Implementing Cisco Unified Communications Voice over IP and QoS (CVOICE 642-437)

Guide Version 1.0

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CVOICE Guide 1.0 (71 Pages)

CVOICE CONTENTS

- Analog Voice Ports
- Analog Signaling
- Configuring Analog Phones (dial-peer)
- Analog Voice Port Configuration
- POTS Dial Peer
- VoIP Dial Peer
- Wildcards (Destination Pattern)
- Inbound Dial-peer match
- Outbound Dial-peer Match
- Dial-peer Preference
- Dial-Peer hunt
- PLAR & PLAR OPX
- Digit Manipulation
- CCME
- CCME Features
- Single Number Reach
- CME Call Flow
- Digital Signal
- Digital Voice Interfaces
- Digital Trunks (T1, E1)
- Digital Signaling Methods (CAS, CCS)
- ISDN
- ISDN Call Flow
- COR (Analog Phones)
- COR (IP Phones)
- CVOICE Real Labs

Analog Voice Ports RJ11 (1 Call per Port)

- What is the relevance? Availability
- VIC (RJ11) : 1FXS, 1FXO, 2FXS, 2FXO, 4FXS

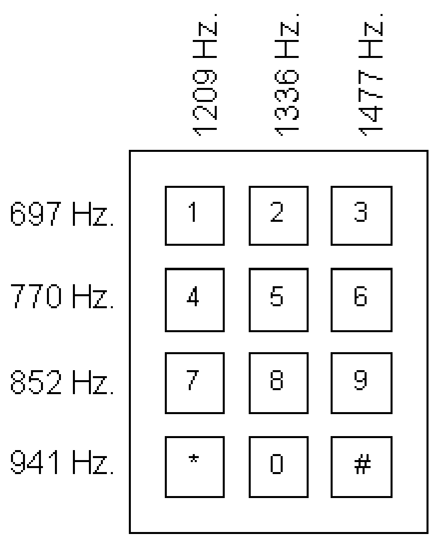
FXS : Foreign Exchange Station (To connect Analog end devices, Phone, Fax, Modem etc)

FXO : Foreign Exchange Office (To bridge analog connection to PSTN, 911 Call)

- FXO - FXS Trunking
- Directly Install on a router or add NM HDV (High Density Voice Network Module)
- Using this module we are giving power to our router to process Voice (20mSec)
- DSP : Digital Signal Processor (Packetizing the voice)

Analog of Signaling

1. Supervisory Signaling (ON HOOK, OFF HOOK, RING)
2. Address Signaling (PULSE, DTMF)



3. Informational Signaling (Dial tone, Busy, Ring Back, Congestion, Conformation)

FXO, FXS Signaling Methods

(Less important, What matches the other side)

Loop Start (PSTN)

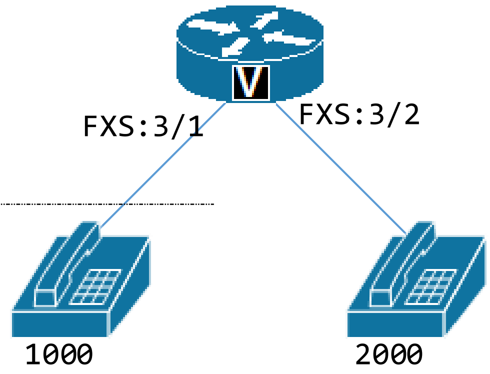
Ground Start (PBX)

<u>Analog Connectivity Issues</u>

Distance limitation (Signal strength low, need repeater, amplify noice)

Wiring limitation (1 Call per line)

Configuring Analog Phones (Dial-Peers)

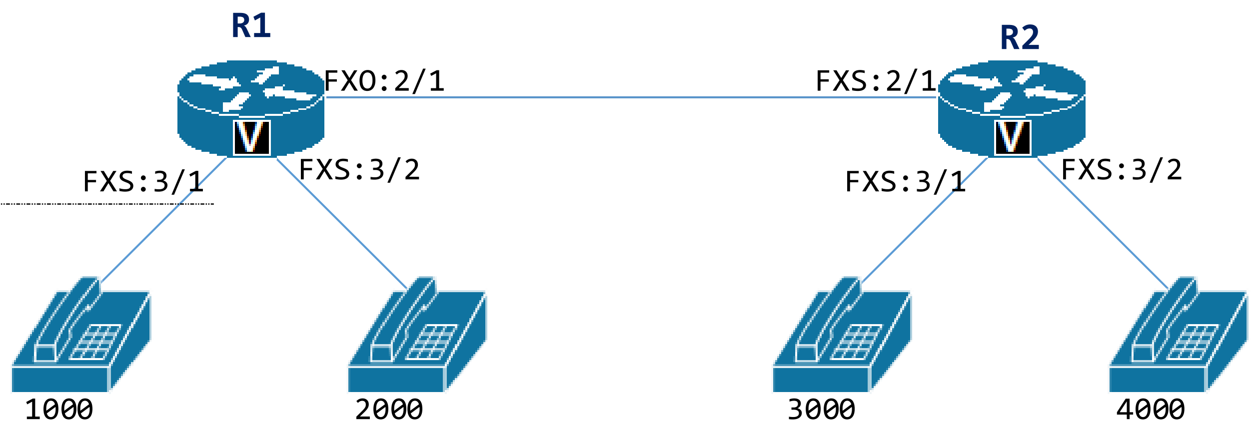



```
#dial-peer voice 1 pots
#destination-pattern 1000
#port 3/1

#dial-peer voice 1 pots
#destination-pattern 1000
#port 3/2
```

Analog Voice Port Configurations

```
#show voice-port summary
#voice-port 3/1
#signalling? (Loop or Ground)
#cptone ?
#ring cadence ?
#shutdown
#no shutdown
#CSIM start <DN>
```

POTS Dial Peers (Inbound & Outbound, Call Legs)



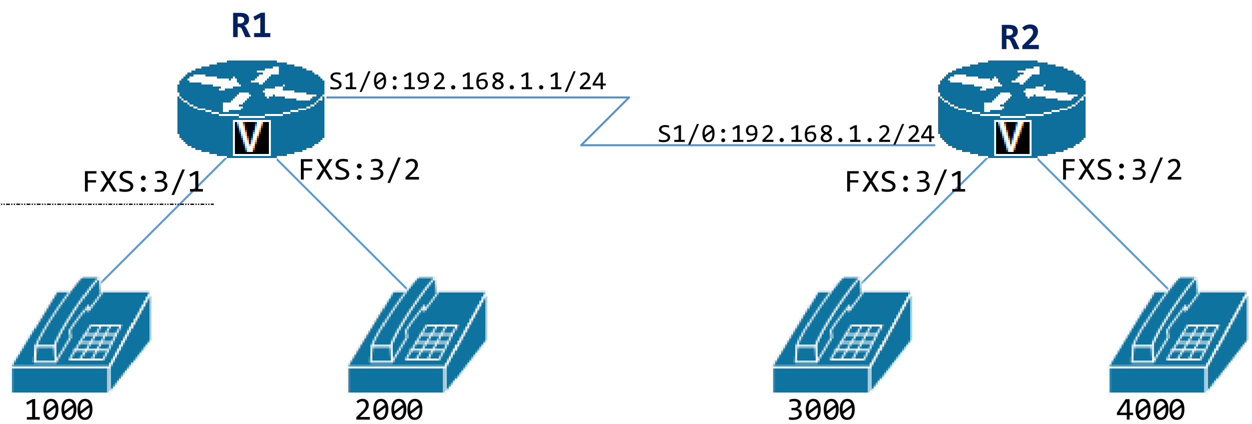
R1 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#description LOCAL_PHONE1
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 3/1

R1(config)#dial-peer voice 2 pots
R1(config-dial-peer)#description LOCAL_PHONE2
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)#port 3/2

R1(config)#dial-peer voice 3 pots
R1(config-dial-peer)#description REMOTE_CALL_3000
R1(config-dial-peer)#destination-pattern 3000
R1(config-dial-peer)#port 2/1
R1(config-dial-peer)#no digit-strip

R1(config)#dial-peer voice 4 pots
R1(config-dial-peer)#description REMOTE_CALL_4000
R1(config-dial-peer)#destination-pattern 4000
R1(config-dial-peer)#port 2/1
R1(config-dial-peer)#no digit-strip
```

R2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE1
R2(config-dial-peer)#destination-pattern 3000
R2(config-dial-peer)#port 3/1

R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE2
R2(config-dial-peer)#destination-pattern 4000
R2(config-dial-peer)#port 3/2

R2(config)#dial-peer voice 3 pots
R2(config-dial-peer)#description REMOTE_CALL_1000
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)#port 2/1
R2(config-dial-peer)#no digit-strip

R2(config)#dial-peer voice 4 pots
R2(config-dial-peer)#description REMOTE_CALL_2000
R2(config-dial-peer)#destination-pattern 2000
R2(config-dial-peer)#port 2/1
R2(config-dial-peer)#no digit-strip
```

VoIP Dial Peers



R1 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#description LOCAL_PHONE1
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 3/1

R1(config)#dial-peer voice 2 pots
R1(config-dial-peer)#description LOCAL_PHONE2
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)#port 3/2

R1(config)#dial-peer voice 3 voip
R1(config-dial-peer)#description REMOTE_VoIP_CALL_3000
R1(config-dial-peer)#destination-pattern 3000
R1(config-dial-peer)# session target ipv4:192.168.1.2

R1(config)#dial-peer voice 4 voip
R1(config-dial-peer)#description REMOTE_VoIP_CALL_4000
R1(config-dial-peer)#destination-pattern 4000
R1(config-dial-peer)# session target ipv4:192.168.1.2
```

R2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE1
```

```
R2(config-dial-peer)#destination-pattern 3000
R2(config-dial-peer)#port 3/1

R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE2
R2(config-dial-peer)#destination-pattern 4000
R2(config-dial-peer)#port 3/2

R2(config)#dial-peer voice 3 voip
R2(config-dial-peer)#description REMOTE_VoIP_CALL_1000
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)# session target ipv4:192.168.1.1

R2(config)#dial-peer voice 4 voip
R2(config-dial-peer)#description REMOTE_VoIP_CALL_2000
R2(config-dial-peer)#destination-pattern 2000
R2(config-dial-peer)# session target ipv4:192.168.1.1
```

Call Legs

1. Inbound Call Leg
2. Outbound Call Leg
3. VOIP, POTS Call Leg

Wild Cards (For Destination Pattern)

Wild Card	Use	Example	Result
.	0 to 9 Match	100.	1001, 1002, 1003, ………, 1009
[X-X]	Single Digit from the range	[4-7]000	4000, 5000, 6000, 7000
[]	Single Digit from the group	[47]000	4000, 7000
[^X]	Negation, Anything but not X	93[^3]	931, 932, 934, 935, ……, 939
(X,XX,X)	Pattern from the group	3(41,88,6)	341, 388, 36
?	Match character to the left 0 or 1 time	83?	83, 833
%	Match character to the left 0 or more time	83%	
+	Match character to the left 1 or more time	83+	
T	Anything up to 32 digits (Use # to stop Inter Digit Timeout)	53T	53X, 53XX, 53XXX, 53XX……XX

Inbound Dial-Peer Match

Inbound call leg says action to take on an inbound call (Codec, Future)

DNIS : Dialed Number Identification Service : Called Number

ANI : Automatic Number Identifier : Calling Number, My Number

1. **incoming called-number** match with DNIS
2. **answer-address** match with ANI
3. **destination-pattern** match with ANI
4. Port march
5. **dial-peer 0**

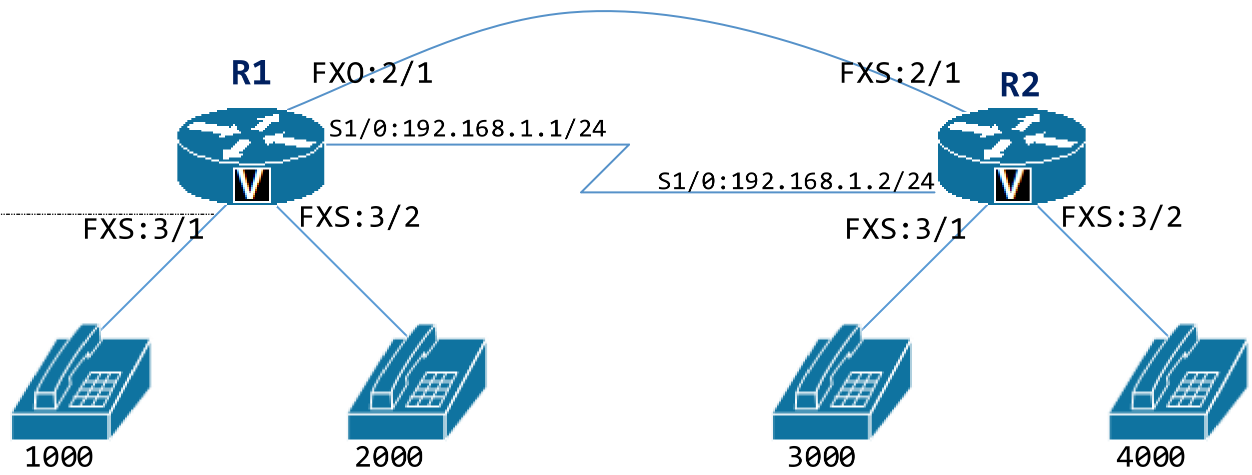
Dial-Peer 0 (Default Dial-Peer)

- Unable to configure
- Works on any codec
- VAD Enabled
- No quality of service (QoS)
- No Application Support
- No DID support
- DTMF Relay doesn’t support

Outbound Dial-Peer Match

Destination-pattern match with DNIS

Dial-Peer Preference



R1 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#description LOCAL_PHONE1
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 3/1

R1(config)#dial-peer voice 2 pots
R1(config-dial-peer)#description LOCAL_PHONE2
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)#port 3/2

R1(config)#dial-peer voice 3 voip
R1(config-dial-peer)#description REMOTE_VoIP_CALL_3000
R1(config-dial-peer)#destination-pattern 3000
R1(config-dial-peer)# session target ipv4:192.168.1.2
R1(config-dial-peer)#preference 0
R1(config)#dial-peer voice 4 voip
R1(config-dial-peer)#description REMOTE_VoIP_CALL_4000
R1(config-dial-peer)#destination-pattern 4000
R1(config-dial-peer)# session target ipv4:192.168.1.2
R1(config-dial-peer)#preference 0

R1(config)#dial-peer voice 5 pots
```

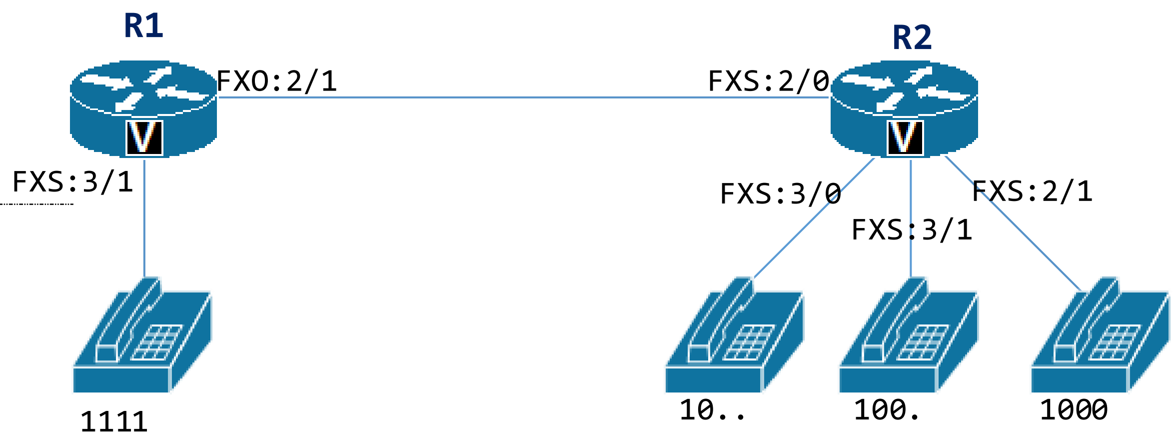
```
R1(config-dial-peer)#description REMOTE_POTS_CALL_3000
R1(config-dial-peer)#destination-pattern 3000
R1(config-dial-peer)#port 2/1
R1(config-dial-peer)#no digit-strip
R1(config-dial-peer)#preference 1

R1(config)#dial-peer voice 6 pots
R1(config-dial-peer)#description REMOTE_POTS_CALL_4000
R1(config-dial-peer)#destination-pattern 4000
R1(config-dial-peer)#port 2/1
R1(config-dial-peer)#no digit-strip
R1(config-dial-peer)#preference 1
```

R2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE1
R2(config-dial-peer)#destination-pattern 3000
R2(config-dial-peer)#port 3/1

R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#description LOCAL_PHONE2
R2(config-dial-peer)#destination-pattern 4000
R2(config-dial-peer)#port 3/2

R2(config)#dial-peer voice 3 voip
R2(config-dial-peer)#description REMOTE_VoIP_CALL_1000
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)# session target ipv4:192.168.1.1
R2(config-dial-peer)#preference 0

R2(config)#dial-peer voice 4 voip
R2(config-dial-peer)#description REMOTE_VoIP_CALL_2000
R2(config-dial-peer)#destination-pattern 2000
```

```
R2(config-dial-peer)# session target ipv4:192.168.1.1
R2(config-dial-peer)#preference 0

R2(config)#dial-peer voice 5 pots
R2(config-dial-peer)#description REMOTE_POTS_CALL_1000
R2(config-dial-peer)#destination-pattern 3000
R2(config-dial-peer)#port 2/1
R2(config-dial-peer)#no digit-strip
R2(config-dial-peer)#preference 1

R2(config)#dial-peer voice 6 pots
R2(config-dial-peer)#description REMOTE_POTS_CALL_1000
R2(config-dial-peer)#destination-pattern 4000
R2(config-dial-peer)#port 2/1
R2(config-dial-peer)#no digit-strip
R2(config-dial-peer)#preference 1
```

Dial-Peer Hunt



R2 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)# destination-pattern 1111
R1(config-dial-peer)# port 1/1

R1(config)#dial-peer voice 2 pots
R1(config-dial-peer)#description OUTBOUND_DIAL-PEER_TO_R2
R1(config-dial-peer)# destination-pattern 10..
R1(config-dial-peer)# port 2/1
```

R2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)# destination-pattern 10..
R2(config-dial-peer)# port 3/0

R2(config-dial-peer)#dial-peer voice 2 pots
R2(config-dial-peer)# destination-pattern 100.
R2(config-dial-peer)# port 3/1

R2(config-dial-peer)#dial-peer voice 3 pots
R2(config-dial-peer)# destination-pattern 1000
R2(config-dial-peer)# port 2/1

R2(config)#dial-peer hunt 5
```

PLAR & PLAR OPX

PLAR (Private Line Automatic Ring Down)

Voice port configured with PLAR capabilities automatically dial a number as soon as the port detect off-hook signal. If the calling party hangs up, you can end up with the called phone continuously ringing.

Uses: Emergency phones in company elevators, parking garages

Configuration

```
Router# voice port 0/0
Router# connection plar 1001
```

PLAR OPX (Off Premise Exchange)

Connection PLAR OPX: The router places call to configured number when ring voltage is detected on the port but the port itself does not go off-hook. Once the called party answers, the FXO port goes off-hook. If ringing stops (ie calling party hangs up) prior to the called party answering, call is dropped. Hence no infinite ringing.

In case of OPX, the router holds the call control until the far end/called party answers. The same reason why it is called Off Premises Extension. You would typically use this when you have a voice gateway VS a CME or a hierarchy of voice gateways just communicating via H.323 or SIP over something like MPLS.

Configuration

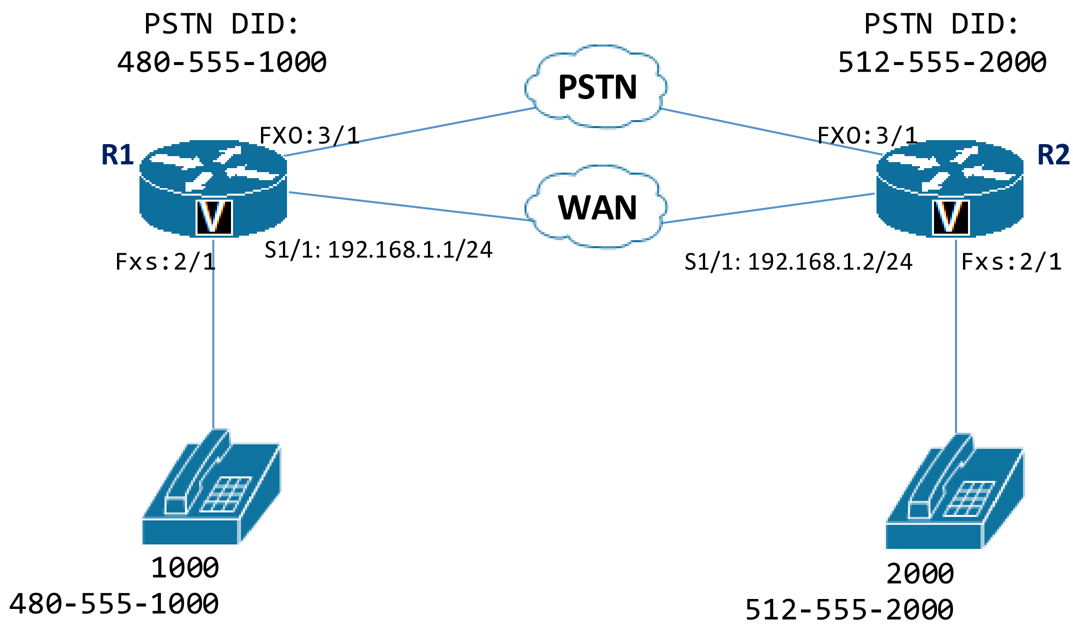
```
Router# voice port 0/0
Router# connection plar opx 1001
```

Digit Manipulation

Digit manipulation is a process of adding or removing digits from a dialed number to help a call reach intended destination properly.

1. Number Expansion (#num-exp 5125552000 200)

It applies on global configuration mode and this will change a number to another number.



When WAN goes down user should dial 512-555-2000 to reach destination 2000 via PSTN. As soon as R2 receive 512-555-2000, it convert the received digits to 2000 using #num-exp command. Similarly in R1 480-555-1000 converted to 1000 when it receive a call via PSTN. The command applied in Global configuration mode.

Router 1 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 2/1

R1(config)#dial-peer voice 2 voip
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)# session target ipv4:192.168.1.2

R1(config)#dial-peer voice 3 pots
R1(config-dial-peer)#destination-pattern 5125552000
```

```
R1(config-dial-peer)#description PSTN_FAIL_OVER
R1(config-dial-peer)#port 3/1
R1(config-dial-peer)#no digit-strip

R1(config)#num-exp 4805551000 1000
```

Router 2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#destination-pattern 2000
R2(config-dial-peer)#port 2/1

R2(config)#dial-peer voice 2 voip
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)# session target ipv4:192.168.1.1

R2(config)#dial-peer voice 3 pots
R2(config-dial-peer)#destination-pattern 4805551000
R2(config-dial-peer)#description PSTN_FAIL_OVER
R2(config-dial-peer)#port 3/1
R2(config-dial-peer)#no digit-strip

R1(config)#num-exp 5125552000 2000
```

2. Automatic Digit Strip (#no digit-strip)

Enable or disable automatic digit stripping behavior of POTS dial peer. Already shown in above example.

3. Voice Translation Profile (Rule) VTR

Applied under Inbound, Outbound dial-peer and under voice port configuration. VTR Works on POTS and VoIP dial-peer. Using VTR we can change Called Number (DNIS) and Calling Number (ANI)

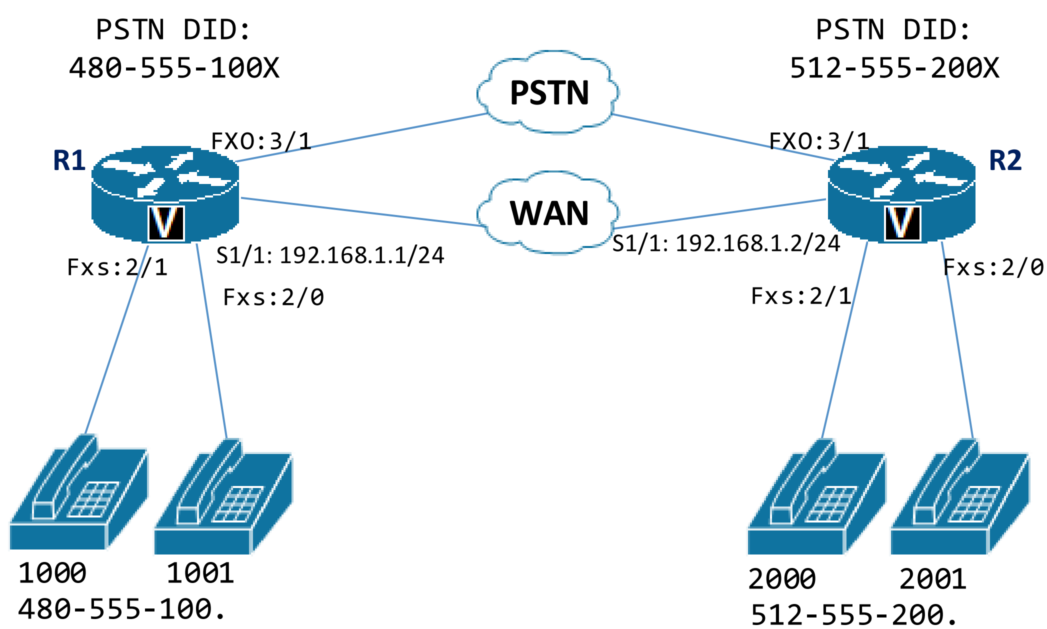
Steps

1. Create VTR Rules
2. Create VTR Profile
3. Apply under dial-peer

VTR Wild Cards

Wild Cards	Use	Example	Result
.	0 to 9 Match		
*	Match the character to the left 0 or more time	.*	1., 2., . will repeat (Full Match)
^	Starting Match	/28^14821/ /00123/	/28 00123/
$	Ending Match	/28148$21/ /00123/	/00123 21/
^XXX$	Exact Match	/28^148$21/ /00123/	/28 00123 21/
[^X-Y]	Do not match a single digit from the range		

Scenario



R1 Configuration

```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 2/1

R1(config)#dial-peer voice 2 pots
```

```
R1(config-dial-peer)#destination-pattern 1001
R1(config-dial-peer)#port 2/0

R1(config)#dial-peer voice 3 voip
R1(config-dial-peer)#destination-pattern 200.
R1(config-dial-peer)#session target ipv4:192.168.1.2

R1(config)#dial-peer voice 4 pots
R1(config-dial-peer)#destination-pattern 200.
R1(config-dial-peer)#port 2/1
R1(config-dial-peer)#no digit-strip
```

Creating VT Rules

Outgoing VT Rule

```
R1(config)#voice translation-rule 1
R1(cfg-translation-rule)#rule 1 /^2\(00.\)$/ /5125552\1/    [DNIS : 200. → 512-555-200.]
R1(cfg-translation-rule)#rule 2 /^1\(00.\)$/ /4805551\1/    [ANI  : 100. → 480-555-100.]
[Note: ANI Changing is optional]
```

Incoming VT Rule

```
R1(config)#voice translation-rule 2
R1(cfg-translation-rule)#rule 1 /^4805551\(00.\)$/ /1\1/    [DNIS : 480-555-100. → 100.]
R1(cfg-translation-rule)#rule 2 /^5125552\(00.\)$/ /2\1/    [ANI  : 512-555-200. → 200.]
[Note: ANI Changing is optional]
```

Creating VT Profiles

Outgoing VT Profile

```
R1(config)#voice translation-profile PSTN_OUTGOING
R1(cfg-translation-profile)#translate called 1          [Outgoing DNIS changing]
R1(cfg-translation-profile)#translate calling 1         [Outgoing ANI changing]
```

Incoming VT Profile

```
R11(config)#voice translation-profile PSTN_INCOMING
R1(cfg-translation-profile)#translate called 2          [Incoming DNIS changing]
R1(cfg-translation-profile)#translate calling 2         [Incoming ANI changing]
```

Apply under Dial-peer

```
R1(config)#dial-peer voice 2
R1(config-dial-peer)#translation-profile outgoing PSTN_OUTGOING
```

```
R1(config)#dial-peer voice 5 pots
R1(config-dial-peer)#incoming called-number 4805551000
R1(config-dial-peer)#port 3/1
R1(config-dial-peer)#translation-profile incoming PSTN_INCOMING
```

Testing VTR Rules

```
CME1#test voice translation-rule 1 2001
Matched with rule 1
Original number: 2001   Translated number: 6195552001
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none

CME1#test voice translation-rule 1 1001
Matched with rule 2
Original number: 1001   Translated number: 4805551001
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none
```

R2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#destination-pattern 2000
R2(config-dial-peer)#port 2/1

R2(config)#dial-peer voice 2 pots
R2(config-dial-peer)#destination-pattern 2001
R2(config-dial-peer)#port 2/0

R2(config)#dial-peer voice 1 voip
R2(config-dial-peer)#destination-pattern 100.
R2(config-dial-peer)#session target ipv4:192.168.1.1

R2(config)#dial-peer voice 2 pots
R2(config-dial-peer)#destination-pattern 100.
R2(config-dial-peer)#port 3/0
R2(config-dial-peer)#no digit-strip
```

Creating VT Rules

Outgoing VT Rule

```
R2(config)#voice translation-rule 1
R2(cfg-translation-rule)#rule 1 /^1\(00.\)$/ /4805551\1/    [DNIS : 100. → 480555100.]
R2(cfg-translation-rule)#rule 2 /^2\(00.\)$/ /5125552\1/    [ANI  : 200. → 512555200.]
```

Incoming VT Rule

```
R2(config)#voice translation-rule 2
R2(cfg-translation-rule)#rule 1 /^5125552\(00.\)$/ /2\1/    [DNIS : 512555200. → 200.]
R2(cfg-translation-rule)#rule 2 /^4805551\(00.\)$/ /1\1/    [ANI  : 480555100. → 100.]
```

Creating VT Profiles

Outgoing VT Profile

```
R2(config)#voice translation-profile PSTN_OUTGOING
R2(cfg-translation-profile)#translate called 1             [Outgoing DNIS Changing]
R2(cfg-translation-profile)#translate calling 1            [Outgoing ANI Changing]
```

Incoming VT Profile

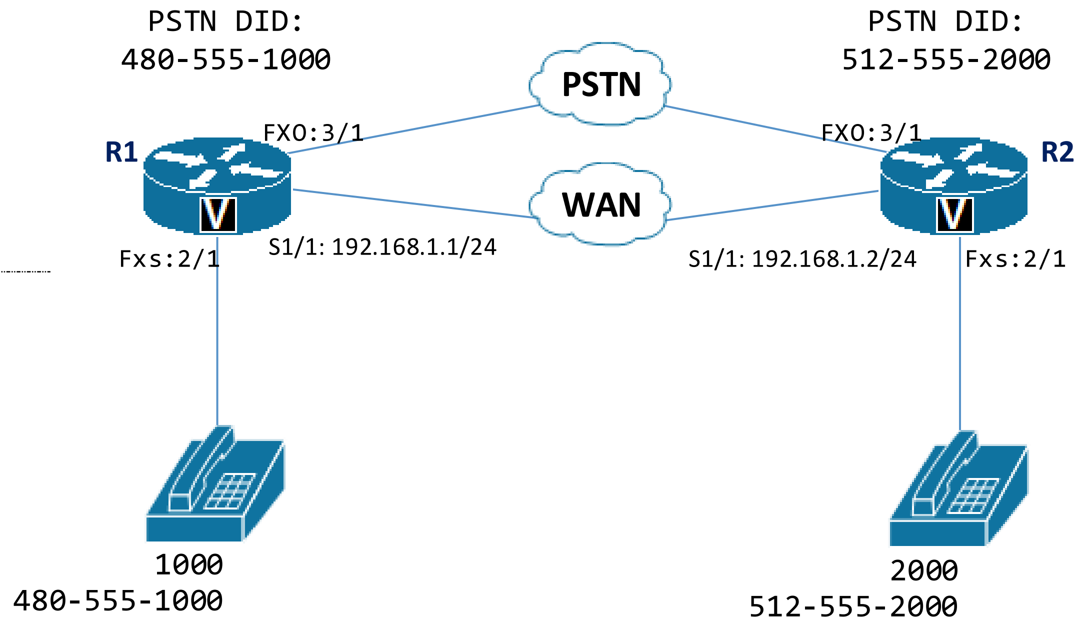
```
R2(config)#voice translation-profile PSTN_INCOMING
R2(cfg-translation-profile)#translate called 2             [Incoming DNIS Changing]
R2(cfg-translation-profile)#translate calling 2            [Incoming ANI Changing]
```

Apply under Dial-peer

```
R2(config)#dial-peer voice 2
R2(config-dial-peer)#translation-profile outgoing PSTN_OUTGOING

R2(config)#dial-peer voice 5 pots
R2(config-dial-peer)#incoming called-number 512555100.
R2(config-dial-peer)#port 3/1
R2(config-dial-peer)#translation-profile incoming PSTN_INCOMING
```

4. Prefix

Allows you to add extra digits before a dialed number.



Even when WAN goes down user may dial 2000 to reach destination 2000 via PSTN but the router R1 will change the DNIS to full DID number (2000 → 512-555-2000). As soon as R2 receive 512-555-2000, it convert the received digits to 2000 using #num-exp command. The command applied under dia-peer configuration mode.

Router 1 Configuration

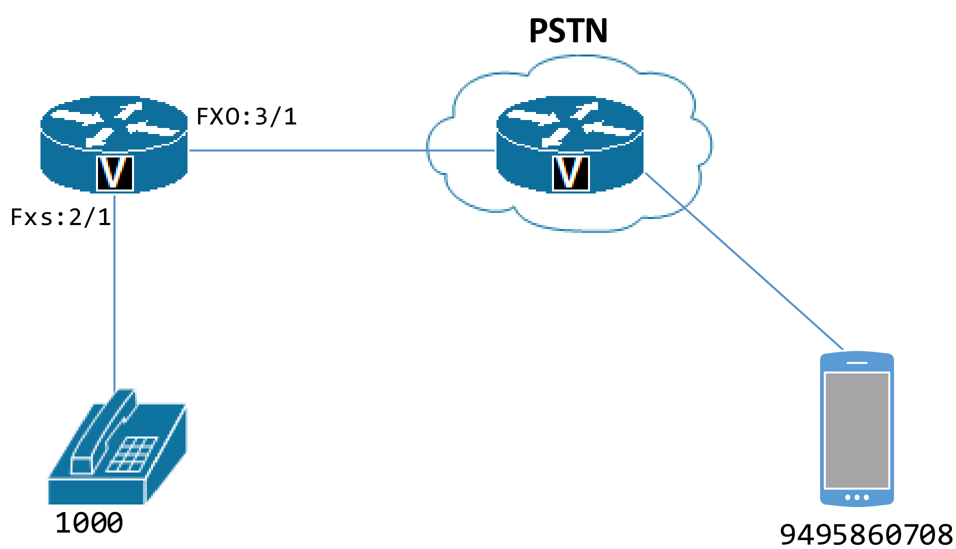
```
R1(config)#dial-peer voice 1 pots
R1(config-dial-peer)#destination-pattern 1000
R1(config-dial-peer)#port 2/1

R1(config)#dial-peer voice 2 voip
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)# session target ipv4:192.168.1.2

R1(config)#dial-peer voice 3 pots
R1(config-dial-peer)#description PSTN_CALL
R1(config-dial-peer)#destination-pattern 2000
R1(config-dial-peer)prefix 512555
R1(config-dial-peer)#port 3/1
R1(config-dial-peer)#no digit-strip

R1(config)#num-exp 4805551000 1000
```

Router 2 Configuration

```
R2(config)#dial-peer voice 1 pots
R2(config-dial-peer)#destination-pattern 2000
R2(config-dial-peer)#port 2/1

R2(config)#dial-peer voice 2 voip
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)# session target ipv4:192.168.1.1

R2(config)#dial-peer voice 3 pots
R2(config-dial-peer)#description PSTN_CALL
R2(config-dial-peer)#destination-pattern 1000
R2(config-dial-peer)#prefix 480555
R2(config-dial-peer)#port 3/1
R2(config-dial-peer)#no digit-strip

R1(config)#num-exp 5125552000 2000
```

5. Forward Digit

Allows you to specify number of right most digits to forward.

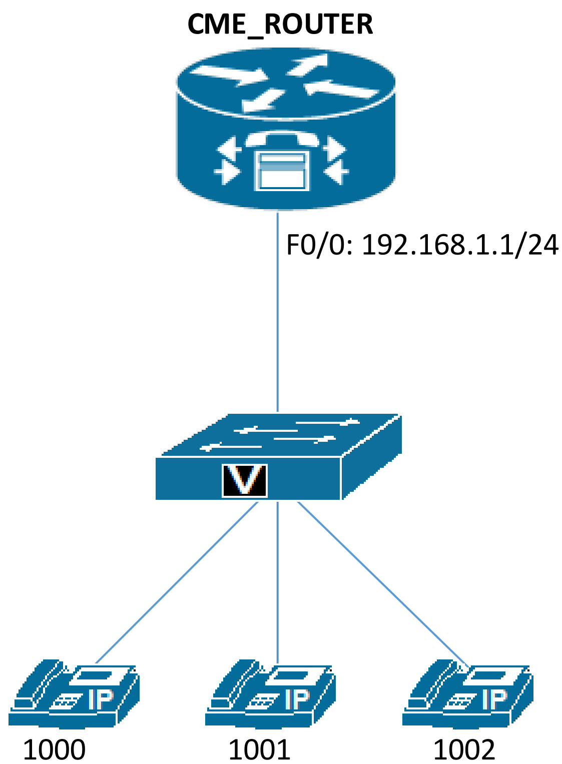



```
Router(config)#dial-peer voice 1 pots
Router(config-dial-peer)#destination-pattern 1000
Router(config-dial-peer)#port 2/1

Router(config)#dial-peer voice 2 pots
```

```
Router(config-dial-peer)#destination-pattern 89495. . . . . .
Router(config-dial-peer)#forward-digits 10
Router(config-dial-peer)#port 3/1
```

Instead of digit-striping command I can able to use forward-digits X command also.

CUCME (Cisco Unified Call Manager Express)



DHCP Configuration (Optional)

```
Router(config)#ip dhcp pool CME1
Router(dhcp-config)#network 192.168.1.0 255.255.255.0
Router(dhcp-config)#default-router 192.168.1.1
Router(dhcp-config)#option 150 ip 192.168.1.1
Router(dhcp-config)#exi
```

CME Configuration

```
Router(config)#telephony-service
Router(config-telephony)#max-dn 10
Router(config-telephony)#max-ephones 10
Router(config-telephony)#ip source-address 192.168.1.1
Router(config-telephony)#create cnf-files
```

CME Optional Commands

```
Router(config-telephony)#user-locale US
Router(config-telephony)#network-locale US
Router(config-telephony)#date-format dd-mm-yy
Router(config-telephony)#time-format 12
```

```
Router(config-telephony)#keepalive 30
Router(config-telephony)#system message TEST_CME
Router# show telephony-service tftp-bindings
```

Creating ephone-dn

```
Router(config)#ephone-dn 1 dual-line
Router(config-ephone-dn)#description MANAGER_EXT
Router(config-ephone-dn)#number 1000
etc
```

ephone Manual Registration

```
Router(config)#ephone 1
Router(config-ephone)#description MANAGER_PHONE
Router(config-ephone)#mac-address 0200.4C4F.4F51
Router(config-ephone)#button 1:1
-----------------------------------------------------------------------------------
:     Normal Ring
B     Beep
S     Silent Ring
M     Monitor Mode
-----------------------------------------------------------------------------------
```

ephone Partial Auto Registration

```
Router(config)#ephone-dn 1 dual-line
Router(config-ephone-dn)#number 1000
Router(config)#ephone-dn 2 dual-line
Router(config-ephone-dn)#number 1001

Router(config)#telephony-service
Router(config-telephony)#auto assign 1 to 2 type 7960
```

ephone Auto Registration (telephony-service setup)

```
Router(config)#telephony-service setup
 --- Cisco IOS Telephony Services Setup ---
Do you want to setup DHCP service for your IP Phones? [yes/no]: yes
Configuring DHCP Pool for Cisco IOS Telephony Services :
```

```
  IP network for telephony-service DHCP Pool     :192.168.1.0
  Subnet mask for DHCP network                   :255.255.255.0
  TFTP Server IP address (Option 150)            :192.168.1.1
  Default Router for DHCP Pool                   :192.168.1.1
```

```
Do you want to start telephony-service setup? [yes/no]: yes
Configuring Cisco IOS Telephony Services :
  Enter the IP source address for Cisco IOS Telephony Services :142.100.0.254
  Enter the Skinny Port for Cisco IOS Telephony Services :  [2000]: 2000
  How many IP phones do you want to configure :  [0]: 10
  Do you want dual-line extensions assigned to phones? [yes/no]: yes
  What Language do you want on IP phones :
      0  English
      1  French
      2  German
[0]: 0
  Which Call Progress tone set do you want on IP phones :
      0  United States
      1  France
      2  Germany
      3  Russia
 [0]: 0
  What is the first extension number you want to configure : 1000
Do you have Direct-Inward-Dial service for all your phones? [yes/no]: no
Do you want to forward calls to a voice message service? [yes/no]: no
Do you wish to change any of the above information? [yes/no]: no
 ---- Setup completed config ---
```

CME Soft key Template

```
Router(config)#ephone-template 1
Router(config-ephone-template)#softkeys connected Endcall Park Trnsfer
Router(config-ephone-template)#exit

Router(config)#ephone 1
Router(config-ephone)#ephone-template 1
Router(config-ephone)#exit
```

- ***alerting***: State after the telephone number has been dialed and the user is waiting (Ring Back)
- ***connected***: Softkey order for connected state
- ***idle***: Softkey order for IDLE state
- ***seized***: Initial dial tone state after going off-hook to place an outbound call

----

CME Button Template

Here the administrator defines the button one by one, hence there is no BUTTON TEMPLATE option in CME. For example Speed Dial configuration is given below.

Speed Dial

```
Router(config)#ephone 3
Router(config-ephone)# speed-dial 1 1001 label HEAD OFFICE
Router(config-ephone)# speed-dial 2 1002 label BRANCH OFFICE
```

Here the first free button will become speed dial to 1001, next free button will be 1002

CME Features

Call Forward

1. From Phone (CFwdAll)

Enable CFwdAll Softkey while idle state. To perform a forward Hit CFwdAll + DN to Forward + #. To cancel hit the CFwdAll again. CFwdAll + Message button to forward to voice mail

Configuration

```
Router(config)#ephone-template 1
Router(config-ephone-template)#softkeys idle CFwdAll
Router(config-ephone-template)#exit

Router(config)#ephone 1
Router(config-ephone)#ephone-template 1
Router(config-ephone)#exit
```

2. From CLI (More Customizable)

```
Router(config)#ephone-dn 3
Router(config-ephone-dn)#call-forward all 2000
Router(config-ephone-dn)#call-forward busy 2001
Router(config-ephone-dn)#call-forward noan 2002 timeout 20
```

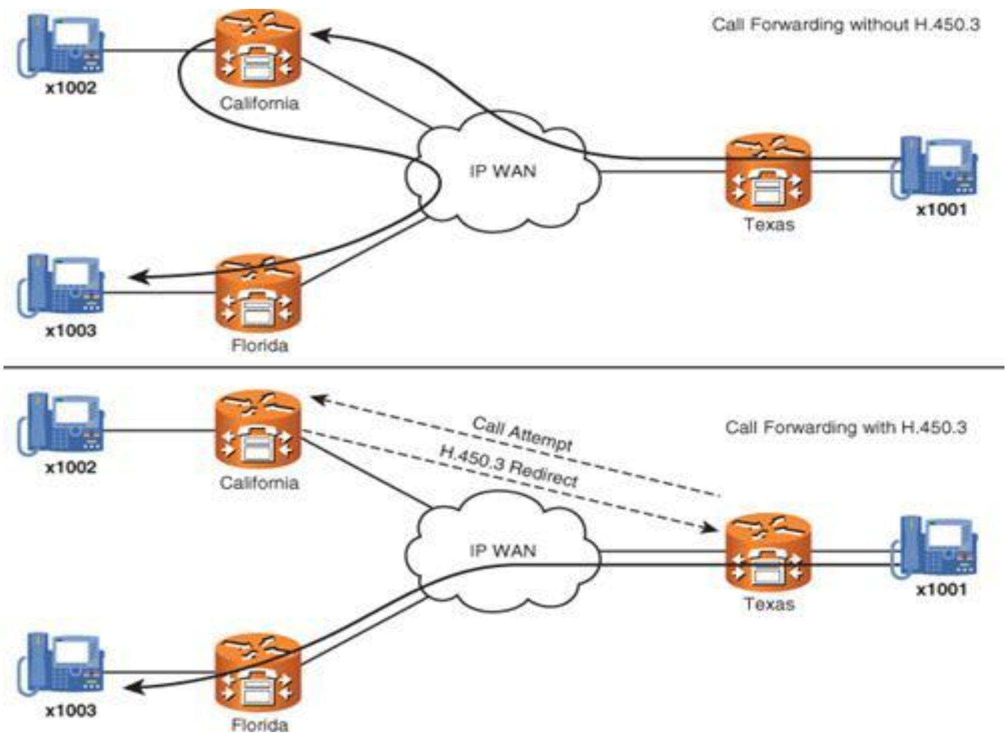
- ***all***: forward all calls
- ***busy***: forward call on busy
- ***max-length***: max number of digits allowed for CFwdAll from IP phone. Value 0 disable the CFwdAll feature from IP Phone
- ***noan***: forward call on no-answer

H.450.3 Standard Forwarding (To avoid rotary forwarding)

While forwarding outside to the network, the forwarding acts as a new call leg, then the call becomes a tandem hop (one behind the other) in the call flow. It leads to some distortion, Call drop, etc. such symptoms called ‘hairpinning’ the call.

H.450.3 standard forward allows CME to redirect the call directly to final destination.

```
Router(config)#telephony-service
Router(config-telephony)#call-forward pattern ....     - Any 4 digit extension
Router(config-telephony)# call-forward pattern .T      - Any number
```



Call Transfer

Enable a Transfer Softkey while connected state.

```
Router(config)#ephone-template 1
Router(config-ephone-template)#softkeys connected Trnsfer Endcall Park
Router(config-ephone-template)#exit

Router(config)#ephone 1
Router(config-ephone)#ephone-template 1
Router(config-ephone)#exit
```

It allows user to perform Consult Transfer by default.

```
Router(config)#telephony-service
Router(config-telephony)#transfer-pattern ....
Router(config-telephony)#transfer-system full-consult
```

- ***blind***: Perform blind call transfers (without consultation) with single phone line using Cisco proprietary method
- ***full-blind***: Perform call transfers without consultation using H.450.2 or SIP REFER standard methods
- ***full-consult***: Perform H.450.2/SIP call transfers with consultation using second phone line (dual-line) if available, fallback to full-blind if second line unavailable. This is the recommended mode for most systems. See also 'supplementary-service' commands under 'voice service voip' and dial-peer.
- ***local-consult***: Perform call transfers with local consultation using second phone line if available, fallback to blind for non-local consultation/transfer target. Uses Cisco proprietary method.
- Consult Transfer need *'dual-line'* DN. We should press Trnsfer + DN + Trnsfer
- Blind, Local Consult leads to Harpinned calls if non Cisco devices exist

----

Call Park

Allows you to retrieve a held call from any phone in the organization. To retrieve the call dial the Park Slot Number.

Step 1 Optional: Enable Park Soft key while connected

```
Router(config)#ephone-template 1
Router(config-ephone-template)#softkeys connected Park Endcall
Router(config-ephone-template)#exit

Router(config)#ephone 1
Router(config-ephone)#ephone-template 1
Router(config-ephone)#exit
```

After completing Step 2 the Park Softkey will appear automatically

Step 2: Create Park Slot DN

```
Router(config)#ephone-dn 10
Router(config-ephone-dn)#number 1111
Router(config-ephone-dn)#description CALL_PARK_SLOT_NUMBER
Router(config-ephone-dn)#park-slot reserved-for 1000
Router(config-ephone-dn)#park-slot timeout 10 limit 3
Router(config-ephone-dn)#park-slot timeout 10 limit 3 notify 3000
Router(config-ephone-dn)#park-slot timeout 10 limit 3 notify 3000 only
Router(config-ephone-dn)#park-slot timeout 10 limit 3 recall
Router(config-ephone-dn)#park-slot timeout 10 limit 3 recall alternate 3001
Router(config-ephone-dn)#park-slot timeout 10 limit 3 transfer 3000

Router(config-ephone-dn)#park-slot timeout 10 limit 3 recall retry 3
-----------------------------------------------------------------------------------
```

- ***Reserved-for <DN>:*** Allows to reserve park slot for a specific number.
- ***Timeout <sec>:*** Specifies number of seconds CME should wait before notifying the original phone where the call is still parked. To notify, CME rings the phone 1 second with a display
- ***Limit <count>:*** After this much timeout the call become dropped.
- ***Notify <DN>:*** Notifies a different DN during each timeout. Here Other DN and the phone from which is parked will be notified.
- ***Notify only:*** Only the other DN will be notified
- ***Recall:*** Call back the original phone after timeout
- ***Alternate <DN>:*** If original phone is busy, alternate DN will called back after timeout
- ***Retry<sec>:*** Set the amount of seconds before CME attempts to transfer a parked call again
- ***Transfer <DN>:*** Call transferred to another DN after timeout

To park a call user hit Park Softkey, CME allocates 1st available park slot to the call. If you want to park to a specific slot use softkey Transfer + Park slot DN

To retrieve Call from Park slot:

From Same Phone → PickUp + * Key

From Other Phones → Dial park slot number or PickUP + Park slot number

---

Call Pickup

Call pickup allows you to answer someone else ringing phone from your local phone.

```
Router(config)#ephone-dn 1
Router(config-ephone-dn)#pickup-group 12
Router(config)#ephone-dn 2
Router(config-ephone-dn)#pickup-group 12
Router(config)#ephone-dn 3
Router(config-ephone-dn)#pickup-group 34
Router(config)#ephone-dn 4
Router(config-ephone-dn)#pickup-group 34
Router(config)#ephone-dn 5
Router(config-ephone-dn)#pickup-group 55
Router(config)#ephone-dn 6
Router(config-ephone-dn)#pickup-group 56
```

- ***Direct PickUp*** : PickUp Softkey + directory number (Pick any Phone regardless of the group) no 'service directed-pickup' command disable this feature.
- ***Local Group PcikUp*** : GPickUp + * (Pick phone with in a same Group)
- ***Other PickUp*** : GPickUp + Group Number (Pick other group phone)

Intercom

Configured between manager and assistant. When assistant press Intercom button, it speed dial manager's number, which auto answer the call on muted speaker phone. It is a 1 way communication. To establish 2 way manager should deactivate mute.

```
Router(config)#ephone-dn 10
Router(config-ephone-dn)#number A1000
Router(config-ephone-dn)#description MANAGER_INTERCOM_NUMBER
Router(config-ephone-dn)#intercom A1001 label ASSISTANT_INTERCOM_SPEED_DIAL

Router(config)#ephone-dn 11
```

```
Router(config-ephone-dn)#number A1001
Router(config-ephone-dn)#description ASSISTANT_INTERCOM_NUMBER
Router(config-ephone-dn)#intercom A1000 label MANAGER_INTERCOM_SPEED_DIAL

Router(config)#ephone 1
Router(config-ephone)#button 2:10

Router(config-ephone)#ephone 9
Router(config-ephone)#button 2:11
```

Paging

Allows to broadcast messages such as emergency notification (fire, etc.). Unicast paging limit is 10. Group paging limit 10 group

1. Unicast Paging Single Group

```
Router(config)#ephone-dn 13
Router(config-ephone-dn)#number 1111
Router(config-ephone-dn)#description ACCOUNTS_PAGING_NUMBER
Router(config-ephone-dn)#paging

Router(config)#ephone-dn 14
Router(config-ephone-dn)#number 2222
Router(config-ephone-dn)#description SALES_PAGING_NUMBER
Router(config-ephone-dn)#paging

Router(config)#ephone 1
Router(config-ephone)#paging-dn 13
Router(config)#ephone 2
Router(config-ephone)#paging-dn 13
Router(config)#ephone 3
Router(config-ephone)#paging-dn 14
Router(config)#ephone 4
Router(config-ephone)#paging-dn 14
```

2. Multicast Paging

```
Router(config)#ephone-dn 13
```

```
Router(config-ephone-dn)#number 1111
Router(config-ephone-dn)#description PAGING_NUMBER
Router(config-ephone-dn)#paging ip 239.0.1.20 port 2000

Router(config)#ephone 1
Router(config-ephone)#paging-dn 13
Router(config)#ephone 2
Router(config-ephone)#paging-dn 13
Router(config)#ephone 3
Router(config-ephone)#paging-dn 13
```

3. Multiple Group Unicast Paging

```
Router(config)#ephone-dn 13
Router(config-ephone-dn)#number 1111
Router(config-ephone-dn)#description ACCOUNTS_PAGING_NUMBER
Router(config-ephone-dn)#paging

Router(config)#ephone-dn 14
Router(config-ephone-dn)#number 2222
Router(config-ephone-dn)#description SALES_PAGING_NUMBER
Router(config-ephone-dn)#paging

Router(config)#ephone-dn 15
Router(config-ephone-dn)#number 3333
Router(config-ephone-dn)#description ACCOUNTS&SALES_PAGING
Router(config-ephone-dn)#paging
Router(config-ephone-dn)#paging group 13,14

Router(config)#ephone 1
Router(config-ephone)#paging-dn 13
Router(config)#ephone 2
Router(config-ephone)#paging-dn 13
Router(config)#ephone 3
Router(config-ephone)#paging-dn 14
```

```
Router(config)#ephone 4
Router(config-ephone)#paging-dn 14
```

4. Multiple Group Multicast Paging

```
Router(config)#ephone-dn 13
Router(config-ephone-dn)#number 1111
Router(config-ephone-dn)#description ACCOUNTS_PAGING_NUMBER
Router(config-ephone-dn)#paging ip 239.0.1.20 port 2000
Router(config)#ephone-dn 14
Router(config-ephone-dn)#number 2222
Router(config-ephone-dn)#description SALES_PAGING_NUMBER
Router(config-ephone-dn)#paging ip 239.0.1.20 port 2000

Router(config)#ephone-dn 15
Router(config-ephone-dn)#number 3333
Router(config-ephone-dn)#description ACCOUNTS&SALES_PAGING
Router(config-ephone-dn)#paging
Router(config-ephone-dn)#paging group 13,14

Router(config)#ephone 1
Router(config-ephone)#paging-dn 13
Router(config)#ephone 2
Router(config-ephone)#paging-dn 13
Router(config)#ephone 3
Router(config-ephone)#paging-dn 14
Router(config)#ephone 4
Router(config-ephone)#paging-dn 14
```

- One way audio
- Speed dial
- Auto answer, Speaker Phone
- IP phones do not support multicast at 224.x.x.x addresses

After Hour Call Blocking

Allows you to define ranges of time specified as After-hour. You can then list number of patterns that are disallowed during those intervals.

Note: Configure clock if there is no NTP server

Router# clock set 18:58:10 04 Oct 2015

Step 1: Define after hours (Mon-Fri: 5PM to 8AM, Sat, Sun: Full Off, Jan 1 full off – New Year day)

```
Router(config)# telephony-service
Router(config-telephony)# after-hours day mon 17:00 8:00
Router(config-telephony)# after-hours day tue 17:00 8:00
Router(config-telephony)# after-hours day wed 17:00 8:00
Router(config-telephony)# after-hours day thu 17:00 8:00
Router(config-telephony)# after-hours day fri 17:00 8:00
Router(config-telephony)# after-hours day sat 00:00 00:00
Router(config-telephony)# after-hours day sun 00:00 00:00
Router(config-telephony)# after-hours date jan 1 00:00 00:00
```

Step 2: Specify patterns to Block (Max 32 patterns)

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 94........
Router(config-telephony)# after-hours block pattern 2 97........
Router(config-telephony)# after-hours block pattern 2 98........ 7-24
```

- 7-24 Block the call all time24/7

Step 3: Exemptions (Optional)

We can add Exemptions on per IP phone basis or using a PIN to bypass the block pattern.

```
Router(config)# ephone 1
Router(config-ephone)# after-hour exempt
Router(config-ephone)# exit

Router(config)# ephone 1
```

```
Router(config-ephone)# pin 1234
Router(config-ephone)# exit

Router(config)# telephony-service
Router(config-telephony)# login timeout 120 clear 23:00
```

- Login command enable login softkey automatically in idle state. If you have created softkey template include Login in idle state. Press login key again to logout
- Timeout: Amount of idle time before the phone automatically revokes the last pin
- Clear: Absolute time at which the last pin becomes invalid

Single Number Reach (SNR)

Single Number reach allows you to link additional devices to a "parent DN". For example you could link your mobile phone with your desk IP phone. When call comes to your DN, the desk IP phone begins to ring, after a small timeout your mobile phone begins to ring along with your office phone. If nobody answered the call CME transfer the call to voice mail, other DN.

Mobility

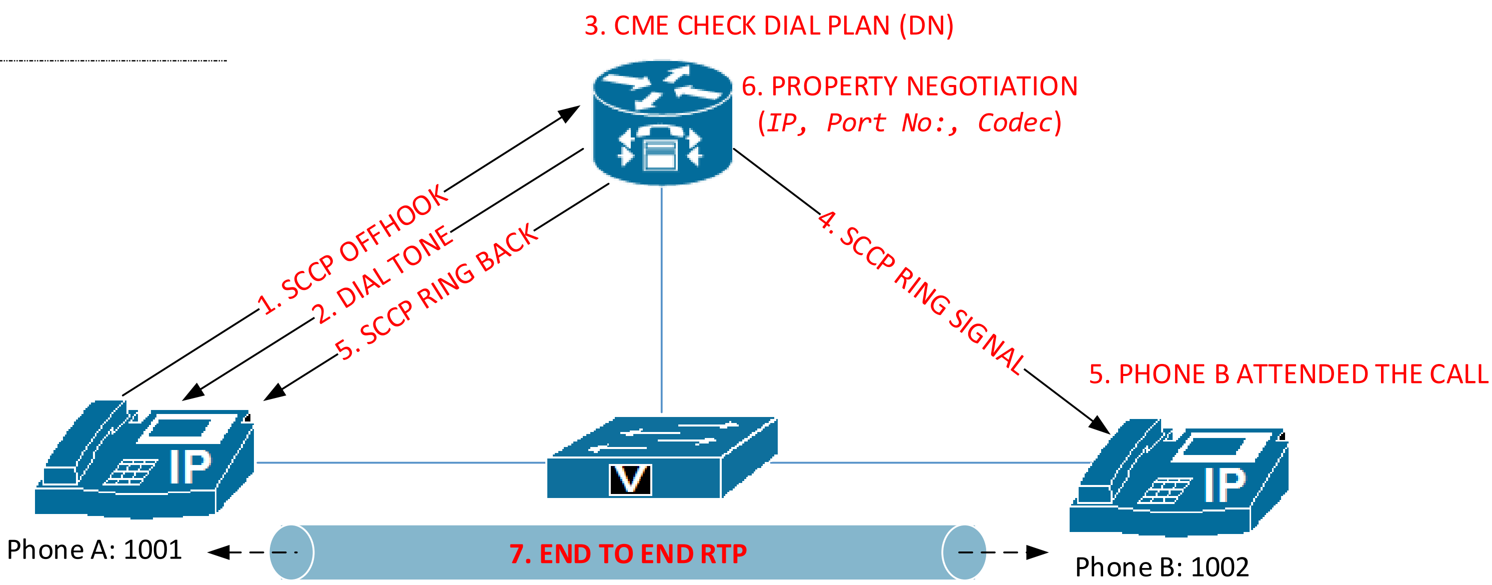
SNR allows a mid-call transfer between your mobile phone and IP Phone. Simply press the Mobility softkey to transfer and Resume to reverse transfer.

```
CME_ROUTER(config)#ephone-dn 1 dual-line
CME_ROUTER(config-ephone-dn)#snr 1234 delay 8 timeout 10 cfwd-noan 1003
CME_ROUTER(config-ephone-dn)#mobility

CME_ROUTER(config)#ephone-template 1
CME_ROUTER(config-ephone-template)#softkeys connected Mobility
CME_ROUTER(config-ephone-template)#exi

CME_ROUTER(config)#ephone 1
CME_ROUTER(config-ephone)#ephone-template 1
```

Delay: remote number ring after this much time

Timeout: The amount of time I seconds cme should wait before call transfer

CME 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

Digital Signal

- Supports more than one call at a time, Voice clarity, Feature rich
- Problems in analog connection: Distance Limitation, 1 call per line, Wiring difficulties
- Signaling used to send called ID information, Transfer, Hold etc

Digital Voice Interfaces

VWIC Cards

- Digital Interfaces broken down in T1, E1, ISDN
- Used to Connect PSTN, PBX
- Digital voice ports are formed at the intersection of a packet switched network and circuit switched network.

Three Types of voice circuits are supported by Cisco voice gateways

1. T1 is an interface having 24 logical voice channel for audio (US, Jappan, Canada). Uses TDM to transmit digital data using CAS
2. E1 is an interface having 32 channels (30 voice channels for audio + 2 Channels for signaling & Framing). Uses TDM to transmit digital data
3. ISDN Integrated Service Digital Network. Is a circuit switched telephone network using CCS

It has 3 variants

1. BRI (Basic rate Interface): 2B+1D
2. T1 PRI: 23B+1D (Primary Rate Interface)
3. E1 PRI: 30B+1D+1F

Digital Trunks

Digital trunks connects PSTN to VoIP network, PSTN to PBX, PBX to VoIP networks, it is widely available in worldwide. A trunk is a digital interface that contains several logical interfaces that connects to a single destination.

T1 CAS	Analog Signaling over Digital T1 line
E1 R2 CAS	Analog Signaling over Digital E1 line
ISDN T1 PRI	Digital signaling over digital T1. Separate signaling channel
ISDN E1 PRI	Digital signaling over digital E1. Separate signaling channel

Digital Signaling Methods

CAS (Channel Associated Signaling)

- Taking the signaling from the Audio Bandwidth
- A T1 line has 24 channels each having 64Kbps bandwidth called DS0 Group
- CAS method steal one bit from these channels hence it is also called RBS (Robed Bit Signaling)
- Advantage: All channel are available for audio (24 or 32 call at a time)
- Disadvantage: User bandwidth reduced (Less Audio clarity)
- We may not get the feature that we need (Transfer) because less efficient signaling

T1 CAS, Configuration & Bandwidth Calculation

- A T1 circuit bundles 24 time slots
- Using CAS, whole 24 slots carry voice traffic

Bandwidth Calculation

24 Channels each having 8 bits	= 24 x 8	= 192
1 Bit for framing & synchronization	= 192 + 1	= 193 bits Called a FRAME
8000 Frames per second	= 8000 x 193	= 1,544,000 = **1.544Mbps**

- A group of 12 FRAME is called SUPPER FRAME where bits from 6^{th} & 12^{th} frame are robbed for signaling.
- A group of 24 FRAME is called SUPPER FRAME where bits from 6^{th}, 12^{th}, 18^{th} & 24^{th} frame are robbed signaling.
- T1CAS provides ANI and DNIS informations in addition to call setup and tear down

- Not all 8000 bit used for framing, 2000 for framing, 2000 CRC, 4000 intelligent supervisory channel (Error detection from network layer & correction in transport layer)

Configuration

Step 1: Controller Configuration

```
Router# show controllers
Router(config)# controllers T1 0/0
Router(config-controller)# framing sf      (or esf)
Router(config-controller)# linecoding b8zs  (or ami)
Router(config-controller)# clock source external (or internal)
Router(config-controller)# ds0-group 1 timeslots 1-12
Router(config-controller)# ds0-group 2 timeslots 13-24
Router(config-controller)# no shutdown
```

Step 2: Verify the Digital Voice Ports

```
Router# show voice port summery
0/0:1      1
0/0:1      2
0/0:1      3
.          .
.          .
.          .
0/0:1      11
0/0:1      12

0/0:2      13
0/0:2      14
0/0:2      15
.          .
.          .
.          .
0/0:2      23
0/0:2      24
```

Step 3: Activate the Digital Voice Port

```
Router(config)# voice port 0/0:1
Router(config-voice-port)# no shutdown

Router(config)# voice port 0/0:2
Router(config-voice-port)# no shutdown
```

Step 4: Inbound match Dial-Peer to Accept Calls

```
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# incoming called-number .
Router(config-dial-peer)# port 0/0:1
Router(config-dial-peer)# direct-inward-dial
```

Step 5: Outbound Dial-Peer for External Calls

```
Router(config)# dial-peer voice 2 pots
Router(config-dial-peer)# destination-pattern .T
Router(config-dial-peer)# port 0/0:2
Router(config-dial-peer)# no digit-strip
```

Here 1 to 12 timeslots (ds0-group 1) used to accept calls and 13 to 24 timeslots (ds0-group 2) used to call external number.

To delete a ds0-group, first shutdown the logical voice port.

```
Router(config)# voice port 0/0:1
Router(config-voice-port)# shutdown
exit
Router(config)# controllers T1 0/0
Router(config-controller)# no ds0-group 1
```

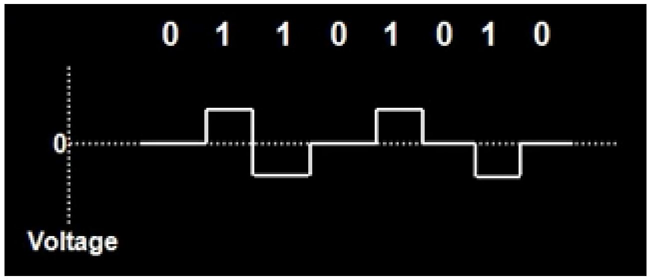
FRAMING: Describes the way bits are robed for signaling. It should match with the other end (PSTN or PBX).

1. **SF**: Supper Framing

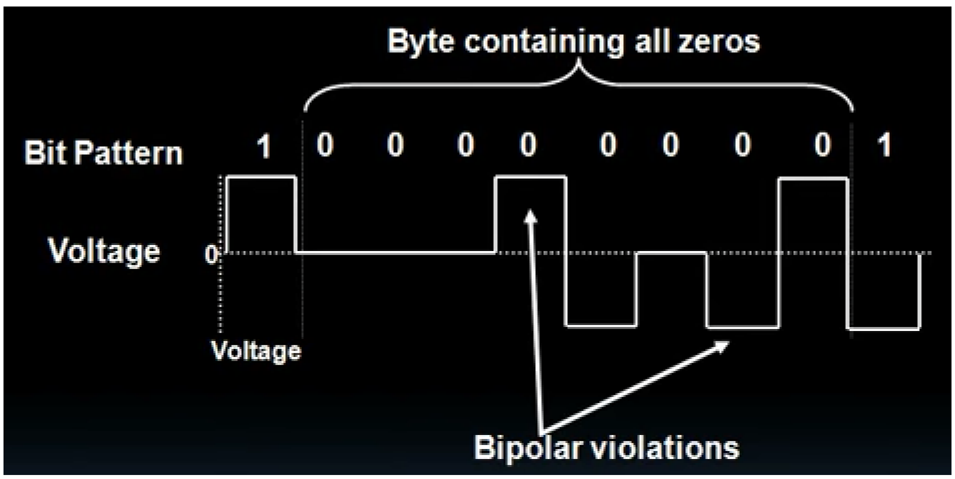
2. **ESF**: Extended Supper Framing

LINECODING: Specifies the type of framing used. It should match with other end.

1. **AMI**: Alternate Mark Inversion: Old method, Alternates polarity when sending binary 1 to maintain average zero voltage.



2. **B8ZS**: Bipolar 8 Zero Substitution: Recommended method



CLOCK SOURCE: Digital interfaces uses timers to ensure voice packets are delivered and assembled properly.

1. **Internal**: Timing derived from on board Phase Locked Loop (PLL) chip present in the Digital interface
2. **External**: Timing derives from PSTN or PBX (External devices) to which the voice port is connected. Recommended method Bcz PSTN clock is more accurate.

If two or more controllers are configured, one should be designed as primary clock source, it will drive other controllers

----

E1 R2 CAS, Configuration & Bandwidth Calculation

- E1 circuit bundles 32 time slots
- Slot 0 for Framing.
- 1 to 31 carry voice traffic (31 time slots available for user data)
- Group of 16 frame called Multiframe

Bandwidth calculation

32 Channels each having 8 bits	= 32 x 8	= 256
8000 Frames per second	= 8000 x 256	= 2,048,000 = **2.048Mbps**

Configuration

Step 1: Controller Configuration

```
Router# show controllers
Router(config)# controllers E1 0/0
Router(config-controller)# framing crc4          (or no-crc4)
Router(config-controller)# linecoding hdb3  (or ami)
Router(config-controller)# clock source external (or internal)
Router(config-controller)# ds0-group 1 timeslots 1-15
Router(config-controller)# ds0-group 2 timeslots 16-31
Router(config-controller)# no shutdown
```

Step 2: Verify the Digital Voice Ports

```
Router# show voice port summery
0/0:1      1
0/0:1      2
0/0:1      3
.          .
.          .
.          .
0/0:1      14
0/0:1      15

0/0:2      16
0/0:2      14
0/0:2      15
.          .
.          .
.          .
0/0:2      29
0/0:2      31
```

Step 3: Activate the Digital Voice Port

```
Router(config)# voice port 0/0:1
Router(config-voice-port)# no shutdown

Router(config)# voice port 0/0:2
Router(config-voice-port)# no shutdown
```

Step 4: Inbound match Dial-Peer to Accept Calls

```
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# incoming called-number .
Router(config-dial-peer)# port 0/0:1
Router(config-dial-peer)# direct-inward-dial
```

Step 5: Outbound Dial-Peer for External Calls

```
Router(config)# dial-peer voice 2 pots
Router(config-dial-peer)# destination-pattern .T
Router(config-dial-peer)# port 0/0:2
Router(config-dial-peer)# no digit-strip
```

Here 1 to 15 timeslots (ds0-group 1) used to accept calls and 16 to 30 timeslots (ds0-group 2) used to call external number.

To delete a ds0-group, first shutdown the logical voice port.

```
Router(config)# voice port 0/0:1
Router(config-voice-port)# shutdown
exit
Router(config)# controllers E1 0/0
Router(config-controller)# no ds0-group 1
```

FRAMING: Describes with or without Cyclic Redundancy Check

1. **CRC4**: With CRC
2. **NO-CRC4**: Without CRC

LINECODING: Specifies the type of framing used. It should match with other end.

1. **AMI** Alternate Mark Inversion: Old method, signal transition with 1

2. **HDBP3** High Density Bipolar 3 recommended method

CLOCK SOURCE: Digital interfaces uses timers to ensure voice packets are delivered and assembled properly.

1. **Internal**: Timing derived from on board Phase Locked Loop (PLL) chip present in the Digital interface
2. **External**: Timing derives from PSTN or PBX (External devices) to which the voice port is connected. Recommended method Bcz PSTN clock is more accurate.

If two or more controllers are configured, one should be designed as primary clock source, it will drive other controllers

----

Network Clock Timing

If a common clock source is not used between devices the binary values in the bit stream night me miss interpreted because the device samples the signal at the wrong moment.

If timing between devices not configured, Clock Slip may occur. Clock Slip is a repetition or deletion of a block of bit stream. It arises due to inability of an equipment buffer to store incoming and outgoing bit stream because of time mis matching.

To eliminate the problem use,

```
Router(config)# network-clock-participates <WIC_SLOT_NO>
Router(config)# network-clock-select <PRIORITY_VALUE> bri/T1/E1
```

CCS (Common Channel Signaling)

- Most people call it as PRI (Primary Rate Interface)
- It dedicate a single channel for Signaling
- *Advantage:* Efficient Signaling means we have 64Kb for signaling only
- *Disadvantage:* We lose 1 channel thus in a T1 line we may have 23 calls, E1 line may have 30 calls simultaneously.

ISDN Integrate Service Digital Network

- ISDN is a circuit switched telephone network designed to allow digital transmission of Data, Voice, and Video over ordinary copper wire.
- ISDN also offers Call waiting, Do not disturb, Speed Dialing, features.
- It uses CCS signaling standard
- ISDN perfect for G.711 calls because each channel having 64Kbps without any robbed bit
- It has inbuilt call control protocol known as ITU/T Q.931
- ISDN organized as 3 layers
  1. ***Layer 1:*** Physical layer – Defines the physical connection between the terminal equipment (TE) and the network termination (NT).
  2. ***Layer 2:*** Q.921 – Also called LAPD (Link Access Protocol, D channel)- which ensures that messages are error free and executed in the right sequence
  3. ***Layer 3:*** Q.931 – Defines the signaling messages for the initial call setup and the termination of the call as defined in Q.931. ITU/T Q.930 & ITU/T Q.931 together support Packet switched & Circuit switched connection

T1 PRI CCS (ISDN) Configuration

Dedicated signaling channel will be 24th (Config X/X:23)

Step 1: Specify ISDN Switch-type

Router(config)#isdn switch-type primary-net5

Step 2: Controller Configuration

Router#show controllers T1 0/0

Router(config)#controller T1 0/0

Router(config-controller)# framing sf (or esf)

Router(config-controller)# linecoding b8zs

Router(config-controller)# clock source external (or internal)

Router(config-controller)#pri-group timeslots 1-23

To delete the pri-group, shutdown the digital voice port

Step 3: Verify Voice Ports

Router#show voice port summary

```
0/0:23     1
0/0:23     2
.          .
.          .
.          .
0/0:23     22
0/0:23     23
```

Step 4: Voice Port Configuration (X/X:23)

Router(config)#interface serial 0/0:23

Router(config-if)#isdn switch-type primary-net5

Router(config-if)#isdn protocol-emulate **network** (other side will be **user**)

Router(config-if)#isdn bchan-number-order ascending (other side will be **descending**)

Router(config-if)#isdn incoming-voice voice

Router(config-if)#isdn overlap-receiving

```
Router(config-if)# no shutdown
```

Step 5: Inbound dial-peer to Accept Calls

```
Router(config)#dial-peer voice 1 pots
Router(config-dial-peer)#incoming called-number .
Router(config-dial-peer)#direct-inward-dial
Router(config-dial-peer)#port 0/0:23
```

Step 6: Outbound for External Calls

```
Router(config)#dial-peer voice 2 pots
Router(config-dial-peer)#destination-pattern .T
Router(config-dial-peer)#port 0/0:23
Router(config-dial-peer)#no digit-strip
```

- ***isdn switch-type***: This should match with same as in PSTN or PBX side
  1. primary-qsig: Supports QSIG signaling per Q.931. Network side functionality is assigned with the isdn protocol-emulate command.
  2. *primary-net5*: NET5 ISDN PRI switch types for Asia, Australia, and New Zealand; ETSI-compliant switches for Euro-ISDN E-DSS1 signaling system.
  3. primary-ntt: Japanese NTT ISDN PRI switches.
  4. primary-4ess: Lucent (AT&T) 4ESS switch type for the United States.
  5. primary-5ess: Lucent (AT&T) 5ESS switch type for the United States.
  6. primary-dms100: Nortel DMS-100 switch type for the United States.
  7. primary-ni: National ISDN switch type
- ***pri-group***: Configure timeslots for ISDN circuit. T1 allows 1 to 23 timeslots for voice (B Channel) & timeslot 24 allocated to the signaling (D Channel)
- ***isdn protocol-emulate***:
  1. network: One side of the ISDN circuit configured as network (e.g. PSTN)
  2. user: Other side must be configure as user (e.g. our side)
- ***isdn bchan-number-order***: Defines the order which the B channel organized for calls
  1. ascending
  2. descending

- ***isdn incoming-voice***: Configure the interface to pass all the calls to DSP card for processing instead of internal MODEMs
- ***overlap-receiving***: Incoming number to be sent digit by digit (not en block)

E1 CCS, Configuration

- E1 circuit bundles 32 time slots
- Slot 0 for Framing and Slot 16 for signaling.
- 1 to 15 (15 time slots) and 17 to 31 (15 time slots) carry voice traffic (30 time slots available for user data)
- Group of 16 frame called Multiframe
- Time slot 16 (8 bit) used for signaling

0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	23	25	26	27	28	29	30	31

- 16^{th} time slot from the 1^{st} frame indicates the start of multi frame
- 16^{th} time slot from the 2^{nd} frame signals 1^{st} & 17^{th} time slots (4 bit for time slot 1 & remaining 4 bit for time slot 17)
- 16^{th} time slot from the 3^{rd} frame signals 2^{nd} & 18^{th} time slots
- 16^{th} time slot from the 4^{th} frame signals 3^{rd} & 19^{th} time slots
- 16^{th} time slot from the 5^{th} frame signals 4^{th} & 20^{th} time slots
- 16^{th} time slot from the 6^{th} frame signals 5^{th} & 21^{st} time slots
- 16^{th} time slot from the 7^{th} frame signals 6^{th} & 22^{nd} time slots
- 16^{th} time slot from the 8^{th} frame signals 7^{th} & 23^{rd} time slots
- 16^{th} time slot from the 9^{th} frame signals 8^{th} & 24^{th} time slots
- 16^{th} time slot from the 10^{th} frame signals 9^{th} & 25^{th} time slots
- 16^{th} time slot from the 11^{th} frame signals 10^{th} & 26^{th} time slots
- 16^{th} time slot from the 12^{th} frame signals 11^{th} & 27^{th} time slots
- 16^{th} time slot from the 13^{th} frame signals 12^{th} & 28^{th} time slots
- 16^{th} time slot from the 14^{th} frame signals 13^{th} & 29^{th} time slots
- 16^{th} time slot from the 15^{th} frame signals 14^{th} & 30^{th} time slots

16^{th} time slot from the 16^{th} frame signals 15^{th} & 31^{th} time slots

Dedicated signaling channel will be 15^{th} (Config X/X:15)

Step 1: Specify ISDN Switch-type

```
Router(config)#isdn switch-type primary-net5
```

Step 2: Controller Configuration

```
Router#show controllers E1 0/0
Router(config)#controller E1 0/0
Router(config-controller)# framing crc4              (or no-crc4)
Router(config-controller)# linecoding hdb3           (or ami)
Router(config-controller)# clock source external     (or internal)
Router(config-controller)#pri-group timeslots 1-31
```

To delete the pri-group, shutdown the digital voice port

Step 3: Verify Voice Ports

```
Router#show voice port summary
0/0:15     1
0/0:15     2
.          .
.          .
.          .
.          .
.          .
.          .
0/0:15     29
0/0:15     30
```

15th channel will be up always

Step 4: Voice Port Configuration (X/X:15)

```
Router(config)#interface serial 0/0:15
Router(config-if)#isdn switch-type primary-net5
Router(config-if)#isdn protocol-emulate network        (other side will be user)
Router(config-if)#isdn bchan-number-order ascending    (other side will be descending)
Router(config-if)#isdn incoming-voice voice
Router(config-if)#isdn overlap-receiving
Router(config-if)# no shutdown
```

Step 5: Inbound dial-peer to Accept Calls

```
Router(config)#dial-peer voice 1 pots
Router(config-dial-peer)#incoming called-number .
Router(config-dial-peer)#direct-inward-dial
Router(config-dial-peer)#port 0/0:15
```

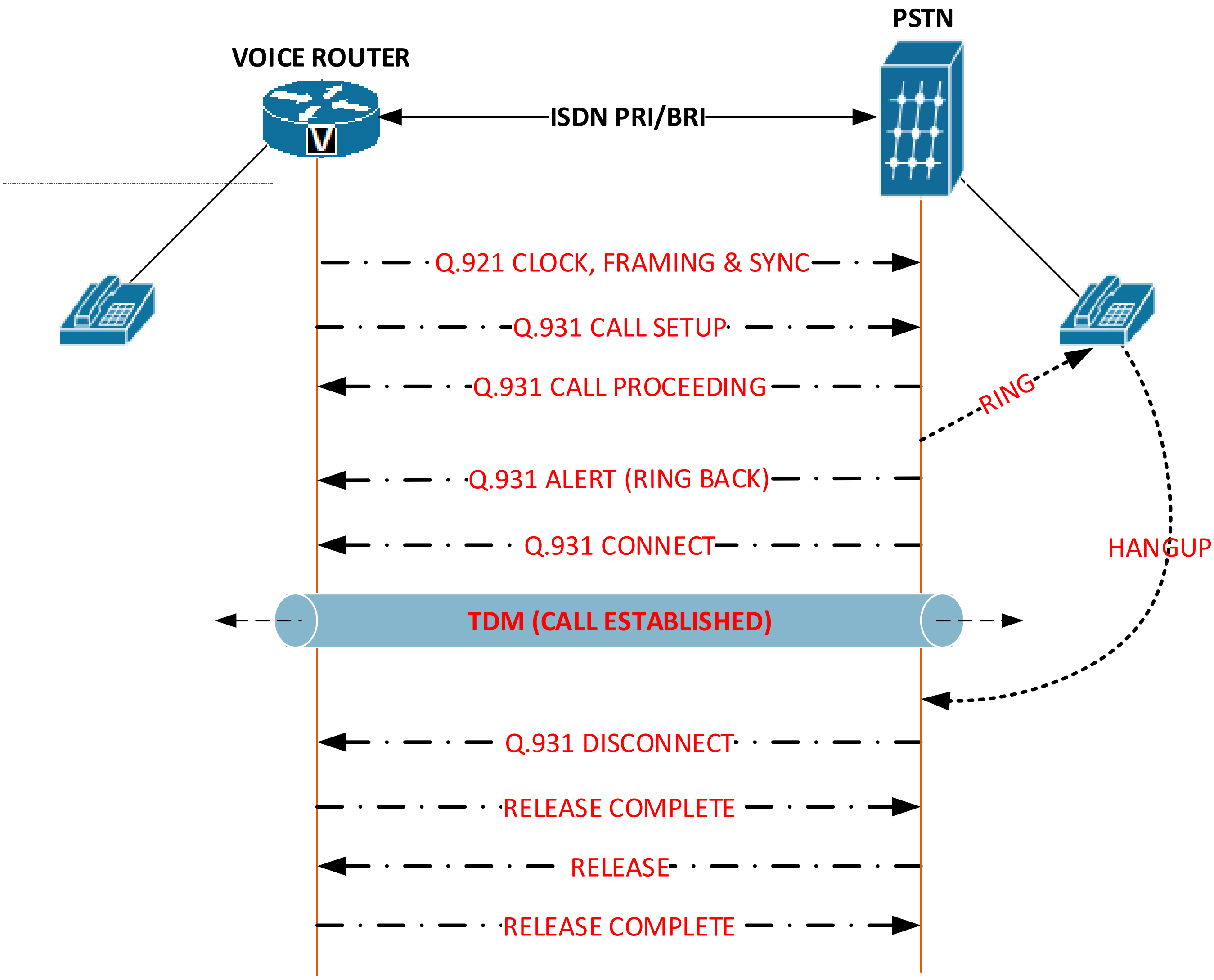
Step 6: Outbound for External Calls

```
Router(config)#dial-peer voice 2 pots
Router(config-dial-peer)#destination-pattern .T
Router(config-dial-peer)#port 0/0:15
Router(config-dial-peer)#no digit-strip
```

ISDN BRI (Basic Rate Interface)

2B+1D = 128Kbps

ISDN Call Flow



Verifying & Troubleshooting Digital Voice Ports

Router# show voice port summary	Identify voice port numbers
Router# show voice port	Voice port parameter settings
Router# show controller T1 0/0	Controller information
Router# show voice dsp	Voice channel configuration info for DSPs
Router# show call summary	Call status for all voice ports
Router# show call active voice	Active call table
Router# show call history voice	History call table
Router# show isdn status	ISDN status (3 layers)
Router# debug isdn q931	Q.931 Call Flow
Router# debug voip ccapi inout	

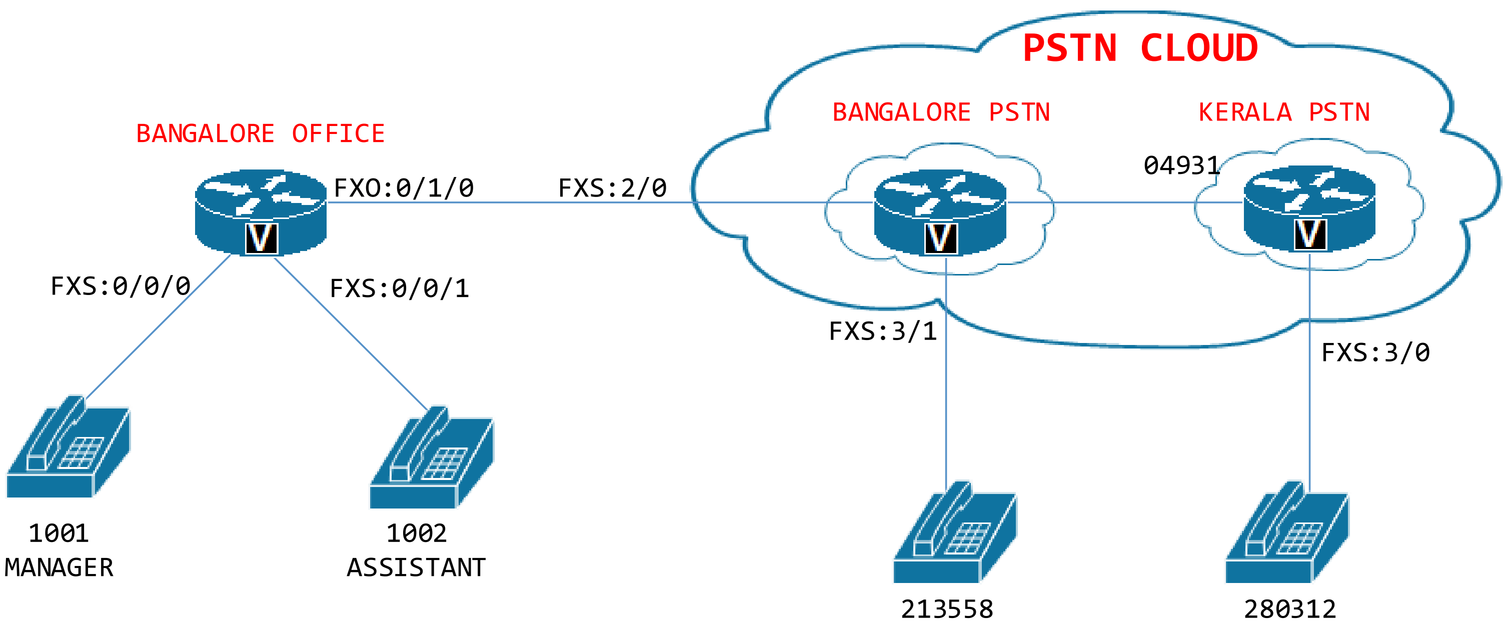
Class of Restrictions (COR)

Dial-Peer Analog Phone COR (Class of Restrictions)

Prevent some users from calling certain number. COR provides calling restrictions in a VoIP network.

COR can be compared to Partition and CSS in CUCM

Scenario:



Restriction: Manager able to call LOCAL & STD, Assistant only LOCAL calls

PSTN CLOUD is a ROUTER

```
PSTN_CLOUD(config)#dial-peer voice 1 pots
PSTN_CLOUD(config-dial-peer)#destination-pattern 213558
PSTN_CLOUD(config-dial-peer)#port 3/1

PSTN_CLOUD(config)#dial-peer voice 2 pots
PSTN_CLOUD(config-dial-peer)#destination-pattern 04931280312
PSTN_CLOUD(config-dial-peer)#port 3/0
```

BANGALORE OFFICE Router

```
BANGLORE_OFFICE(config)#dial-peer voice 1 pots
BANGLORE_OFFICE(config-dial-peer)#destination-pattern 1001
BANGLORE_OFFICE (config-dial-peer)#port 0/0/0
```

```
BANGLORE_OFFICE(config)#dial-peer voice 2 pots
BANGLORE_OFFICE(config-dial-peer)#destination-pattern 1002
BANGLORE_OFFICE(config-dial-peer)#port 0/0/1

BANGLORE_OFFICE(config)#dial-peer voice 3 pots
BANGLORE_OFFICE(config-dial-peer)#destination-pattern 213...
BANGLORE_OFFICE(config-dial-peer)#description LOCAL_CALL
BANGLORE_OFFICE (config-dial-peer)#port 0/1/0
BANGLORE_OFFICE (config-dial-peer)#no digit-strip

BANGLORE_OFFICE(config)#dial-peer voice 4 pots
BANGLORE_OFFICE(config-dial-peer)#destination-pattern 04931......
BANGLORE_OFFICE(config-dial-peer)#description STD_CALL
BANGLORE_OFFICE (config-dial-peer)#port 0/1/0
BANGLORE_OFFICE (config-dial-peer)#no digit-strip
```

Step 1: Create COR-CUSTOM Member

```
BANGLORE_OFFICE(config)#dial-peer cor custom
BANGLORE_OFFICE(config-dp-cor)#name LOCAL_MEMBER
BANGLORE_OFFICE(config-dp-cor)#name STD_MEMBER
```

Step 2: Create Outgoing COR List

```
BANGLORE_OFFICE(config)#dial-peer cor list LOCAL_LIST
BANGLORE_OFFICE(config-dp-corlist)#member LOCAL_MEMBER

BANGLORE_OFFICE(config)#dial-peer cor list STD_LIST
BANGLORE_OFFICE(config-dp-corlist)#member STD_MEMBER
```

Step 3: Create Incoming COR List

```
BANGLORE_OFFICE(config)#dial-peer cor list MANAGER_LIST
BANGLORE_OFFICE(config-dp-corlist)#member LOCAL_MEMBER
BANGLORE_OFFICE(config-dp-corlist)#member STD_MEMBER
```

BANGLORE_OFFICE(config)#dial-peer cor list ASST_LIST

BANGLORE_OFFICE(config-dp-corlist)#member LOCAL_MEMBER

<u>Step 4: Apply Outgoing COR list under Outgoing Dial Peer</u>

BANGLORE_OFFICE (config)#dial-peer voice 3 pots

BANGLORE_OFFICE (config-dial-peer)#corlist outgoing LOCAL_LIST

BANGLORE_OFFICE (config)#dial-peer voice 4 pots

BANGLORE_OFFICE r(config-dial-peer)#corlist outgoing STD_LIST

<u>Step 5: Apply Incoming COR list under incoming Dial Peer</u>

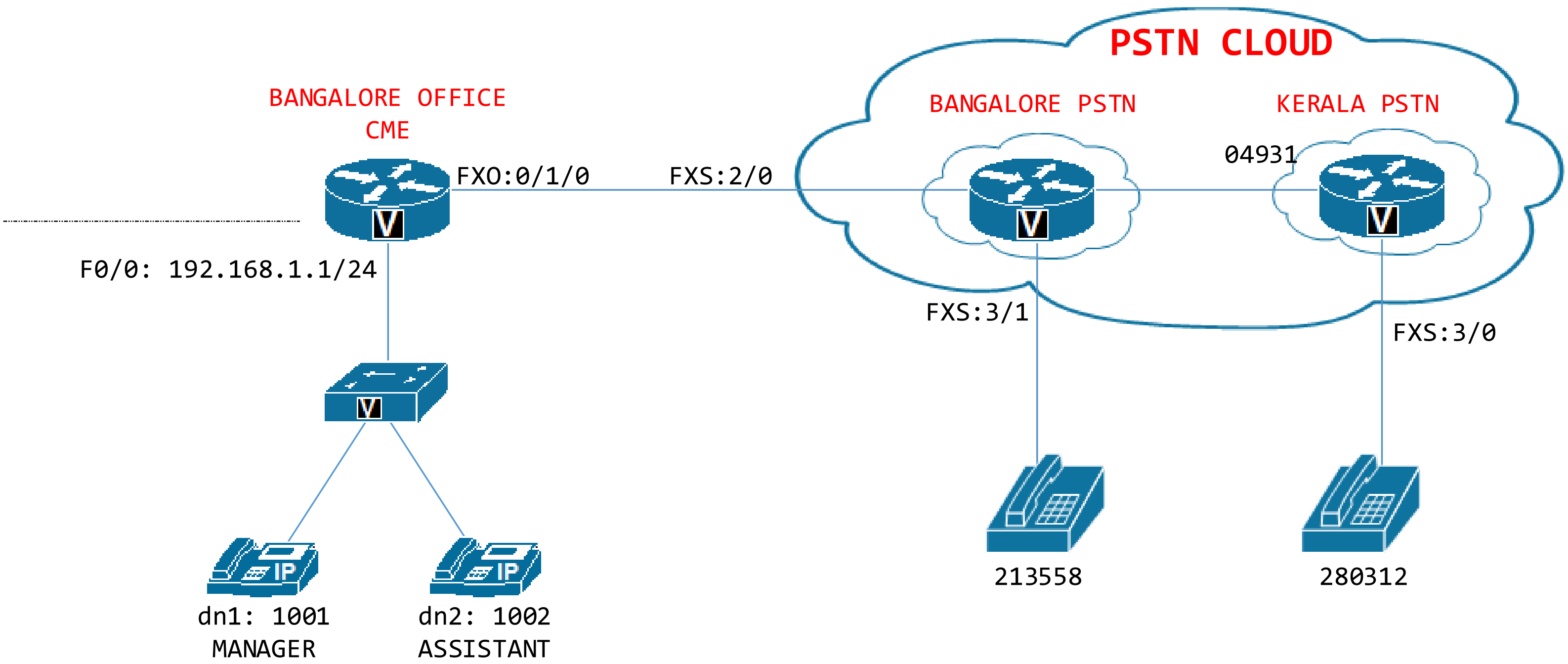
BANGLORE_OFFICE (config)#dial-peer voice 1 pots

BANGLORE_OFFICE (config-dial-peer)#corlist incoming MANAGER_LIST

BANGLORE_OFFICE (config)#dial-peer voice 4 pots

BANGLORE_OFFICE (config-dial-peer)#corlist incoming ASST_LIST

CME IP Phone COR (Class of Restrictions)




```
Restriction: Manager able to call LOCAL & STD, Assistant only LOCAL calls
```

PSTN CLOUD is a ROUTER

```
PSTN_CLOUD(config)#dial-peer voice 1 pots
PSTN_CLOUD(config-dial-peer)#destination-pattern 213558
PSTN_CLOUD(config-dial-peer)#port 3/1

PSTN_CLOUD(config)#dial-peer voice 2 pots
PSTN_CLOUD(config-dial-peer)#destination-pattern 04931280312
PSTN_CLOUD(config-dial-peer)#port 3/0
```

Configure CME

```
ephone-dn 1 dual-line
number 1001
description MANAGER

ephone-dn 2 dual-line
number 1002
description ASSISTANT
```

BANGALORE OFFICE CME

```
BANGLORE_OFFICE_CME(config)#dial-peer voice 3 pots
BANGLORE_OFFICE_CME(config-dial-peer)#destination-pattern 213...
BANGLORE_OFFICE_CME(config-dial-peer)#description LOCAL_CALL
BANGLORE_OFFICE_CME(config-dial-peer)#port 0/1/0
BANGLORE_OFFICE_CME(config-dial-peer)#no digit-strip

BANGLORE_OFFICE_CME(config)#dial-peer voice 4 pots
BANGLORE_OFFICE_CME(config-dial-peer)#destination-pattern 04931......
BANGLORE_OFFICE_CME(config-dial-peer)#description STD_CALL
BANGLORE_OFFICE_CME(config-dial-peer)#port 0/1/0
BANGLORE_OFFICE_CME(config-dial-peer)#no digit-strip
```

Step 1: Create COR-CUSTOM Member

```
BANGLORE_OFFICE_CME(config)#dial-peer cor custom
BANGLORE_OFFICE_CME(config-dp-cor)#name LOCAL_MEMBER
BANGLORE_OFFICE_CME(config-dp-cor)#name STD_MEMBER
```

Step 2: Create Outgoing COR List

```
BANGLORE_OFFICE_CME(config)#dial-peer cor list LOCAL_LIST
BANGLORE_OFFICE_CME(config-dp-corlist)#member LOCAL_MEMBER

BANGLORE_OFFICE_CME(config)#dial-peer cor list STD_LIST
BANGLORE_OFFICE_CME(config-dp-corlist)#member STD_MEMBER
```

Step 3: Create Incoming COR List

```
BANGLORE_OFFICE_CME(config)#dial-peer cor list MANAGER_LIST
BANGLORE_OFFICE_CME(config-dp-corlist)#member LOCAL_MEMBER
BANGLORE_OFFICE_CME(config-dp-corlist)#member STD_MEMBER

BANGLORE_OFFICE_CME(config)#dial-peer cor list ASST_LIST
BANGLORE_OFFICE_CME(config-dp-corlist)#member LOCAL_MEMBER
```

Step 4: Apply Outgoing COR list under Outgoing Dial Peer

BANGLORE_OFFICE (config)#dial-peer voice 3 pots

BANGLORE_OFFICE (config-dial-peer)#corlist outgoing LOCAL_LIST

BANGLORE_OFFICE (config)#dial-peer voice 4 pots

BANGLORE_OFFICE r(config-dial-peer)#corlist outgoing STD_LIST

Step 5: Apply Incoming COR list under ePhone-dn

BANGLORE_OFFICE_CME(config)#ephone-dn 1

BANGLORE_OFFICE_CME(config-ephone-dn)#corlist incoming MANAGER_LIST

BANGLORE_OFFICE_CME(config)#ephone-dn 2

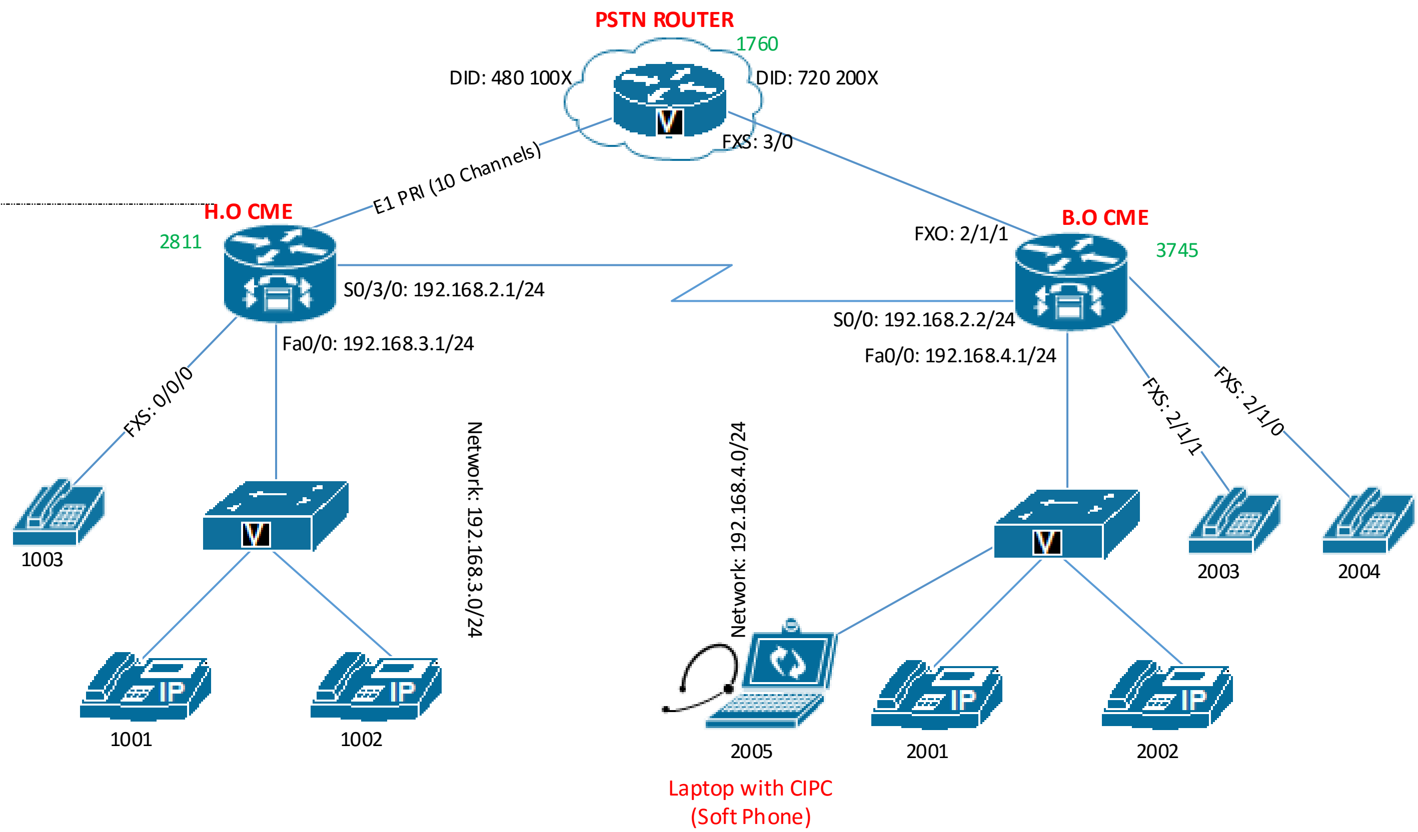
BANGLORE_OFFICE_CME(config-ephone-dn)#corlist incoming ASST_LIST

Notes:

If there is no outgoing COR list applied, the call is always routed

If there is no incoming COR list applied, the call is always routed

CVOICE Real Lab



Conditions

When WAN goes down, user dial only 200X, but the call should go via PSTN using proper digit manipulations

Lab Instructions

PSTN Router

1. Configure ISDN E1 PRI circuit in PSTN Router (Cisco 1760)
2. Configure required dial-peer in PSTN Router (Cisco 1760)

HO Router

3. Configure interfaces (FastEthernet0/0, Serial0/3/0) and EIGRP Routing
4. Configure two DHCP Pools (one for HO and other for BO)
5. Configure ISDN E1 PRI circuit to PSTN
6. Configure Dial-peers
7. Configure Digit Manipulation (VTR & num-exp)
8. Configure CME
9. Configure Call Forward No Answer from 1001, 1002 to 1003
10. Configure PAGING 1004 in 1001 & 1002
11. Configure INTERCOM between 1001 & 1002

12. Configure SNR & MOBILITY in 1001 to 1003

BO Router

13. Configure interfaces (FastEthernet0/0, Serial0/0) and EIGRP Routing
14. Configure ip helper-address to get DHCP
15. Configure Dial-peers
16. Configure Digit Manipulation (Prefix & num-exp)
17. Configure CME
18. Configure PLAR in voice port 2/1/0 connection to 2001
19. Configure COR in BO (2001 able to call 2003 & 2004; 2002 able to call only 2003 but not 2004)
20. Configure Laptop with CIPS and register in B.O CME

1. PSTN Router Configuration

Step 1: ISDN Configuration

```
PSTN(config)#isdn switch-type primary-net5
PSTN(config)#controller E1 1/0
PSTN(config-controller)# framing crc4
PSTN(config-controller)# linecode hdb3
PSTN(config-controller)# clock source internal
PSTN(config-controller)# pri-group timeslots 1-10,16
PSTN(config-controller)# no shutdown

PSTN(config)# interface Serial1/0:15
PSTN(config-if)# isdn switch-type primary-net5
PSTN(config-if)# isdn protocol-emulate network
PSTN(config-if)# isdn bchan-number-order ascending
PSTN(config-if)# isdn incoming-voice voice
PSTN(config-if)# no shutdown
```

Step 2: Dial-Peer Configurations

```
PSTN(config)# dial-peer voice 1 pots
PSTN(config-dial-peer)# incoming called-number .
PSTN(config-dial-peer)# direct-inward-dial
PSTN(config-dial-peer)# port 1/0:15
```

```
PSTN(config)# dial-peer voice 2 pots
PSTN(config-dial-peer)# destination-pattern 480100.
PSTN(config-dial-peer)# no digit-strip
PSTN(config-dial-peer)# port 1/0:15
```

```
PSTN(config)# dial-peer voice 4 pots
PSTN(config-dial-peer)# destination-pattern 720200.
PSTN(config-dial-peer)# no digit-strip
PSTN(config-dial-peer)# port 3/1
```

2. H.O CME Configurations

Step 1 Basic Configurations

```
HO(config)#interface FastEthernet0/0
HO(config-if)#ip address 192.168.3.1 255.255.255.0
HO(config-if)# no shutdown

HO(config)# interface Serial0/3/0
HO(config-if)# ip address 192.168.2.1 255.255.255.0
HO(config-if)# no shutdown

HO(config)# router eigrp 1
HO(config-router)# network 192.168.2.0
HO(config-router)# network 192.168.3.0
HO(config-router)# no auto-summary
```

Step 2: DHCP Pool Configuration

```
HO(dhcp)# ip dhcp pool CME1
HO(dhcp-config)# network 192.168.3.0 255.255.255.0
HO(dhcp-config)# default-router 192.168.3.1
HO(dhcp-config)# option 150 ip 192.168.3.1

HO(dhcp)# ip dhcp pool CME2
```

```
HO(dhcp-config)# network 192.168.4.0 255.255.255.0
HO(dhcp-config)# option 150 ip 192.168.4.1
HO(dhcp-config)# default-router 192.168.4.1
```

Step 3: ISDN Configuration

```
HO(dhcp)# isdn switch-type primary-net5

HO(config)#controller E1 0/2/0
HO(config-controller)# framing crc4
HO(config-controller)# linecode hdb3
HO(config-controller)# clock source line
HO(config-controller)#pri-group timeslots 1-10,16
HO(config-controller)# no shutdown

HO(config)# interface Serial0/2/0:15
HO(config-if)# isdn protocol-emulate user
HO(config-if)# isdn bchan-number-order descending
HO(config-if)#isdn switch-type primary-net5
HO(config-if)#isdn incoming-voice voice
HO(config-if)# no shutdown
```

Step 4: Dial-Peer Configurations

```
HO(config)# dial-peer voice 1 pots
HO(config-dial-peer)# description ISDN_DIAL_PEER_TO_ACCEPT_INCOMING_CALL
HO(config-dial-peer)# incoming called-number .
HO(config-dial-peer)# direct-inward-dial
HO(config-dial-peer)# port 0/2/0:15

HO(config)# dial-peer voice 2 pots
HO(config-dial-peer)# description FOR_1003_ANALOG_PHONE
HO(config-dial-peer)# destination-pattern 1003
HO(config-dial-peer)# no digit-strip
HO(config-dial-peer)# port 0/0/0
```

```
HO(config)# dial-peer voice 3 voip
HO(config-dial-peer)# description FOR_CALLING_BO_CME_OVER_WAN
HO(config-dial-peer)# destination-pattern 200.
HO(config-dial-peer)# session target ipv4:192.168.2.2

HO(config)# dial-peer voice 4 pots
HO(config-dial-peer)# description FOR_CALLING_BO_CME_OVER_PSTN
HO(config-dial-peer)# destination-pattern 200.
HO(config-dial-peer)# no digit-strip
HO(config-dial-peer)# port 0/2/0:15
```

Step 5: Digit Manipulation

```
HO(config)# voice translation-rule 1
HO(cfg-translation-rule)# rule 1 /200\(.\)/ /720200\1/
HO(cfg-translation-rule)# rule 2 /100\(.\)/ /480100\1/

HO(profile)# voice translation-profile PSTN
HO(cfg-translation-profile)# translate calling 1
HO(cfg-translation-profile)# translate called 1

HO(config)# num-exp 480100. 100.

HO(config)# dial-peer voice 4
HO(config-dial-peer)# translation-profile outgoing PSTN
```

Step 6: CME Configurations

```
HO(config)# telephony-service
HO(config-telephony)# max-ephones 20
HO(config-telephony)#max-dn 20
HO(config-telephony)#ip source-address 192.168.3.1 port 2000
HO(config-telephony)#create cnf-files
```

```
HO(config)#ephone-dn  1  dual-line
HO(config-ephone-dn)#number 1001
HO(config-ephone-dn)#call-forward noan 1003 timeout 10

HO(config)# ephone-dn  2  dual-line
HO(config-ephone-dn)# number 1002
HO(config-ephone-dn)#call-forward noan 1003 timeout 10

HO(config)#ephone  1
HO(config-ephone)#mac-address 001A.2F35.B1D8
HO(config-ephone)#button  1:1

HO(config)#ephone  2
HO(config-ephone)#mac-address 001E.4A0B.FE22
HO(config-ephone)#button  1:2
```

Step 7: Call Forward

```
HO(config)#ephone-dn  1
HO(config-ephone-dn)#call-forward noan 1003 timeout 10

HO(config)# ephone-dn  2
HO(config-ephone-dn)#call-forward noan 1003 timeout 10
```

Step 8: Paging Configuration

```
HO(config)# ephone-dn  5
HO(config)# descriptio PAGING_DN
HO(config-ephone-dn)# number 1004
HO(config-ephone-dn)# paging

HO(config)#ephone  1
HO(config-ephone)#paging-dn 5

HO(config)#ephone  2
```

```
HO(config-ephone)#paging-dn 5
```

Step 8: Intercom Configuration

```
HO(config)# ephone-dn  3
HO(config-ephone-dn)# number A1001
HO(config-ephone-dn)# description MANAGER_INTERCOM_NUMBER
HO(config-ephone-dn)# intercom A1002 label "ASSISTANT"

HO(config)# ephone-dn  4
HO(config-ephone-dn)# number A1002
HO(config-ephone-dn)# description ASSISTANT_INTERCOM_NUMBER
HO(config-ephone-dn)# intercom A1001 label "MANGER"

HO(config)#ephone  1
HO(config-ephone)#button 2:3
HO(config-ephone)#restart

HO(config)#ephone  2
HO(config-ephone)#button 2:4
HO(config-ephone)#restart
```

Step 9: SNR and Mobility

```
HO(config)# ephone-template  1
HO(config-ephone-template)# softkeys connected  Mobility Trnsfer Hold

HO(config)#ephone  1
HO(config-ephone)# ephone-template 1
HO(config-ephone)# restart

HO(config)#ephone-dn  1
HO(config-ephone-dn)#snr 1003 delay 5 timeout 10 cfwd-noan 1002
HO(config-ephone-dn)# mobility
```

BO Router Configurations

Step1: Basic Configurations

```
BO(config)#interface FastEthernet0/0
BO(config-if)#ip address 192.168.4.1 255.255.255.0
BO(config-if)#ip helper-address 192.168.2.1
BO(config-if)# no shutdown

BO(config)#interface Serial0/0
BO(config-if)#ip address 192.168.2.2 255.255.255.0
BO(config-if)#no fair-queue
BO(config-if)#clock rate 64000
BO(config-if)# no shutdown

BO(config)#router eigrp 1
BO(config-router)#network 192.168.2.0
BO(config-router)#network 192.168.4.0
BO(config-router)#no auto-summary
```

Step 2: ip helper-address Configuration

```
BO(config)#interface FastEthernet0/0
BO(config-if)#ip helper-address 192.168.2.1
```

Step 3: Dial-Peer Configurations

```
BO(config)#dial-peer voice 1 pots
BO(config-dial-peer)# description FOR_ANALOG_PHONE1
BO(config-dial-peer)#destination-pattern 2003
BO(config-dial-peer)#port 2/1/0

BO(config)#dial-peer voice 2 pots
BO(config-dial-peer)# description FOR_ANALOG_PHONE2
BO(config-dial-peer)#destination-pattern 2004
BO(config-dial-peer)#port 2/1/1
```

```
BO(config)#dial-peer voice 3 voip
BO(config-dial-peer)# description FOR_CALLING_HO_CME_OVER_WAN
BO(config-dial-peer)#destination-pattern 100.
BO(config-dial-peer)#session target ipv4:192.168.2.1

BO(config)#dial-peer voice 4 pots
BO(config-dial-peer)# description FOR_CALLING_HO_CME_OVER_PSTN
BO(config-dial-peer)#destination-pattern 100.
BO(config-dial-peer)#no digit-strip
BO(config-dial-peer)#port 2/0/0
```

Step 4: Digit Manipulations (Prefix & num-exp)

```
BO(config)#dial-peer voice 4
BO(config-dial-peer)#prefix 4804

BO(config)#num-exp 720200. 200.
```

Step 5: CME Configurations

```
BO(config)#telephony-service
BO(config-telephony)# max-ephones 20
BO(config-telephony)#max-dn 20
BO(config-telephony)#ip source-address 192.168.4.1 port 2000
BO(config-telephony)#system message B.O CME
BO(config-telephony)#create cnf-files

BO(config)#ephone-dn  1  dual-line
BO(config-ephone-dn)#number 2001

BO(config)#ephone-dn  2  dual-line
BO(config-ephone-dn)#number 2002

BO(config)#ephone  1
BO(config-ephone)#mac-address 0013.7F73.79E0
```

```
BO(config-ephone)#button  1:1

BO(config)#ephone  2
BO(config-ephone)#mac-address 001A.A1CF.142C
BO(config-ephone)#button  1:2
```

Step 6: Configure PLAR

```
BO(config)#voice-port 2/1/1
BO(config-voice-port)# connection plar 1003
```

Step 7: COR Configurations

```
BO(config)#dial-peer cor custom
BO(config-dp-cor)#name 2003-COR
BO(config-dp-cor)#name 2004-COR

BO(config)#dial-peer cor list 2003ONLY
BO(config-dp-corlist)#member 2003-COR

BO(config)#dial-peer cor list 2004ONLY
BO(config-dp-corlist)#member 2004-COR

BO(config)#dial-peer cor list 2003-2004
BO(config-dp-corlist)#member 2003-COR
BO(config-dp-corlist)#member 2004-COR

BO(config)#dial-peer cor list 2003
BO(config-dp-corlist)#member 2003-COR

BO(config)#dial-peer voice 1 pots
BO(config-dial-peer)#corlist outgoing 2003ONLY

BO(config)#dial-peer voice 2 pots
BO(config-dial-peer)#corlist outgoing 2004ONLY
```

```
BO(config)#ephone-dn  1  dual-line
BO(config-ephone-dn)#corlist incoming 2003-2004
```

```
BO(config)#ephone-dn  2  dual-line
BO(config-ephone-dn)#corlist incoming 2003
```

Step 8: CIPC Configurations

Install CIPC in laptop.

Run as Administrator → Settings → Preferences → Network →

TFTP Server 1: 192.168.4.1 → OK

Connect laptop to switch in branch office. Check the IP address obtained by the laptop. *(If it is not getting any IP, put manual IP 192.168.4.X/24)*

```
BO(config)#ephone-dn  3  dual-line
BO(config-ephone-dn)#number 2005
```

```
BO(config)#ephone  3
BO(config-ephone)#mac-address 1C65.9DAA.CB76
BO(config-ephone)#button  1:3
```

END