Cisco Voice Gateways

PacNOG6 VoIP Workshop
Nadi, Fiji. November 2009

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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)
E1 Configuration

! This configuration works with Telecom NZ E1 circuits
!
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
!
!
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-npe
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-npe
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
Translation Pattern Examples ... ctd

! translate any number to 0212304323
!
translation-rule 120
  Rule 1 any 0212304323

! Normalise numbers into a standard format
!
translation-rule 150
  Rule 1 ^644498.... 498 ! 6444981234 --> 4981234
  Rule 2 ^04498.... 498 ! 044981234 --> 4981234
  Rule 3 ^00644498.... 498 ! 006444981234 --> 4981234
!
Apply the Translation Pattern

! dial-peer voice 44989560 voip
destination-pattern 4989560
translate-outgoing calling 100 ! translated the CALLING number
translate-outgoing called 200 ! translate the CALLED number
session protocol sipv2
session target ipv4:203.114.148.130
dtmf-relay rtp-nce
codec g711ulaw
no vad
!