Cisco Voice Gateways

PacNOG 3 VoIP Workshop
June 2007, Cook Islands

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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                      ! rather than data
  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
  cptone NZ
  timeouts interdigit 3
  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-n-te          ! 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-n-te
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
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-nte
codec g711ulaw
no vad
!
!