Distributed IP-PBX

Tele-convergence of IP-PBX / PSTN / FAX / legacy PABX
And
Distributed network approach with area isolation
AGENDA

• Background
• Application and Appliances
• Architecture & Benefits
• Case Study
• Key Findings
BACKGROUND
An organization with a headquarter located centrally which had 15+ zonal areas and had more than 150+ branches located remotely with zones.

What do they have and practice:
- They have a central internet connectivity.
- They had zones connected with E1.
- They had a central IP-PBX soft switch.
- They had application systems, mailing etc. running centrally.
- Had PABX for inter-telecommunication and PSTN dropped to call out and have calls in for all places.
BACKGROUND: Realization

- Having communication through Datacom.
- Having independent system for zones to be operational even if data link unavailable.
- Having PSTN to be trunked to remote locations.
- Having PABX lines to be trunked to few locations.

So they asked for:

1. MUX for all locations.
2. Routers to integrate with data & MUX.
3. PABX unit for independent dialer.
4. Microwave setups for connectivity.
BACKGROUND: Realization

Complexity !!!
Expensive !!!
Mess !!!
Burden !!!
BACKGROUND: Realization

- Packet switching.
- Layer-3 network for connectivity nodes.
- Single box solution.
- Integrated communication equipment.
- Complexity minimization.
BACKGROUND: Telecommunication

- A huge mix of evolved technologies.
- Divided in circuit and packet switched network.

Visualization is from the Opte Project of the various routes through a portion of the Internet.
BACKGROUND: Circuit Vs Packet switching

Figure 9.1
Circuit Switching

Figure 9.2
Packet Switching
BACKGROUND: PABX, PSTN & IP-PBX

- **PBX**: Switchboard operator managed system using cord circuits.

- **PABX (Private Automatic Branch Exchange)**: Availability of electromechanical switches gradually replaced manual switchboard - PBX.

- **PSTN (Public Switched Telephone Network)**: Aggregates circuit-switched telephone networks that are operated by national, regional, or local telephony operators.

- **IP PBX**: Handles voice signals over Internet protocol, bringing benefits for computer telephony integration (CTI).
**BACKGROUND: Codecs**

<table>
<thead>
<tr>
<th>Codecs</th>
<th>Payload Bitrate</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>64 kbit/s</td>
</tr>
<tr>
<td>G.726</td>
<td>16, 24 or 32 kbit/s</td>
</tr>
<tr>
<td>G.723.1</td>
<td>5.3 or 6.3 kbit/s</td>
</tr>
<tr>
<td>G.729</td>
<td>8 kbit/s</td>
</tr>
<tr>
<td>GSM</td>
<td>13 kbit/s</td>
</tr>
</tbody>
</table>

G.711 is freely available as well gives highest quality.

G.72x is proprietary codecs required purchasing.

GSM is very popular due to good CPU and bandwidth tradeoff.
APPLICATION & APPLIANCES
APPLICATION & APPLIANCES: Contents

- Asterisk
- Cisco voice enabled routers
- Traditional phone sets
- IP Phone
- PABX stations
APPLICATION & APPLIANCES: Asterisk

• A software implementation of a telephone private branch exchange (PBX).
• Allows attached telephones to make calls to one another.
• Connects other telephone services, such as the PSTN and VoIP services through media gateway.

For features and details visit
http://www.asterisk.org/get-started/features
APPLICATION & APPLIANCES:
Cisco voice enabled routers

Integrated services
• Routing
• PBX
APPLICATION & APPLIANCES:
Traditional phone, IP-phone & PABX station
Considering Scenarios

• Nationwide distributed work areas.
• Implemented legacy PABX for internal communication.
• Data & Internet connectivity.
• Single box solution.
• Area with lower resources considering rural area and environment.
Common Organization Scenario
Transformation Scenario 1
Transformation Scenario 2
CASE SCENARIO:
Zone 1 Network Diagram
Why Such integration?

1. Using the customers existing circuit setup PABX or PSTN.
2. Using cross site communication.
4. Going ahead in advanced communication technology.
5. Cost benefit while transforming technology.
CASE STUDY
CASE STUDY: A sample Lab for such scenario

Prime Integrations:
1. Asterisk Server.
2. Cisco Routers.
3. Legacy PABX integration.
4. PSTN connectivity integration.
CASE STUDY: Asterisk Server Configuration

vi /etc/asterisk/extensions.conf

```
[cmeterunk]
Exten => _X.,1,Set(do_Voicemail=no)
Exten => _X.,n,NoOp(${CALLERID(num)})
Exten => _X.,n,Dial(SIP/\${EXTEN})
Exten => _X.,n,NoOp(${HANGUPCAUSE} DAN ${DIALSTATUS})
Exten => _X.,n,Hangup()

###Router01###
Exten => _20XX.,1,Set(do_Voicemail=no)
Exten => _20XX.,n,NoOp(${CALLERID(num)})
Exten => _20XX.,n,Dial(SIP/Router01/\${EXTEN},,tTw)
Exten => _20XX.,n,NoOp(${HANGUPCAUSE} DAN ${DIALSTATUS})
Exten => _20XX.,n,Hangup()

###Router02###
Exten => _21XX.,1,Set(do_Voicemail=no)
Exten => _21XX.,n,NoOp(${CALLERID(num)})
Exten => _21XX.,n,Dial(SIP/Router02/\${EXTEN},,tTw)
Exten => _21XX.,n,NoOp(${HANGUPCAUSE} DAN ${DIALSTATUS})
Exten => _21XX.,n,Hangup()
```

vi /etc/asterisk/sip.conf

```
[Router01]
type=friend
host=10.11.121.2
dtmfmode=rfc2833
relaxdtmf=yes
canreinvite=no
insecure=port,invite
context=cmeterunk
quality=yes
nat=yes
Disallow=all
Allow=ulaw
Allow=alaw

[Router02]
type=friend
host=10.11.121.6
dtmfmode=rfc2833
relaxdtmf=yes
canreinvite=no
insecure=port,invite
context=cmeterunk
quality=yes
nat=yes
Disallow=all
Allow=ulaw
Allow=alaw
```

Context "cmeterunk" actually allows the router peer configurations at sip.conf file. As for router01 information in sip.conf it gives informations for peering router like IP, codec etc. As when the dial pattern needed the use each other simultaneously. As for router01 dial pattern you can find that it is seeking Router01 name from sip.conf.
CASE STUDY: **Cisco Router Configuration**  
(Basic Configuration for voice)

```plaintext
voice service voip // Declaring the voice service over which mode. In our case its IP.
   allow-connections h323 to h323
   allow-connections h323 to sip
   allow-connections sip to h323
   allow-connections sip to sip

   supplementary-service h450.12 advertise-only // Common Information Additional Network Feature for H.323
   fax-protocol t38 is-redundancy 0 bi-redundancy 0 failback pass-through g711ulaw // Declaring fax protocol enabling low and high signal and giving option if fax protocol not available to go through an audio codec

   sip // sip configuration
      reflex disable // Reliable provisional response support disabled to stop error code
      registrar server expires max 1200 min 60 // Enabling SIP registry server and mentioning its expire time

   voice class codec 1 // declaring a codec group tag
      codec preference 1 g711ulaw
      codec preference 2 g711ulaw

   voice register global // Global registry information declaration
      mode cme // mode defining to CME of Cisco
      source-address 10.11.121.2 port 5060 // Defining a registry server IP and port
      max-dn 10 // maximum dial no defining
      max-pool 10 // maximum pool defining
      tftp-path flash: // configuration loaded from flash with tftp
      create-profile sync 0004028852090405 // Creating profile for IP phones

   ccm-manager application redundant port 5060 // Call manager application redundant port declaration

   dspfarm profile 1 transcoding universal // Digital Signal Processor (DSP) profile for codec transformation for IP to IP media gateway
      description Transcoding
      codec g711ulaw
      codec g711alaw
      maximum sessions 5
      associate application SCCP

   sip-ua // SIP user agent informations
      registrar ipv4:10.11.121.2 expired 3600
      sip-server ipv4:10.11.121.2

   telephony-service // CUCME configuration for router
      max-ephones 10
      max-dn 10
      system message #Router01
      time-zone 21
      max-conferences 8 gain -6
      transfer-system full-consult
```

This case study covers the basic configuration for voice services on a Cisco router, detailing the setup of voice services, in-band signaling, and the configuration of a DSP profile for codec transformation.
CASE STUDY: Cisco Router Configuration
( Registering IP Phone / Pots Phone / Voice peer )

dial-peer voice 1 pots // POTS number declaration
destination-pattern 2002
incoming called-number .%
port 0/0/0
!
dial-peer voice 2 voip // Peering with asterisk server
description Router01-Asterisk
destination-pattern 1...
session protocol sipv2
session target ipv4:10.11.120.100:5060
dtmf-relay sip-notify
codec g711ulaw
no vad
!
dial-peer voice 3 voip // Peering with routers
description Router01-Router02
destination-pattern 21..
session protocol sipv2
session target ipv4:10.11.121.6:5060
dtmf-relay sip-notify
codec g711ulaw
no vad

voice register dn 1 // Dial number declaration
number 2001
allow watch
name 2001
!
voice register pool 1 // Pool profile information for the number
id mac 0030.4F7B.E3F9
number 1 dn 1
dtmf-relay sip-notify
username 2001 password 123456
codec g711ulaw
no vad
!
dial-peer voice 1 pots // POTS number declaration
destination-pattern 2002
incoming called-number .%
port 0/0/0
!
dial-peer voice 2 voip // Peering with asterisk server
description Router01-Asterisk
destination-pattern 1...
session protocol sipv2
session target ipv4:10.11.120.100:5060
dtmf-relay sip-notify
codec g711ulaw
no vad
!
dial-peer voice 3 voip // Peering with routers
description Router01-Router02
destination-pattern 21..
session protocol sipv2
session target ipv4:10.11.121.6:5060
dtmf-relay sip-notify
codec g711ulaw
no vad
CASE STUDY: Legacy PABX integration

! dial-peer voice 1 pots
  preference 1
  destination-pattern 2000
  incoming called-number .%
  port 0/0/0
!
! dial-peer voice 2 pots
  preference 2
  destination-pattern 2000
  incoming called-number .%
  port 0/0/1
!
! dial-peer voice 3 pots
  preference 3
  destination-pattern 2000
  incoming called-number .%
  port 0/0/2
!
! dial-peer voice 4 pots
  preference 4
  destination-pattern 2000
  incoming called-number .%
  port 0/0/3
!
CASE STUDY: PSTN Connectivity Integration
(Call in/out for PSTN if considered legacy PBX)

An incoming call to the PSTN numbers are forwarded to 2000

Call out numbers beginning with 0
Through PSTN ports
CASE STUDY: PSTN Connectivity Integration
(Call in/out for PSTN if considered legacy PBX)

This will create a loop circuit for which all ports will be off-hook.

! voice-port 0/3/1
connection plar opx 2000
description PSTN-FXO-Router1-PABX 029117891
caller-id enable
!
voice-port 0/3/2
connection plar opx 2000
description PSTN-FXO-Router1-PABX 029117892
caller-id enable
!

dial-peer voice 1 pots
description PSTN-Out
destination-pattern 0T
direct-inward-dial
forward-digits all
port 0/3/1
!
dial-peer voice 2 pots
description PSTN-Out
destination-pattern 0T
direct-inward-dial
forward-digits all
port 0/3/2
!

This will allow all to use outgoing through PSTN
CASE STUDY: PSTN Connectivity Integration

(Call in for PSTN if considered legacy PBX)

In such case its better if we use hunt group for PABX integration

! voice hunt-group 1 sequential
   list 2097, 2098, 2099
   timeout 15
   pilot 2000
!
   dial-peer voice 1 pots
   destination-pattern 2097
   incoming called-number .%
   port 0/0/0
!
   dial-peer voice 2 pots
   destination-pattern 2098
   incoming called-number .%
   port 0/0/1
!
   dial-peer voice 3 pots
   destination-pattern 2099
   incoming called-number .%
   port 0/0/2
!

15 sec delay if not picked
CALL STUDY: PSTN Connectivity Integration (Dedicated PSTN and using it for remote router)

Call in for Router1 (Resided in Router2)
! voice-port 0/3/1
  connection plar opx 2010
  description PSTN-FXO-Router1-PABX 029117891
caller-id enable
!
voice-port 0/3/2
  connection plar opx 2012
description PSTN-FXO-Router1-PABX 029117892
caller-id enable
!

Call out for Router1 (Resided in Router1)
! voice translation-rule 1
  rule 1 /^0/ /A20100/
! voice translation-rule 2
  rule 1 /^0/ /A20120/
! voice translation-profile phone2010
  translate called 1
! voice translation-profile phone2012
  translate called 2
!
voice register dn 1
  translation-profile incoming phone2010
  number 2010
  allow watch
  name 2010
!

Call out for Router1 (Resided in Router2)
dial-peer voice 1 pots
  translation-profile incoming phone2012
destination-pattern 2012
  incoming called-number .%
  port 0/0/0
!
dial-peer voice 2 voip
  description PSTN-OUT-2010
destination-pattern A2010T
  session protocol sipv2
  session target ipv4:10.11.121.2
dtmf-relay sip-notify
  codec g711ulaw
  no vad
!
dial-peer voice 3 voip
  description PSTN-OUT-2012
destination-pattern A2012T
  session protocol sipv2
  session target ipv4:10.11.121.2
dtmf-relay sip-notify
  codec g711ulaw
  no vad
!
Call in for Router1 (Resided in Router2)
- voice-port 0/3/1
- connection plan opx 2010
- description PSTN-FXO-Router1-PABX 029117891
- caller-id enable

Call out for Router1 (Resided in Router1)
- dial-peer voice 1 pots
- description Call-Out-For-2010
- destination-pattern A2010T
- direct-inward-dial
- port 0/3/1

Call out for Router1 (Resided in Router2)
- dial voice 1 pots
- description Call-Out-For-2010
- destination-pattern A2010T
- direct-inward-dial
- port 0/3/0

As the router gets A2010T where T = 01711234567
It will only forward T to its mentioned port of PSTN

PSTN

2010 dials 01711234567 and that transforms in A201001711234567 which is mentioned as A2010T to transport to Router 2

Call going to 01711234567

A call coming to 029117891

Router 2

Router 1

2010

Call going to 01711234567

Router 2

Router 1

2010

Call out for Router1 (Resided in Router1)
- voice translation-rule 1
- rule 1 /0/ /A20100/
- voice translation-profile phone2010
- translate called 1
- dial-peer voice 1 pots
- translation-profile incoming phone2012
- destination-pattern 2012
- incoming called-number .%
- port 0/0/0
- dial-peer voice 2 voip
- description PSTN-OUT-2010
- destination-pattern A2010T
- session protocol sipv2
- session target ipv4:10.11.121.2
dtmf-relay sip-notify
codec g711ulaw
no vad
CASE STUDY: Remote FXO activity due to local device or data network failure

Reasons:

While we disconnect lines through phone a tone is recognized to reset the port to tell that it should be onhook to establish calls.

Each country uses their own tones to perform this operation.

So just like circuit connectivity Packet connectivity is required to carry its actual tone to destination port to tell it to onhook for link breakage.

For a help we can try the following site where custom tones are listed by country usage.
http://www.3amsystems.com/World_Tone_Database
CASE STUDY: Remote FXO activity due to local device or data network failure

voice class dualtone-detect-params 1
freq-max-deviation 20
cadence-variation 20
!

voice class custom-cptone BD-CPTONE
dualtone disconnect
  frequency 450
  cadence 200 300 700 800 3000 10000 250
!

voice-port 0/3/1
supervisory disconnect dualtone mid-call
supervisory custom-cptone BD-CPTONE
supervisory dualtone-detect-params 1
compand-type a-law
cptone GB
timeouts call-disconnect 1
timeouts wait-release 1
connection plar opx 2010
description PSTN-FXO-Router1-PABX 029117891
caller-id enable
!

voice-port 0/3/2
supervisory disconnect dualtone mid-call
supervisory custom-cptone BD-CPTONE
supervisory dualtone-detect-params 1
compand-type a-law
cptone GB
timeouts call-disconnect 1
timeouts wait-release 1
connection plar opx 2012
description PSTN-FXO-Router1-PABX 029117892
caller-id enable
!
Conclusion: Final Achievements and Future

1. A distributed organization can operate on their own communication pattern.
2. While merging to this technology their existing legacy technology PABX is integrated.
3. Zone based registry allows you to have your own area communication due to unavailability of data connection.
4. Local or remote PSTN integration.

Scope generated due to implementation:

1. FAX integration or personal FAX network.
2. Achieving a single area of communication management for voice, video, data, internet etc.
3. Achieving IP-PABX smart features.
4. Using data communication more efficiently sharing it with QOS.
5. Getting IP-TSP trunks as an alternative option of PSTN.
Thank you

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