Contents

Introduction

This is the basic config for any ISR router 2800 / 3800 series with PRI connection to PSTN / PBX and H.323 protocol on IP side.

Design

The example toplogy:

Tandberg VT - E1 PRI - PSTN - E1 PRI - ISR router - H.323 GK - Tandberg VT

Configuration

1. First we need to register GW and the video endpoint to the GK, we can run GK feature on the same router if we have IOS with GK feature for example - C2800NM-IPVOICE_IVS-M. The most simple setup will be if we register endpoing with full E164 number and GW with the tech prefix, chousen in a way it will allow dialing to PSTN without any number transformation. Assume PSTN will requre 00 for international call, in this case we can use the following config on GK:

!

```
gatekeeper
 zone local video-98 mappets.com 10.52.218.98
 no use-proxy video-98 default inbound-to terminal
 no use-proxy video-98 default outbound-from terminal
no shutdown
```

!

There is no need to configure zone prefixes as GW will register with the tech prefix. And VT will register with the full E164 number, so GK will always be able to resolve the them to IP. For VT we need to allow direct calls to / from H.323 GW, so we disabling proxy requirement.

The matching GW config will be:

!

```
voice service voip
 h323
  emptycapability
  h225 h245-address on-connect
  call start slow
```

```
  h245 tunnel disable
  h245 timeout olc 30
```

!

Formally you will need only slow start and OLC timeout tuning. Because it may take some time for all ISDN secondary calls to get connected and for the BAS and codec negotiation to start, we need allow voip side to wait for some time. The 30 second timeout is reasonable setting, if we are talking about some VT like Tandberg or Polycom, they will have that settings by default. But if you have CallManager on IP side, you need to tune that timeouts via Service parameters:

H245 TCS Timeout 30

Media Exchange Interface Capability Timer 30

Media Exchange Timer 30

You may also consider to un-check the check-box on the GW config page:

"Wait for Far End H.245 Terminal Capability Set"

It usually good practice explicitly specify which interface will be used for H.323 signaling and if you register the GW to GK you will need to configure:

!

```
interface FastEthernet0/0
 ip address 10.52.218.91 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id video-98 ipaddr 10.52.218.98 1719
 h323-gateway voip h323-id ios-91-to-98
 h323-gateway voip tech-prefix 00
 h323-gateway voip bind srcaddr 10.52.218.91
```

!

```
gateway
```

!

Next we can proceed with the H.320 video GW configuration.

1. Specify the card type you will have:

```
card type e1 1 1
```

2. As video calls are actually the DATA calls by the nature of video content, we need to ensure the clocking is configured corectcly:

```
network-clock-participate slot 1
network-clock-participate wic 1
```

! ! Choose the one which required by your hardware setup. !

```
network-clock-select 1 E1 1/0
```

3. You will need to specify the ISDN switch type before you can proceed with PRI group config on the controller:

```
isdn switch-type primary-net5
```

4. The controller config is the same as for any PRI connection:

```
controller E1 1/0
 pri-group timeslots 1-31
 description >>>>> Test of NEW BT PRI ports - out 338898 *** <<<<<

!
```

5. Now we need to configure D-channel to accept / send Data calls:

```
!

interface Serial1/0:15
 description >>>>> Test of NEW BT PRI ports - out 338898 *** <<<<<
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn map address .* plan unknown type unknown
 isdn negotiate-bchan resend-setup
 isdn bchan-number-order ascending
 isdn sending-complete
 isdn integrate calltype all
 no cdp enable

!
```

6. Now we can proceed with the preparation for the dial-peer config, first let define the codecs we will use on IP side. Please keep in mind that GW will actually do the filtering of the codes it will announce to the endpoints on both sides.

```
!

voice class codec 1000
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
 codec preference 3 g728
 codec preference 4 g722-64
 codec preference 5 g722-56
 codec preference 6 g722-48
 video codec h261
 video codec h263
 video codec h263+
 video codec h264

!
```

7. This part is required to accept the incoming video calls, we need to have to provide the secondary numbers to the calling party, so they can call us on that numbers to add more bandwidth to the video call, that numbers will be sent out during ISO 13871 bonding stage. According to ISO 13871 only 7 digits will be sent. So there is no need or sense to configure more then 7 digits in the pool config. But you can configure less then 7 digits. The secondary called number will be crated from the primary called number by replacing the last digits with the numbers from the pool. So in case when only last 1 or 2 digits are different, you pool may

have only 1 or 2 digit numbers in it.

!

```
voice class called number pool 3151
 index 1 315140 - 315149
```

8. Now we need to apply that pool to the voice port to activate it:

!

```
voice-port 1/0:15
 voice-class called-number-pool 3151
```

!

9. Now we are ready to proceed with the dial-peer configuration we need 4 of them, two incoming and two outgoing. It is possible to use one POTS dial-peer as incoming and outgoing at the same time, but we will provide config for two of them for the sake of clarity:

!

```
dial-peer voice 99 pots
 description default incoming video dial-peer
 information-type video
 incoming called-number 313300
 bandwidth maximum 384
 direct-inward-dial
```

!

!

```
dial-peer voice 9001 pots
 description default outgoing video dial-peer
 destination-pattern 00
 information-type video
 bandwidth maximum 384
 port 1/0:15
 forward-digits all
```

!

10. here is the example of the VOIP dial-peer, now we configured it as both - incoming and outgoing:

!

```
dial-peer voice 770 voip
 description default incoming/outgoing voip dial-peer for all calls from GK w
 destination-pattern 701...
 voice-class codec 1000
 session target ras
 incoming called-number 00.
 dtmf-relay h245-alphanumeric
```

!

Related show Commands

This section provides information you can use to confirm your configuration is working properly.

Certain show commands are supported by the Output Interpreter Tool (registered customers only), which allows you to view an analysis of show command output.

```
show network-clocks
show isdn status
show dial-peer voice summary
show voice call status
show voice dsp
```

There also some useful commands to add and they can be used on All routers in production as well:

```
service nagle
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec
service timestamps log datetime msec
```

!

```
logging message-counter syslog
logging queue-limit 100000
logging buffered 5000000
logging rate-limit 10000
no logging console
```

And some very useful aliases :)

!

```
alias exec c conf t
alias exec r sh run
alias exec i sh ip ro
alias exec ib sh ip int brie
alias exec gs show run | s gatekeeper
alias exec ge sho gatekeeper end
alias exec sg sho gatekeeper
alias exec rs sh run | s
alias exec ri sh run | i
alias exec rb sh run | b
alias exec dp sh run | be dial-p
```

!

In case you will need to contact TAC, you can get the debugs following this template:

1. 1. Add to the config or make sure you have it already:

!

```
service sequence-numbers
service timestamps debug datetime msec
service timestamps log datetime msec
```

!

```
logging buffered 5000000
logging rate-limit 10000
logging queue-limit 100000
no logging console
```

!

1. 2. Then start logging to a file from your telnet application and get:

!

```
U all
Term no mon
Term len 0
```

!

```
Sh ver
Sh run
```

!

1. 3. Then enable debugs (you can copy / paste all of them to the telnet session window)

!

```
Deb voip ccapi inout
Deb isdn q931
Deb h225 q931
Deb h225 asn1
Deb h245 asn1
Deb cch323 h245
deb voip tsp dialpeer
deb voip h221 raw decode

Sh deb
Term no mon
```

Clear log before placing call:

```
Clear logg
```

Then place the call and get content of the buffer as:

```
Show logg
```

Please send the log in simple TXT format attached to the e-mail in simple TXT format with extension".TXT". Please include attach@cisco.com in CC to get logs and e-mail attached to case notes.

Enjoy :)

Show running-config

```
Add show running config of your device
```

Related Information

Technical Support & Documentation - Cisco Systems