Cisco Voice Gateways

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Jonny Martin
jonny@jonnynet.net
Voice Gateways

- Any device with one or more TDM PSTN interfaces on them
  - TDM - Time Division Multiplexing (i.e. traditional telephony)
  - PSTN - Public Switched Telephone Network
  - To be really useful, gateways also need an IP interface on them
- Many vendors, we’ll concentrate on Cisco IOS based voice gateways
- Both analog and digital interfaces, we’ll look at the more common ones
Interface Types - Digital

- ISDN primary rate circuits (there are others, but we will look at ISDN)

- E1 (primarily used in Europe and Oceania)
  - 2 Mbit/s bearer
  - 32x 64kbit/s channels. 30 for voice, 1 for signalling (timeslot 16), 1 framing

- T1 (primarily used in North America)
  - 1.5 Mbit/s bearer
  - 24x 64kbit/s channels. 23 for voice, 1 for signalling (timeslot 24)

- Common interfaces for ISP dial-in, PBX to carrier trunks, etc.
Interface Types - Digital

• Basic Rate ISDN
  • 144kbit/s bearer
  • 2x 64kbit/s channels + 1x 16kbit/s signalling channel
• 2B + D
  • B channels = 64kbit/s voice/data channels
  • D channel(s) = signalling data channels
Interface Types - Analog

- Only really two types:

- FXO interface - plugs into your telco (Foreign eXchange central Office)
  - uses FXS signalling!

- FXS interface - plugs into a telephone. e.g. ATAs (Foreign eXchange Station)
  - uses FXO signalling!

- Uses analog signalling, limited to one DDI per line

- Signalling is generally more ambiguous and harder to work with than digital signalling
AS5300 / AS5350 / AS 5400

- Multi-port E1/T1 access servers
- Popular ISP dial-in boxes
- 5300 - can be used for VoIP when loaded with DSP cards
- 5350/5400 has universal ports - modem or VoIP
- Dial-up ISPs often well placed to provide VoIP services
  - POPs in many locations, with the right hardware!
IOS Voice Configuration

• For VoIP we need to configure:
  • voice-port - the voice ‘interface’
    • FXS / FXO - e.g. voice-port 1/0/0
    • E1/T1 signalling channel - e.g. voice-port 1/0:D
  • dial-peer - tells the gateway how to connect voice ports to VoIP call legs

• For E1/T1 links we also need to configure the physical bearer
  • controller E1 / controller T1
  • interface serial 0:15 (the signalling timeslot for an E1, 0:23 for T1)
This configuration works with Telecom NZ E1 circuits

```text
isdn switch-type primary-net5
controller E1 0
  clock source line primary
  pri-group timeslots 1-10,16  ! note, timeslots count from 1.
  description Link to Telecom

interface Serial0:15  ! note, serial channels count from 0.
  no ip address
  isdn switch-type primary-net5
  isdn incoming-voice modem  ! treats incoming calls as modem or voice
  !  ! rather than data

voice-port 0:D
  echo-cancel coverage 64
  cptone NZ  ! returns NZ progress tones
  bearer-cap Speech
```

T1 Configuration

!
isdn switch-type primary-ni
!
!
controller T1 1/0
  framing esf
  linecode b8zs
  pri-group timeslots 1-24
!
!
interface Serial1/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn incoming-voice modem
!
!
voice-port 1/0:D
  echo-cancel coverage 64
! default cptone is US
!
FXS / FXO Configuration

! Some useful settings
!
voice-port 1/0/0
  no comfort-noise                   ! needs ‘no vad’ on VoIP dial-peer
  cptone NZ
  timeouts interdigit 3             ! timeout when gathering dialled digits
  description Analog phone line
!

! Or, if you’re just having a play, the defaults will work:
!
voice-port 1/0/1
!
Dial Peers

• Basic building block on Cisco voice gateways, the dial-peer

• All calls consists of at least two call legs:
  • Originating device to originating gateway (POTS)
  • Originating gateway to IP network (VoIP)
  • ...and/or
  • IP network to destination gateway
  • Destination gateway to destination device
Dial Peers ...ctd

- Most hardware will also allow TDM switching, i.e. POTS to POTS
- But not typically VoIP media proxying (i.e. no VoIP-VoIP)
Dial Peer Syntax

! POTS dial peer
!

dial-peer voice tag pots
destination-pattern number
port voiceport#
other configurable options
!

! VoIP dial peer
!

dial-peer voice tag voip
destination-pattern number
session target data address
other configurable options
!

! Destination pattern = E.164 number (i.e. a telephone number)
Dial Peer Matching

- When a call is made, IOS will select the appropriate dial-peer for an outbound leg depending on call direction
  - voip --> pots
  - pots --> voip
- Longest match for destination-pattern is chosen
- If multiple longest matches exist, the dial-peer with the lowest preference will be chosen
Example POTS Dial Peers

! Outbound send-everything-to-the-pstn POTS dial-peer:
!
dial-peer voice 1 pots
    destination-pattern T  ! T = digit timeout, i.e. any string of digits
direct-inward-dial  ! allow incoming calls from the POTS port also
    port 0:D
!

! Only send numbers prefixed with 021 out the POTS port:
!
dial-peer voice 1 pots
    destination-pattern 021T  ! T = digit timeout, i.e. any string of digits
direct-inward-dial
    port 1:D
!

! Only send seven digit numbers prefixed by 04
!
dial-peer voice 1 pots
    destination-pattern 04....... ! . = a single digit
direct-inward-dial
    port 2:D
!
Example VoIP dial-peers

! Send calls to 4989560 to a VoIP PABX or phone at IP address a.b.c.d
!
dial-peer voice 44989560 voip
destination-pattern 4989560
session protocol sipv2
session target ipv4:a.b.c.d
dtmf-relay rtp-nte                   ! RFC2833 out of band DTMF signalling
codec g729br8
no vad
!

!
dial-peer voice 2001 voip
   huntstop                           ! Don’t search for a match past this dial-peer
   preference 2
   destination-pattern 2001
   session protocol sipv2
   session target ipv4:202.53.189.62
dtmf-relay rtp-nte
playout-delay mode fixed             ! sets a fixed jitter buffer, useful for Fax
codec g711ulaw
no vad                               ! always use this for fax!
!
Failover Routing

- Failover routing is achieved by ‘hunting’ on busy, no answer, and a myriad of other causes

- Works for both *pots* and *voip* dial-peers

- Use *preference* to step through dial-peers
  - 0 is best and the default, 9 is worst

- Use *huntstop* on the ‘last’ dial-peer

- Often used in conjunction with *translation-patterns* to ensure correct dial string for different trunks
Failover Example

Incoming POTS calls first try one VoIP server, then failover to another if that server doesn’t answer or is busy.

voice hunt user-busy
voice hunt no-answer

dial-peer voice 49896411 voip
destination-pattern 4989641
session protocol sipv2
session target ipv4:a.b.c.1
dtmf-relay rtp-nte
codec g711ulaw

! dial-peer voice 49896412 voip
huntstop
preference 1
destination-pattern 4989641
session protocol sipv2
session target ipv4:a.b.c.2
dtmf-relay rtp-nte
codec g711ulaw
!
Translation Patterns

• Used to translate called and calling numbers

• Uses basic translation rules to prepend / strip digits, translate one number into a completely different number

• Some basic examples...
Translation Pattern Examples

! strip 644 from the start of the number for numbers starting 6442 - 6449
!
translation-rule 100
  Rule 2 ^6442...... 2
  Rule 3 ^6443...... 3
  Rule 4 ^6444...... 4
  Rule 5 ^6445...... 5
  Rule 6 ^6446...... 6
  Rule 7 ^6447...... 7
  Rule 8 ^6448...... 8
  Rule 9 ^6449...... 9
!

! Prefix 04 to the beginning of any number
!
translation-rule 101
  Rule 1 ^.% 04
Failover Scenario

- Calls come in from PSTN
- Sent to PABX
- Eventually sent on to one of the phones
- What happens if the something breaks?
  - network
  - server
  - phone
Failover Scenario

- Create primary dialpeer from voice router to PABX
- Create secondary path to trombone back out the PSTN, after translating the called number to the oncall engineers cellphone number
- Now if any of the network fails, NOC calls can still get through to someone
Failover Implementation

! translate any number to 0212304323
!
translation-rule 120
  Rule 1 any 0212304323
translation-rule 121
  Rule 1 any [other engineers number]
!
dial-peer voice 100 voip
  destination-pattern 0800123456 ! our incoming NOC number
  preference 0
  session protocol sipv2
  session target ipv4: my.pabx.ip.address
!
dial-peer voice 110 pots
  destination-pattern 0800123456 ! only try this dialpeer if the above fails
  preference 1
  translate-outgoing called 120 ! translate the CALLED number
  port 0:D ! send the call out a PSTN voice port
!
Changing the failover target

- Write a script to change the failover target to the appropriate on-call engineer
  
  ```
  telnet voice-router
  enable
  conf t
  dial-peer voice 110 pots
     translate-outgoing called 12X
  exit
  exit
  ```

- This can be called by an existing script which changes a SMS or email target in NAGIOS, to change them in sync

- One translation rule per on-call engineer