficient and productive (WebEX example)
- Reachability / Be everywhere at once
- Reduces costs

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	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.		
Cisco Unified Communications Applications	Cisco Unified Communications Manager Express	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
Call Processing	Up to 104 phones	Up to 450 phones	Up to 500 phones	30,000 + phones
Solution Components	Cisco Unified Communications 500 Series for Small Business	Cisco Integrated Services Routers	Cisco 7800 Series Media Convergence Servers	Cisco 7800 Series Media Convergence Servers
	Cisco Smart Business Communications System	Cisco Unified Communications Manager Express	Cisco Unified Communications Manager Business Edition	Cisco Unified Communications Manager

Cisco live!

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

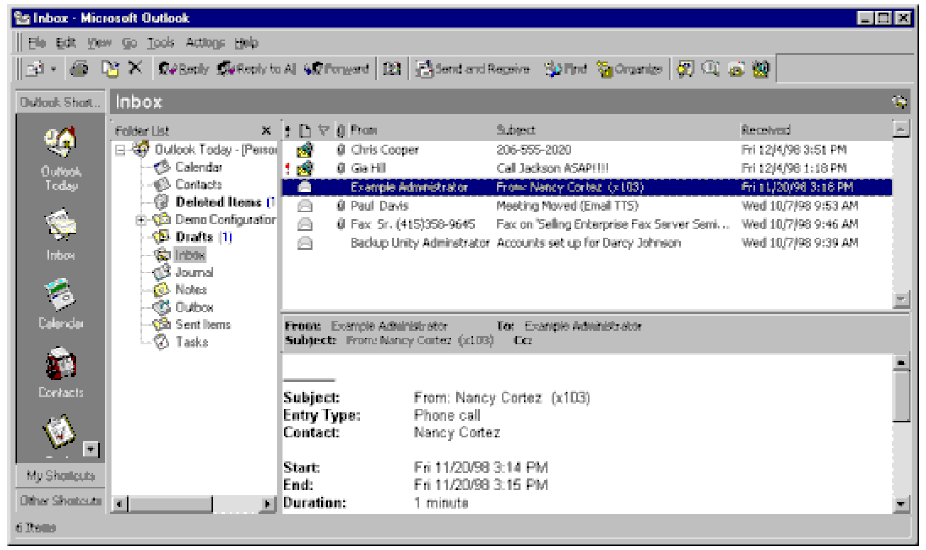
Cisco Unified Communications Manager Express

- Part of Cisco Routers (ISRs); up to 450 phones
- Ideal for small business, enterprise branch office, or service provider-managed service
- Voice mail and auto attendant with integrated Cisco Unity Express
- PBX or key switch configuration

Call Processing

CUCM Business Edition

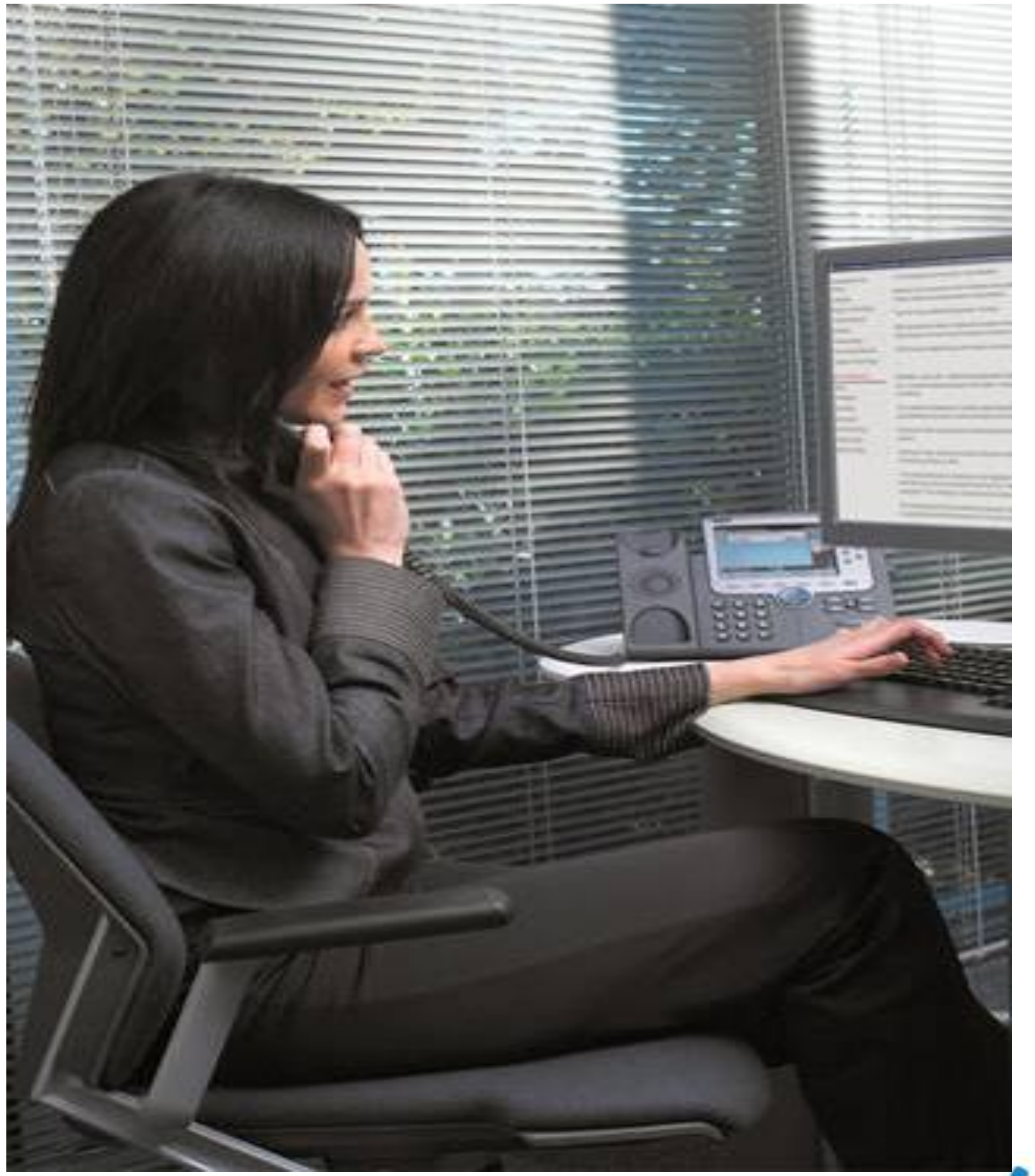
- Three key Cisco Unified Communications applications on a single server:
  - Cisco Unified Communications Manager
  - Cisco Unified Mobility
  - Cisco Unity Connection
- Up to 500 users
- Single or multisite centralised configurations



Call Control

Cisco Unified Communications Manager

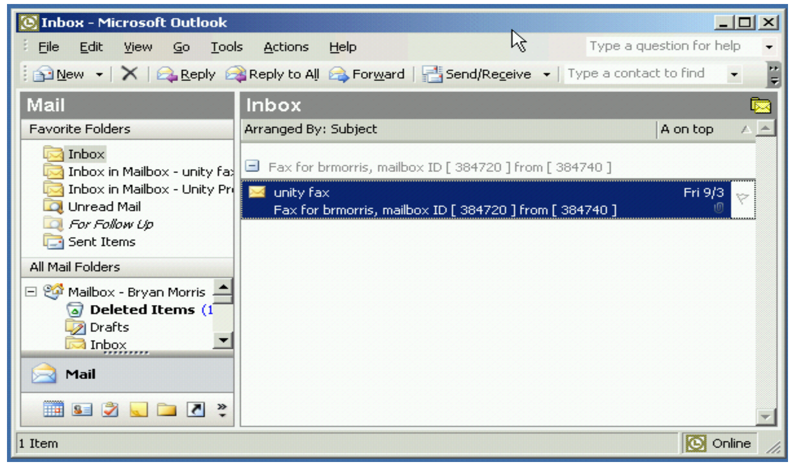
- Handles signalling and device control; up to 30,000 seats per cluster
- GUI-based administration
- 4.X – Windows-based
- 5.X and later - Appliance-based
- Directory services that can be standalone or integrated with an existing directory (i.e AD)
- A standardised interface to external applications 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
- Integrates easily with CUCM and CME



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

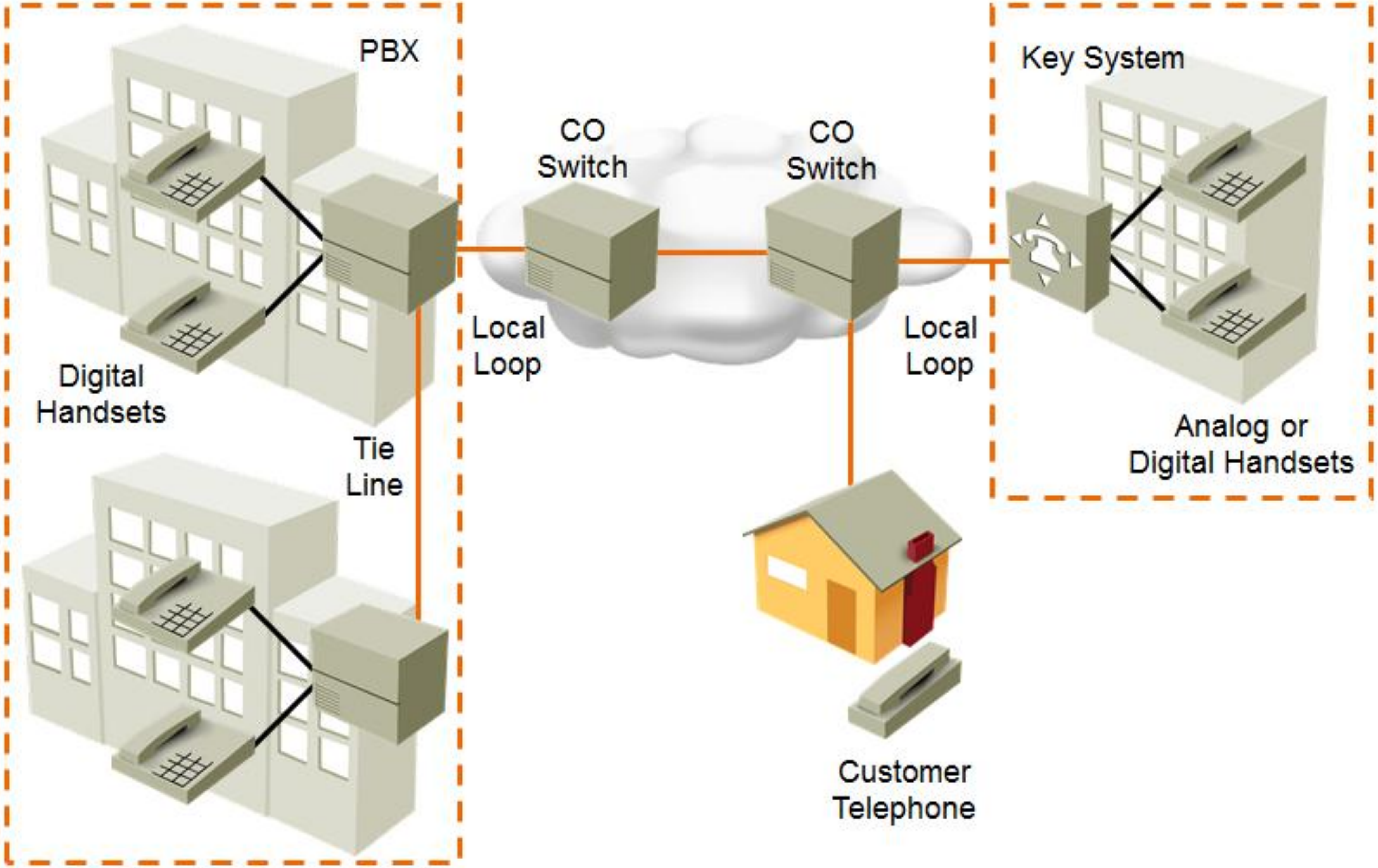
- Cisco Unified Contact Centre Express
  - Advanced small- to medium-sized call centre functions (queuing)
- Cisco Unified Contact Centre
  - Advanced medium- to large-sized call centre functions
- Cisco Unified Meeting Place
  - Conferencing bridge
- Cisco Emergency Responder
  - Enhanced 911 services
- 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

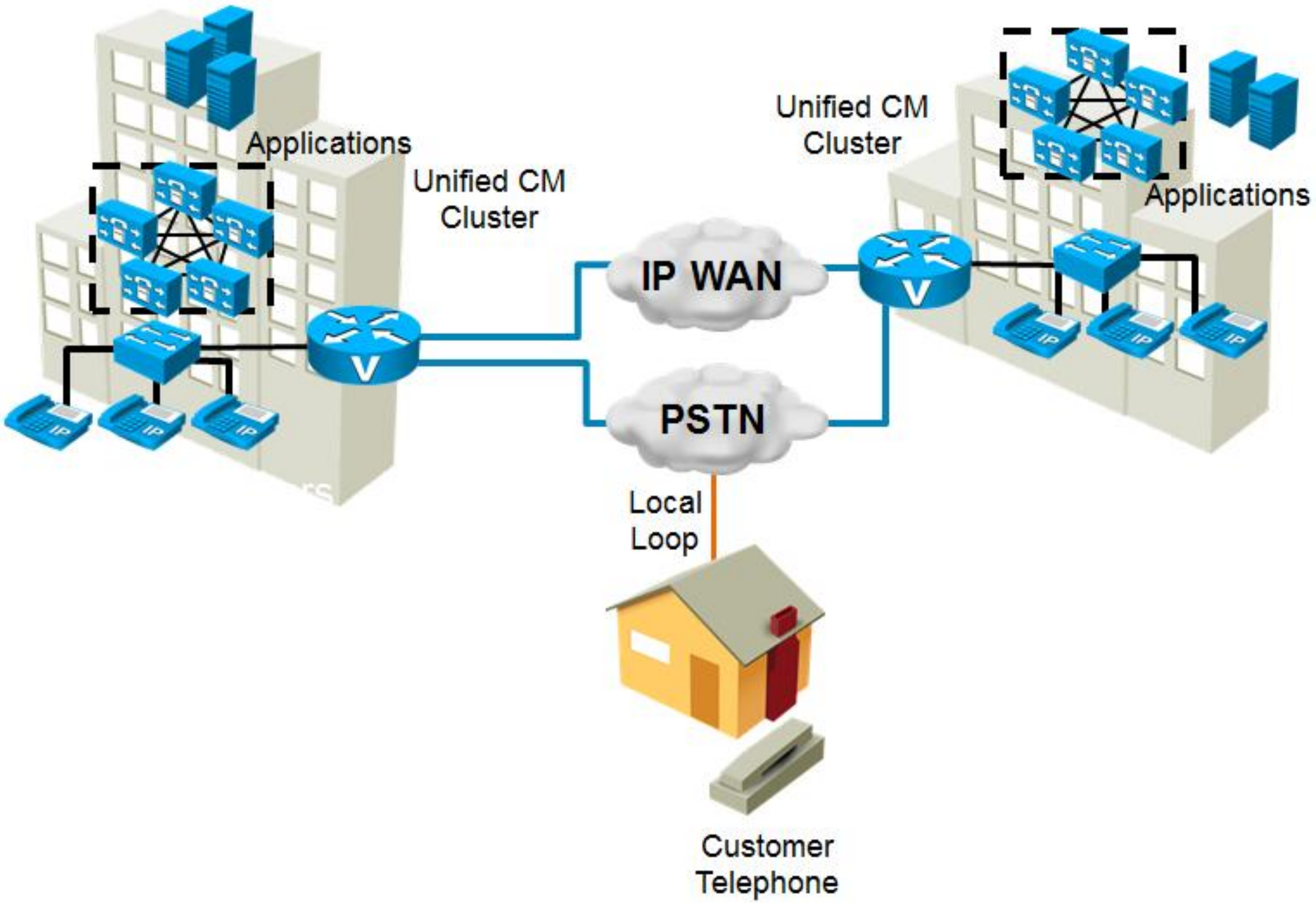
Commercial/Retail Cisco Unified IP Phones: 7931G/7921G 	Color Touch Cisco Unified IP Phones: 7970G/7971G-GE/7975G 
Mobility Cisco Unified IP Phones: 7921G/IPC 	Video Cisco Unified IP Phones: 7985G Software: Cisco Unified Video Advantage 
Business Class Cisco Unified IP Phones: 7940G/7960G/7941G/7961G 	Conference Cisco Unified IP Phones: 7936G/7937G 
Advanced Media Cisco Unified IP Phones: 7942G/7945G/7962G/7965G/7975G 	Impress Your Friends! Cisco Unified IP Phones: 8961/9951/9971 

Making the Technical Transition to VoIP

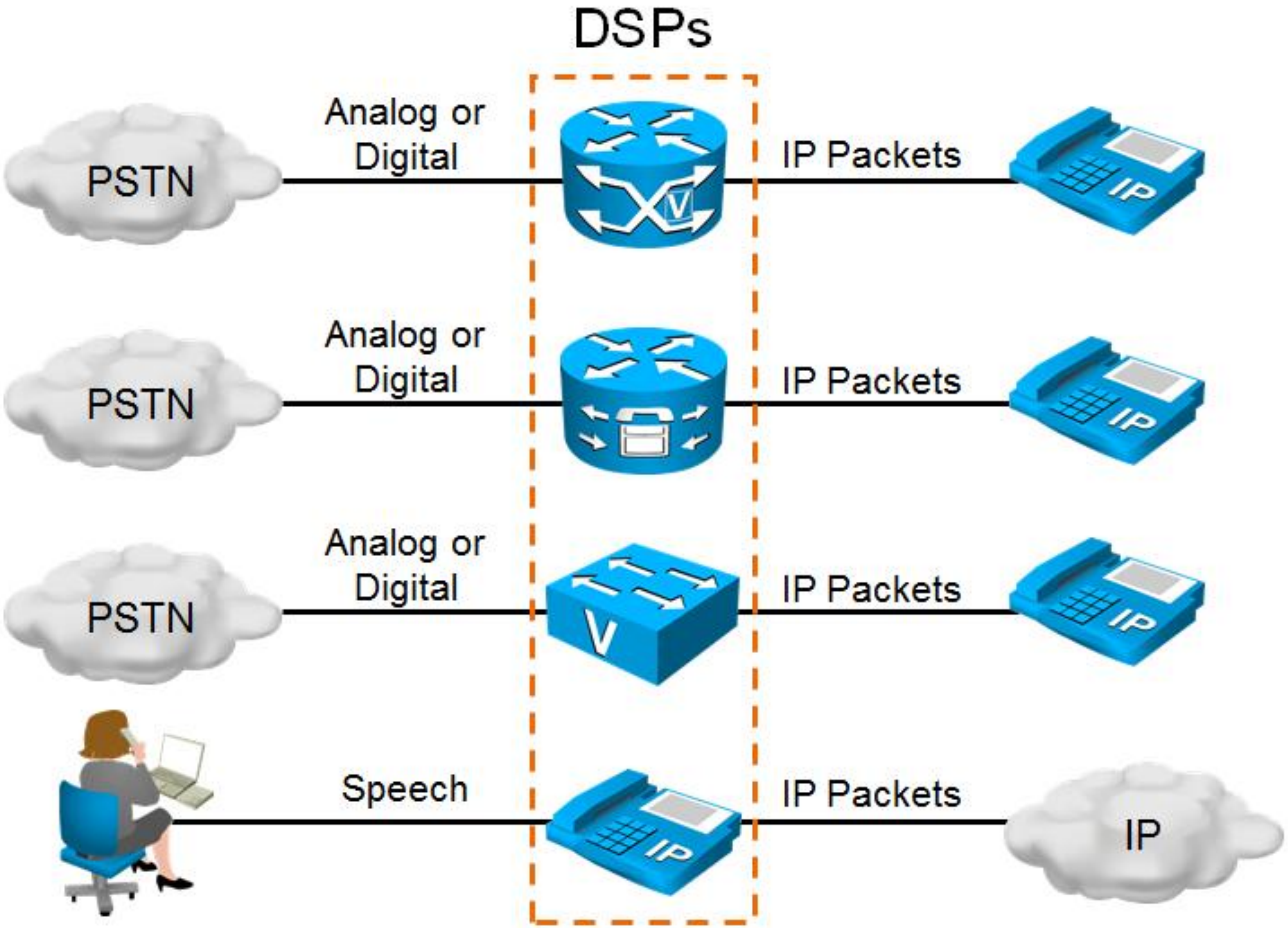
Traditional Business Phone System



Cisco VoIP Phone System



Digital Signal Processors



Digital Signal Processors (Cont.)

- The DSP chip performs the sampling, quantisation, encoding, and optional compression step of digitisation
- 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

Payload Type	Sequence Number	Time Stamp	Payload

- 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
- Exchanges the following information between hosts:
  - 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)

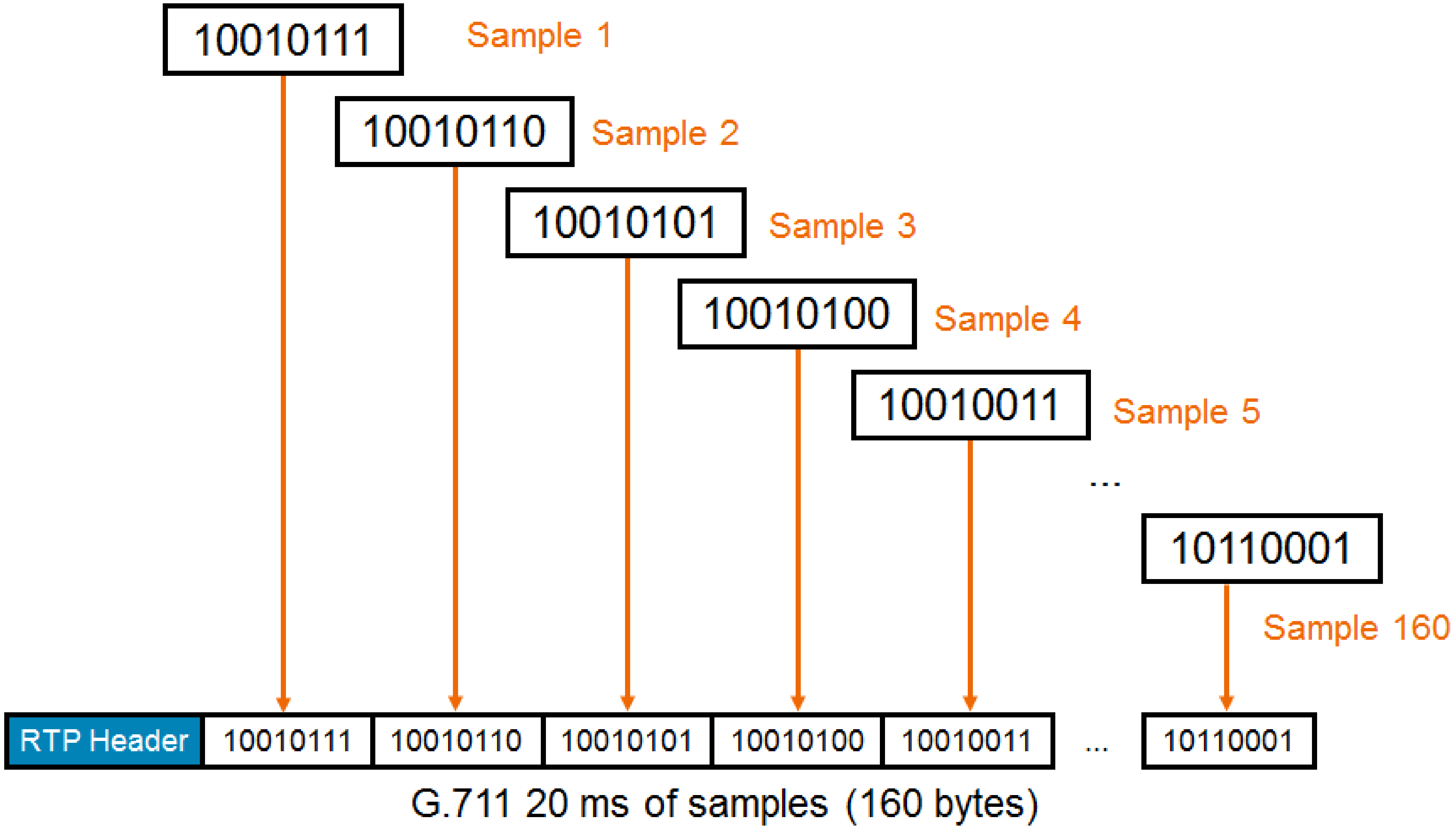
Double-tap to watch RTCP in action!

Packetisation

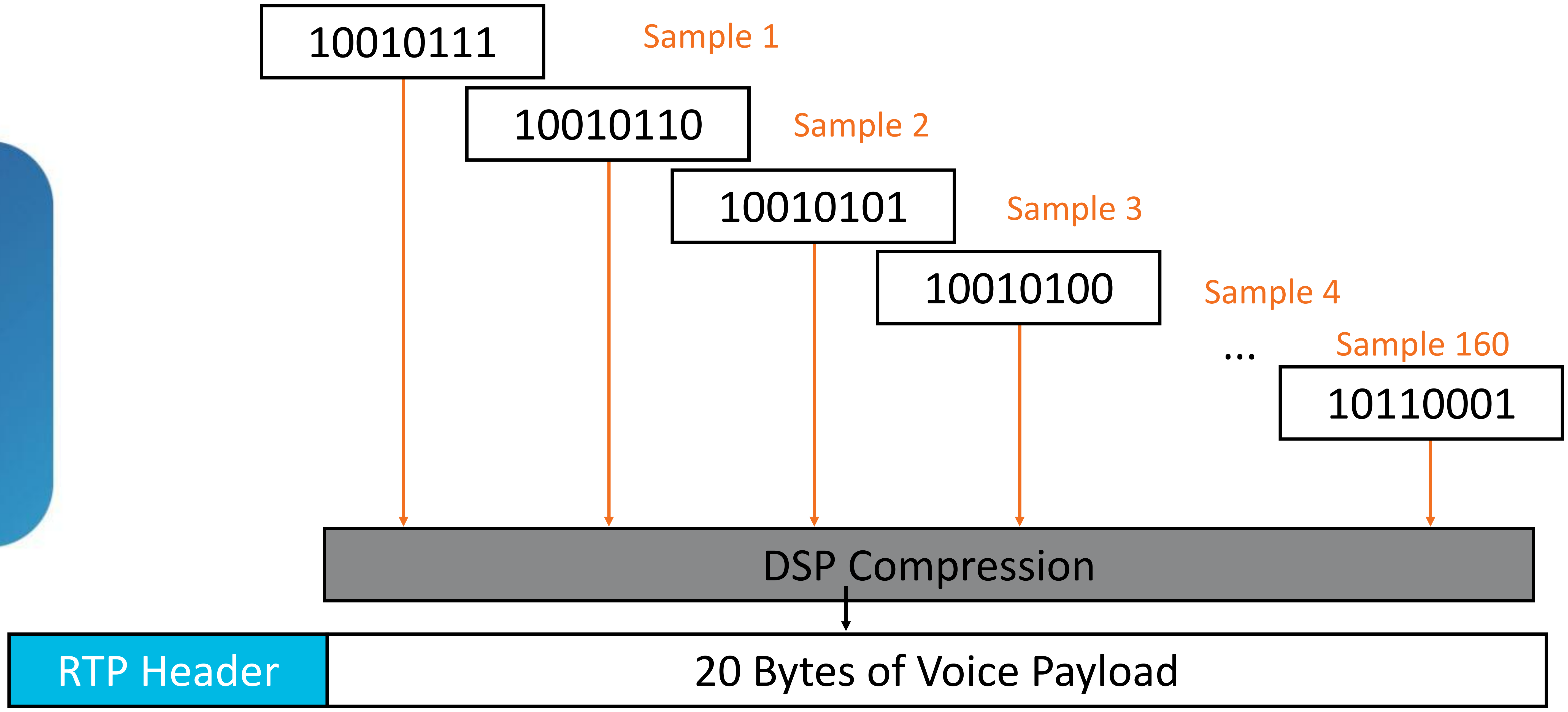
Layer 2 Header	IP Header	UDP Header	RTP Header	Voice Payload

- The DSP resources 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

Packetisation—G.711 Example



Packetisation—G.729 Example



G.729 20 ms of voice contained in packet

Codecs—Bandwidth Implications

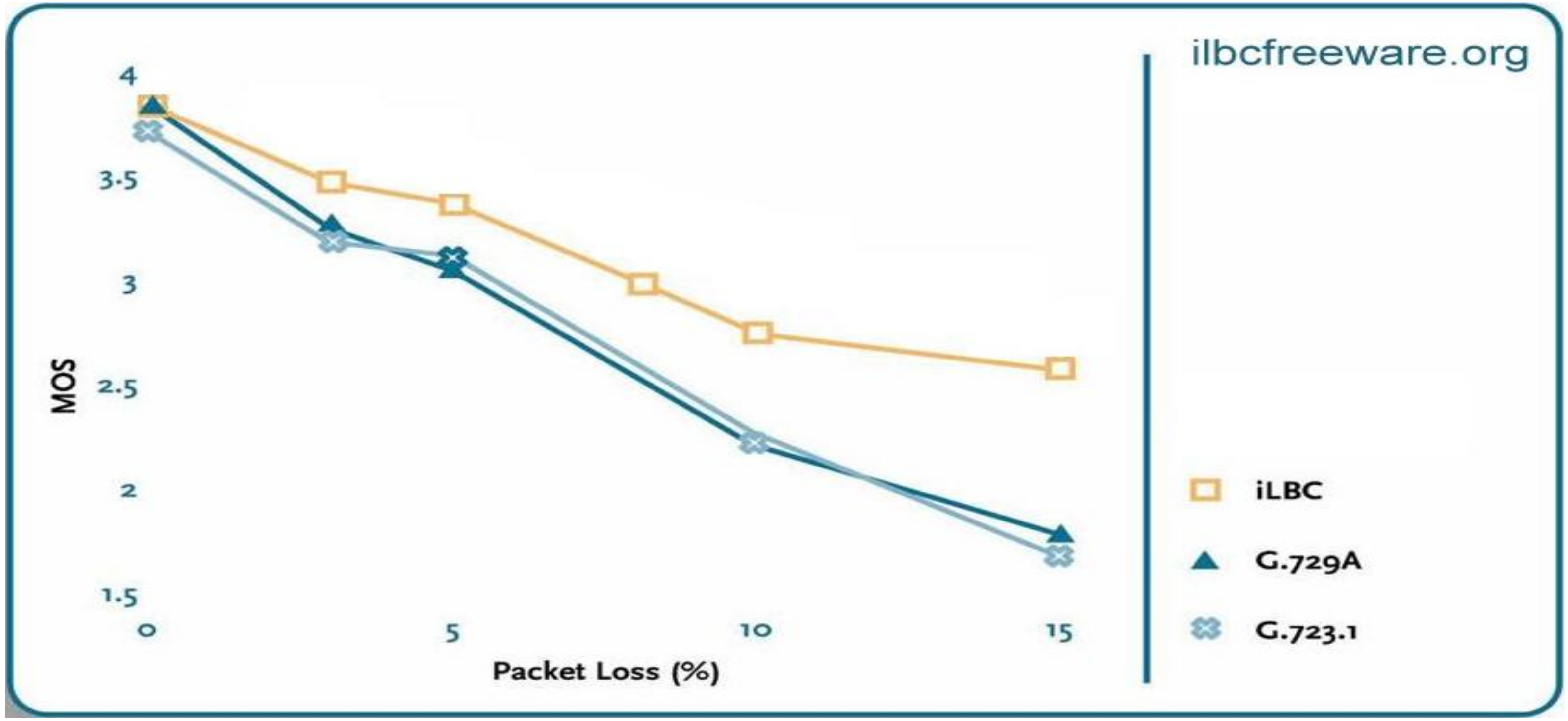
*G.711, G.729, and iLBC are the most Common Codecs.

Codec	G.711 *	G.726 r32	G.726 r24	G.726 r16	G.728	iLBC *	G.729 *	G.723 r63	G.723 r53
Bandwidth not including overhead	64 kb/s	32 kb/s	24 kb/s	16 kb/s	16 kb/s	13.3 kb/s	8 kb/s	6.3 kb/s	5.3 kb/s

Internet Low Bitrate Codec

- Was designed for packetised 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



VoIP Signalling Protocols

- Signalling generates and monitors the call control information between two endpoints to:
  - Establish the connection
  - Monitor the connection
  - Release the connection
- The signalling protocol must pass supervisory, informational, and address signalling
- Signalling 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 centralised intelligence point

VoIP Signalling Protocols Comparison

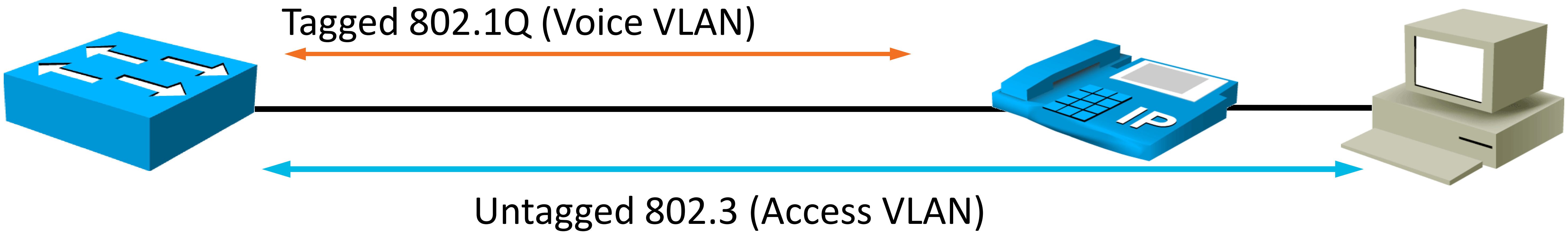
	Standards Body	Vendor Neutrality	Used on Gateways	Used on Cisco Unified IP Phones	Architecture
H.323	ITU	Very Good	Yes	No	Peer-to-peer
MGCP	IETF	Good	Yes	Yes, limited	Client/server
SIP	IETF	Basic	Yes	Yes, Cisco Unified IP Phones and third-party phones	Peer-to-peer
SCCP	None	Proprietary	Yes, limited	Yes, Cisco Unified IP Phones only	Client/server

Voice VLANs

- Separates voice and data traffic
- Minimises cabling: one drop for phone/PC
- Prevents unnecessary IP address renumbering
- Simplifies QoS configurations
- Requires two VLANs / subnets: one for data traffic and one for voice traffic

Voice VLANs (Cont.)

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



Configuring Voice VLANs

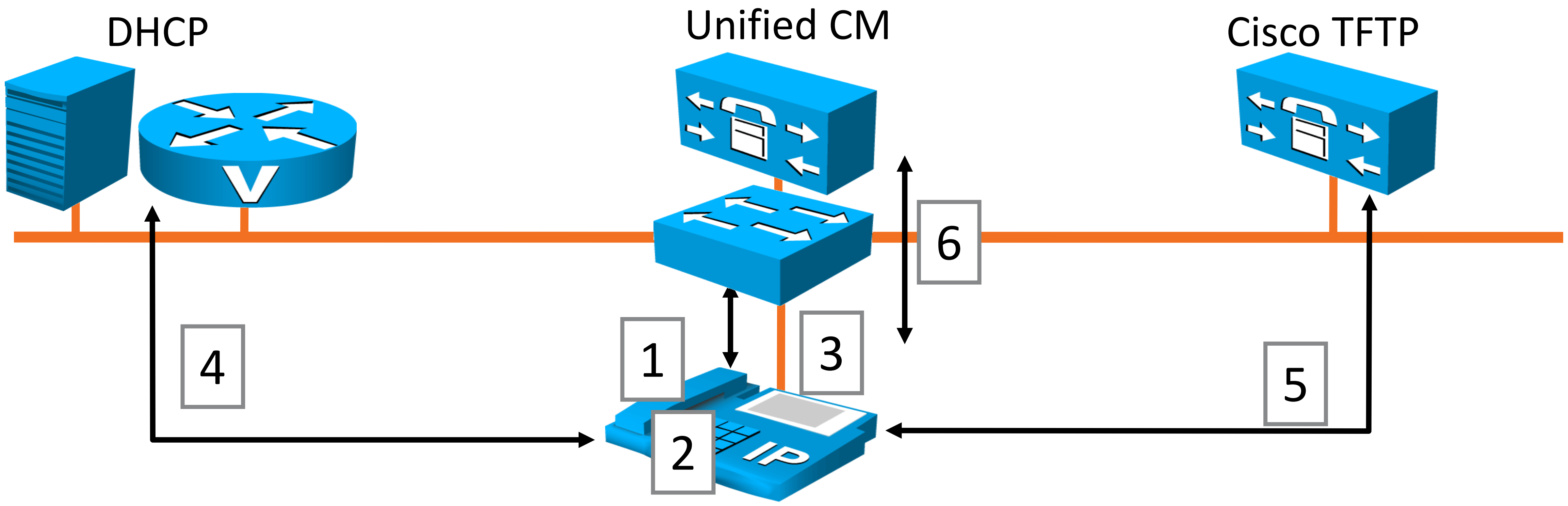
```
Console(config)#interface FastEthernet0/1
Console(config-if)#switchport access vlan 12
Console(config-if)#switchport mode access
Console(config-if)#switchport voice vlan 112
Console(config-if)#spanning-tree portfast
```

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

– The voice VLAN is used for voice and signalling that originates and terminates on the Cisco IP phone

– Spanning-tree PortFast allows STP to enable the port quickly

Cisco 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

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

CME: Core Configuration

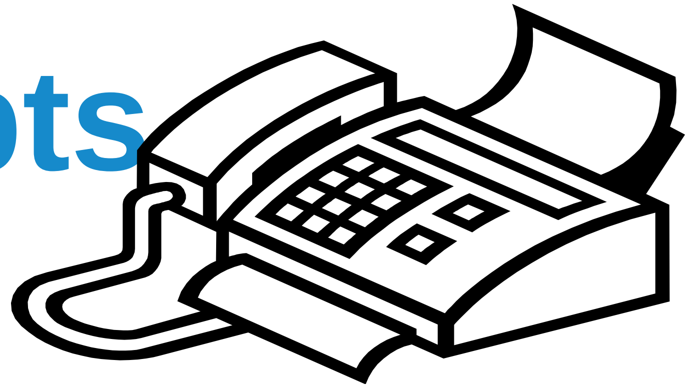
- The “Big 3” commands to enable CME Services

```
CMERouter(config)#telephony-service
CMERouter(config-telephony)#ip source-address X.X.X.X
CMERouter(config-telephony)#max-ephone XX
CMERouter(config-telephony)#max-dn XX
```

- Defines the IP address CME should use to receive calls
- Defines the maximum IP phones supported (should match license)
- Depending on IOS Version, may also need this:

```
CMERouter(config-telephony)#create cnf-files
```

Ephone and Ephone-dn Concepts



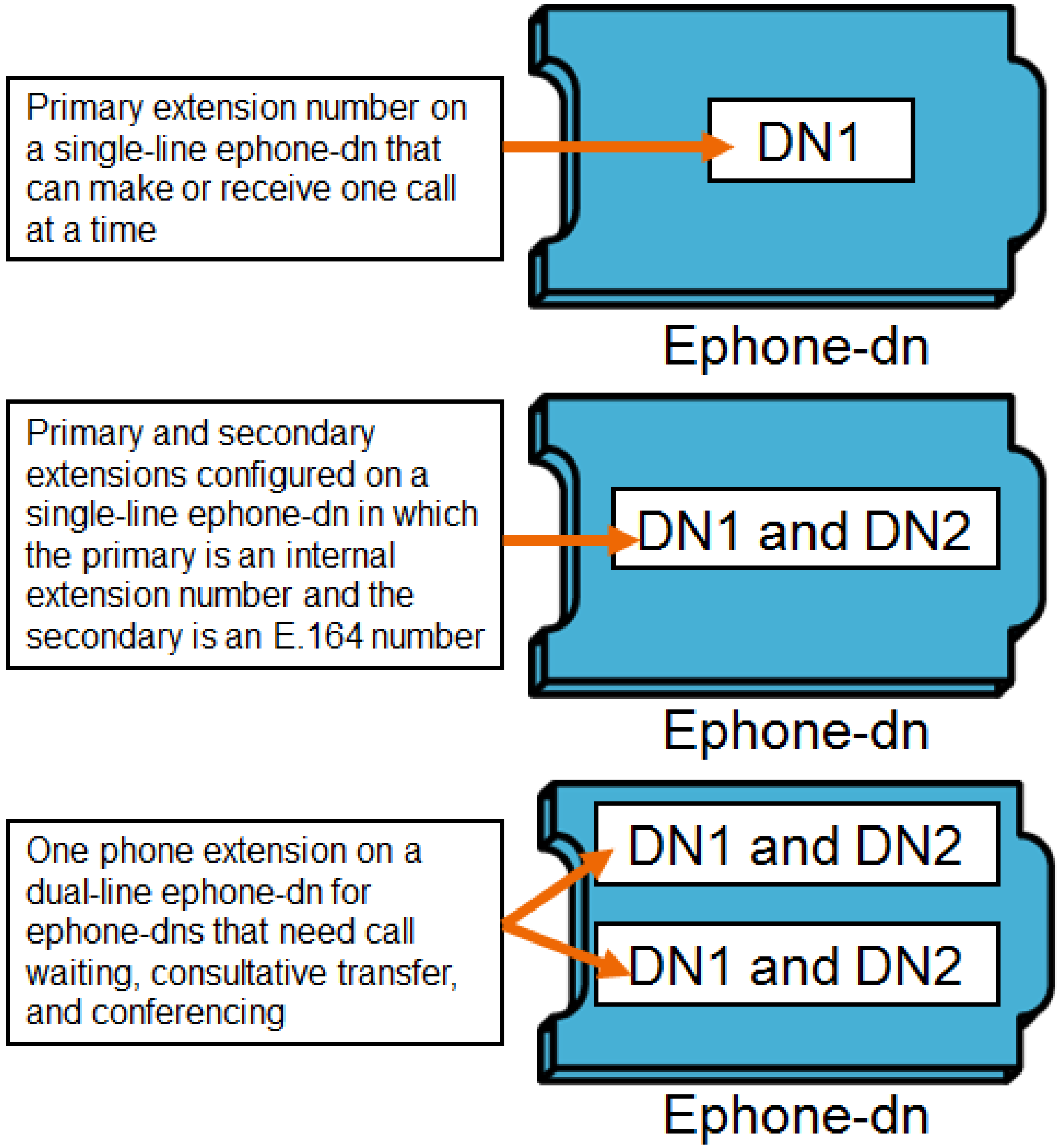
- Ephones and ephone-dns are modular Cisco IOS software constructs
  - An ephone represents the configuration and setting of the physical phone
  - An ephone-dn is a numeric destination that can be associated with one or more ephones
- The maximum number of supported ephones is determined by the license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

Ephone-dn Features

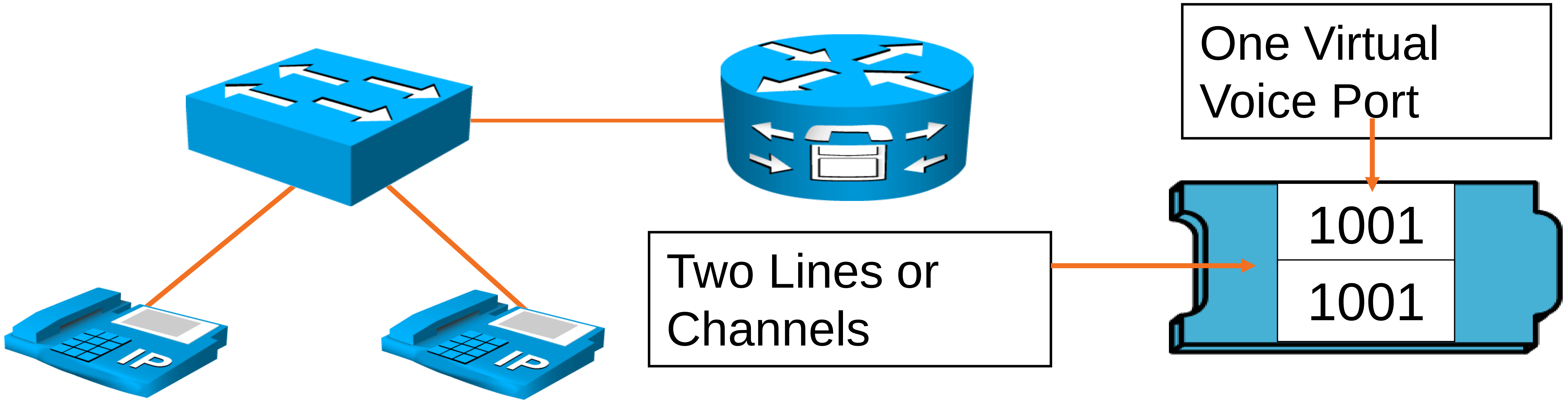
- An ephone-dn has a primary directory number assigned to it and can have an optional secondary number
- A dn-tag is a unique value that is assigned when the ephone-dn is created
- An ephone-dn can be single line or dual line

  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



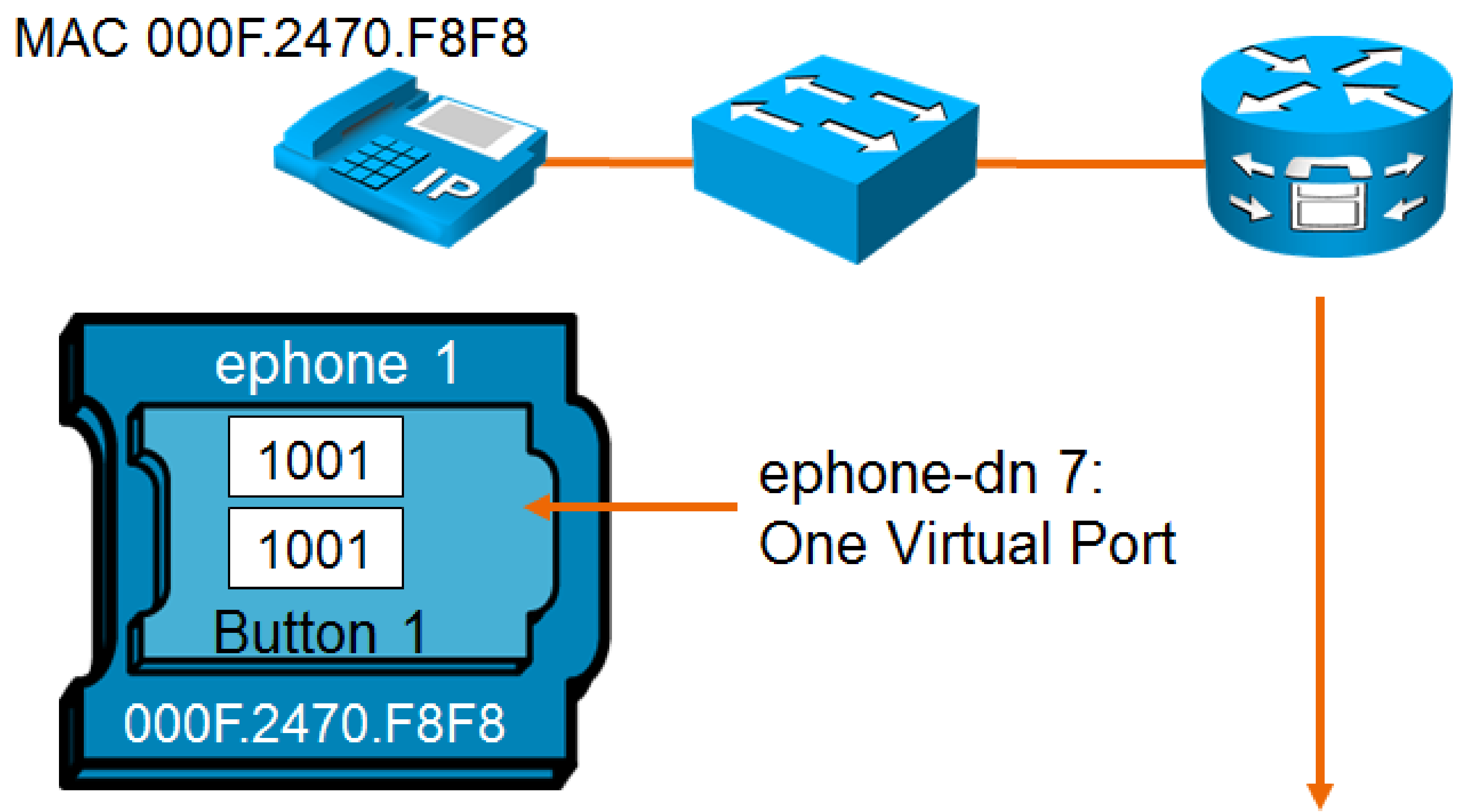
Basic Ephone-dn Configuration




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

- Assigns a primary extension number to an ephone-dn

Example: Basic Ephone Configuration

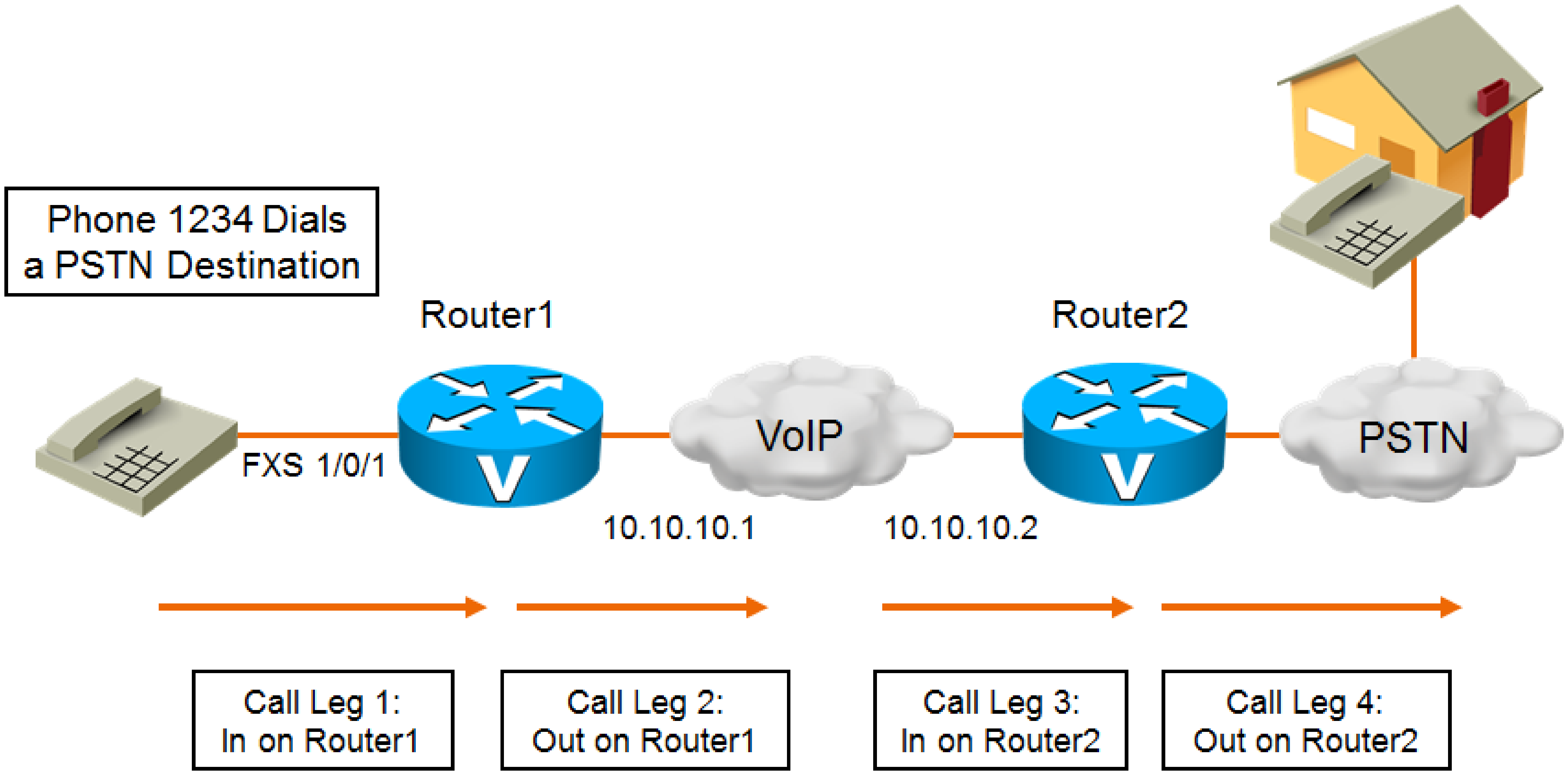



```
CMERouter(config)#ephone-dn 7 dual-line
CMERouter(config-ephone-dn)#number 1001
CMERouter(config-ephone-dn)#exit
CMERouter(config)#ephone 1
CMERouter(config-ephone)#mac-address 000F.2470.F8F8
CMERouter(config-ephone)#button 1:7
```

Example: 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



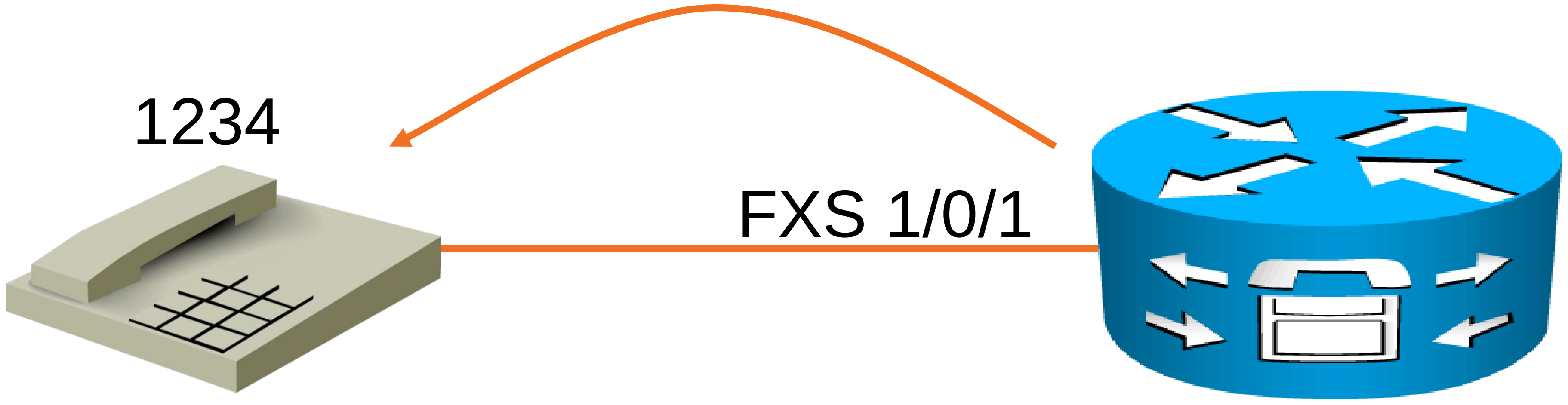
Phone 1234 Dials
a PSTN Destination
Router1
Router2
FXS 1/0/1
VoIP
PSTN
10.10.10.1
10.10.10.2
Call Leg 1:
In on Router1
Call Leg 2:
Out on Router1
Call Leg 3:
In on Router2
Call Leg 4:
Out on Router2

Dial Peers

- Dial peers are an addressable call endpoint; represents a call leg
- You can use dial peers inbound, outbound, or both
- Dial peers define the properties of the call leg…for example:
  – Destination pattern, destination port or IP address
  – Codec
  – QoS markings
  – VAD

- Cisco voice-enabled routers use two types of dial peers:
  – POTS dial peers—connect to a traditional telephony network: FXO, FXS, E&M, BRI, PRI T1/E1, and CAS T1/E1
  – VoIP dial peers—connect over an IP network using an IP address

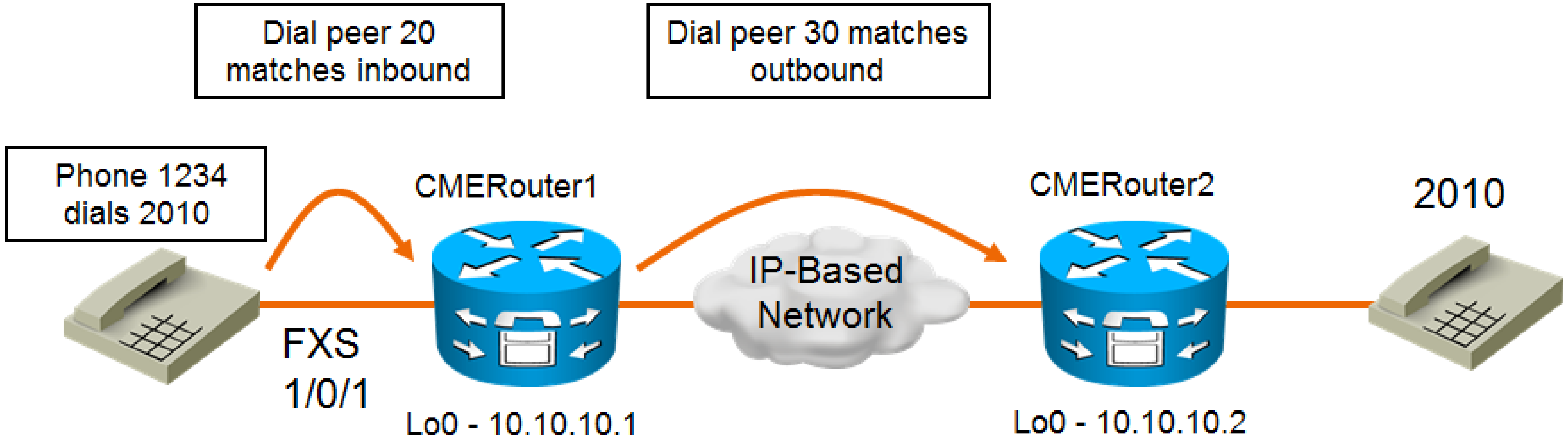
POTS Dial Peers



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




```
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:
  – **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

```
dial-peer voice 1 voip
destination-pattern .T
session target ipv4:10.1.1.1

dial-peer voice 2 voip
destination-pattern 555[2-3]...
session target ipv4:10.2.2.2

dial-peer voice 3 voip
destination-pattern 5551...
session target ipv4:10.3.3.3

dial-peer voice 4 voip
destination-pattern 5551234
session target ipv4:10.4.4.4
```

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

Preference

```
dial-peer voice 5 pots
destination-pattern 5552...
preference 1
port 1/0/0

dial-peer voice 6 voip
destination-pattern 5552...
preference 0
session target ipv4:10.3.3.3
```

- 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

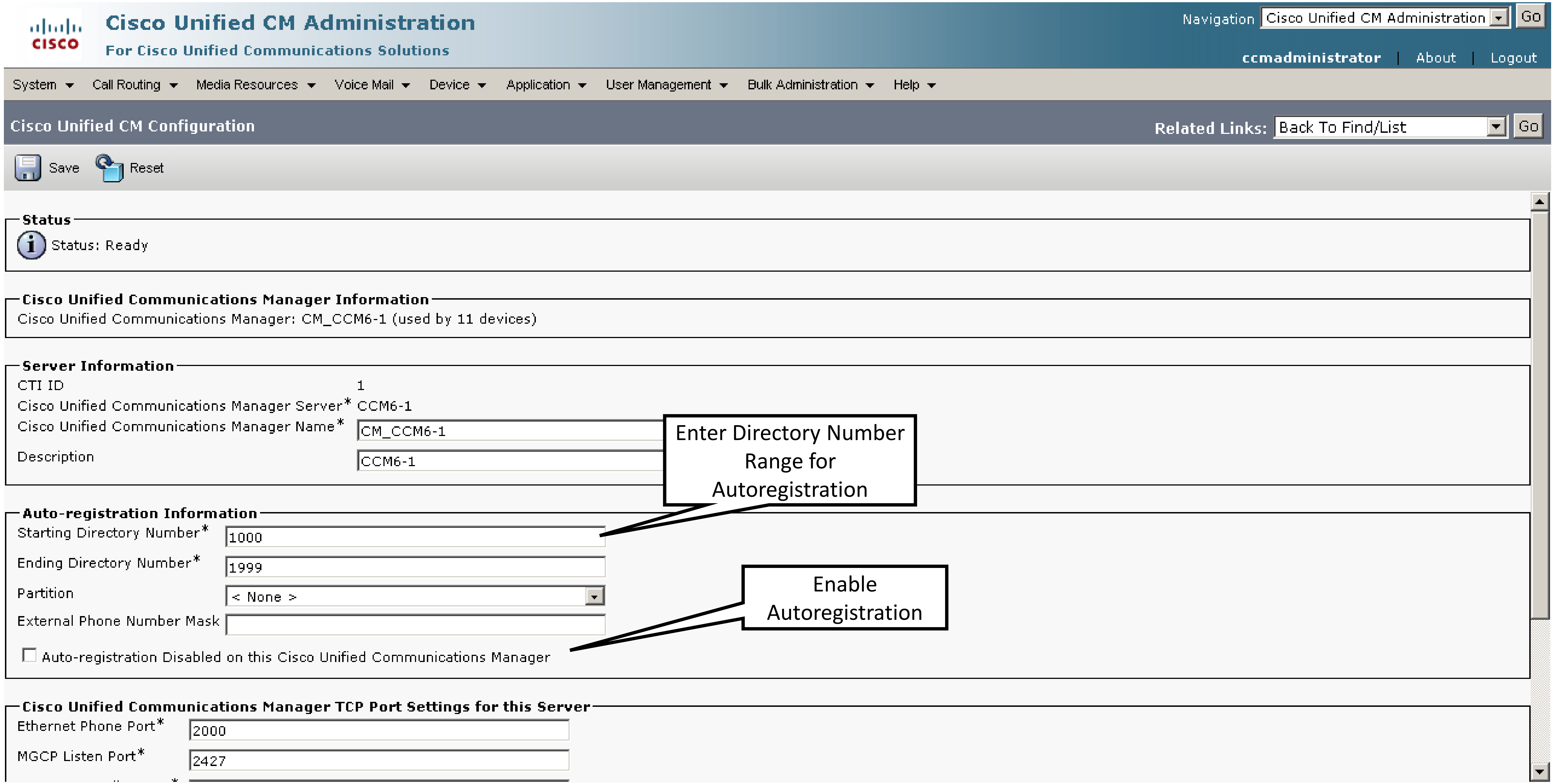
CUCM: Endpoint Configuration Methods

Method for Adding IP Phones	Advantages	Disadvantages
Autoregistration	▪ Devices automatically added	▪ Default Settings, random DN ▪ Modifications needed
Unified CM BAT	▪ Bulk add	▪ MAC addresses required in BAT files
Unified CM Auto-Register Phone Tool	▪ Very scalable ▪ MAC addresses not required	▪ Cisco CRS required ▪ Complex configuration
Manual Configuration	▪ Simple	▪ MAC addresses required ▪ Time-consuming

My Favorite: Autoregistration

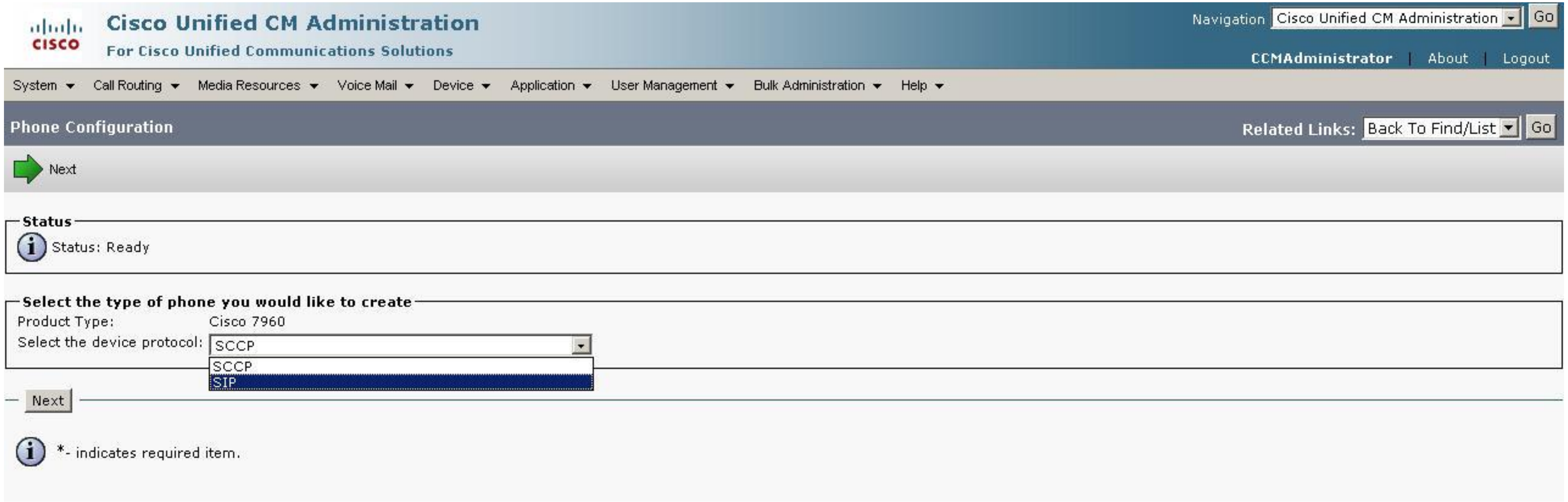
- Supported by all Cisco IP phones
- Existing endpoints are not affected
- Automatically adds Cisco IP phones not found in database (based on MAC addresses)
- Phones assigned the next available directory number of the configured range
- Cisco Unified Communications Manager BAT can be used to make bulk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

CUCM Autoregistration Configuration



Manual IP Phone Configuration

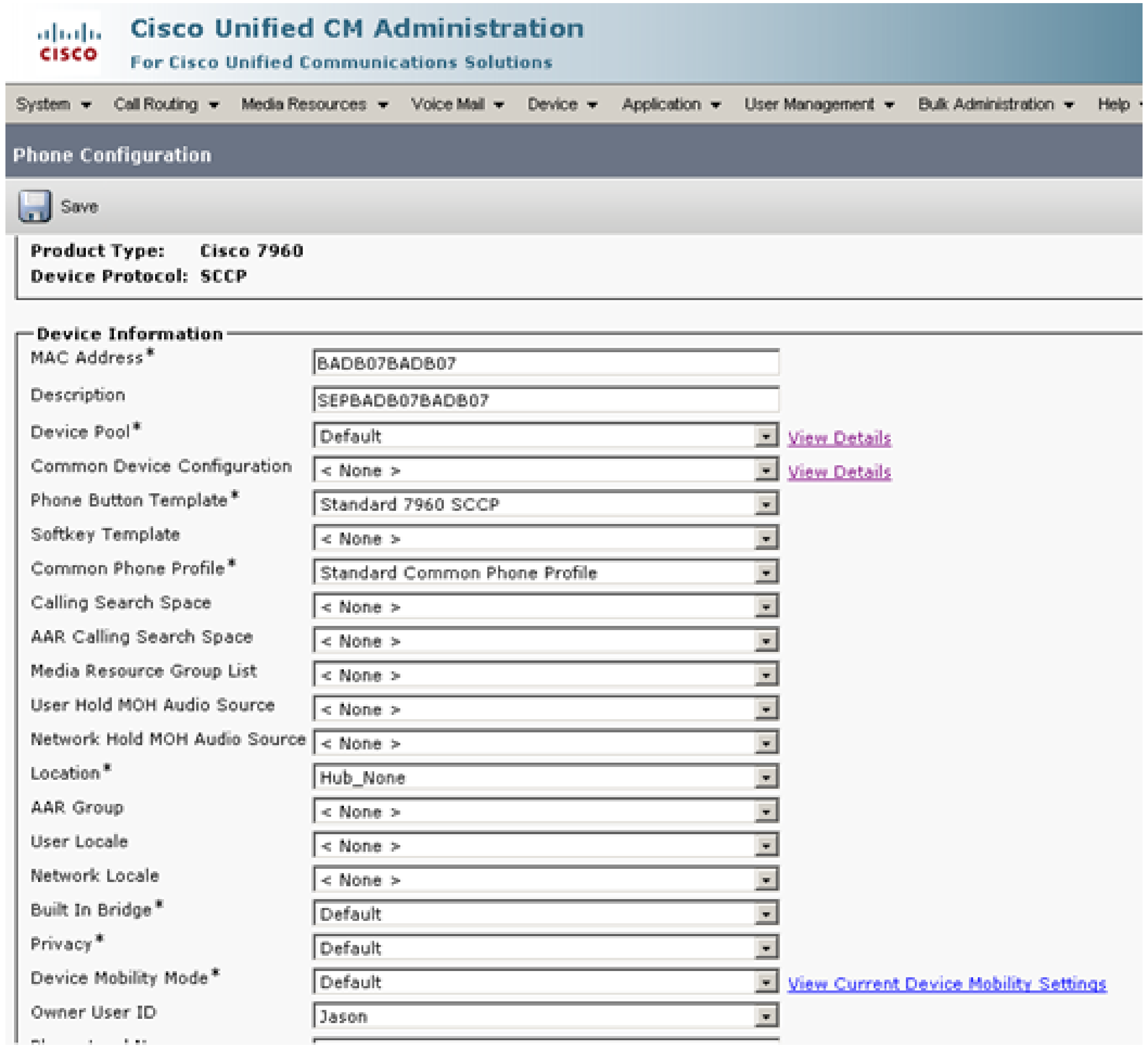
Step 1: Adding the IP Phone



- Select phone type, e.g. 7960
- Select phone protocol (SIP or SCCP)

Manual IP Phone Configuration

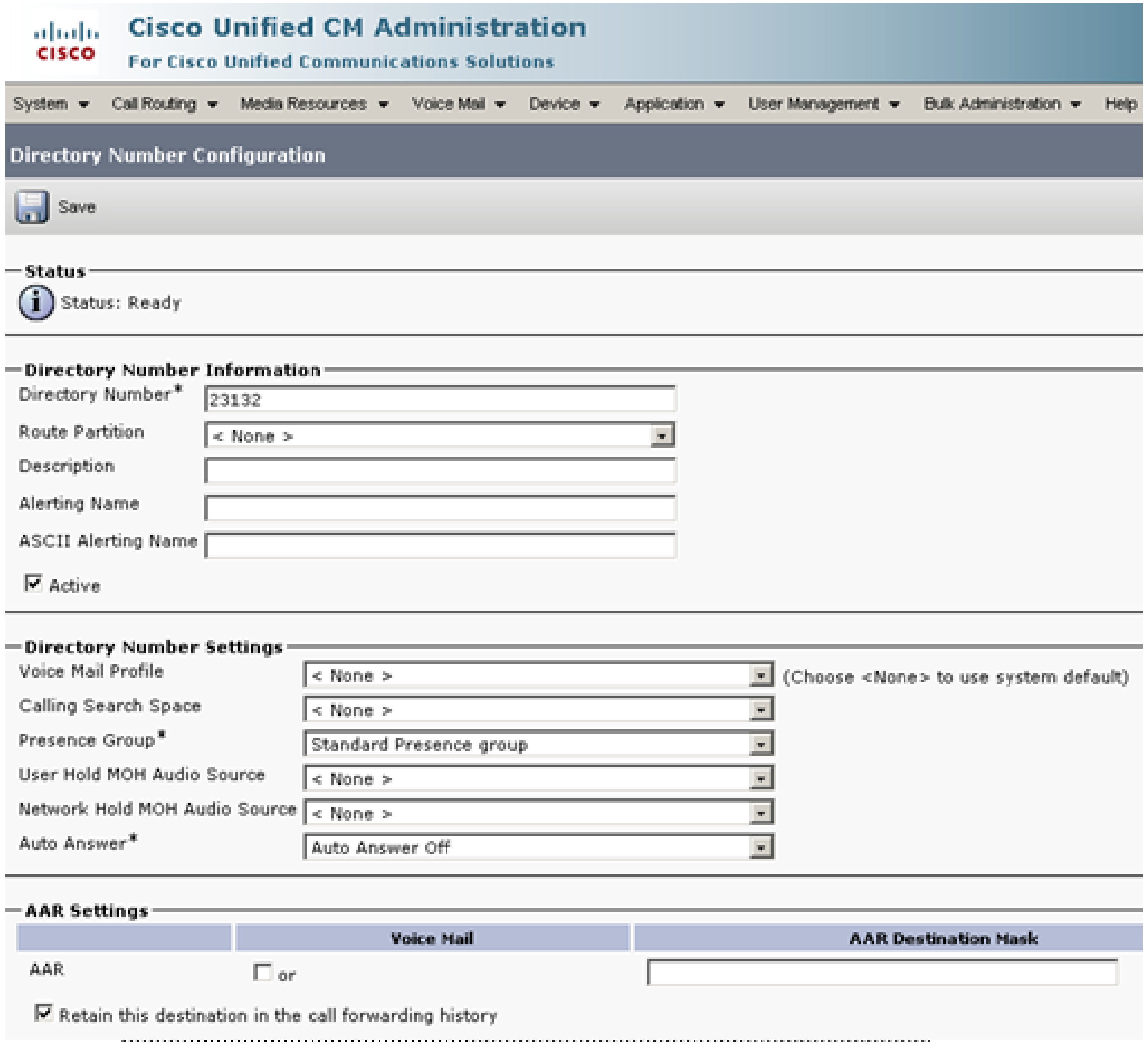
Step 2: Phone Configuration



- 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

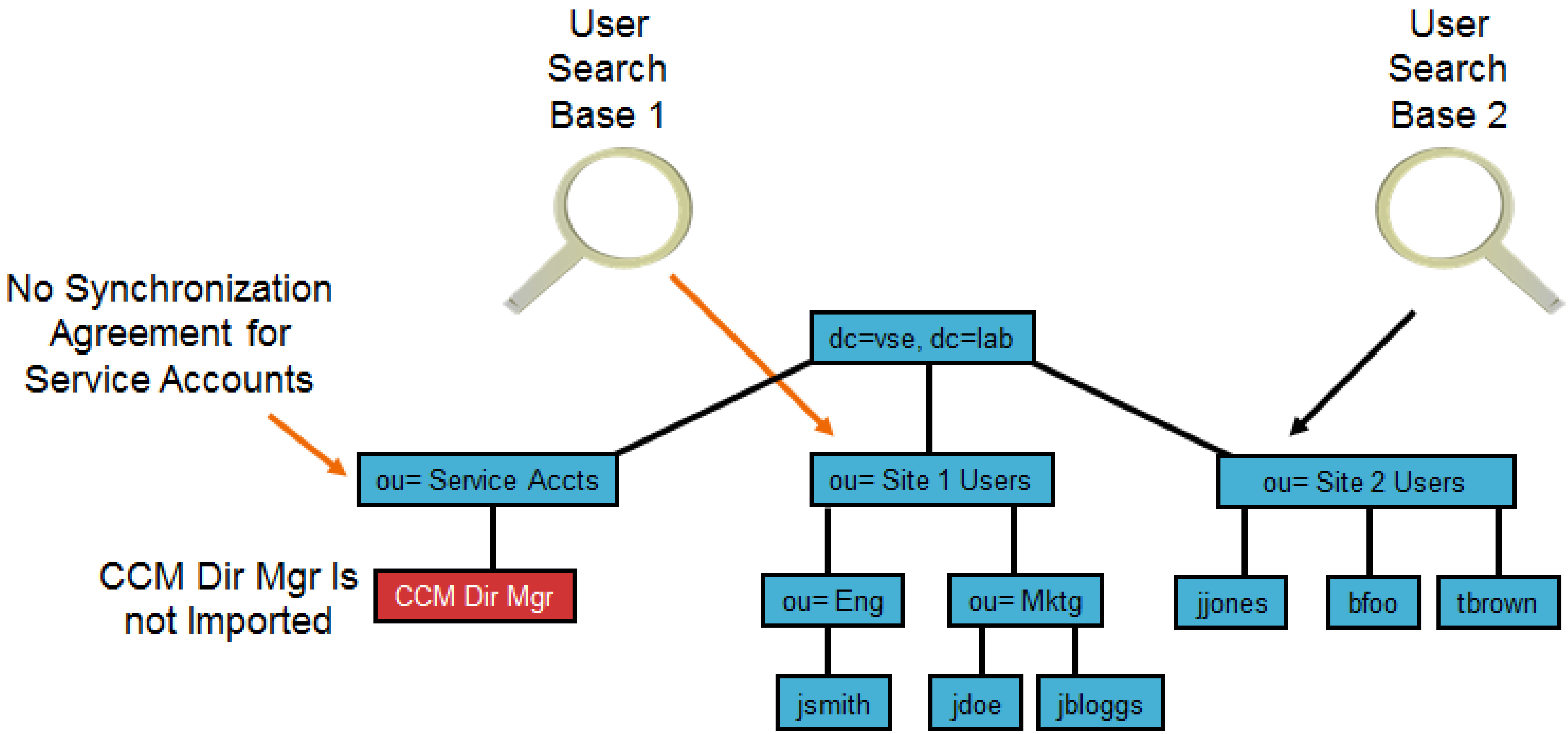
User Management Options

- 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 synchronisation
    - For user provisioning
    - Personal and organisational user data are managed in LDAP
  - LDAP authentication
    - For user authentication
    - Passwords managed in LDAP

Cisco Unified CM End-User Data Location

	No LDAP Integration	LDAP Synchronisation	LDAP Authentication
Personal and Organisational Settings: User ID First, Middle, and Last Name Manager User ID and Department Phone Number and Mail ID	Local	LDAP (replicated to local)	LDAP (replicated to local) or Local
Password	Local	Local	LDAP
Cisco Unified CM Settings: PIN and Digest Credentials Groups and Roles Associated PCs Controlled Devices Extension Mobility Profile and CAPF Presence Group and Mobility	Local	Local	Local

User Search Bases



- Two synchronization agreements are used

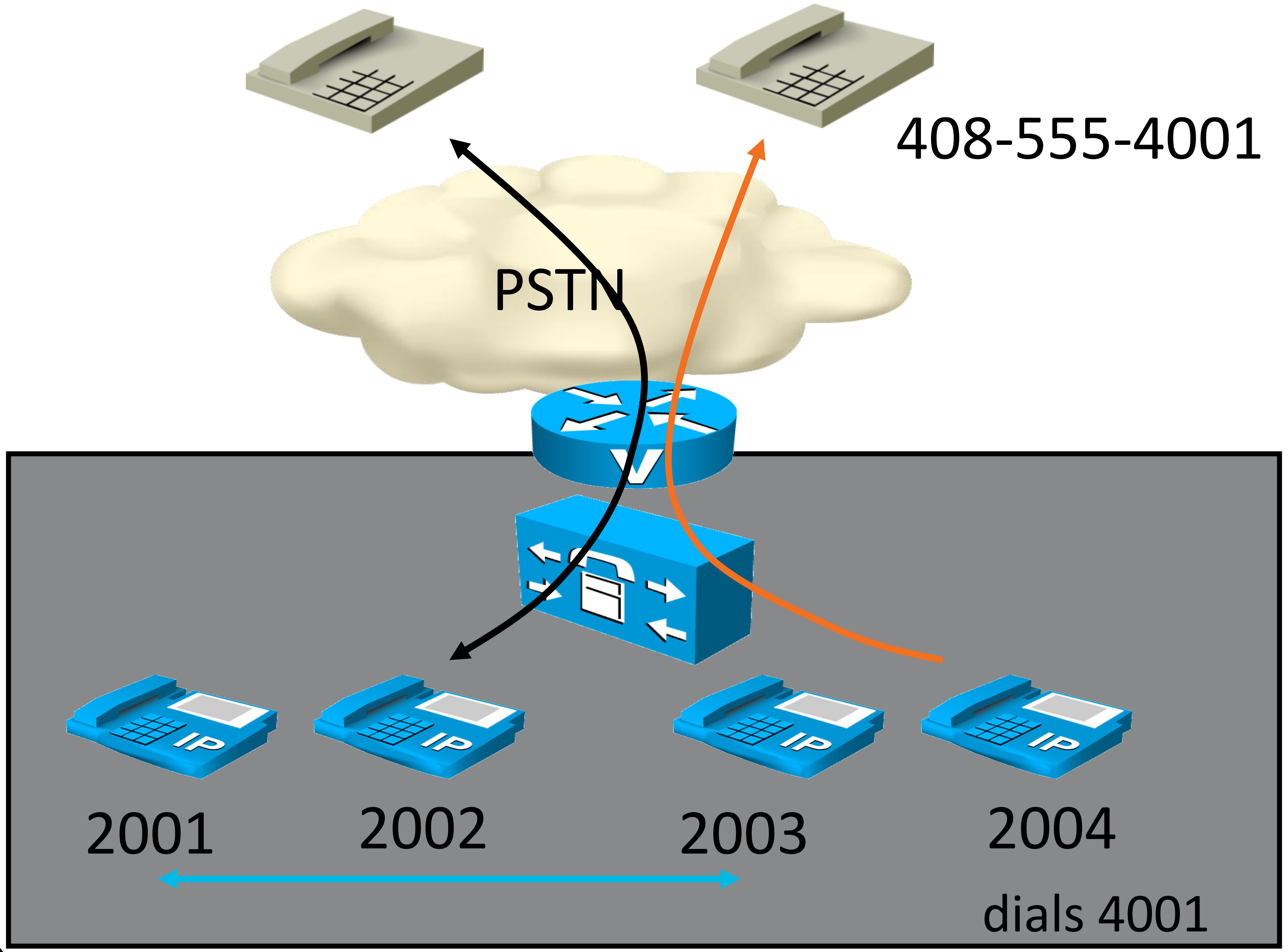
  One synchronization agreement specifies User Search Base 1 and imports users jsmith, jdoe, and jbloggs

  The second synchronization agreement specifies User Search Base 2 and imports users jjones, bfoo, and tbrown

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)



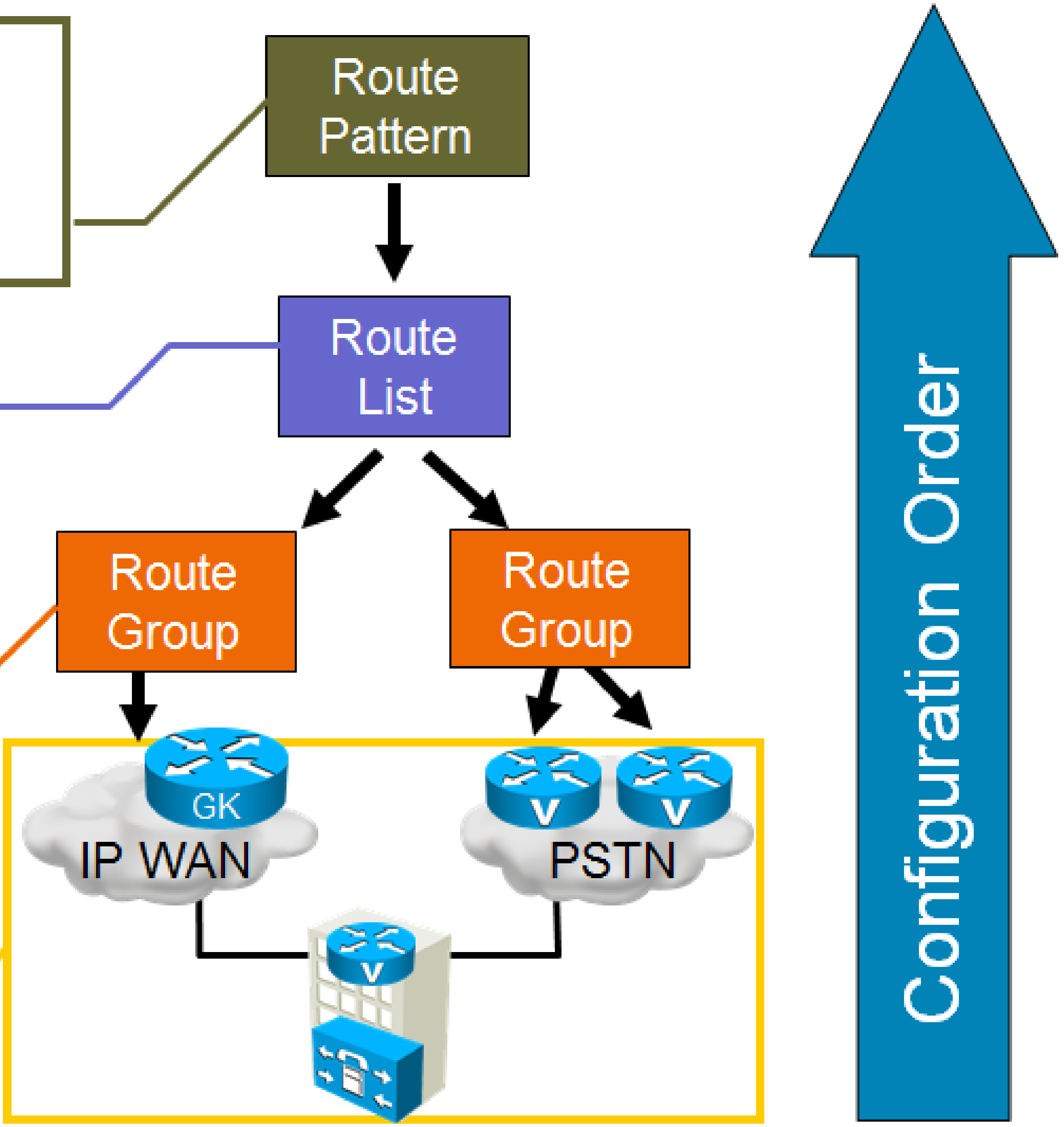
Dial-Plan Configuration in Cisco Unified Communications Manager

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

Wildcard	Description
x	Single digit (0–9, *, #)
@	North American Numbering Plan
!	One or more digits (0–9)
[x-y]	Generic range notation
[^x-y]	Exclusion range notation
.	Terminates access code
#	Terminates interdigit timeout
<wildcard>?	Matches zero or more occurrences of any digit that matches the previous wildcard
<wildcard>+	Matches one or more occurrences of any digit that matches the previous wildcard

Route Pattern Examples

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

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

Lobby x1001

Employee x1002

Manager x1003

Route Patterns:
Local: 9.[2-9]XXXXXXXXX
LD: 9.1[2-9]XXXXXXXXX
INTL: 9.011!

Q & A

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