

CCVP (Voice)

Cisco CCVP Test 1

Introduction to Voice Technologies

The most up-to-date version of this test is at:

<http://networksims.com/v01.html>

Cisco Router Challenge 11

Outline

This challenge involves the configuration of the S0 port on a router.

Objectives

The objectives of this challenge are to:

- Setup the IP address on S0 port.
- Setup the subnet mask on S0 port.
- Enable the S0 port.
- Set the description for the S0 port.
- Define the speed of the S0 port.
- Define EIGRP routing.

Commands

```
> enable
# config t
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# no shutdown
(config-if)# description students
(config-if)# encapsulation ppp
(config-if)# ppp authentication chap
(config-if)# clock rate 56000
(config-if)# carrier-delay 8
(config-if)# bandwidth 198
(config-if)# exit
(config)# router eigrp 111
(config-network)# network 10.0.0.1
```

Example

```
> enable
# config t
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# no shutdown
(config-if)# description students
(config-if)# encapsulation ?
  atm-dxi      ATM-DXI encapsulation
  frame-relay  Frame Relay networks
  hdlc         Serial HDLC synchronous
  lapb        LAPB (X.25 Level 2)
  ppp         Point-to-Point protocol
  smds        Switched Megabit Data Service (SMDS)
  x25         X.25
(config-if)# encapsulation ppp
(config-if)# ppp ?
  accm        Set initial Async Control Character Map
  acfc        Options for HDLC Address & Control Field Compression
  authentication  Set PPP link authentication method
  bridge      Enable PPP bridge translation
  chap        Set CHAP authentication parameters
  ipcp        Set IPCP negotiation options
  lcp         PPP LCP configuration
  link        Set miscellaneous link parameters
  max-bad-auth  Allow multiple authentication failures
  multilink   Make interface multilink capable
  pap         Set PAP authentication parameters
  pfc         Options for Protocol Field Compression
  quality     Set minimum Link Quality before link is down
  reliable-link  Use LAPB with PPP to provide a reliable link
  timeout     Set PPP timeout parameters
  use-tacacs  Use TACACS to verify PPP authentications
(config-if)# ppp authentication?
  chap        Challenge Handshake Authentication Protocol (CHAP)
  ms-chap     Microsoft Challenge Handshake Authentication Protocol (MS-CHAP)
  pap         Password Authentication Protocol (PAP)
(config-if)# ppp authentication chap
(config-if)# clock ?
  rate        Configure serial interface clock speed

(config-if)# clock rate ?
  Speed (bits per second)
  1200
  2400
  4800
  9600
  14400
  19200
  28800
  32000
  38400
  56000
  57600
  64000
  72000
  115200
  125000
  128000
  148000
```

```
192000
250000
256000
384000
500000
512000
768000
800000
1000000
1300000
2000000
4000000
8000000
```

```
<300-4000000>    Choose clockrate from list above
(config-if)# clock rate 56000
(config-if)# carrier-delay 8
(config-if)# bandwidth 198
(config-if)# exit
(config)# router eigrp 111
(config-network)# network 10.0.0.1
# sh running
```

Cisco Router Challenge 12

Outline

This challenge involves the configuration of the S1 port on a router.

Objectives

The objectives of this challenge are to:

- Setup the IP address on S1 port.
- Setup encapsulation on the S1 port.
- Setup authentication on the S1 port.
- Define other S1 parameters.

Example

```
> enable
# config t
(config)# int s1
(config-if)# ip address 46.187.202.5 254.0.0.0
(config-if)# no shutdown
(config-if)# description academics
(config-if)# encapsulation ppp
(config-if)# ppp authentication pap
(config-if)# clock rate 56000
(config-if)# bandwidth 63
(config-if)# exit
(config)# router eigrp 111
(config-network)# network 10.0.0.1
# sh running
```

Cisco CCVP Test 2

Introduction to Voice Technologies

The most up-to-date version of this test is at:

<http://networksims.com/v02.html>

Cisco Router Challenge 211

Outline

This challenge involves the configuration of **FXS** (Foreign Exchange Station) Voice Port Configuration on a Voice-enabled router. Some routers have FXS interfaces which can connect to a standard telephone, fax machine, or similar device and thus must provide ringing, voltage supplies, and a dial tone. Normally the FXS interface uses an RJ-11 connector to connect to telephone equipment.

Objectives

The objectives of this challenge are to:

- Define FXS Voice Port details.
- Define FXS country details.

Outline

```
> enable
# sh version
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal groudstart
(config-voiceport)# cptone GB
(config-voiceport)# ring cadanece pattern01
(config-voiceport)# exit
(config)# exit
# show voice port
# show voice dsp
```

Example

```
> enable
# sh version
```

Cisco IOS Software, C2600 Software (C2600-ADVENTERPRISEK9-M), Version 12.4(12), RELEASE SOFTWARE (fc1)
Technical Support: <http://www.cisco.com/techsupport>
Copyright (c) 1986-2006 by Cisco Systems, Inc.
Compiled Fri 17-Nov-06 11:18 by prod_rel_team

ROM: System Bootstrap, Version 12.2(7r) [cmong 7r], RELEASE SOFTWARE (fc1)

Router uptime is 5 hours, 38 minutes
System returned to ROM by power-on
System image file is "flash:c2600-testk9-mz.124-12.bin"

Cisco 2611XM (MPC860P) processor (revision 1.0) with 111616K/19456K bytes of memory.
Processor board ID JAD07130QPE
M860 processor: part number 5, mask 2
2 FastEthernet interfaces
2 Serial(sync/async) interfaces
2 Voice FXO interfaces
2 Voice FXS interfaces
32K bytes of NVRAM.
49152K bytes of processor board System flash (Read/Write)

Configuration register is 0x3162

config t

(config)# voice-port ?

(config)# voice-port 1/0/0

(config-voiceport)# signal ?

groundStart Ground Start

loopStart Loop Start

(config-voiceport)# signal groudstart

(config-voiceport)# cptone ?

locale 2 letter ISO-3166 country code

AR Argentina	IS Iceland	PE Peru
AU Australia	IN India	PH Philippines
AT Austria	ID Indonesia	PL Poland
BE Belgium	IE Ireland	PT Portugal
BR Brazil	IL Israel	RU Russian Federation
CA Canada	IT Italy	SA Saudi Arabia
CN China	JP Japan	SG Singapore
CO Colombia	JO Jordan	SK Slovakia
C1 Custom1	KE Kenya	SI Slovenia
C2 Custom2	KR Korea Republic	ZA South Africa
CY Cyprus	LB Lebanon	ES Spain
CZ Czech Republic	LU Luxembourg	SE Sweden
DK Denmark	MY Malaysia	CH Switzerland
EG Egypt	MX Mexico	TW Taiwan
FI Finland	NP Nepal	TH Thailand
FR France	NL Netherlands	TR Turkey
DE Germany	NZ New Zealand	GB United Kingdom
GH Ghana	NG Nigeria	US United States
GR Greece	NO Norway	VE Venezuela
HK Hong Kong	PK Pakistan	ZW Zimbabwe
HU Hungary	PA Panama	

(config-voiceport)# cptone GB

(config-voiceport)# ring ?

cadence Ringing cadence on/off durations

frequency The ring frequency to be used in the FXS interface

(config-voiceport)# ring cadence ?

```

define      User Defined Cadence
pattern01  2sec on 4sec off
pattern02  1sec on 4sec off
pattern03  1.5sec on 3.5sec off
pattern04  1sec on 2sec off
pattern05  1sec on 5sec off
pattern06  1sec on 3sec off
pattern07  .8sec on 3.2sec off
pattern08  1.5sec on 3sec off
pattern09  1.2sec on 3.7sec off
pattern10  1.2sec on 4.7sec off
pattern11  .4sec on .2sec off .4sec on 2sec off
pattern12  .4sec on .2sec off .4sec on 2.6sec off

(config-voiceport)# ring cadence pattern01
(config-voiceport)# exit
(config)# exit
# sh voice port
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 3 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Echo Cancel worst case ERL is set to 6 dB
Playout-delay Mode is set to adaptive
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 250 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set

Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Supervisory Disconnect Time Out is set to 750 ms
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US

Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None
Translation profile (Incoming):
Translation profile (Outgoing):

Voice card specific Info Follows:
Signal Type is loopStart

```

Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is active
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Hookflash-in Timing is set to max=1000 ms, min=150 ms
Hookflash-out Timing is set to 400 ms
No disconnect acknowledge
Ring Cadence is defined by CPTone Selection
Ring Cadence are [20 40] * 100 msec
Ringer Equivalence Number is set to 1

Foreign Exchange Station 1/0/1 Slot is 1, Sub-unit is 0, Port is 1

Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 3 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Echo Cancel worst case ERL is set to 6 dB
Playout-delay Mode is set to adaptive
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 250 ms
Playout-delay Minimum mode is set to default, value 40 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Supervisory Disconnect Time Out is set to 750 ms
Ringing Time Out is set to 180 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US

Analog Info Follows:

Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm
Station name None, Station number None
Translation profile (Incoming):
Translation profile (Outgoing):

Voice card specific Info Follows:

Signal Type is loopStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive

```

Ring Ground Status is inactive
Tip Ground Status is active
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Hookflash-in Timing is set to max=1000 ms, min=150 ms
Hookflash-out Timing is set to 400 ms
No disconnect acknowledge
Ring Cadence is defined by CPTone Selection
Ring Cadence are [20 40] * 100 msec
Ringer Equivalence Number is set to 1
Playout-delay Minimum mode is set to default, value 40 ms

```

```
# sh voice dsp
```

```

DSP   DSP           DSPWARE CURR  BOOT           PAK   TX/RX
TYPE  NUM CH  CODEC      VERSION STATE  STATE      RST AI VOICEPORT TS  ABORT  PACK COUNT
=====
C542  001  01  None      4.4.21 IDLE  idle      0  0  1/0/0    NA   0      604/613
C542  002  01  None      4.4.21 IDLE  idle      0  0  1/0/1    NA   0      597/594

```

Cisco Router Challenge 212

Outline

This challenge involves the configuration of **FXO** (Foreign Exchange Office) Voice Port Configuration on a Voice-enabled router. FXS are edge devices whereas the FXO port connects to the PBX.

Objectives

The objectives of this challenge are to:

- Define FXO Voice Port details.

Outline

```

> enable
# sh version
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal loopstart
(config-voiceport)# ring number 3
(config-voiceport)# dial-type dtmf

```

Example

```

> enable
# sh version
# config t
(config)# voice-port 1/0/0
(config-voiceport)# ?
Voice-port configuration commands:
  battery-reversal      Enable FXS battery-reversal generation

```

bearer-cap	Specify the bear capability
busyout	Configure busyout trigger event & procedure
caller-id	Configure port caller id parameters
comfort-noise	Use fill-silence option
connection	Specify Trunking Parameters
cptone	Configure voice call progress tone locale
default	Set a command to its defaults
description	Description of what this port is connected to
disc_pi_off	close voice path when disconnect with PI received
disconnect-ack	FXS sending disconnect acknowledge
echo-cancel	Echo-cancellation option
exit	Exit from voice-port configuration mode
impedance	Specifies the terminating impedance of the interface
input	Configure input gain for voice
music-threshold	Threshold for Music on Hold
mwi	Enable MWI on this port
no	Negate a command or set its defaults
non-linear	Use non-linear processing during echo cancellation
output	Configure output attenuation for voice
playout-delay	Configure voice playout delay buffer
ren	Ringer Equivalence Number
ring	Ring frequency Parameters
shutdown	Take voice-port offline
signal	The signaling type for the interface FXS or FXO
snmp	Modify SNMP voice port parameters
station-id	Configure station ID
supervisory	Configure supervisory disconnect lcfo
threshold	Threshold [noise] for voice port
timeouts	Configure voice timeout parameters
timing	Configure voice timing parameters
translate	Translation rule
translation-profile	Translation profile
trunk-group	Configure interface to be in a trunk group
voice-class	Set voiceport voice class control parameters

(config-voiceport)# signal loopstart

(config-voiceport)# ?

Voice-port configuration commands:

battery-reversal	Enable FXO battery-reversal detection
bearer-cap	Specify the bear capability
busyout	Configure busyout trigger event & procedure
comfort-noise	Use fill-silence option
connection	Specify Trunking Parameters
cptone	Configure voice call progress tone locale
default	Set a command to its defaults
description	Description of what this port is connected to
dial-type	Configure type of dialer for voice
disc_pi_off	close voice path when disconnect with PI received
echo-cancel	Echo-cancellation option
exit	Exit from voice-port configuration mode
impedance	Specifies the terminating impedance of the interface
input	Configure input gain for voice
music-threshold	Threshold for Music on Hold
no	Negate a command or set its defaults
non-linear	Use non-linear processing during echo cancellation
output	Configure output attenuation for voice
playout-delay	Configure voice playout delay buffer
pre-dial-delay	FXO Pre-dial Delay
ring	Number of rings
shutdown	Take voice-port offline
signal	The signaling type for the interface FXS or FXO
snmp	Modify SNMP voice port parameters
station-id	Configure station ID
supervisory	Configure answer + disconnect supervision options

```

threshold          Threshold [noise] for voice port
timeouts           Configure voice timeout parameters
timing             Configure voice timing parameters
translate          Translation rule
translation-profile Translation profile
trunk-group        Configure interface to be in a trunk group
voice-class        Set voiceport voice class control parameters
(config-voiceport)# ring ?
    number          Number of rings for the FXO interface

(config-voiceport)# ring number ?
    <1-10>          The number of rings detected before closing loop

(config-voiceport)# ring number 3
(config-voiceport)# dial ?
    dtmf           touch-tone dialer
    mf             mf-tone dialer
    pulse          pulse dialer
(config-voiceport)# dial-type dtmf

```

This answers on the third ring and uses DTMF (touch-tone dialer) ring tone type.

Cisco Router Challenge 213

Outline

This challenge involves the configuration of FXS (Foreign Exchange Station) Voice Port Configuration for the main configuration timers, especially in situations where users require more time to dial numbers. The main timeouts are **initial timeout**, **interdigit timeout**, **ringing timeout** and the **hookflash-in timer**.

Objectives

The objectives of this challenge are to:

- Define FXS Voice Port details.
- Define FXS country details.
- Define voice port timer settings, especially for initial timeout, interdigit timeout, ringing timeout and the hookflash-in timer.

Outline

```

> enable
# sh version
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal groundstart
(config-voiceport)# cptone GB
(config-voiceport)# ring cadancece pattern01
(config-voiceport)# timeout call-disconnect 10
(config-voiceport)# timeout initial 15
(config-voiceport)# timeout interdigit 20
(config-voiceport)# timeout ringing 60

```

```
(config-voiceport)# timeout hookflash-in 500
```

Example

```
> enable
# sh version
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal groundstart
(config-voiceport)# cptone GB
(config-voiceport)# ring cadancece pattern01
(config-voiceport)# timeout ?
  call-disconnect  Call Disconnect Timeout after Destination Hangs Up in
                  seconds
  hookflash-in    Define hookflash-in delay in milliseconds
  initial         Initial Timeout duration in seconds
  interdigit      Interdigit Timeout duration in seconds
  power-denial    Duration for which power-denial is applied
  ringing         Ringing no answer timeout duration in seconds
  wait-release    Wait release timeout duration in seconds

(config-voiceport)# timeout call-disconnect ?
<0-120>  seconds
infinity infinite timeout

(config-voiceport)# timeout initial ?
<0-120>  seconds

(config-voiceport)# timeout interdigit ?
<0-120>  seconds

(config-voiceport)# timeout power-denial ?
<0-1500> milliseconds

(config-voiceport)# timeout ringing ?
<5-60000> seconds
infinity  infinite timeout

(config-voiceport)# timeout wait-release ?
<1-3600> seconds
infinity infinite timeout
(config-voiceport)# timeout call-disconnect 10
(config-voiceport)# timeout initial 15
(config-voiceport)# timeout interdigit 20
(config-voiceport)# timeout ringing 60
(config-voiceport)# timeout hookflash-in 500
```

This sets the ringing timeout to 60 seconds, which gives the user up to one minute to answer the call. It also increases the interdigit timeout to 20 seconds, which again gives users a maximum time between dialing digits of 20 seconds.

The hookflash is a brief interruption in the loop current when a trunk route starts, and is not taken as a call disconnect. It is caused by momentarily pressing down the cradle on a telephone. Also, some telephones reserve a button (such as 'recall') that sends a *timed loop break*.

Cisco Router Challenge 214

Outline

There are two main base TDM (Time Division Multiplexing) streams, these are E1 (mainly used in Europe) and T1 (mainly used in the USA). These streams give 2.048Mbps (for E1 with 32 channels) and 1.544Mbps (for T1 with 24 channels). This challenge involves defining the parameters for a T1 connection.

Objectives

The objectives of this challenge are to:

- Define T1 configuration details.
- Define the framing type.
- Define the clock source.
- Define the line code used.
- Define how many channels are used (DS0 group), and the signaling type (E&M Wink Start signaling).

Outline

```
> enable
# sh version
# config t
(config)# controller t1
(config-controller)# framing esf
(config-controller)# clock source line
(config-controller)# linecode b8zs
(config-controller)# ds0-group 1 timeslots 1-2 type e&m-wink-start
(config-controller)# pri-group timeslots 1-24
(config-controller)# no shutdown
(config-controller)# exit
(config)# isdn switch-type primary-qsig
```

Example

```
> enable
# sh version
# config t
(config)# controller t1
(config-controller)# ?
Controller configuration commands:
cablelength    Specify cable length for a DS1 link
cas-group      Configure the specified timeslots for CAS(Channel Associate Signals)
channel-group  Specify timeslots to channel-group mapping for an interface
clock          Specify the clock source for a DS1 link
default        Set a command to its defaults
description    Controller specific description
ds0            ds0 commands
exit           Exit from controller configuration mode
fdl            Specify the FDL standard for a DS1 data link
framing        Specify the type of Framing on a DS1 link
help           Description of the interactive help system
```

```

linecode          Specify line encoding method for a DS1 link
loopback          Put the entire T1 line into loopback
no                Negate a command or set its defaults
pri-group         Configure specified timeslots for PRI
shutdown         Shut down a DS1 link (send Blue Alarm)
(config-controller)# framing esf
(config-controller)# clock source line
(config-controller)# linecode b8zs
(config-controller)# ds0-group 1 timeslots 1-2 type e&m-wink-start
(config-controller)# no shutdown
(config-controller)# exit
(config)# isdn switch-type primary-qsig
(config)# exit
# sh controller t1
T1 is up.
  Applique type is Channelized T1
  Cablelength is long gain36 0db
  No alarms detected.
alarm-trigger is not set
Slot 4 CSU Serial #09480883 Model TEB HWVersion 6.00 RX level = 0DB
Configured clock mode swapped to Loop-timed by priority clocking!!
Framing is ESF, Line Code is B8ZS, Clock Source is Line.

```

Cisco Router Challenge 215

Outline

There are two main base TDM (Time Division Multiplexing) streams, these are E1 (mainly used in Europe) and T1 (mainly used in the USA). These streams give 2.048Mbps (for E1 with 32 channels) and 1.544Mbps (for T1 with 24 channels). This challenge involves defining the parameters for a E1 connection.

Objectives

The objectives of this challenge are to:

- Define E1 configuration details.
- Define the framing type.
- Define the clock source.
- Define the line code used.

Outline

```

> enable
# sh version
# config t
(config)# controller e1
(config-controller)# framing esf
(config-controller)# clock source line
(config-controller)# linecode crc4
(config-controller)# no shutdown

```

```
(config-controller)# exit
(config)# isdn switch-type primary-qsig
```

Example

```
> enable
# sh version
# config t
(config)# controller e1
(config-controller)# framing crc4
(config-controller)# clock source line
(config-controller)# linecode hdb3
(config-controller)# no shutdown
(config-controller)# exit
(config)# isdn switch-type primary-qsig
(config)# exit
# show controllers e1
E1 is up.
  Applique type is Channelized E1 - balanced
  No alarms detected.
  Version info of Slot 0:  HW: 2, Firmware: 4, PLD Rev: 2
Manufacture Cookie is not programmed.
Framing is CRC4, Line Code is HDB3, Clock Source is Line Primary.
Data in current interval (251 seconds elapsed):
  0 Line Code Violations, 0 Path Code Violations
  0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
  0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
Total Data (last 24 hours)
  0 Line Code Violations, 0 Path Code Violations,
  0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins,
  0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

Cisco Router Challenge 216

Outline

This challenge involves the configuration of **FXO** (Foreign Exchange Office) Voice Port Configuration on a Voice-enabled router, and fine tuning the voice port details, such as for the input gain and the output attenuation.

Objectives

The objectives of this challenge are to:

- Define FXO Voice Port details.
- Define the input gain for the voice.
- Define the input impedance.
- Define the output attenuation.

- Disable echo cancellation.

Outline

```
> enable
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal loopstart
(config-voiceport)# ring number 3
(config-voiceport)# dial-type dtmf
(config-voiceport)# input gain 1
(config-voiceport)# impedance 600r
(config-voiceport)# output attenuation 0
```

Example

```
> enable
# config t
(config)# voice-port 1/0/0
(config-voiceport)# signal loopstart
(config-voiceport)# ring number 3
(config-voiceport)# dial-type dtmf
(config-voiceport)# impedance ?
    600r  600 Ohms real
(config-voiceport)# impedance 600r

(config-voiceport)# input ?
    gain  Configure gain in db for voice input

Router(config-voiceport)# input gain ?
    <-6 - 14>  gain in db

(config-voiceport)# input gain 1
(config-voiceport)# output ?
    attenuation  Amount of attenuation inserted at transmit side
                  of the interface

(config-voiceport)# output attenuation ?
    <-6 - 14>  attenuation in db
```

```
(config-voiceport)# output attenuation 0
(config-voiceport)# no ?
```

Voice-port configuration commands:

battery-reversal	Enable FXO battery-reversal detection
bearer-cap	Specify the bear capability
busyout	Configure busyout trigger event & procedure
comfort-noise	Use fill-silence option
connection	Specify Trunking Parameters
cptone	Configure voice call progress tone locale
default	Set a command to its defaults
description	Description of what this port is connected to
dial-type	Configure type of dialer for voice
disc_pi_off	close voice path when disconnect with PI received
echo-cancel	Echo-cancellation option
exit	Exit from voice-port configuration mode
impedance	Specifies the terminating impedance of the interface
input	Configure input gain for voice
music-threshold	Threshold for Music on Hold
no	Negate a command or set its defaults
non-linear	Use non-linear processing during echo cancellation

output	Configure output attenuation for voice
playout-delay	Configure voice playout delay buffer
pre-dial-delay	FXO Pre-dial Delay
ring	Number of rings
shutdown	Take voice-port offline
signal	The signaling type for the interface FXS or FXO
snmp	Modify SNMP voice port parameters
station-id	Configure station ID
supervisory	Configure answer + disconnect supervision options
threshold	Threshold [noise] for voice port
timeout	Configure voice timeout parameters
timing	Configure voice timing parameters
translate	Translation rule
translation-profile	Translation profile
trunk-group	Configure interface to be in a trunk group
voice-class	Set voiceport voice class control parameters

```
(config-voiceport)# no echo-cancel ?
coverage    Echo Cancel Coverage
enable      Echo Cancel Enable
suppressor  Echo Suppressor
(config-voiceport)# no echo-cancel enable
```

Cisco CCVP Test 3

Voice Interface Configuration

The most up-to-date version of this test is at:

<http://networksims.com/v03.html>

Cisco Router Challenge 217

Outline

This challenge involves the configuration of a dial peer. When a call is initiated the router must decide where the call is to be routed to. A dial peer is an addressable end point, which is a destination pattern with explicit digits and wildcards to define a single telephone number or a range of ones. The main dial peers in VoIP are:

- **POTs dial peers.** This connects to a traditional phone network (POTs), such as for PBX or PSTN. The dial peers have a telephone number and a specific voice port for the edge device.
- **VoIP dial peers.** This connects to an IP network, and the dial peers have a destination address, and a next-hop router. Normally its destination is defined as the loopback address of the remote device.

Objectives

The objectives of this challenge are to:

- Define a POTS dial peer.
- Define a VoIP dial peer.

Outline

```
> enable
# config t
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.12
```

In the first example the phone is on a POTS network, thus when the phone dials Extension 11 it will get sent to Port 1/0/0 (where it should find the remote connection). In the second example, the phone connects to an IP network thus if it dials Extension 22, it will be directed to the IP address of 88.10.11.12.

Example

```
> enable
# config t
(config)# dial-peer ?
  cor          Class of Restriction
  hunt         Define the dial peer hunting choice
  outbound     Define the outbound options
  terminator   Define the address terminate character
  voice       Voice type
(config)# dial-peer voice ?
  <1-2147483647> Voice dial-peer tag

(config)# dial-peer voice 1 ?
  mmoip  Multi Media Over IP
  pots   Telephony
  voatm  Voice over ATM
  vofr   Voice over Frame Relay
  voip   Voice over IP
(config)# dial-p voice 1 pots
(config-dial-peer)# ?
DIALPEER configuration commands:
  answer-address      The Call Destination Number
  authentication      SIP Digest Authentication Configuration
  call-block          Incoming Call Blocking
  capacity            capacity update timer config
  carrier-id          Configure Carrier ID
  clid                Caller ID option
  corlist             set the Class of Restriction lists
  default             Set a command to its defaults
  description         Dialpeer specific description
  destination         Outbound dial-peer match config
  destination-pattern A full E.164 telephone number prefix
```

```

digit-strip          Use digit strip option for the POTS digits replacement
direct-inward-dial  Use Called Number as final call destination
dnis-map            The name of a configured dnis-map
exit                Exit from dial-peer configuration mode
fax                 Configure fax
forward-digits      Configure the destination digits forward of this
                    dialpeer
group-name           Configure parameter group
huntstop            Stop hunting on Dial-Peers
incoming            Incoming called number
information-type     Information type for dialpeer
max-conn            Sets the maximum connections per peer, negation sets
                    to unlimited
no                  Negate a command or set its defaults
numbering-type      The calling/called party numbering type
paramspace          Define parameter space
permission          set the call orig/term permission of this dialpeer
port                Voice port associated with this peer
preference           Configure the preference order of this dialpeer
prefix              The pattern to be dialed before the dialed num
register             Register the E.164 number of this dial peer with
                    gatekeeper
resource            Resource allocation policy
service             The selected service
session             The session [ target | protocol | transport ] for this
                    peer
shutdown            Change the Admin State of this peer to down (no->up)
supplementary-service Config supplementary service features
supported-language  Language(s) supported by the peer
tgrep               TGREP config
tone                Generate tones
translate-outgoing  Translation rule
translation-profile Translation profile
trunk-group-label   Configure Trunk Group Label
trunkgroup          trunk groups associated with this peer
voice               Configure GATEWAY dial-peer for voice services
voice-class         Set Dial-peer voice class control parameters
(config-dial-peer)# destination-pattern ?
    WORD A sequence of digits - representing the prefix or full telephone number
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port ?
    <1-1> Voice interface slot #
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-p voice 1 voip
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# session ?
    protocol The session protocol to be used in getting to this peer
    target    The session target for this peer
    transport The transport layer protocol used for this peer

(config-dial-peer)# session target ?
    WORD A string specifying the session target
(config-dial-peer)# session target ipv4:88.10.11.12

```

Cisco Router Challenge 218

Outline

This challenge involves the configuration of a dial peer for a default destination pattern, and to define a preference for the dial-peer.

Objectives

The objectives of this challenge are to:

- Define a POTs dial peer.
- Define a VoIP dial peer.
- Define a default VoIP dial peer.
- Define a preference for the dial-peer.

Outline

```
> enable
# config t
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 3 voip
(config-dial-peer)# destination-pattern .T
(config-dial-peer)# session target ipv4:88.10.11.12
(config-dial-peer)# preference 1
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.13
(config-dial-peer)# preference 2
```

The .T option matches at least one digit, and is typically used as a default destination pattern, where it will execute this one if none of the others match. With the preference command the device will pick the dial-peer with the highest preference.

Example

```
> enable
# config t
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 3 voip
(config-dial-peer)# destination-pattern .T
(config-dial-peer)# session target ipv4:88.10.11.12
(config-dial-peer)# prefe ?
<0-10> Preference order
(config-dial-peer)# preference 1
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.13
(config-dial-peer)# preference 2
```

Cisco Router Challenge 219

Outline

This challenge involves the configuration of a PLAR (Private Line, Automated Ringdown) connection. With PLAR, if a telephone goes off-hook, the router will select a predefined dial peer to setup a call to a PBX or a destination telephone. The user does not hear a dial tone, and there is an automated connection. Typically examples of this are in a hotel reception where a visitor might pick up the phone and be directed to the telephone in Reception.

Objectives

The objectives of this challenge are to:

- Define a POTS dial peer.
- Define a VoIP dial peer.
- Define a PLAR connection which will connect a telephone on a certain voice port to a destination phone, automatically.

Outline

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection plar 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.12
```

In this example when the telephone connected to voice port 1/0/0 is picked-up, this router (Remote Router) will automatically generate the digits for 22 for a dial peer lookup. It will then match these digits to the Dial-peer number 2, and send the call automatically to a destination of 88.10.11.12 (the loopback address of the Central Router), where the device there will send it to the correct voice port:

	Remote	Central	
Telephone --Voice1/0/0-	Router ----- Router		--- Telephone
(Ext. 11)	10.0.0.1	88.10.11.12	(Ext. 22)

Thus the user will connect automatically to a certain telephone.

Example

```

> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection ?
    plar      Private Line Auto Ringdown
    tie-line  A tie line
    trunk    A Straight Tie Line
(config-voiceport)# connection plar ?
    WORD  A string of digits including wild cards
    tied  dedicated tie to this number
(config-voiceport)# connection plar 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.12

```

Cisco Router Challenge 220

Outline

This challenge involves the configuration of a **trunk** route. A trunk route will remain permanent, even in the absence of any calls. The ports on either side will thus be permanently allocated to the trunk route. A trunk line is a little like a hot-line which is permanently connected, no matter what. There is no dialing involved, at all.

Objectives

The objectives of this challenge are to:

- Define a POTs dial peer.
- Define a VoIP dial peer.
- Define a trunk connection.
-

Outline

```

> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection trunk 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.12

```

In this example there will be a direct connection from the phone connected to the destination, which makes a connection back through dial-peer 1:

```

| Remote |           | Central   |
Telephone --Voice1/0/0-| Router |-----| Router   |--- Telephone
(Ext. 11)                |10.0.0.1|         | 88.10.11.12 |         (Ext. 22)
```

In this example, both dial-peers are required, one for the outbound connection (dial-peer 2), and the other to map the connection back to the same port (dial-peer 1).

In this case the dial-peer on the other side (Central Router) will be:

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection trunk 11
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# session target ipv4:10.0.0.1
```

Example

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection ?
  plar      Private Line Auto Ringdown
  tie-line  A tie line
  trunk     A Straight Tie Line
(config-voiceport)# connection trunk ?
  WORD     A string of digits including wild cards

(config-voiceport)# conn tr 22 ?
  answer-mode  Slave mode trunking
  retry-timer  timer value for retry connetion
  <cr>
(config-voiceport)# connection trunk 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22
(config-dial-peer)# session target ipv4:88.10.11.12
```

Cisco Router Challenge 221

Outline

This challenge involves the configuration of a **tie-line** route. A tie-line are often used to assign a decided circuit between two PBXs. With this there will be an IP network in-between the PBX connections, thus two remote sites with PBXs can be connected via a tie-line over an IP network.

Objectives

The objectives of this challenge are to:

- Define a POTs dial peer.
- Define a VoIP dial peer.
- Define a tie-line connection.
-

Outline

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection tie-line 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11..
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22..
(config-dial-peer)# session target ipv4:88.10.11.12
```

With tie-line there is a direct connection from the two telephone connections. Any phone from the 11.. extension, will be able to connect direct to a phone on the 22.. telephone system.

	Remote	Central	
Telephone --Voice1/0/0-	Router ----- Router	--- Telephone	
(Ext. 11..)	10.0.0.1	88.10.11.12	(Ext. 22..)

For example, if a user phones Ext 2211 from the 11... network, the call will be routed to the 22.. network, and the same goes for the 22.. network, where a call to the 11.. network will be routed to the 11.. telephone network. In this case the dial-peer on the other side (Central Router) will be:

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection tie-line 11
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 22..
```

```
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 11..
(config-dial-peer)# session target ipv4:10.0.0.1
```

Example

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# connection tie-line 22
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# destination-pattern 11..
(config-dial-peer)# port 1/0/0
(config-dial-peer)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 22..
(config-dial-peer)# session target ipv4:88.10.11.12
```

Cisco Router Challenge 222

Outline

This challenge involves the of a translation-rule which is a regular expression format of the dial pattern.

Objectives

The objectives of this challenge are to:

- Define a translation-rule.
- Apply translation-rule.

Outline

```
> enable
# config t
(config)# voice translation-rule 111
(cfg-translation-rule)# rule 1 /^666/ /444\1/
(cfg-translation-rule)# exit
(config)# dial-peer 10 pots
(config-dial-peer)# destination-pattern 99..
(config-dial-peer)# translate-outgoing called 111
(config-dial-peer)# forward-digits all
(config-dial-peer)# exit
(config)# voice translation-profile 111
```

Outline

```

> enable
# config t
(config)# voice ?
  call                Voice call related configuration.
  cause-code          Sets the internal Q850 cause code mapping
  class               Control parameters class
  disc-pi-incoming-on disconn with PI from incoming leg is maintained
  dnis-map            Create or add to a dnis-map
  dsp                 DSP functions
  enum-match-table    enum match table entry
  hpi                 Host port interface
  hunt                Dialpeer hunt conditions.
  iec                 Configure Internal Error Code behavior
  register            voice register commands
  rtp                 enable to open RTP in both directions.
  service             Global packet telephony service commands
  source-group        Source Group configuration commands
  statistics          Voice Statistics
  translation-profile Translation profile configuration commands
  translation-rule     Translation Rule configuration commands
  vad-time            Voice activity detection hangover period
(config)# voice translation-rule ?
  <1-2147483647> Translation rule tag
(config)# voice translation-rule 111
(cfg-translation-rule)# ?
Translation rule configuration commands:
  default Set a command to its defaults
  exit    Exit from Translation rule configuration mode
  help    Description of the interactive help system
  no      Negate a command or set its defaults
  rule    Translation rule

(cfg-translation-rule)# rule ?
  <1-15> Translation rule tag

(cfg-translation-rule)# rule 1 ?
  /WORD/ Matching pattern
  reject Call block rule

(cfg-translation-rule)# rule 1 /^666/ ?
  /WORD/ Replacement pattern

(cfg-translation-rule)# rule 1 /^666/ /444\1/ ?
  plan Match and replace the number plan
  type Match and replace the number type
  <cr>

(cfg-translation-rule)# rule 1 /^666/ /444\1/
(cfg-translation-rule)# exit
(config)# dial-peer voice 10 pots
(config-dial-peer)# ?
DIALPEER configuration commands:
  answer-address The Call Destination Number
  authentication SIP Digest Authentication Configuration
  call-block     Incoming Call Blocking
  capacity       capacity update timer config
  carrier-id     Configure Carrier ID
  clid           Caller ID option
  corlist        set the Class of Restriction lists
  default        Set a command to its defaults
  description    Dialpeer specific description
  destination    Outbound dial-peer match config
  destination-pattern A full E.164 telephone number prefix

```

```

digit-strip          Use digit strip option for the POTS digits replacement
direct-inward-dial  Use Called Number as final call destination
dnis-map            The name of a configured dnis-map
exit                Exit from dial-peer configuration mode
fax                Configure fax
forward-digits      Configure the destination digits forward of this
                    dialpeer
group-name          Configure parameter group
huntstop           Stop hunting on Dial-Peers
incoming           Incoming called number
information-type    Information type for dialpeer
max-conn           Sets the maximum connections per peer, negation sets
                    to unlimited
no                 Negate a command or set its defaults
numbering-type     The calling/called party numbering type
paramspace         Define parameter space
permission         set the call orig/term permission of this dialpeer
port              Voice port associated with this peer
preference         Configure the preference order of this dialpeer
prefix            The pattern to be dialed before the dialed num
register           Register the E.164 number of this dial peer with
                    gatekeeper
resource           Resource allocation policy
service           The selected service
session           The session [ target | protocol | transport ] for this
                    peer
shutdown           Change the Admin State of this peer to down (no->up)
supplementary-service Config supplementary service features
supported-language Language(s) supported by the peer
tgrep             TGREP config
tone              Generate tones
translate-outgoing Translation rule
translation-profile Translation profile
trunk-group-label Configure Trunk Group Label
trunkgroup        trunk groups associated with this peer
voice             Configure GATEWAY dial-peer for voice services
voice-class       Set Dial-peer voice class control parameters

(config-dial-peer)# destination-pattern 99..
(config-dial-peer)# translate-outgoing ?
    called    called party number will required translate
    calling   calling party number will required translate

(config-dial-peer)# translate-outgoing ?
    incoming  Translation Profile for incoming call leg
    outgoing  Translation Profile for outgoing call leg

(config-dial-peer)# translate-outgoing called 111

(config-dial-peer)# forward-digits ?
    <0-32>    number of right-justified dialed digits to be forwarded
    all       forward all destination digits
    extra     extra dialed digits to be forwarded
(config-dial-peer)# forward-digits all
(config-dial-peer)# exit
(config)# voice translation-profile ?
    WORD      Translation profile name
(config)# voice translation-profile 111
(cfg-translation-profile)# ?
Translation Profile configuration commands:
    default   Set a command to its defaults
    exit      Exit from translation profile configuration mode
    help      Description of the interactive help system
    no        Negate a command or set its defaults

```

translate Specify numbers that should be translated

Cisco CCVP Test 4

Voice Dial Peer Configuration

The most up-to-date version of this test is at:

<http://networksims.com/v04.html>

Cisco Router Challenge 162

Outline

This challenge involves compression the RTP header for a serial interface.

> CCNP ONT Area: Unit 5: Congestion Avoidance, Policing, Shaping and Link Efficiency Mechanisms

Objectives

The objectives of this challenge are to:

- Define RTP header compression.

Example

```
> en
# config t
(config)# int e0
(config-if)# ip ?
Interface IP configuration subcommands:
  access-group      Specify access control for packets
  accounting        Enable IP accounting on this interface
  address          Set the IP address of an interface
  audit            Apply IDS audit name
  auth-proxy       Apply authentication proxy
  authentication    authentication subcommands
  bandwidth-percent Set EIGRP bandwidth limit
  broadcast-address Set the broadcast address of an interface
  cef              Cisco Express Forwarding interface commands
  cgmp             Enable/disable CGMP
  dhcp             Configure DHCP parameters for this interface
  directed-broadcast Enable forwarding of directed broadcasts
  dvmrp           DVMRP interface commands
  flow            NetFlow related commands
  header-compression IPHC options
```

```

hello-interval      Configures IP-EIGRP hello interval
helper-address      Specify a destination address for UDP broadcasts
hold-time           Configures IP-EIGRP hold time
idle-group          Specify interesting packets for idle-timer
igmp                IGMP interface commands
information-reply    Enable sending ICMP Information Reply messages
inspect             Apply inspect name
irdp                ICMP Router Discovery Protocol
load-sharing        Style of load sharing
local-proxy-arp     Enable local-proxy ARP
mask-reply          Enable sending ICMP Mask Reply messages
mobile             Mobile IP support
mrm                Configure IP Multicast Routing Monitor tester
mrout-cache        Enable switching cache for incoming multicast packets
mtu                Set IP Maximum Transmission Unit
multicast           IP multicast interface commands
nat                NAT interface commands
nbar               Network-Based Application Recognition
next-hop-self      Configures IP-EIGRP next-hop-self
nhrp               NHRP interface subcommands
ospf               OSPF interface commands
pgm                PGM Reliable Transport Protocol
pim                PIM interface commands
policy             Enable policy routing
proxy-arp          Enable proxy ARP
rarp-server        Enable RARP server for static arp entries
redirects          Enable sending ICMP Redirect messages
rgmp               Enable/disable RGMP
rip                Router Information Protocol
route-cache        Enable fast-switching cache for outgoing packets
router             IP router interface commands
rsvp               RSVP Interface Commands
rtp                RTP parameters
sap                Session Announcement Protocol interface commands
security           DDN IP Security Option
split-horizon      Perform split horizon
summary-address    Perform address summarization
tcp                TCP header compression and other parameters
unnumbered         Enable IP processing without an explicit address
unreachables       Enable sending ICMP Unreachable messages
urd                Configure URL Rendezvousing
verify             Enable per packet validation
vrf                VPN Routing/Forwarding parameters on the interface
wccp               WCCP interface commands

(config-if)# ip rtp ?
  compression-connections  Maximum number of compressed connections
  header-compression        Enable RTP header compression
  priority                  Assign a priority queue for RTP streams
  reserve                   Assign a reserved queue for RTP streams

(config-if)# ip rtp header-compression
(config-if)# encapsulation ppp
(config-if)# ip rtp compression-connections ?
  <3-1000>  Number of connections
(config-if)# ip rtp compression-connections 20

```

Cisco Router Challenge 163

Outline

This challenge involves compression the RTP header for a frame relay connection.

> CCNP ONT Area: Unit 5: Congestion Avoidance, Policing, Shaping and Link Efficiency Mechanisms

Objectives

The objectives of this challenge are to:

- Define RTP header compression for a frame-relay connection.

Example

```
> en
# config t
(config)# int s0
(config-if)# encapsulate ?
  atm-dxi      ATM-DXI encapsulation
  frame-relay  Frame Relay networks
  hdlc         Serial HDLC synchronous
  lapb        LAPB (X.25 Level 2)
  ppp         Point-to-Point protocol
  smds        Switched Megabit Data Service (SMDS)
  x25         X.25
(config-if)# encapsulate frame-relay

(config-if)# clock ?
  rate  Configure serial interface clock speed

(config-if)# clock rate ?
  Speed (bits per second)
  1200
  2400
  4800
  9600
  14400
  19200
  28800
  32000
  38400
  56000
  57600
  64000
  72000
  115200
  125000
  128000
  148000
  192000
  250000
  256000
  384000
  500000
  512000
  768000
  800000
  1000000
```

1300000
2000000
4000000
8000000

<300-4000000> Choose clockrate from list above

(config-if)# clock rate 1200

(config-if)# frame-relay ?

accounting	Special accounting instruction
address-reg	ELMI address registration
broadcast-queue	Define a broadcast queue and transmit rate
class	Define a map class on the interface
congestion-management	Enable Frame Relay congestion management
de-group	Associate a DE group with a DLCI
fragment	Enable end-to-end fragmentation for all PVCs
fragmentation	Adaptive fragmentation
ifmib-counter64	Support IF-MIB's total packet/byte counts of Counter64 on FR if/subif when main interface's ifSpeed < 20 Mbps
interface-dlci	Define a DLCI on an interface/subinterface
interface-queue	configure PVC interface queueing
intf-type	Configure a FR DTE/DCE/NNI interface
inverse-arp	Enable/disable FR inverse ARP
ip	Frame Relay Internet Protocol config commands
lmi-n391dte	set full status polling counter
lmi-n392dce	LMI error threshold
lmi-n392dte	LMI error threshold
lmi-n393dce	set LMI monitored event count
lmi-n393dte	set LMI monitored event count
lmi-t392dce	set DCE polling verification timer
lmi-type	Use CISCO-ANSI-CCITT type LMI
local-dlci	Set source DLCI when LMI is not supported
map	Map a protocol address to a DLCI address
multicast-dlci	Set DLCI of a multicast group
policing	Enable Frame Relay policing
priority-dlci-group	Define a priority group of DLCIs
qos-autosense	enable QOS autosense
route	frame relay route for pvc switching
traffic-shaping	Enable Frame Relay Traffic Shaping
traps-maximum	set max traps FR generates at link up or when getting LMI Full Status message

(config-if)# frame-relay map ?

bridge	Bridging
bstun	Block Serial Tunnel
dls	Data Link Switching (Direct encapsulation only)
ip	IP
ipv6	IPV6
llc2	llc2
pppoe	PPP over Ethernet
qlc	qlc protocol
rsrb	Remote Source-Route Bridging
stun	Serial Tunnel

(config-if)# frame-relay map ip ?

A.B.C.D Protocol specific address

(config-if)# frame-relay map ip 1.2.3.4 ?

<16-1007> DLCI

(config-if)# frame-relay map ip 1.2.3.4 111 ?

broadcast	Broadcasts should be forwarded to this address
cisco	Use CISCO Encapsulation
compress	Enable TCP/IP and RTP/IP header compression

```

ietf                Use RFC1490/RFC2427 Encapsulation
nocompress          Do not compress TCP/IP headers
payload-compression Use payload compression
rtp                 RTP header compression parameters
tcp                 TCP header compression parameters
<cr>

(config-if)# frame-relay map ip 1.2.3.4 111 broadcast ?
cisco               Use CISCO Encapsulation
compress            Enable TCP/IP and RTP/IP header compression
ietf                Use RFC1490/RFC2427 Encapsulation
nocompress          Do not compress TCP/IP headers
payload-compression Use payload compression
rtp                 RTP header compression parameters
tcp                 TCP header compression parameters
<cr>

(config-if)# frame-relay map ip 1.2.3.4 111 broadcast rtp ?
header-compression Enable RTP/IP compression

(config-if)# frame-relay map ip 1.2.3.4 111 broadcast rtp header-compression ?
active              Always compress RTP headers
connections         Maximum number of compressed RTP connections
passive             Compress for destinations sending compressed RTP headers
<cr>
(config-if)# frame-relay map ip 1.2.3.4 111 b r header-compression

```

Cisco CCVP Test 5

VoIP Fundamentals

The most up-to-date version of this test is at:

<http://networksims.com/v05.html>

Cisco Router Challenge 223

Outline

This challenge involves the configuration of an H.323 gateway.

Objectives

The objectives of this challenge are to:

- Define H.323

Outline

```

> enable
# config t
(config)# int e0
(config-if)# ip address 1.2.3.4 255.255.0.0
(config-if)# h323-gateway voip interface
(config-if)# h323-gateway voip h323-id gw_1
(config-if)# h323-gateway voip id gk.testing.com ipaddr 1.2.3.5 1718
(config-if)# h323-gateway voip bind srcaddr 1.2.3.4
(config-if)# h323-gateway voip tech-prefix 1#
(config-if)# exit
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 1166..
(config-dial-peer)# session target ras
(config-dial-peer)# exit
(config)# dial-peer voice 3 pots
(config-dial-peer)# destination-pattern 911
(config-dial-peer)# port 1/0/0
(config-dial-peer)# no register e164
(config-dial-peer)# exit
(config)# gateway

```

Where gk.test.com is the Gatekeeper name.
1.2.3.5 is the IP address to bind with, and 1718 is the port.

Example

```

> enable
# config t
(config)# int e0
(config-if)# ip address 1.2.3.4 255.255.0.0
(config-if)# h323-gateway ?
    voip    Configure H323 Gateway Voip Interface

(config-if)# h323-gateway voip ?
    bind          Configure Bind IP Address
    h323-id       Specify an H.323 ID for this interface
    id            Gatekeeper identifier
    interface     Configure H323 Gateway Voip Interface
    tech-prefix  Specify a technology prefix
(config-if)# h323-gateway voip interface
(config-if)# h323-gateway voip h323-id ?
    WORD         Specify the h323 id
(config-if)# h323-gateway voip h323-id gw_1
(config-if)# h323-gateway voip id ?
    WORD         An ASCII string up to 128 bytes

(config-if)# h323-gateway voip id gk.testing.com ?
    ipaddr       IP address of the gatekeeper this gateway wants to register with
    multicast    Use multicast discovery to register the gateway with a gatekeeper

(config-if)# h323-gateway voip id gk.testing.com ipaddr ?
    A.B.C.D     An IP address

(config-if)# h323-gateway voip id gk.testing.com ipaddr 1.2.3.4 ?
    <1-65535>   Port number
    <cr>

(config-if)# h323-gateway voip id gk.testing.com ipaddr 1.2.3.5 1718
(config-if)# h323- v b ?
    srcaddr     IP address of this interface that will be used as source addr

```

```

(config-if)# h323-gateway voip bind srcaddr ?
    A.B.C.D An IP address
(config-if)# h323-gateway voip bind srcaddr 1.2.3.4
(config-if)# h323-gateway voip tech-prefix ?
    WORD A technology prefix that the interface will register with the
        gatekeeper
(config-if)# h323-gateway voip tech-prefix #1
(config)# dial-peer voice 2 voip
(config-dial-peer)# destination-pattern 1166..
(config-dial-peer)# session target ras
(config-dial-peer)# exit
(config)# dial-peer voice 3 pots
(config-dial-peer)# destination-pattern 911
(config-dial-peer)# port 1/0/0
(config-dial-peer)# no register e164
(config-dial-peer)# exit
(config)# gateway

```

The main commands:

- **h323-gateway voip interface.** This enables the router interface for H.323 processing.
- **h323-gateway voip h323-id gw_1.** This defines the H323 ID for the router.
- **h323-gateway voip id gk.testing.com ipaddr 1.2.3.5 1718.** This defines the ID of the gatekeeper for its IP address and TCP port number.

And the optional ones are:

- **h323- voip tech-prefix 10#.** This registers a technology prefix which tells the gateway that this gateway can handle 1# destinations (see explanation below)
- **h323- voip bind srcaddr 1.2.3.4.** This defines the source address for H.323 packets (1.2.3.4).

With **no register e164**, the router, when communicating with the gateway, does not register the destination pattern and thus defines that it must use an alternative method for gaining it.

For a technology-prefix, the administrator defines different classes of gateway, such as:

- 1# - voice gateway.
- 2# - voicemail gateway.
- 3# H.320 gateway.
- And so on.

The tech-pref is then added to the number that is required, so that it reaches the right gateway. For example a caller might use 1#1112222 for a telephone at 1112222 for a voice gateway. On receiving this, the voice gateway will strip-off the tech-prefix, and sends it to the telephone at 1112222.

Cisco Router Challenge 224

Outline

This challenge involves the configuration of SIP

Objectives

The objectives of this challenge are to:

- Enable SIP, and optional parameters.
- Define a dial-peer for SIP Version 2.

Outline

```
> enable
# config t
(config)# sip-ua
(config-sip-ua)# retry invite 10
(config-sip-ua)# retry response 10
(config-sip-ua)# retry cancel 10
(config-sip-ua)# retry bye 10
(config-sip-ua)# sip-server dns:test
(config-sip-ua)# exit
(config)# dial-peer voice 66 voip
(config-dial-peer)# destination-pattern 111
(config-dial-peer)# session protocol sipv2
(config-dial-peer)# session target ipv4:1.2.3.4
(config-dial-peer)# exit
(config)# dial-peer voice 66 voip
(config-dial-peer)# destination-pattern 111
(config-dial-peer)# session protocol sipv2
(config-dial-peer)# session target sip-server
(config-dial-peer)# exit
(config)# exit
# sh sip-ua status
# sh sip-ua timers
```

Example

```
> enable
# config t
(config)# sip-ua
(config-sip-ua)# ?
SIP UA configuration commands:
  aaa                sip-ua AAA related configuration
  authentication     Digest Authentication Configuration
  calling-info       Specify treatment of calling information
  default            Set a command to its defaults
  disable-early-media Disable early-media cut through
  exit              Exit from sip-ua configuration mode
  max-forwards       Change number of max-forwards for SIP Methods
  mwi-server         Configure a mwi Server
```

```

nat          Enable NAT(Network Address Traversal) settings for the
             SIP User Agent
no          Negate a command or set its defaults
notify      SIP Signaling Notify Configuration
offer       Configure settings for Offers made from the Gateway
reason-header  Configure settings for supporting SIP Reason Header
redirection Enable call redirection (3xx) handling
registrar   Configure SIP registrar VoIP Interface
remote-party-id  Enable Remote-Party-ID support in SIP User Agent
retry       Change default retries for each SIP Method
set         Sets the PSTN cause to SIP status code (and vice versa)
             and sets the PSTN cause to SIP requests

sip-server  Configure a SIP Server Interface
srv         DNS SRV Query Type
suspend-resume  Enable support for ISDN SUSPEND/RESUME
timers      SIP Signaling Timers Configuration
transport   Enable SIP UA transport for TCP/UDP

(config-sip-ua)# retry ?
bye         BYE retry value
cancel      CANCEL retry value
comet       COMET retry value
info        INFO retry value
invite      INVITE retry value
notify      NOTIFY retry value
prack       PRACK retry value
refer       REFER retry value
register    REGISTER retry value
(config-sip-ua)# retry invite 10
(config-sip-ua)# retry response 10
(config-sip-ua)# retry cancel 10
(config-sip-ua)# retry bye 10
(config-sip-ua)# sip-server dns:test
(config-sip-ua)# exit
(config)# dial-peer voice 66 voip
(config-dial-peer)# destination-pattern 111
(config-dial-peer)# session ?
protocol    The session protocol to be used in getting to this peer
target      The session target for this peer
transport    The transport layer protocol used for this peer

(config-dial-peer)# session protocol ?
cisco       Cisco Session Protocol
multicast   Multicast Session Protocol(voice conferencing)
sipv2       IETF Session Inititation Protocol

(config-dial-peer)# session protocol sipv2
(config-dial-peer)# session target ?
WORD        A string specifying the session target
(config-dial-peer)# session target ipv4:1.2.3.4
(config-dial-peer)# exit
(config)# dial-peer voice 66 voip
(config-dial-peer)# destination-pattern 111
(config-dial-peer)# session protocol sipv2
(config-dial-peer)# session target sip-server
(config-dial-peer)# exit
(config)# exit
# sh sip-ua timers
SIP UA Timer Values (millisecs unless noted)
trying 10000, expires 180000, connect 10000, disconnect 500
comet 500, prack 500, rellxx 500, notify 500
refer 500, register 500, info 500, hold 2880 minutes, aging 5 minutes
# sh sip-ua status
SIP User Agent Status

```

```
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
```

```
SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1
```

Cisco Router Challenge 225

Outline

This challenge involves the configuration of an MGCP Residential Gateway.

Objectives

The objectives of this challenge are to:

- Enable Call Manager Application MGCP.
- Define the MGCP call-agent.
- Enable MGCP.

Outline

```
> enable
# config t
(config)# ccm-manager mgcp
(config)# mgcp
(config)# mgcp call-agent 192.168.0.1
(config)# voice 1/0/0
(config-voiceport)# exit
(config)# voice 1/0/1
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# application mgcpapp
(config-dial-peer)# voice 1/0/0
(config)# dial-peer voice 2 pots
(config-dial-peer)# application mgcpapp
(config-dial-peer)# voice 1/0/1
```

```

(config-dial-peer)# exit
(config)# exit
# show mgcp statistics
# show call application voice summary
# show mgcp
# show call active voice brief
# show call history voice

```

Example

```

> enable
# config t
(config)# ccm ?
  application          application specific
  config               MGCP download configuration
  download-tones      Enable Tone Download from TFTP server
  fallback-mgcp       Enable Fallback from MGCP to H.323 mode if no Call
                    Manager is available
  fax                  Enable fax protocol for MGCP
  mgcp                 Enable Call Manager Application MGCP mode
  music-on-hold       Enable multicast Music-on-hold
  redundant-host      Redundant host list
  sccp                 Enable Call Manager Application SCCP mode
  shut-backhaul-interfaces Shutdown the backhauled interfaces if no Call
                    Manager is available
  switchback          Configure switchback options for rehomeing to
                    higher-order Call Manager

(config)# ccm-manager mgcp
(config)# mgcp ?
  <1025-65535>         Enable MGCP with user specified UDP port number
  behavior            Set MGCP message behavior
  bind                MGCP bind command
  block-newcalls      Take down active connections in an orderly way
  call-agent          Specify address of call-agent
  codec               The codec rate to be attempted for MGCP controlled
                    connections
  default-package     Select the Default Package Capability to be supported by
                    MGCP
  dtmf-relay          configure mgcp dtmf-relay
  endpoint            Configure endpoint handling
  explicit            MGCP Level to disable/enable explicit detections
  fax                 Configure MGCP Fax Parameters
  ip                  Configure IP parameters for MGCP-controlled connections
  max-waiting-delay   Specify Maximum Waiting Delay(MWD), prevents restart
                    avalanches
  modem               Configure MGCP Modem Parameters
  package-capability  Select the Package Capabilities to be supported by MGCP
  persistent          Configure persistents events handling
  piggyback           Configure piggyback message
  layout              The jitter buffer packet size attempted for MGCP
                    controlled connections
  profile              MGCP profile configuration mode
  quality-threshold   Specify voice quality related threshold values
  quarantine          Configuration for event quarantine buffer handling
  request             Configuration for MGCP requests sent by this gateway
  restart-delay       Specify the Restart Delay timer value
  rtp                 configuration for MGCP rtp timer
  rtrcac              Enable rtr-based VoIP CAC for MGCP
  sched-time          Specify the Scheduler timer value
  sdp                 Specify SDP operation for MGCP
  sgcp                Configuration for SGCP running in MGCP stack
  src-cac             Enable system resource check CAC for MGCP

```

```

timer                configure MGCP timers
vad                  Enable VoiceActivityDetection(Silence Suppression) for
                    MGCP
validate            Validation of MGCP messaging
voice-quality-stats Enable Voice Quality related stats reporting for MGCP
<cr>

(config)# mgcp
(config)# mgcp call-agent ?
WORD Hostname or IP address of the call-agent
(config)# mgcp call-agent 192.168.0.1
(config)# voice 1/0/0
(config-voiceport)# exit
(config)# voice 1/0/1
(config-voiceport)# exit
(config)# dial-peer voice 1 pots
(config-dial-peer)# ap ?
WORD Application name (Use show call application voice summary for list)

(config-dial-peer)# application mgcpapp
(config-dial-peer)# voice 1/0/0
(config)# dial-peer voice 2 pots
(config-dial-peer)# application mgcpapp
(config-dial-peer)# voice 1/0/1

# show mgcp statistics
UDP pkts rx 0, tx 0
Unrecognized rx pkts 0, MGCP message parsing errors 0
Duplicate MGCP ack tx 0, Invalid versions count 0
CreateConn rx 0, successful 0, failed 0
DeleteConn rx 0, successful 0, failed 0
ModifyConn rx 0, successful 0, failed 0
DeleteConn tx 0, successful 0, failed 0
NotifyRequest rx 0, successful 0, failed 0
AuditConnection rx 0, successful 0, failed 0
AuditEndpoint rx 0, successful 0, failed 0
RestartInProgress tx 0, successful 0, failed 0
Notify tx 0, successful 0, failed 0
ACK tx 0, NACK tx 0
ACK rx 0, NACK rx 0

IP address based Call Agents statistics:
No Call Agent message.

System resource check is DISABLED. No available statistic

DSO Resource Statistics
-----
Utilization: 0.00 percent
Total channels: 0
Addressable channels: 0
Inuse channels: 0
Disabled channels: 0
Free channels: 0
# show call application voice summary
SERVICES (standalone applications):
name                type                description
ipsla-responder     Tcl Script          builtin:app_test_rcvr_script.tcl
clid_authen         Tcl Script          builtin:app_clid_authen_script.tcl
clid_col_npw_npw    Tcl Script          builtin:app_clid_col_npw_npw_script.tcl
DEFAULT             C Script            builtin:Session_Service.C
CTAPP               C Script            builtin:CallTreatment_Service.C
fax_hop_on          Tcl Script          builtin:app_fax_hop_on_script.tcl

```

ipsla-testcall	Tcl Script	builtin:app_test_place_script.tcl
clid_authen_npw	Tcl Script	builtin:app_clid_authen_npw_script.tcl
session	Tcl Script	builtin:app_session_script.tcl
clid_col_npw_3	Tcl Script	builtin:app_clid_col_npw_3_script.tcl
lib_off_app	CCAPI	Libretto Offramp
stcapp	CCAPI	SCCP Call Control Application
MGCPAPP	CCAPI	MGCP Application

show mgcp

```

MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP block-newcalls DISABLED
MGCP validate domain name DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
MGCP quarantine mode discard/step
MGCP quarantine of persistent events is ENABLED
MGCP dtmf-relay for VoIP disabled for all codec types
MGCP dtmf-relay for VoAAL2 disabled for all codec types
MGCP voip modem passthrough disabled
MGCP voaal2 modem passthrough disabled
MGCP voip modem relay: Disabled.
MGCP TSE payload: 100
MGCP T.38 Named Signalling Event (NSE) response timer: 200
MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer: 5
MGCP request timeout 500
MGCP maximum exponential request timeout 4000
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp DISABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 40, 200 in msec
MGCP Fax Playout Buffer is 300 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: line-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
                        hs-package atm-package ms-package dt-package res-package
                        mt-package fxr-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Fax is ENABLED
MGCP T.38 Fax ECM is ENABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0MGCP T.38 Fax High Speed Redundancy: 0
MGCP control bind :DISABLED
MGCP media bind :DISABLED
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Dynamic payload type for G.726-16K codec
MGCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec
MGCP Guaranteed scheduler time is disabled

```

show call active voice brief

```
<ID>:<start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
  IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%>
<multipkt>/<corrected>
  last <buf event time>s dur:<Min>/<Max>s
  FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
  Tele <int>:tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l>
dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent>
total:<rcvd>/<sent>/<drops>
  Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt:
<type>/<manf>
  bw:<req>/<act> codec:<audio>/<video>
  tx:<audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120
pkts>/<t120 bytes>
  rx:<audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120
pkts>/<t120 bytes>
Total call-legs:2
1269 :7587246hs.1 +260 pid:0 Answer active
  dur 00:07:14 tx:590/11550 rx:21721/434420
IP 172.29.248.111:17394 rtt:3ms pl:431850/0ms lost:0/0/0 dela
y:69/69/70ms g729r8
1269 :7587246hs.2 +259 pid:133001 Originate 133001 active
  dur 00:07:14 tx:21717/434340 rx:590/11550
  Tele 1/0:1 (2):tx:434350/11640/0ms g729r8 noise:-44 acom:-19
i/0:-45/-45 dBm
```

show call history voice

```
GENERIC:
SetupTime=104648 ms
Index=1
PeerAddress=55240
PeerSubAddress=
PeerId=2
PeerIfIndex=105
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconectTime=143329
CallDuration=00:06:23
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=37668
TransmitBytes=6157536
ReceivePackets=37717
ReceiveBytes=6158452

VOIP:
ConnectionId[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
RemoteIPAddress=1.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
```

```
FastConnect=TRUE
SessionProtocol=cisco
SessionTarget=ipv4:1.14.82.14
OnTimeRvPayout=40
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPayoutDelay=67 ms
LoWaterPayoutDelay=67 ms
ReceiveDelay=67 ms
LostPackets=0 ms
EarlyPackets=0 ms
LatePackets=0 ms
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas
Modem passthrough signaling method is nse
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 373sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec
```

GENERIC:

```
SetupTime=104443 ms
Index=2
PeerAddress=50110
PeerSubAddress=
PeerId=100
PeerIfIndex=104
LogicalIfIndex=10
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconnectTime=143330
CallDuration=00:06:23
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=37717
TransmitBytes=5706436
ReceivePackets=37668
ReceiveBytes=6609552
```

TELE:

```
ConnectionId=[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
TxDuration=375300 ms
VoiceTxDuration=375300 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-75
ACOMLevel=11
SessionTarget=
ImgPages=0
```

Cisco Router Challenge 195

Outline

This challenge involves the configuration of SIP with a Cisco SIP Gateway FXO setup. Some routers have Foreign Exchange Station (FXS) interfaces which can connect to a standard telephone, fax machine, or similar device and thus must provide ringing, voltage supplies, and a dial tone. Normally the FXS interface uses an RJ-11 connector to connect to telephone equipment.

Objectives

The objectives of this challenge are to:

- Define SIP details.
- Define the voice port settings

Outline

```
> enable
# sh version
# config t
(config)# sip-au
(config-sip-ua)# ?
(config-sip-ua)# exit
(config)# voice-port 1/0/0
(config-voiceport)# description testing
(config-voiceport)# input gain 8
(config-voiceport)# caller-id enable
(config-voiceport)# exit
(config)# dial-peer voice 200 voip
(config-dial-peer)# exit
(config)# gateway
```

Example

```
> enable
# sh version
Cisco IOS Software, C2600 Software (C2600-ADVENTERPRISEK9-M), Version 12.4(12), RELEASE
SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2006 by Cisco Systems, Inc.
Compiled Fri 17-Nov-06 11:18 by prod_rel_team
```

```
ROM: System Bootstrap, Version 12.2(7r) [cmong 7r], RELEASE SOFTWARE (fc1)
```

```
Router uptime is 5 hours, 38 minutes
System returned to ROM by power-on
System image file is "flash:c2600-testk9-mz.124-12.bin"
```

```
Cisco 2611XM (MPC860P) processor (revision 1.0) with 111616K/19456K bytes of memory.
Processor board ID JAD07130QPE
M860 processor: part number 5, mask 2
2 FastEthernet interfaces
2 Serial(sync/async) interfaces
2 Voice FXO interfaces
```

2 Voice FXS interfaces

32K bytes of NVRAM.

49152K bytes of processor board System flash (Read/Write)

Configuration register is 0x3162

config t

(config)# sip-au

(config-sip-ua)# ?

SIP UA configuration commands:

aaa	sip-ua AAA related configuration
authentication	Digest Authentication Configuration
calling-info	Specify treatment of calling information
default	Set a command to its defaults
disable-early-media	Disable early-media cut through
exit	Exit from sip-ua configuration mode
max-forwards	Change number of max-forwards for SIP Methods
mwi-server	Configure a mwi Server
nat	Enable NAT(Network Address Traversal) settings for the SIP User Agent
no	Negate a command or set its defaults
notify	SIP Signaling Notify Configuration
offer	Configure settings for Offers made from the Gateway
reason-header	Configure settings for supporting SIP Reason Header
redirection	Enable call redirection (3xx) handling
registrar	Configure SIP registrar VoIP Interface
remote-party-id	Enable Remote-Party-ID support in SIP User Agent
retry	Change default retries for each SIP Method
set	Sets the PSTN cause to SIP status code (and vice versa) and sets the PSTN cause to SIP requests
sip-server	Configure a SIP Server Interface
srv	DNS SRV Query Type
suspend-resume	Enable support for ISDN SUSPEND/RESUME
timers	SIP Signaling Timers Configuration
transport	Enable SIP UA transport for TCP/UDP

(config-sip-ua)# exit

(config)# voice-port ?

<1-1> Voice interface slot

(config)# voice-port 1/0/0

<1-1> Voice interface slot

(config-voiceport)# ?

Voice-port configuration commands:

battery-reversal	Enable FXS battery-reversal generation
bearer-cap	Specify the bear capability
busyout	Configure busyout trigger event & procedure
caller-id	Configure port caller id parameters
comfort-noise	Use fill-silence option
connection	Specify Trunking Parameters
cptone	Configure voice call progress tone locale
default	Set a command to its defaults
description	Description of what this port is connected to
disc_pi_off	close voice path when disconnect with PI received
disconnect-ack	FXS sending disconnect acknowledge
echo-cancel	Echo-cancellation option
exit	Exit from voice-port configuration mode
impedance	Specifies the terminating impedance of the interface
input	Configure input gain for voice
music-threshold	Threshold for Music on Hold
mwi	Enable MWI on this port
no	Negate a command or set its defaults
non-linear	Use non-linear processing during echo cancellation
output	Configure output attenuation for voice
playout-delay	Configure voice playout delay buffer

```

ren                Ringer Equivalence Number
ring               Ring frequency Parameters
shutdown           Take voice-port offline
signal             The signaling type for the interface FXS or FXO
snmp               Modify SNMP voice port parameters
station-id         Configure station ID
supervisory        Configure supervisory disconnect lcfo
threshold          Threshold [noise] for voice port
timeouts           Configure voice timeout parameters
timing             Configure voice timing parameters
translate          Translation rule
translation-profile Translation profile
trunk-group        Configure interface to be in a trunk group
voice-class        Set voiceport voice class control parameters
(config-voiceport)# description ?
LINE A string (up to 64 characters) describing the port connection (e.g.
pbx1)
(config-voiceport)# description testing

(config-voiceport)# input ?
gain Configure gain in db for voice input

(config-voiceport)# input gain ?
<-6 - 14> gain in db

(config-voiceport)# input gain 8
(config-voiceport)# caller-id ?
alerting          Define caller id alerting method
attenuation        Configure caller id tx attenuation
block             Block the caller id of the calls made from this port
enable            Enable caller id on this port
format            Change caller id format
(config-voiceport)# caller-id enable
(config-voiceport)# exit
(config)# dial-peer ?
cor              Class of Restriction
hunt             Define the dial peer hunting choice
outbound         Define the outbound options
terminator       Define the address terminate character
voice            Voice type

(config)# dial-peer voice ?
<1-2147483647> Voice dial-peer tag

(config)# dial-peer voice 200 ?
mmoip           Multi Media Over IP
pots            Telephony
voatm           Voice over ATM
vofr            Voice over Frame Relay
voip            Voice over IP

(config)# dial-peer voice 200 voip

(config-dial-peer)# ?
DIALPEER configuration commands:
acc-qos         The Minimally Acceptable Quality of Service to be
                used in getting to this peer
answer-address  The Call Destination Number
application      The selected application
call            Per Voip dial-peer Call configuration
call-block      Incoming Call Blocking
carrier-id      Configure Carrier ID
clid            Caller ID option

```

```

codec          The codec rate to be attempted in getting to this peer
corlist        set the Class of Restriction lists
default        Set a command to its defaults
description    Dialpeer specific description
destination-pattern A full E.164 telephone number prefix
dnis-map       The name of a configured dnis-map
dtmf-relay     Transport DTMF digits across IP link
exit           Exit from dial-peer configuration mode
expect-factor  Expectation Factor of voice quality
fax            Configure fax
fax-relay      fax-relay options
huntstop       Stop hunting on Dial-Peers
icpif          Calculated Planning Impairment Factor
incoming       Incoming called number
ip             Set ip packet options
max-conn       Sets the maximum connections per peer, negation sets
               to unlimited
max-redirects Configure the max number of redirects for this
               dialpeer
modem          Modem commands through this peer
no             Negate a command or set its defaults
numbering-type The calling/called party numbering type
permission     set the call orig/term permission of this dialpeer
playout-delay Configure voice playout delay buffer
preference     Configure the preference order of this dialpeer
req-qos        The desired Quality of Service to be used in
               getting to this peer
roaming        Use roaming server
rtp            RTP config
session        The session [ target | protocol | transport ] for this
               peer
settle-call    Use settlement server
shutdown       Change the Admin State of this peer to down (no->up)
signal-type    The signaling type to be used when getting to this
               peer
signaling      Signaling payload handling
snmp           Modify SNMP voice peer parameters
supplementary-service Config supplementary service features
tech-prefix    The H.323 gateway technology prefix
tone           Generate tones
translate-outgoing Translation rule
translation-profile Translation profile
trunk-group-label Configure Trunk Group Label
trunkgroup     trunk groups associated with this peer
vad            Use VoiceActivityDetection as necessary option
voice          Configure GATEWAY dial-peer for voice services
voice-class    Set Dial-peer voice class control parameters
(config-dial-peer)# exit
(config)# gateway

(config-gateway)# ?
GATEWAY configuration commands:
  default      Set a command to its defaults
  emulate      Gateway emulation configuration
  exit         Exit from gateway configuration mode
  no           Negate a command or set its defaults
  security     Gateway security configuration
  timer        Gateway-wide timers

```

Cisco CCVP Test 6

VoIP Signalling and Call Control

The most up-to-date version of this test is at:

<http://networksims.com/v06.html>

Cisco Router Challenge 134

Outline

This challenge involves the configuration of auto QoS on an interface.

Objectives

The objectives of this challenge are to:

- Define CEF (Cisco Express Forwarding), as this is required for Auto QoS.
- Enable NBAR (Network Based Application Recognition), as this is required for Auto QoS.
- Define the bandwidth on an interface.
- Enable Auto QoS.

Example

```
> en
# config t
(config)# ip cef
(config)# int s0
(config-if)# bandwidth ?
    <1-10000000> Bandwidth in kilobits
    inherit      Specify how bandwidth is inherited
(config-if)# bandwidth 256
(config-if)# ip nbar ?
    protocol-discovery Enable NBAR protocol discovery

(config-if)# ip nbar protocol ?
    <cr>
(config-if)# ip nbar protocol
(config-if)# auto ?
    qos Configure AutoQoS

(config-if)# auto qos ?
    voip Configure AutoQoS for VoIP

(config-if)# auto qos voip ?
    trust Trust the DSCP marking
    <cr>
```

```

(config-if)# auto qos voip
(config-if)# exit
(config)# exit
# sh ip nbar pr
Serial0/0

```

Protocol	Input		Output	
	Packet Count	Byte Count	Packet Count	Byte Count
(bps)	5 minute bit rate (bps)		5 minute bit rate	
bgp	0	0	0	0
	0	0	0	0
citrix	0	0	0	0
	0	0	0	0
cuseeme	0	0	0	0
	0	0	0	0
custom-01	0	0	0	0
	0	0	0	0
custom-02	0	0	0	0
	0	0	0	0
custom-03	0	0	0	0
	0	0	0	0
custom-04	0	0	0	0
	0	0	0	0
custom-05	0	0	0	0
	0	0	0	0
custom-06	0	0	0	0
	0	0	0	0
custom-07	0	0	0	0
	0	0	0	0
custom-08	0	0	0	0
	0	0	0	0
custom-09	0	0	0	0
	0	0	0	0
custom-10	0	0	0	0
	0	0	0	0
dhcp	0	0	0	0
	0	0	0	0
dns	0	0	0	0
	0	0	0	0
egp	0	0	0	0

	0	0
	0	0
eigrp	0	0
	0	0
exchange	0	0
	0	0
fasttrack	0	0
	0	0
finger	0	0
	0	0
ftp	0	0
	0	0
gnutella	0	0
	0	0
gopher	0	0
	0	0
gre	0	0
	0	0
http	0	0
	0	0
icmp	0	0
	0	0
imap	0	0
	0	0
ipinip	0	0
	0	0
ipsec	0	0
	0	0
irc	0	0
	0	0
kazaa2	0	0
	0	0
kerberos	0	0
	0	0
l2tp	0	0
	0	0
ldap	0	0
	0	0
napster	0	0

	0	0
	0	0
netbios	0	0
	0	0
	0	0
netshow	0	0
	0	0
	0	0
nfs	0	0
	0	0
	0	0
nntp	0	0
	0	0
	0	0
notes	0	0
	0	0
	0	0
novadigm	0	0
	0	0
	0	0
ntp	0	0
	0	0
	0	0
pcanywhere	0	0
	0	0
	0	0
pop3	0	0
	0	0
	0	0
pptp	0	0
	0	0
	0	0
printer	0	0
	0	0
	0	0
rcmd	0	0
	0	0
	0	0
rip	0	0
	0	0
	0	0
rsvp	0	0
	0	0
	0	0
rtp	0	0
	0	0
	0	0
rtspplayer	0	0
	0	0
	0	0
secure-ftp	0	0
	0	0
	0	0
secure-http	0	0
	0	0
	0	0
secure-imap	0	0

	0	0
	0	0
secure-irc	0	0
	0	0
secure-ldap	0	0
	0	0
secure-nntp	0	0
	0	0
secure-pop3	0	0
	0	0
secure-telnet	0	0
	0	0
smtp	0	0
	0	0
snmp	0	0
	0	0
socks	0	0
	0	0
sqlnet	0	0
	0	0
sqlserver	0	0
	0	0
ssh	0	0
	0	0
streamwork	0	0
	0	0
sunrpc	0	0
	0	0
syslog	0	0
	0	0
telnet	0	0
	0	0
tftp	0	0
	0	0
vdolive	0	0
	0	0
xwindows	0	0
	0	0
unknown	0	0

	0	0
	0	0
Total	0	0
	0	0
	0	0

Explanation

Key facts:

CCNP Objective: QoS Implementation Methods.

- **AutoQoS for the Enterprise** is the next generation of QoS generation, and uses NBAR for traffic discovery and classification. The basic Auto QoS is **Auto QoS VoIP**.
- For Auto QoS to work, **CEF** and **NBAR** must be enabled. Also the **bandwidth** must be correctly defined on the interface.
- AutoQoS automatically generate QoS commands.
- AutoQoS analyzes network traffic and tries to optimize the QoS through traffic classes that the AutoQoS Discovery method to create policies, which are applied to the interface(s).
- AutoQoS simplifies the configuration.
- AutoQoS uses **Classification** (This uses AutoQoS Discovery with NBAR to discover the requirements); **Policy generation** (This uses access-lists, class-maps and policy maps to optimize the setup); **Configuration** (This configures the required interfaces); **Monitoring and reporting** (This continually updates and reports on the operation); and **Consistency** (This allows for consistency across a range of devices).

Cisco Switch Challenge 70

Outline

This challenge involves configuring Auto QoS on a switch.

Objectives

The objectives of this challenge are to:

- Define Auto QoS

Example

```
> en
# config t
(config)# cdp run

(config)# int vlan 10
```

```

(config)# int vlan 10
(config-vlan)# exit
(config)# int vlan 20
(config-vlan)# exit

(config)# int fa0/1
(config-if)# cdp enable
(config-if)# switchport ?
  access          Set access mode characteristics of the interface
  block           Disable forwarding of unknown uni/multi cast addresses
  broadcast        Set broadcast suppression level on this interface
  encapsulation   Set trunking encapsulation when interface is in trunking mode
  host            Set port host
  mode            Set trunking mode of the interface
  multicast        Set multicast suppression level on this interface
  native          Set trunking native characteristics when interface is in
                  trunking mode
  nonegotiate     Device will not engage in negotiation protocol on this
                  interface
  port-security   Security related command
  priority        Set appliance 802.1p priority
  protected       Configure an interface to be a protected port
  pruning         Set pruning VLAN characteristics when interface is in trunking
                  mode
  trunk           Set trunking characteristics of the interface
  unicast         Set unicast suppression level on this interface
  voice           Voice appliance attributes
  <cr>

(config-if)# switchport access vlan 10
(config-if)# switchport voice ?
  vlan           Vlan for voice traffic

(config-if)# switchport voice vlan ?
  <1-4094>       Vlan for voice traffic
  dot1p         Priority tagged on PVID
  none          Don't tell telephone about voice vlan
  untagged      Untagged on PVID
(config-if)# switchport voice vlan 20
(config-if)# au ?
  qos           Configure AutoQoS

(config-if)# auto qos ?
  voip         Configure AutoQoS for VoIP

(config-if)# auto qos voip ?
  cisco-phone  Trust the QoS marking of Cisco IP Phone
  trust        Trust the COS marking

(config-if)# auto qos voip cisco-phone
(config-if)# exit

```

Note:

For Auto QoS VoIP, CDP needs to be enabled.

Cisco Switch Challenge 226

Outline

This challenge involves configuring H.323 CAC (Call Admission Control).

Objectives

The objectives of this challenge are to:

- Define H.323 CAC.

Outline

```
> en
# config t
(config)# call threshold interface e0
(config)# call threshold global cpu-avg low 10 high 50 busyout
(config)# call threshold global total-calls low 15 high 5000 busyout
(config)# call threshold global total-mem low 15 high 5000 busyout
(config)# call threshold global io-mem low 15 high 5000 busyout
(config)# call spike 20 steps 10 size 1000
(config)# call treatment on
(config)# call treatment action hairpin
```

Example

```
> en
# config t
(config)# call thres ?
  global          the global resources of this gateway
  interface       interface triggers for this gateway
  poll-interval  the poll interval for some resources
(config)# call threshold interface e0
(config)# call thres ?
  global          the global resources of this gateway
  interface       interface triggers for this gateway
  poll-interval  the poll interval for some resources

(config)# call th g ?
  cpu-5sec       the CPU utilization in the last 5 seconds
  cpu-avg        the average CPU utilization
  io-mem         the IO memory utilization
  proc-mem       the Processor memory utilization
  total-calls    the total number of calls
  total-mem      the total memory utilization

(config)# call th g cpu-avg ?
  low           the low threshold
```

```

(config)# call th g cpu-avg 15 ?
<1-100> low threshold in %

(config)# call th g cpu-avg 1 15 ?
high the high threshold

(config)# call th g cpu-avg 1 15 h ?
<1-100> high threshold in %

(config)# call th g cpu-avg 1 15 h 50 ?
busyout busyout the voice interfaces if out-of-resource
treatment apply out-of-resource to call treatment
<cr>
(config)# call threshold global cpu-avg low 15 high 50 busyout
(config)# call threshold global total-calls low 15 high 5000 busyout
(config)# call threshold global total-mem low 15 high 5000 busyout
(config)# call threshold global io-mem low 15 high 5000 busyout
(config)# call spike ?
<1-2147483647> Incoming call numbers for spiking threshold
(config)# call spike 1 ?
steps number of steps for spiking sliding window
<cr>
(config)# call spike 1 s ?
<3-10> number of steps
(config)# call spike 1 s 3 ?
size step size in millisecond
(config)# call spike 1 s 3 s ?
<100-250> millisecond
(config)# call spike 1 s 3 s 1000 ?
<cr>
(config)# call spike 20 steps 10 size 1000
(config)# call treat ?
action Action to take when call treatment is triggered
cause-code Select the cause code for disconnection
isdn-reject Select the ISDN reject cause-code
on toggle deny on/off
(config)# call treatment on
(config)# call treat a ?
hairpin Hairpin
playmsg play the selected message
reject Disconnect the call and pass down cause code
(config)# call treatment action hairpin

```

Cisco CCVP Test 7

Improving and Maintaining Voice Quality

The most up-to-date version of this test is at:

<http://networksims.com/v07.html>

1 MPLS

Cisco MPLS

MPLS Introduction

The most up-to-date version of this test is at:

<http://networksims.com/i01.html>

Cisco Router Challenge 227

Outline

This challenge involves basic frame-mode MPLS configuration.

Objectives

The objectives of this challenge are to:

- Enable CEF globally.
- Enable CEF on S0.
- Define IGP routing protocol.
- Assign LDP router ID.
- Enable IPv4 MPLS on an interface (mpls ip).

Commands

```
> enable
# config t
(config)# ip cef
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip route-cache cef
(config-if)# mpls ip
(config-if)# exit
(config)# router ospf 101
(config-router)# network 10.0.0.0 0.0.0.255 area 1
(config)# mpls ldp router-id loopback5
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
```

Example

```
> enable
# config t
(config)# ip cef
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip route- ?
    cef                Enable Cisco Express Forwarding
    flow               Enable Flow fast-switching cache
    policy             Enable fast-switching policy cache for outgoing packets
    same-interface     Enable fast-switching on the same interface
    <cr>

(config-if)# ip route-cache cef
(config-if)# mpls ?
    accounting        Enable MPLS accounting on this interface
    ip                Configure dynamic MPLS forwarding for IP
    label             Label properties
    ldp               Configure Label Distribution Protocol (LDP) parameters
    mtu               Set tag switching Maximum Transmission Unit
    netflow           Configure Egress Netflow Accounting
    traffic-eng       Configure Traffic Engineering parameters
(config-if)# mpls ip

(config-if)# exit
(config)# router ospf 101
(config-router)# network 10.0.0.0 0.0.0.255 area 1
(config-router)# exit
(config)# mpls ?
    atm              Configure ATM options
    ip               Dynamic MPLS forwarding for IP
    ipv6             Dynamic MPLS forwarding for IPv6
    label            Label properties
    ldp              Label Distribution Protocol
    static           Configure static label bindings
    traffic-eng      Configure Traffic Engineering parameters

(config)# mpls ldp ?
    advertise-labels Label advertisements
    atm              Configure ATM MPLS options
    backoff          Set LDP session backoff parameters
    discovery        LDP discovery
    explicit-null    Advertise Explicit Null label in place of Implicit Null
    graceful-restart Configure LDP Graceful Restart
    holdtime         LDP session holdtime
    igp              Configure IGP-related LDP parameters
    logging          Enable LDP logging
    loop-detection   Enable LDP Loop Detection
    maxhops          Limit hop count for LDP LSP setup
    neighbor         Configure neighbor parameters
    path-vector      Path Vector for LDP LSP setup
    request-labels   Access list to specify valid downstream on demand
                    destinations.
    router-id        Select interface to prefer for LDP identifier address
    session          Configure session parameters
    tcp              Set TCP parameters for LDP

(config)# mpls ldp router-id ?
    Async            Async interface
    BVI              Bridge-Group Virtual Interface
    CDMA-Ix          CDMA Ix interface
```

CTunnel	CTunnel interface
Dialer	Dialer interface
Ethernet	IEEE 802.3
FastEthernet	FastEthernet IEEE 802.3
Group-Async	Async Group interface
Lex	Lex interface
Loopback	Loopback interface
MFR	Multilink Frame Relay bundle interface
Multilink	Multilink-group interface
Null	Null interface
Port-channel	Ethernet Channel of interfaces
Serial	Serial
TokenRing	IEEE 802.5
Tunnel	Tunnel interface
Vif	PGM Multicast Host interface
Virtual-PPP	Virtual PPP interface
Virtual-Template	Virtual Template interface
Virtual-TokenRing	Virtual TokenRing

```
(config)# mpls ldp router-id loopback5
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
```

Cisco Router Challenge 228

Outline

This challenge involves configuration of MPLS over a routed PVC in ATM.

Objectives

The objectives of this challenge are to:

- Enable CEF globally.
- Enable MPLS on ATM.
- Define ATM interface parameters.
- Define ATM sub-interface.

Commands

```
> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# exit
(config)# int atm0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config)# int atm0.1 point-to-point
(config-subif)# ip address 138.10.11.1 255.255.255.238
(config-subif)# mpls ip
(config-subif)# pvc 2/100
(config-if-atm-vc# encapsulation aal5snap
(config-if-atm-vc# exit
```

```
(config-if)# exit
(config)# router ospf 101
(config-router)# network 10.0.0.0 0.0.0.255 area 1
```

Example

```
> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# exit
(config)# int atm0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config)# int atm0.1 ?
  mpls          Treat as an MPLS link
  multipoint    Treat as a multipoint link
  point-to-point Treat as a point-to-point link
  tag-switching Treat as a tag switching link (obsolete, use mpls)
<cr>
(config)# int atm0.1 point-to-point
(config-subif)# ip address 138.10.11.1 255.255.255.238
(config-subif)# mpls ?
  atm          Tag controlled ATM parameters
  ip           Configure dynamic MPLS forwarding for IP
  label        Label properties
  ldp          Configure Label Distribution Protocol (LDP) parameters
  mtu          Set tag switching Maximum Transmission Unit
  netflow      Configure Egress Netflow Accounting
  traffic-eng  Configure Traffic Engineering parameters
(config-subif)# mpls ip
(config-subif)# pvc ?
  <0-7>        Enter VPI/VCI value (slash required)
  <1-1023>     Enter VCI value
  WORD         Optional handle to refer to this connection
(config-subif)# pvc 2/100
(config-if-atm-vc)# encapsulation ?
  aal5cisco    Cisco PPP over AAL5 Encapsulation
  aal5mux      AAL5+MUX Encapsulation
  aal5nlpid    AAL5+NLPIID Encapsulation
  aal5snap     AAL5+LLC/SNAP Encapsulation
(config-if-atm-vc)# encapsulation aal5snap
(config-if)# exit
(config)# router ospf 101
(config-router)# network 10.0.0.0 0.0.0.255 area 1
```

Cisco MPLS

MPLS Basics

The most up-to-date version of this test is at:

<http://networksims.com/i02.html>

Cisco Router Challenge 229

Outline

This challenge involves an MPLS VPN configuration, which is often seen in ISP applications, and uses MPLS forwarding and VRF (Virtual Routing and Forwarding). VRF is used to create multiple instances of a routing table within the same router, at the same time. Thus the same, or overlapping, IP addresses can be used without a conflict, as all of these routing instances are independent. VRF uses Forwarding Information Bases (FIBs), which are distinct routing tables. With an MPLS domain, the **Provider Edge (PE)** routing switch is the only device to have knowledge of the multiple different virtual routing engines, and **Customer Edge (CE)** devices then participate in their MPLS VPN routing via **route dissemination (RD)** to and from the PE, using a routing engine such as eBGP, OSPF or static routing. All the other Provider (P) switches in backbone do not have any knowledge of IP routing within this context. The PE traffic travels over the core network using label switching. This challenge involves the configuration of the VRF on the PE routers, and the next challenge sets up BGP PE-PE Routing on PE Routers, which is required to transport the routes over the backbone.

Objectives

The objectives of this challenge are to:

- Configure VRF.
- Configure RD (which is used to make a unique IP address).
- Configure import and export policy (RT).
- Associate VRF with an interface

Commands

```
> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# ip vrf forwarding Testing
(config-if)# exit
(config)# ip vrf Testing
(config-vrf)# rd 1:100
(config-vrf)# route-target both 1:100
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip vrf forwarding Testing
(config-if)# exit
```

Example

```

> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# ip vrf forwarding Testing
(config-if)# exit
(config)# ip vrf ?
WORD VPN Routing/Forwarding instance name
(config)# ip vrf Testing
(config-vrf)# ?
IP VPN Routing/Forwarding instance configuration commands:
bgp          Commands pertaining to BGP
context      Associate SNMP context with this vrf
default      Set a command to its defaults
description  VRF specific description
exit         Exit from VRF configuration mode
maximum      Set a limit
mdt          Backbone Multicast Distribution Tree
no           Negate a command or set its defaults
rd           Specify Route Distinguisher
route-target Specify Target VPN Extended Communities
vpn          Configure VPN ID as specified in rfc2685

(config-vrf)# rd 1:100
(config-vrf)# rd ?
ASN:nn or IP-address:nn VPN Route Distinguisher

(config-vrf)# route-target ?
ASN:nn or IP-address:nn Target VPN Extended Community
both         Both import and export Target-VPN community
export       Export Target-VPN community
import       Import Target-VPN community

(config-vrf)# route-target both ?
ASN:nn or IP-address:nn Target VPN Extended Community
(config-vrf)# route-target both 1:100
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip vrf ?
forwarding  Configure forwarding table
receive     Add Interface Address into VRF Table
sitemap     Configure route-map for routes received from this site

(config-if)# ip vrf forwarding ?
WORD Table name
(config-if)# ip vrf forwarding Testing
(config-if)# exit

```

The RD value creates routing and forwarding tables. It is added at the beginning of customer IP addresses, to convert them to unique IP address. It can either be:

- 16-bit AS number: 32-bit number. For example 1:100, which has an AS of 1, and a 32-bit number of 100.
- 32-bit IP number: 16-bit number. For example 192.168.1.1:1, which has a 16-bit value of 1.

The VRF is associated with an interface with:

```
(config-if)# ip vrf forwarding Testing
```

Cisco Router Challenge 230

Outline

The previous challenge involved the configuration of the VRF on the PE routers, and this next challenge sets up BGP PE-PE Routing on PE Routers, which is required to transport the routes over the backbone.

Objectives

The objectives of this challenge are to:

- Configure VRF.
- Configure RD (which is used to make a unique IP address).
- Configure import and export policy (RT).
- Associate VRF with an interface.
- Configure BGP PE-PE on a PE device.

Commands

```
> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# ip vrf forwarding Testing
(config-if)# exit
(config)# ip vrf Testing
(config-vrf)# rd 1:100
(config-vrf)# route-target both 1:100
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip vrf forwarding Testing
(config-if)# exit
(config)# router bgp 1
(config-router)# neighbor 138.199.17.1 remote-as 1
(config-router)# neighbor 138.199.17.1 update-source loopback5
(config-router)# address-family vpn4
(config-router-af)# neighbor 1.2.3.4 send-community extended
(config-router-af)# neighbor 1.2.3.4 activate
```

Example

```
> enable
# config t
(config)# ip cef
(config)# int loopback5
(config-if)# ip address 138.199.17.2 255.255.255.248
(config-if)# ip vrf forwarding Testing
```

```

(config-if)# exit
(config)# ip vrf ?
    WORD VPN Routing/Forwarding instance name
(config)# ip vrf Testing
(config-vrf)# ?
IP VPN Routing/Forwarding instance configuration commands:
    bgp          Commands pertaining to BGP
    context      Associate SNMP context with this vrf
    default      Set a command to its defaults
    description  VRF specific description
    exit         Exit from VRF configuration mode
    maximum      Set a limit
    mdt          Backbone Multicast Distribution Tree
    no          Negate a command or set its defaults
    rd           Specify Route Distinguisher
    route-target Specify Target VPN Extended Communities
    vpn         Configure VPN ID as specified in rfc2685

(config-vrf)# rd 1:100
(config-vrf)# rd ?
    ASN:nn or IP-address:nn VPN Route Distinguisher

(config-vrf)# route-target ?
    ASN:nn or IP-address:nn Target VPN Extended Community
    both                    Both import and export Target-VPN community
    export                  Export Target-VPN community
    import                  Import Target-VPN community

(config-vrf)# route-target both ?
    ASN:nn or IP-address:nn Target VPN Extended Community
(config-vrf)# route-target both 1:100
(config)# int s0
(config-if)# ip address 138.199.17.1 255.255.255.248
(config-if)# ip vrf ?
    forwarding  Configure forwarding table
    receive     Add Interface Address into VRF Table
    sitemap     Configure route-map for routes received from this site

(config-if)# ip vrf forwarding ?
    WORD Table name
(config-if)# ip vrf forwarding Testing
(config-if)# exit
(config-if)# exit
(config)# router bgp 1
(config-router)# neighbor 138.199.17.1 remote-as 1
(config-router)# neighbor 138.199.17.1 update-source loopback5
(config-router)# address-family vpn4
(config-router-af)# ?
Router Address Family configuration commands:
    auto-summary      Enable automatic network number summarization
    autonomous-system Specify AS number for Address Family
    default           Set a command to its defaults
    default-information Control distribution of default information
    default-metric    Set metric of redistributed routes
    distance          Define an administrative distance
    distribute-list   Filter networks in routing updates
    eigrp             EIGRP specific commands
    exit-address-family Exit from Address Family configuration mode
    help             Description of the interactive help system
    maximum-paths    Forward packets over multiple paths
    metric            Modify EIGRP routing metrics and parameters
    network          Enable routing on an IP network
    no               Negate a command or set its defaults

```

```

offset-list          Add or subtract offset from RIP metrics
redistribute         Redistribute information from another routing protocol
variance            Control load balancing variance
(config-router-af)# neighbor ?
A.B.C.D            Neighbor address
WORD              Neighbor tag
X:X:X:X::X        Neighbor IPv6 address

(config-router-af)# neighbor 1.2.3.4 ?
activate           Enable the Address Family for this Neighbor
advertise-map      specify route-map for conditional advertisement
advertisement-interval Minimum interval between sending BGP routing updates
allowas-in         Accept as-path with my AS present in it
description        Neighbor specific description
distribute-list    Filter updates to/from this neighbor
dmzlink-bw         Propagate the DMZ link bandwidth
ebgp-multihop     Allow EBGp neighbors not on directly connected
                  networks
filter-list        Establish BGP filters
local-as           Specify a local-as number
maximum-prefix     Maximum number of prefix accept from this peer
next-hop-self      Disable the next hop calculation for this neighbor
password           Set a password
peer-group         Member of the peer-group
prefix-list        Filter updates to/from this neighbor
remote-as          Specify a BGP neighbor
remove-private-AS  Remove private AS number from outbound updates
route-map          Apply route map to neighbor
route-reflector-client Configure a neighbor as Route Reflector client
send-community     Send Community attribute to this neighbor
shutdown           Administratively shut down this neighbor
soft-reconfiguration Per neighbor soft reconfiguration
timers             BGP per neighbor timers
unsuppress-map     Route-map to selectively unsuppress suppressed routes
update-source      Source of routing updates
version            Set the BGP version to match a neighbor
weight            Set default weight for routes from this neighbor

(config-router-af)# neighbor 1.2.3.4 send ?
both              Send Standard and Extended Community attributes
extended         Send Extended Community attribute
standard         Send Standard Community attribute
<cr>

(config-router-af)# neighbor 1.2.3.4 send-community extended
(config-router-af)# neighbor 1.2.3.4 activate

```

2 CCNP (Voice Gateway)

Cisco Gateway and Gatekeeper

Gateway and Gatekeeper Introduction

The most up-to-date version of this test is at:

<http://networksims.com/ga01.html>

Cisco Router Challenge 231

Outline

This challenge involves basic MGCP gateway configuration.

Objectives

The objectives of this challenge are to:

- Define MGCP Gateway configuration

Commands

```
> enable
# config t
(config)# mgcp
(config)# mgcp call-agent 192.168.1.1
(config)# ccm-manager mgcp
```

If the MGCP configuration is to be loaded from CallManager, the IP address of the TFTP server (such as CallManager) must be defined, such as:

```
(config)# ccm-manager config
(config)# ccm-manager config server 192.168.1.2
```

And then to bind MGCP to the voice ports:

```
(config)# dial-peer voice 100 pots
(config-dial-peer)# application MGCPAPP
(config-dial-peer)# port 1/0/1
```

And there needs to be at least one dial peer in case CallManager is not available:

```
(config)# dial-peer voice 200 pots
(config-dial-peer)# destination-pattern 123..
(config-dial-peer)# incoming called-number .
(config-dial-peer)# port 1/0/1
(config-dial-peer)# exit
```

Next the IP address that CallManager communicates with is defined:

```
(config)# int loopback15
(config-if)# ip address 192.168.1.1 255.255.255.0
(config-if)# exit
(config)# mgcp bind control source-interface loopback15
```

and to enable DTMF-relay:

```
(config)# mgcp dtmf-relay voip code all mode out-of-band
```

and to enable the MGCP gateway to use the fallback mode:

```
(config)# ccm-manager fallback-mgcp
(config)# ccm-manager redundant-host 192.168.1.1
```

and finally the details can be shown:

```
(config)# exit
# sh ccm-manager
# sh mgcp
```

Example

```
> enable
# config t
> enable
# config t
(config)# mgcp
(config)# mgcp call-agent 192.168.1.1
(config)# ccm-manager mgcp
(config)# ccm-manager config server 192.168.1.2
(config)# ccm-manager control
(config)# dial-peer voice 100 pots
(config-dial-peer)# application mgcpapp
(config-dial-peer)# port 1/0/1

(config-dial-peer)# exit
(config)# dial-peer voice 200 pots
(config-dial-peer)# destination-pattern 123..
(config-dial-peer)# incoming called-number .
(config-dial-peer)# port 1/0/1
(config-dial-peer)# exit
(config)# int loopback15
(config-if)# ip address 192.168.1.1 255.255.255.0
(config-if)# exit
(config)# mgcp bind ?
    control  bind only MGCP control packets
    media    bind only media packets
```

```

(config)# mgcp bind control ?
    source-interface  Specify interface for source address of MGCP packets

(config)# mgcp bind control source-interface ?
    Async            Async interface
    BVI              Bridge-Group Virtual Interface
    CTunnel          CTunnel interface
    Dialer           Dialer interface
    FastEthernet     FastEthernet IEEE 802.3
    Loopback         Loopback interface
    MFR              Multilink Frame Relay bundle interface
    Multilink        Multilink-group interface
    Null             Null interface
    Serial           Serial
    Tunnel           Tunnel interface
    Vif              PGM Multicast Host interface
    Virtual-Template Virtual Template interface
    Virtual-TokenRing Virtual TokenRing
(config)# mgcp bind control source-interface loopback15
(config)# mg dt ?
    voaal2  Enable mgcp dtmf-relay for VoAAL2 Calls (using Annex K Type3
            packets).
    voip    Enable mgcp dtmf-relay for VoIP Calls

(config)# mg dt voi ?
    codec  Configure mgcp dtmf-relay codec

(config)# mg dt voi c ?
    all          Enable mgcp dtmf-relay for all codec
    low-bit-rate Enable mgcp dtmf-relay for low-bit-rate codec

(config)# mg dt voi c a ?
    mode  Set mgcp dtmf-relay mode

(config)# mg dt voi c a m ?
    cisco      Set mgcp dtmf-relay mode to be cisco
    nse        Set mgcp dtmf-relay mode to be nse
    nte-ca     Set mgcp dtmf-relay mode to be nte-ca
    nte-gw     Set mgcp dtmf-relay mode to be nte-gw
    out-of-band Set mgcp dtmf-relay mode to be out-of-band
(config)# mgcp dtmf-relay voip code all mode out-of-band
(config)# ccm ?
    application          application specific
    config               MGCP download configuration
    download-tones       Enable Tone Download from TFTP server
    fallback-mgcp        Enable Fallback from MGCP to H.323 mode if no Call
                        Manager is available
    fax                  Enable fax protocol for MGCP
    mgcp                 Enable Call Manager Application MGCP mode
    music-on-hold        Enable multicast Music-on-hold
    redundant-host       Redundant host list
    sccp                 Enable Call Manager Application SCCP mode
    shut-backhaul-interfaces Shutdown the backhauled interfaces if no Call
                        Manager is available
    switchback           Configure switchback options for rehomeing to

(config)# ccm-manager fallback-mgcp
(config)# ccm-manager redundant-host ?
    WORD  IP address or Domain name of backup host number 1

(config)# ccm-manager redundant-host 1.2.3.4 ?
    WORD  IP address or Domain name of backup host number 2
    <cr>

```

(config)# ccm-manager redundant-host 192.168.1.1

(config)# exit

sh ccm-manager

MGCP Domain Name: Router

Priority	Status	Host
Primary	Down	1.2.3.4
First Backup	None	
Second Backup	None	

Current active Call Manager: None
Backhaul/Redundant link port: 2428
Failover Interval: 30 seconds
Keepalive Interval: 15 seconds
Last keepalive sent: 23:59:59 UTC Feb 28 1993 (elapsed time: 00:05:3
)
Last MGCP traffic time: 00:02:21 UTC Mar 1 1993 (elapsed time: 00:03:10
Last failover time: None
Last switchback time: None
Switchback mode: Graceful
MGCP fallback mode: Enabled/OFF
Last MGCP fallback start time: None
Last MGCP fallback end time: None

Backhaul/Redundant link is down

Configuration Auto-Download Information

=====

No configurations downloaded

Current state: Waiting for commands

Configuration Download statistics:

Download Attempted	: 1
Download Successful	: 0
Download Failed	: 1
Configuration Attempted	: 0
Configuration Successful	: 0
Configuration Failed(Parsing)	: 0
Configuration Failed(config)	: 0

Last config download command:

Configuration Error History:

TFTP File download failed

FAX mode: cisco

Router# sh mg

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE

MGCP call-agent: 1.2.3.4 Initial protocol service is MGCP 0.1

MGCP block-newcalls DISABLED

MGCP validate domain name DISABLED

MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED

MGCP quarantine mode discard/step

MGCP quarantine of persistent events is ENABLED

MGCP dtmf-relay for VoIP disabled for all codec types

MGCP dtmf-relay for VoAAL2 disabled for all codec types

MGCP voip modem passthrough disabled

MGCP voaal2 modem passthrough disabled

MGCP voip modem relay: Disabled.

MGCP TSE payload: 100

MGCP T.38 Named Signalling Event (NSE) response timer: 200

MGCP Network (IP/AAL2) Continuity Test timer: 200

MGCP 'RTP stream loss' timer: 5

MGCP request timeout 500

MGCP maximum exponential request timeout 4000

MGCP gateway port: 2427, MGCP maximum waiting delay 3000

```
MGCP restart delay 0, MGCP vad DISABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp DISABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 40, 200 in msec
MGCP Fax Playout Buffer is 300 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: line-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
                        hs-package atm-package ms-package dt-package res-package
                        mt-package fxr-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Fax is ENABLED
MGCP T.38 Fax ECM is ENABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0MGCP T.38 Fax High Speed Redundancy: 0
MGCP control bind :DISABLED
MGCP media bind :DISABLED
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Dynamic payload type for G.726-16K codec
MGCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec
MGCP Guaranteed scheduler time is disabled
```

Cisco Gateway and Gatekeeper

MGCP Gateway

The most up-to-date version of this test is at:

<http://networksims.com/ga02.html>

Cisco Router Challenge 232

Outline

This challenge involves H.323 configuration.

Objectives

The objectives of this challenge are to:

- Define H.323 voice class configuration.

Commands

```
> enable
# config t
(config)# voice class codec 44
(config-class)# ?
(config-class)# codec preference 1 g728
(config-class)# codec preference 2 g729r8
(config-class)# codec preference 3 g726r32
(config-class)# exit
(config)# dial-peer voice 3 voip
(config-dial-peer)# destination-pattern .T
(config-dial-peer)# session target ipv4:88.10.11.12
(config-dial-peer)# preference 1
(config-dial-peer)# voice-class code 44
```

Example

```
> enable
# config t
(config)# voice class codec 44
(config-class)# ?
VOICECLASS configuration commands:
  codec  Set class codec parameters
  exit   Exit from voice class configuration mode
  help   Description of the interactive help system
  no     Negate a command or set its defaults
(config-class)# codec ?
  preference  Set priority order for using this codec

(config-class)# codec preference ?
  <1-14>  Priority order (1 = Highest)

(config-class)# codec preference 1 ?
  clear-channel  Clear Channel 64000 bps (No voice capabilities: data transport
                 only)
  g711alaw       G.711 A Law 64000 bps
  g711ulaw       G.711 u Law 64000 bps
  g723ar53       G.723.1 ANNEX-A 5300 bps (contains built-in vad that cannot be
                 disabled)
  g723ar63       G.723.1 ANNEX-A 6300 bps (contains built-in vad that cannot be
                 disabled)
  g723r53        G.723.1 5300 bps
  g723r63        G.723.1 6300 bps
  g726r16        G.726 16000 bps
  g726r24        G.726 24000 bps
  g726r32        G.726 32000 bps
  g728           G.728 16000 bps
  g729br8        G.729 ANNEX-B 8000 bps (contains built-in vad that cannot be
                 disabled)
  g729r8         G.729 8000 bps
(config-class)# codec preference 1 g728
(config-class)# codec preference 2 g729r8
(config-class)# codec preference 3 g726r32
(config-class)# exit
(config)# dial-peer voice 3 voip
(config-dial-peer)# destination-pattern .T
```

```
(config-dial-peer)# session target ipv4:88.10.11.12
(config-dial-peer)# preference 1
(config-dial-peer)# voice-class code 44
```

Cisco Router Challenge 233

Outline

This challenge involves Voice Service VoIP Configuration.

Objectives

The objectives of this challenge are to:

- Define Voice Service.

Commands

```
> enable
# config t
(config)# voice service voip
(conf-voi-serv)# allow-connections h323 to h323
(conf-voi-serv)# h323
(conf-serv-h323)# no h225 timeout keepalive
(conf-serv-h323)# call service stop
(conf-serv-h323)# call start slow
```

Example

```
> enable
# config t
(config)# voice service voip
Router(conf-voi-serv)# ?
VOICE SERVICE configuration commands:
  allow-conn  Define connections
  cause-code  Sets the internal cause code for SIP and H323
  default     Set a command to its defaults
  exit        Exit from voice service configuration mode
  fax         Global fax commands
  h323        Global H.323 configuration commands
  modem       Global modem commands
  no          Negate a command or set its defaults
  shutdown    Stop VoIP services gracefully without dropping active calls
  signaling   Global setting for signaling payload handling
  sip         SIP configuration commands
```

```
(conf-voi-serv)# allow-connections h323 to h323
(conf-voi-serv)# h323
(conf-serv-h323)# ?
```

```
VOICE SERVICE VOIP H323 configuration commands:
  bearer-cap-ie  Specify bearer-cap-ie coding
  call           Global setting for H.323 Calls
  default        Set a command to its defaults
  exit          Exit from voice service voip h323 configuration mode
```

```

h225          TCP H225 call signalling channel
h245          H245 Signalling
h450          H450 parameter configuration
no            Negate a command or set its defaults
ras           Gateway RAS configuration
session       H323 Voice Protocol session config
(conf-serv-h323)# no ?
bearer-cap-ie Specify bearer-cap_ie coding
call          Global setting for H.323 Calls
h225          TCP H225 call signalling channel
h245          H245 Signalling
h450          H450 parameter configuration
ras           Gateway RAS configuration
session       H323 Voice Protocol session config

(conf-serv-h323)# no h225 ?
signal        Specify signaling options
timeout       Specify timeout for maintaining connections

(conf-serv-h323)# no h225 t ?
keepalive     KEEPALIVE timeout
setup         SETUP timeout
tcp           H225 CSA connection type

(conf-serv-h323)# no h225 timeout keepalive
(conf-serv-h323)# call ?
service       H.323 service configuration
start        Global setting for H.323 Call start procedures: Fast/Slow Start
              (Default: Fast Start)

(conf-serv-h323)# call service ?
stop          Stop H.323 service

(conf-serv-h323)# call service stop
(conf-serv-h323)# call start ?
fast          Use Fast Start procedures to initiate call
slow         Use Slow Start procedures to initiate call

(conf-serv-h323)# call start slow

```

Cisco Gateway and Gatekeeper

H.323

The most up-to-date version of this test is at:

<http://networksims.com/ga03.html>

Cisco Router Challenge 234

Outline

This challenge involves SIP Dial Peer configuration.

Objectives

The objectives of this challenge are to:

- Define SIP dial peers.

Commands

```
> enable
# config t
(config)# dial-peer voice 1111 voip
(config-dial-peer)# session target ipv4:10.1.1.1
(config-dial-peer)# session protocol sipv4
(config-dial-peer)# session transport tcp
(config-dial-peer)# exit
(config)# dial-peer voice 1112 voip
(config-dial-peer)# session target ipv4:10.1.1.1
(config-dial-peer)# session protocol sipv4
(config-dial-peer)# voice-class sip transport switch udp tcp
(config-dial-peer)# destination-pattern 99..
```

Example

```
> enable
# config t
(config)# dial-peer voice 1111 voip
(config-dial-peer)# session ?
  protocol    The session protocol to be used in getting to this peer
  target      The session target for this peer
  transport    The transport layer protocol used for this peer
(config-dial-peer)# session target ?
  WORD        A string specifying the session target
(config-dial-peer)# session target ipv4:10.1.1.1
(config-dial-peer)# sess protocol ?
  cisco       Cisco Session Protocol
  multicast   Multicast Session Protocol(voice conferencing)
  sipv2       IETF Session Inititation Protocol
(config-dial-peer)# session protocol sipv4
(config-dial-peer)# sess transport ?
  system      defer to voice service voip session transport
  tcp         Transport Layer Protocol - TCP
  udp         Transport Layer Protocol - UDP
(config-dial-peer)# session transport tcp
(config-dial-peer)# exit
(config)# dial-peer voice 1112 voip
(config-dial-peer)# session target ipv4:10.1.1.1
(config-dial-peer)# session protocol sipv4
(config-dial-peer)# voice-class sip transport switch udp tcp
(config-dial-peer)# destination-pattern 99..
```

By default UDP is used as the transport protocol. In the first dial-peer the command:

```
(config-dial-peer)# session transport tcp
```

is used so that SIP switches from UDP to TCP when the voice packets get to within 200 bytes of the MTU (Maximum Transmission Unit), and thus avoid any fragmentation of the UDP segments.

The command:

```
(config-dial-peer)# voice-class sip transport switch udp tcp
```

is used to enable switching between UDP and TCP transport SIP messages in a specific dial peer.

Cisco Router Challenge 235

Outline

This challenge involves SIP UA configuration for the registration of analog phones with a redundant server. In this case the maximum number of hops for SIP is defined.

Objectives

The objectives of this challenge are to:

- Allow a gateway to register E.164 numbers on non-SIP phones with a registrar. For this the **registrar** command is used.
- Specify the IP address of the SIP server (using **sip-server**).
- Define maximum SIP hops (using **max-forwards**). This value can range between 1 and 70 (the default is 70).
- Disable the listening for SIP UA for messages on port 5060 for UDP (**no transport udp**), and will thus listen for TCP messages.
- Show the configured E.164 phone number registration (using **show sip-ua register status**).
- Verify the SIP UA configuration (using **show sip-ua status**).

Commands

```
> enable
# config t
(config)# sip-ua
(config-sip-ua)# registrar ipv4:192.168.1.1 tcp
(config-sip-ua)# registrar ipv4:192.168.1.2 tcp secondary
(config-sip-ua)# sip-server ipv4:192.168.1.3
(config-sip-ua)# no transport udp
(config-sip-ua)# max-forwards 15
(config-sip-ua)# exit
(config)# exit
```

```
# sh sip-ua status
# show sip-ua register status
```

Example

```
> enable
# config t
(config)# sip-ua
(config-sip-ua)# registrar ?
WORD Registrar Server address
(config-sip-ua)# registrar ipv4:192.168.1.1 tcp
(config-sip-ua)# registrar ipv4:192.168.1.2 tcp secondary
(config-sip-ua)# sip ?
WORD Specify the Server address
(config-sip-ua)# sip-server ipv4:192.168.1.3
(config-sip-ua)# no ?
aaa sip-ua AAA related configuration
authentication Digest Authentication Configuration
calling-info Specify treatment of calling information
disable-early-media Disable early-media cut through
max-forwards Change number of max-forwards for SIP Methods
mwi-server Configure a mwi Server
nat Enable NAT(Network Address Traversal) settings for the
SIP User Agent
notify SIP Signaling Notify Configuration
offer Configure settings for Offers made from the Gateway
reason-header Configure settings for supporting SIP Reason Header
redirection Enable call redirection (3xx) handling
registrar Configure SIP registrar VoIP Interface
remote-party-id Enable Remote-Party-ID support in SIP User Agent
retry Change default retries for each SIP Method
set Sets the PSTN cause to SIP status code (and vice versa)
and sets the PSTN cause to SIP requests
sip-server Configure a SIP Server Interface
srv DNS SRV Query Type
suspend-resume Enable support for ISDN SUSPEND/RESUME
timers SIP Signaling Timers Configuration
transport Enable SIP UA transport for TCP/UDP
(config-sip-ua)# no tr ?
tcp Disable SIP User Agent in TCP Mode
udp Disable SIP User Agent in UDP Mode
(config-sip-ua)# no transport udp
(config-sip-ua)# max-forwards ?
<1-70> Number of max-forwards
(config-sip-ua)# max-forwards 15
(config-sip-ua)# exit
(config)# exit
# sh sip-ua status
SIP User Agent Status
SIP User Agent for UDP : DISABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
```

Reason Header will override Response/Request Codes: DISABLED

```
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Timespec line (t=) required
  Media supported: audio image
  Network types supported: IN
  Address types supported: IP4
  Transport types supported: RTP/AVP udpt1
# show sip-ua register status
Line      peer      expires (sec)  registered
=====
4101      20001     120            yes
4102      20005     120            yes
```

Cisco Router Challenge 236

Outline

This challenge involves SIP Voice Service Configuration for hair-pin calls for all dial-peers.

Objectives

The objectives of this challenge are to:

- Allow hairpinned calls for all dial peers with redirect **ip2ip**.
- Set the IP address for all SIP traffic as the local loopback.
- Define that the gateway acts as a registrar server.

Commands

```
> enable
# config t
(config)# sip-ua
(config-sip-ua)# redirect ip2ip
(config-sip-ua)# sip
(config-sip-ua)# bind control source-interface loopback10
(config-sip-ua)# registrar server expires max 1000 min 500
(config-sip-ua)# exit
(config)# exit
# sh sip-ua status
```

Example

```
> enable
# config t
(config)# voice service voip
(config-sip-ua)# redirect ip2ip
(config-sip-ua)# sip
(config-sip-ua)# bind control source-interface loopback10
(config-sip-ua)# registrar server expires max 1000 min 500
(config-sip-ua)# exit
(config)# exit
```

```
# sh sip-ua status
```

Cisco Gateway and Gatekeeper

SIP Gateway

The most up-to-date version of this test is at:

<http://networksims.com/ga04.html>

Cisco Router Challenge 237

Outline

This challenge involves configuration of the supervisory tone disconnect

Objectives

The objectives of this challenge are to:

- Define CP tone.
- Define timeouts for wait-release and call-disconnect.
- Define supervisory disconnect.

Commands

```
> enable
# config t
(config)# voice 1/0/0
(config-voiceport)# timeouts wait-release 10
(config-voiceport)# timeouts call-disconnect 10
(config-voiceport)# cptone us
(config-voiceport)# supervisory disconnect dualtone mid-call
```

Example

```
> enable
# config t
(config)# voice 1/0/0
Router(config-voiceport)# ?
Voice-port configuration commands:
  battery-reversal      Enable FXS battery-reversal generation
  bearer-cap            Specify the bearer capability
  busyout               Configure busyout trigger event & procedure
  caller-id             Configure port caller id parameters
  comfort-noise         Use fill-silence option
  connection            Specify Trunking Parameters
```

```

cptone          Configure voice call progress tone locale
default        Set a command to its defaults
description    Description of what this port is connected to
disc_pi_off    Close voice path when disconnect with PI received
disconnect-ack FXS sending disconnect acknowledge
echo-cancel    Echo-cancellation option
exit           Exit from voice-port configuration mode
impedance      Specifies the terminating impedance of the interface
input          Configure input gain for voice
music-threshold Threshold for Music on Hold
mwi            Enable MWI on this port
no             Negate a command or set its defaults
non-linear     Use non-linear processing during echo cancellation
output         Configure output attenuation for voice
playout-delay  Configure voice playout delay buffer
ren            Ringer Equivalence Number
ring           Ring frequency Parameters
shutdown       Take voice-port offline
signal         The signaling type for the interface FXS or FXO
snmp           Modify SNMP voice port parameters
station-id     Configure station ID
supervisory    Configure supervisory disconnect lcfo
threshold      Threshold [noise] for voice port
timeouts       Configure voice timeout parameters
timing         Configure voice timing parameters
translate      Translation rule
translation-profile Translation profile
trunk-group    Configure interface to be in a trunk group
voice-class    Set voiceport voice class control parameters

(config-voiceport)# timeouts ?
call-disconnect Call Disconnect Timeout after Destination Hangs Up in
seconds
hookflash-in    Define hookflash-in delay in milliseconds
initial         Initial Timeout duration in seconds
interdigit      Interdigit Timeout duration in seconds
power-denial    Duration for which power-denial is applied
ringing         Ringing no answer timeout duration in seconds
wait-release    Wait release timeout duration in seconds

(config-voiceport)# timeout w ?
<1-3600> seconds
infinity infinite timeout

(config-voiceport)# timeouts wait-release 10
(config-voiceport)# timeout call ?
<0-120> seconds
infinity infinite timeout
(config-voiceport)# timeouts call-disconnect 10
(config-voiceport)# cp ?
locale 2 letter ISO-3166 country code

AR Argentina      IS Iceland        PE Peru
AU Australia      IN India          PH Philippines
AT Austria        ID Indonesia     PL Poland
BE Belgium        IE Ireland       PT Portugal
BR Brazil         IL Israel        RU Russian Federation
CA Canada         IT Italy          SA Saudi Arabia
CN China          JP Japan         SG Singapore
CO Colombia       JO Jordan        SK Slovakia
C1 Custom1        KE Kenya       SI Slovenia
C2 Custom2        KR Korea Republic ZA South Africa
CY Cyprus         LB Lebanon       ES Spain
CZ Czech Republic LU Luxembourg     SE Sweden

```

DK Denmark	MY Malaysia	CH Switzerland
EG Egypt	MX Mexico	TW Taiwan
FI Finland	NP Nepal	TH Thailand
FR France	NL Netherlands	TR Turkey
DE Germany	NZ New Zealand	GB United Kingdom
GH Ghana	NG Nigeria	US United States
GR Greece	NO Norway	VE Venezuela
HK Hong Kong	PK Pakistan	ZW Zimbabwe
HU Hungary	PA Panama	

```

(config-voiceport)# cptone us
(config-voiceport)# su ?
    disconnect  Configure supervisory disconnect lcfo
(config-voiceport)# supervisory disconnect dualtone mid-call

```

Cisco Router Challenge 238

Outline

This challenge involves customizing the Supervisory Disconnect Tone.

Objectives

The objectives of this challenge are to:

- Define a tone class.
- Apply the tone class for tone details.

Commands

```

> enable
# config t
(config)# voice 1/0/0
(config-voice-port)# supervisory disconnect dualtone pre-connect voice-class 5
(config-voice-port)# exit
(config)# voice class dualtone 5
(cfg-dual-detect)# freq-max-power 20
(cfg-dual-detect)# freq-min-power 10
(cfg-dual-detect)# cadence-variation 10
(cfg-dual-detect)# freq-max-deviation 10
(cfg-dual-detect)# freq-max-delay 10

```

Example

```

> enable
# config t
(config)# voice 1/0/0
(config-voice-port)# supervisory disconnect dualtone pre-connect voice-class 5
(config-voice-port)# exit
(config)# voice class dualtone 5
(cfg-dual-detect)# ?
VOICECLASS configuration commands:
  cadence-variation  Cadence variation allowed
  exit              Exit from voice class configuration mode

```

```

freq-max-delay      Timing difference between two frequencies
freq-max-deviation  Maximum frequency deviation allowed for each frequency
freq-max-power      Absolute value of upper limit for tone power per
                    frequency
freq-min-power      Absolute value of lower limit for tone power per
                    frequency
freq-power-twist    The difference between the power of two frequencies
help                Description of the interactive help system
no                 Negate a command or set its defaults
(cfg-dual-detect)# freq-max-power ?
<0-20> Unit is dbm0

(cfg-dual-detect)# freq-max-power 20
(cfg-dual-detect)# freq-min-power ?
<10-35> Unit is -dbm0
(cfg-dual-detect)# freq-min-power 10
(cfg-dual-detect)# cadence-variation ?
<0-200> Unit is 10 ms
(cfg-dual-detect)# cadence-variation 10
(cfg-dual-detect)# freq-max-dev ?
<10-125> Unit in Hz
(cfg-dual-detect)# freq-max-deviation 10
(cfg-dual-detect)# freq-max-delay ?
<10-100> Unit is 10 ms
(cfg-dual-detect)# freq-max-delay 10

```

Cisco Router Challenge 239

Outline

This challenge involves configuring FGD for T1.

Objectives

The objectives of this challenge are to:

- Configure T1 for the linecode (B8ZS) and framing (ESF).
- Define E&M-FGD and FGD-ENNA.

Commands

```

> enable
# config t
(config)# controller t1
(config-controller)# framing esf
(config-controller)# linecode b8zs
(config-controller)# pri-group timeslots 1-10
(config-controller)# no shutdown
(config-controller)# clock source line
(config-controller)# ds0-group 1 t 1-4 type e&m-fgd
(config-controller)# ds0-group 2 t 5-24 type fgd-enna

```

Example

```
> enable
# config t
(config)# controller t1
Router(config-controller)# ?
Controller configuration commands:
  cablelength      Specify the cable length for a DS1 link
  channel-group    Specify the timeslots to channel-group mapping for an
                  interface
  clock            Specify the clock source for a DS1 link
  default          Set a command to its defaults
  description      Controller specific description
  ds0-group        DS0 time slots that make up a logical voice port
  exit            Exit from controller configuration mode
  framing          Specify the type of Framing on a DS1 link
  help            Description of the interactive help system
  linecode        Specify the line encoding method for a DS1 link
  loopback        Put the entire T1 line into loopback
  no              Negate a command or set its defaults
  pri-group        Configure the specified timeslots for PRI
  shutdown        Shut down a DS1 link (send Blue Alarm)
(config-controller)# framing ?
  esf  Extended Superframe
  sf   Superframe
(config-controller)# framing esf
(config-controller)# linecode ?
  ami  AMI encoding
  b8zs B8ZS encoding
(config-controller)# linecode b8zs
(config-controller)# pri-group timeslots 1-10
(config-controller)# no shutdown
(config-controller)# clock source line
(config-controller)# ds0-group ?
  <1-11> ds0-group-number

(config-controller)# ds0-group 1 ?
  timeslots  number of timeslots

(config-controller)# ds0-group 1 t ?
  <1-24> timeslot-list

(config-controller)# ds0-group 1 t 1-4 type ?
  e&m-delay-dial
  e&m-fgd
  e&m-immediate-start
  e&m-wink-start
  ext-sig
  fgd-eana
  fxo-ground-start
  fxo-loop-start
  fxs-ground-start
  fxs-loop-start
(config-controller)# ds0-group 1 t 1-4 type e&m-fgd
(config-controller)# ds0-group 2 t 5-24 type fgd-enna
```

Cisco Gateway and Gatekeeper

Circuits

The most up-to-date version of this test is at:

<http://networksims.com/ga05.html>