Cisco Voice Gateways

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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
!                                   ! 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

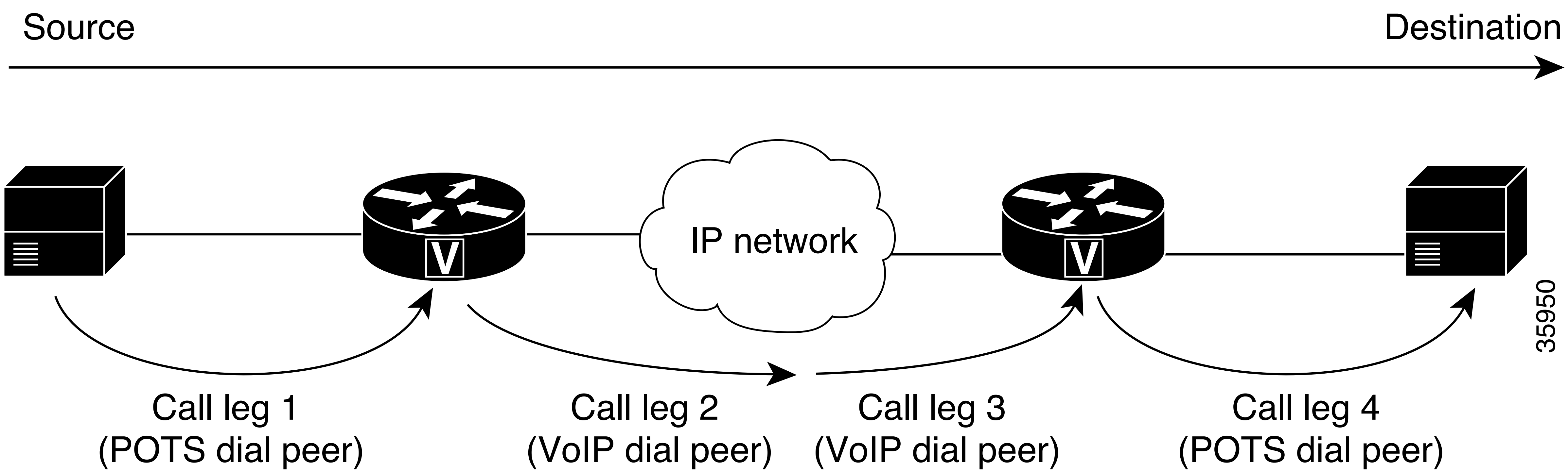
```
! Some useful settings
!
voice-port 1/0/0
 no comfort-noise                    ! needs ‘no vad’ on VoIP dial-peer
 cptone NZ
 timeouts interdigit 3               ! timeout when gathering dialled digits
 description Analog phone line
!


! 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

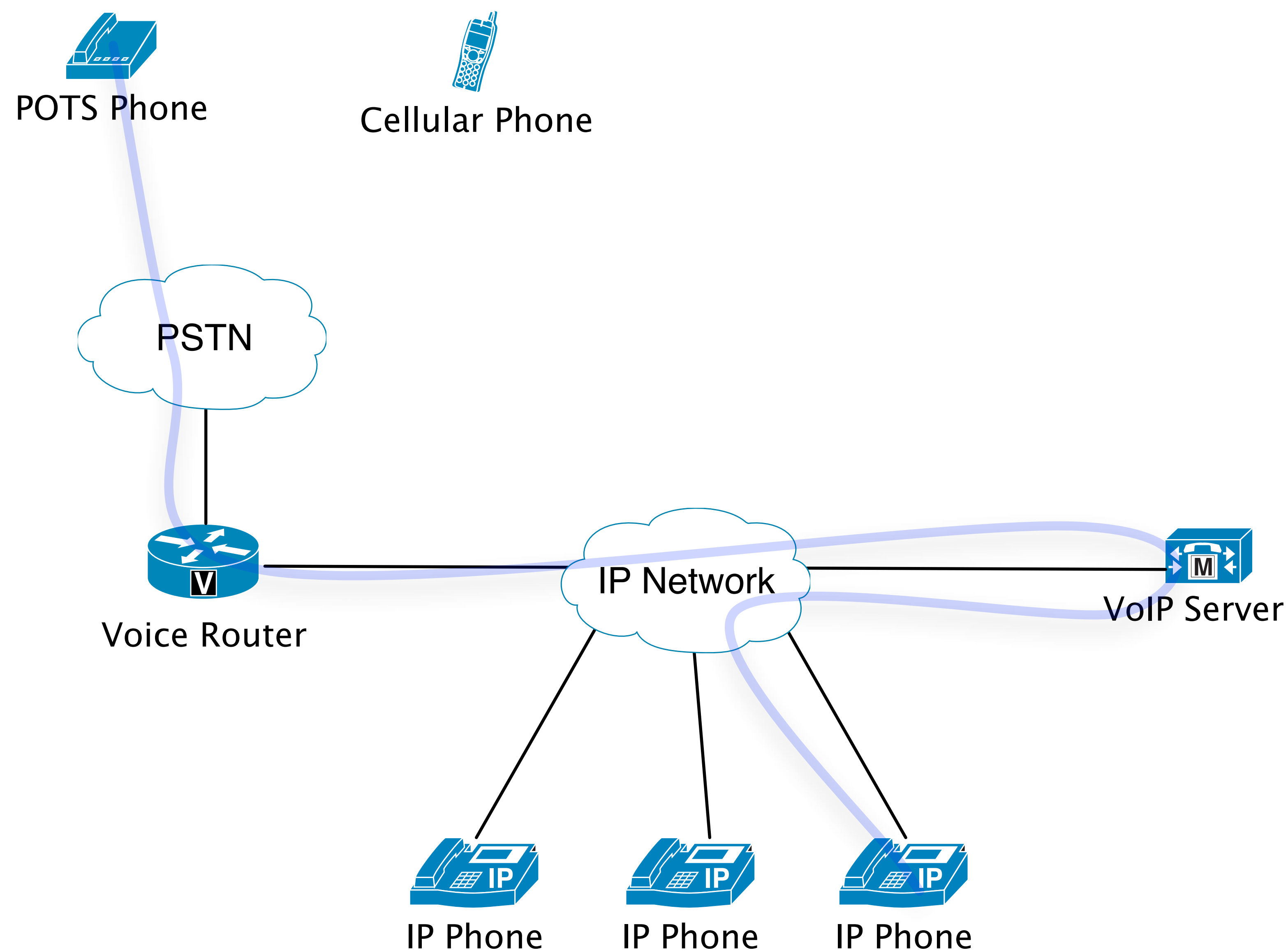
- Used to translate called and calling numbers
- Uses basic translation rules to prepend / strip digits, translate one number into a completely different number
- Some basic examples...

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
```

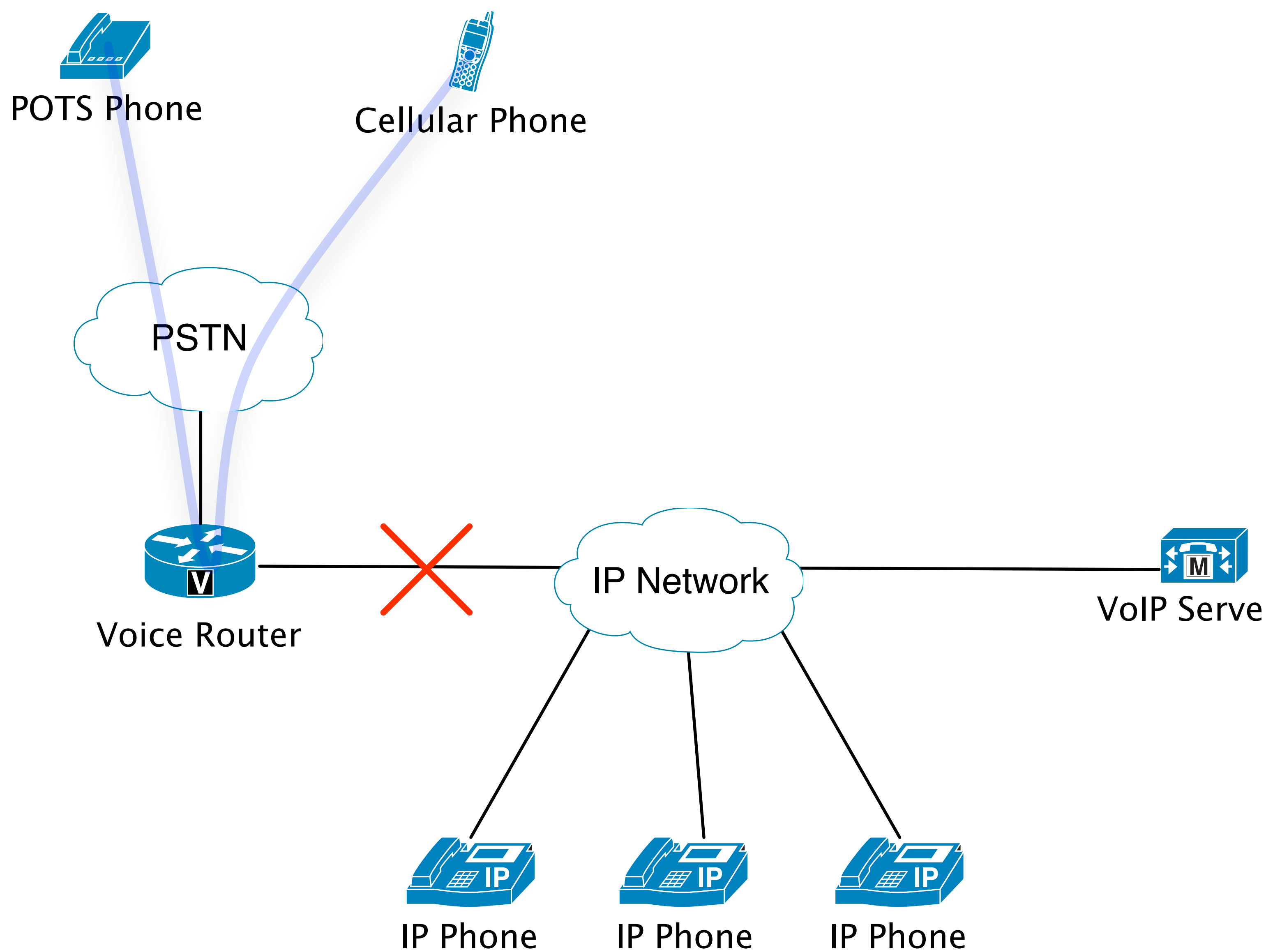
Failover Scenario

- 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
 preference 1                           ! only try this dialpeer if the above fails
 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