

CCVP (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 configuratio for the registration of analog phones with a redundanct 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 udptl
# 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         JP Japan         SG Singapore
CN China          JO Jordan        SK Slovakia
CO Colombia       KE Kenya        SI Slovenia
C1 Custom1        KR Korea Republic ZA South Africa
C2 Custom2        LB Lebanon       ES Spain
CY Cyprus         LU Luxembourg    SE Sweden
CZ Czech Republic

```

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>