Distributed IP-PBX

bdNOG
Bangladesh Network Operators Group

Tele-convergence of IP-PBX / PSTN / FAX / legacy PABX

And

Distributed network approach with area isolation

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Contributor Acknowledgement

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Adnan Howlader

AGENDA

- Background
- Application and Appliances
- Architecture & Benefits
- Case Study
- Key Findings

BACKGROUND

BACKGROUND : The Beginning

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.

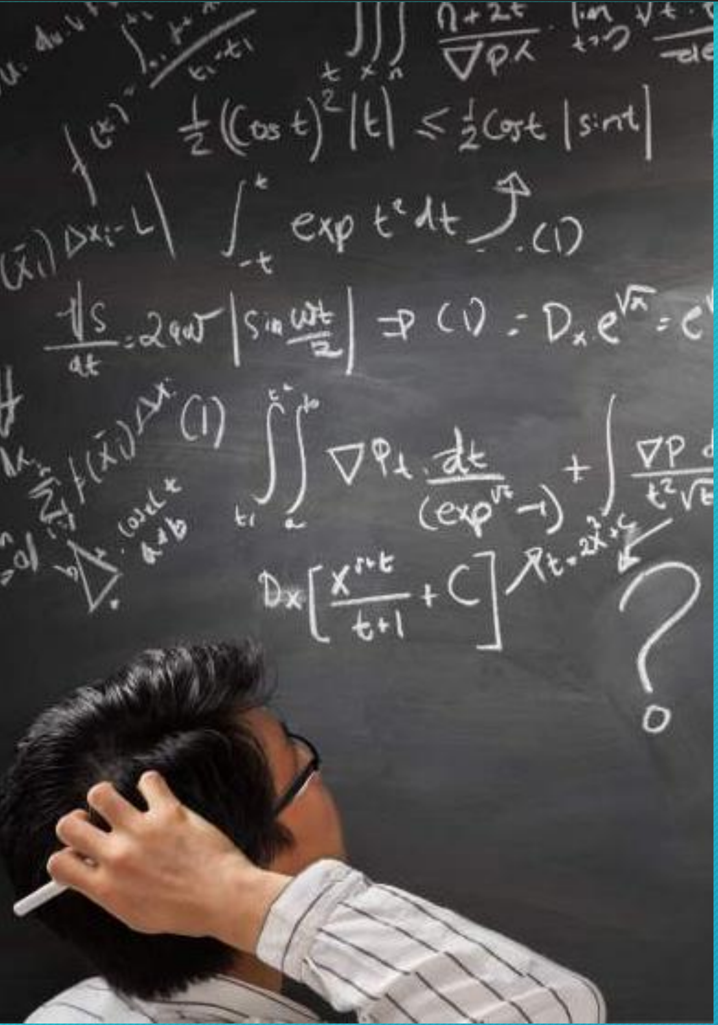
BACKGROUND : Realization

- 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



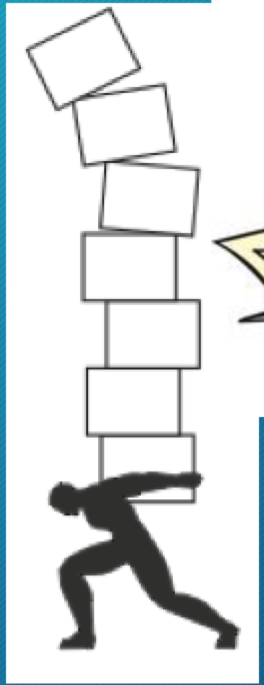
Complexity !!!

Expensive !!!

Mess !!!

Burden !!!

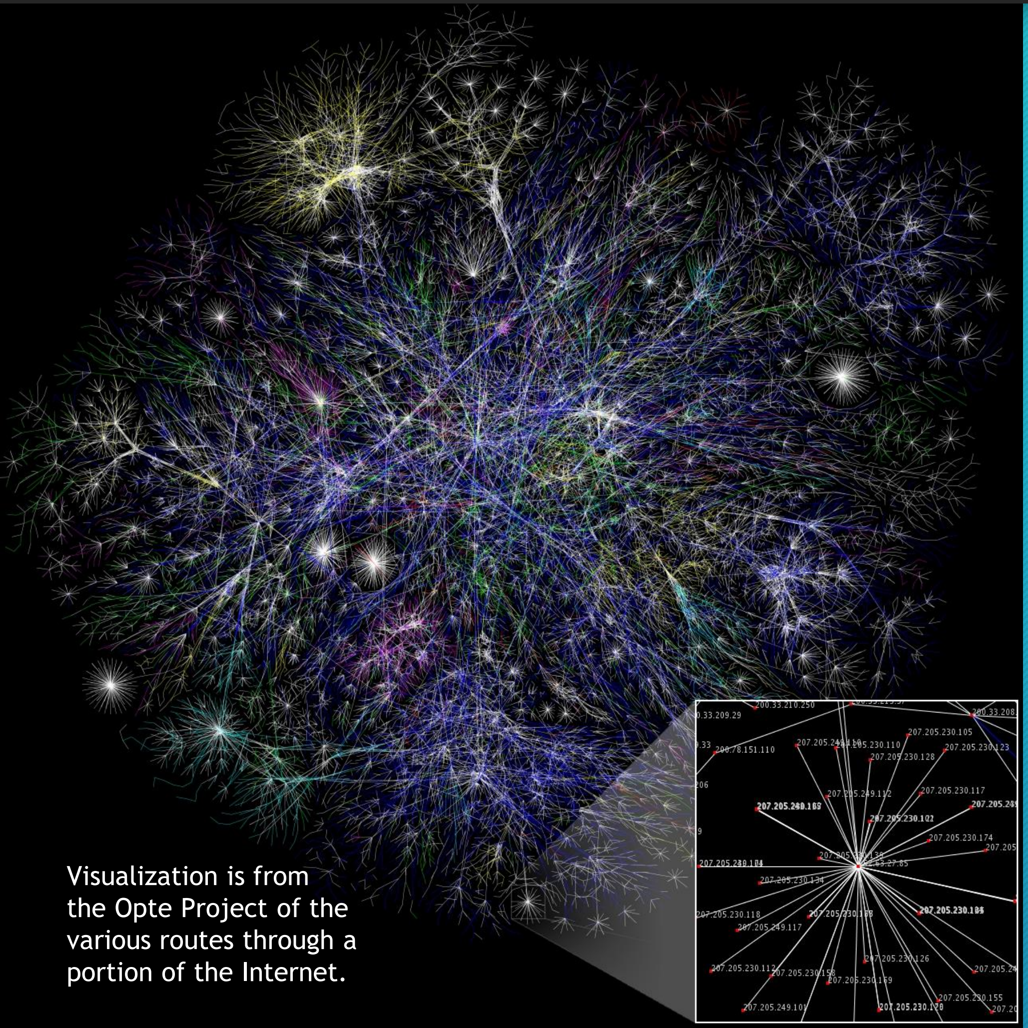
....



BACKGROUND : Realization

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

BACKGROUND : Telecommunication



Visualization is from the Opte Project of the various routes through a portion of the Internet.

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

BACKGROUND : Circuit Vs Packet switching

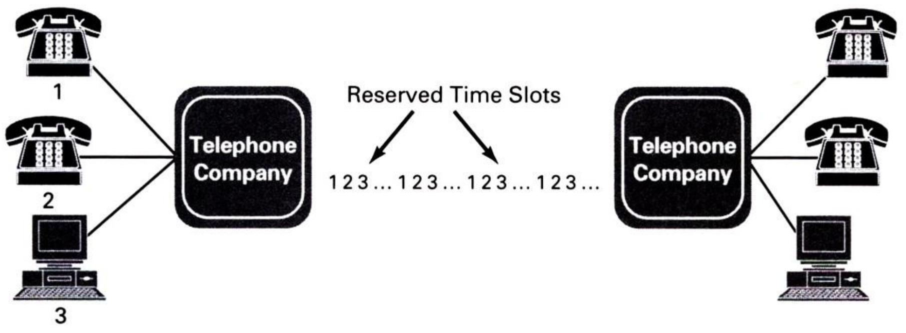


Figure 9.1
Circuit Switching

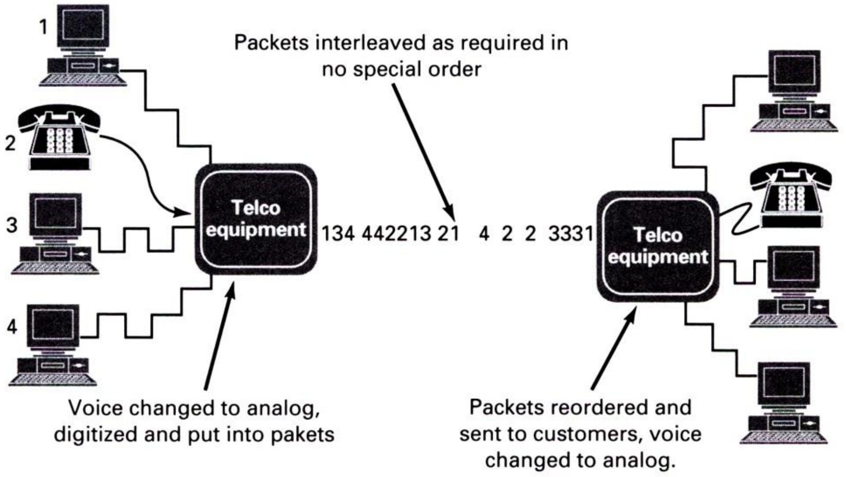


Figure 9.2
Packet Switching

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

Codecs	Payload Bitrate
G.711	64 kbit/s
G.726	16, 24 or 32 kbit/s
G.723.1	5.3 or 6.3 kbit/s
G.729	8 kbit/s
GSM	13 kbit/s

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

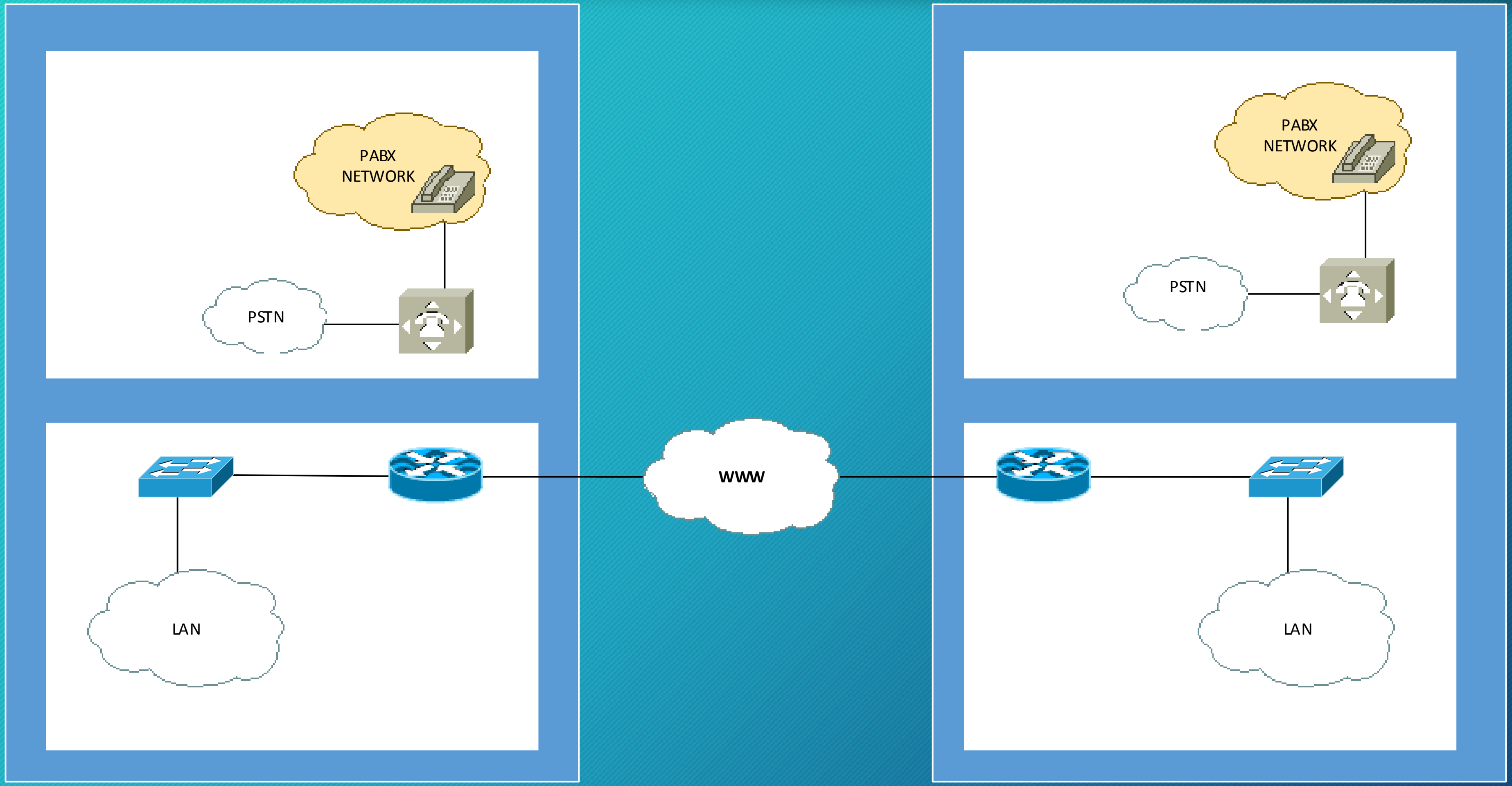
APPLICATION & APPLIANCES :
Traditional phone, IP-phone & PABX station

ARCHITECTURE & BENEFITS

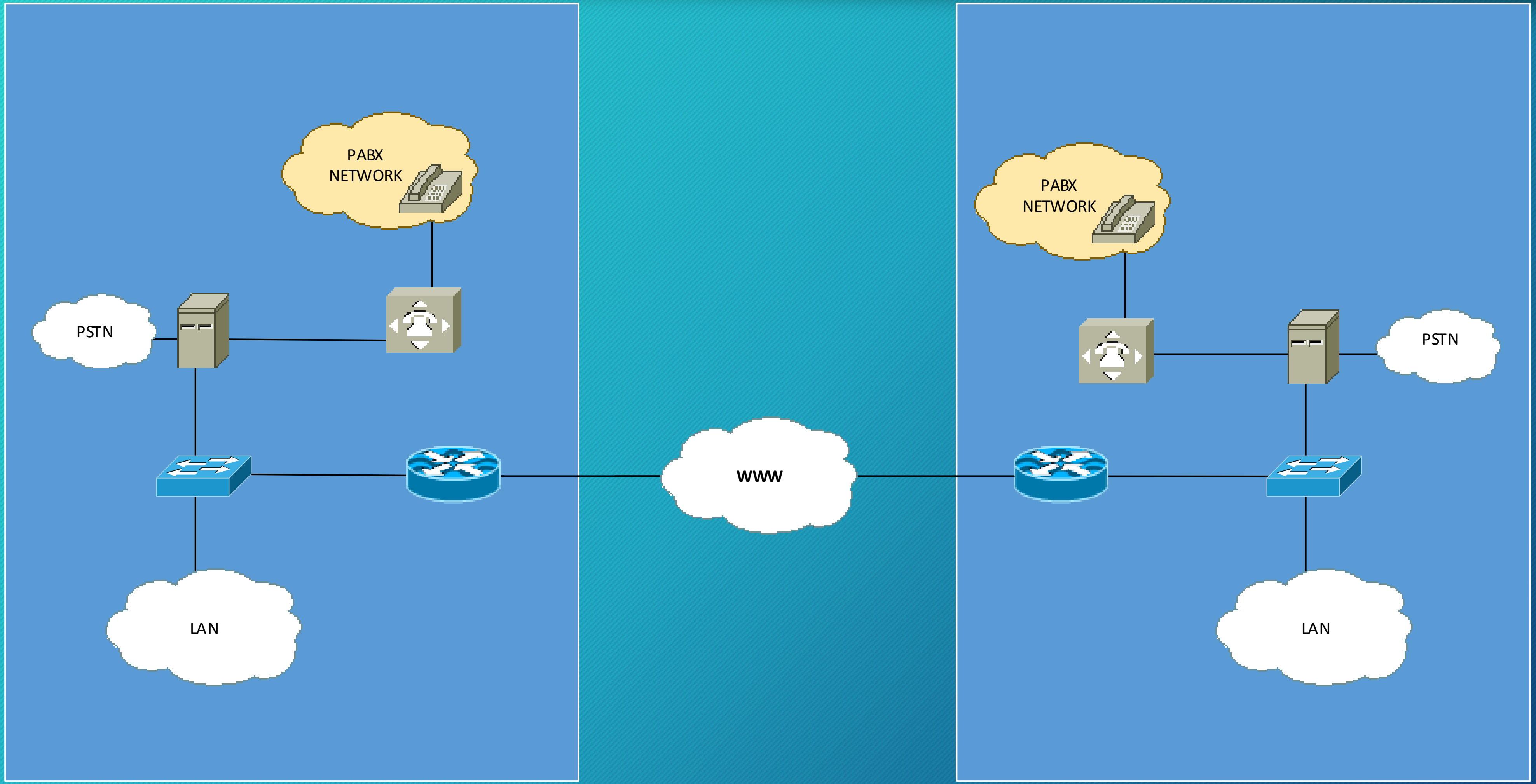
Considering Scenarios

- 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.

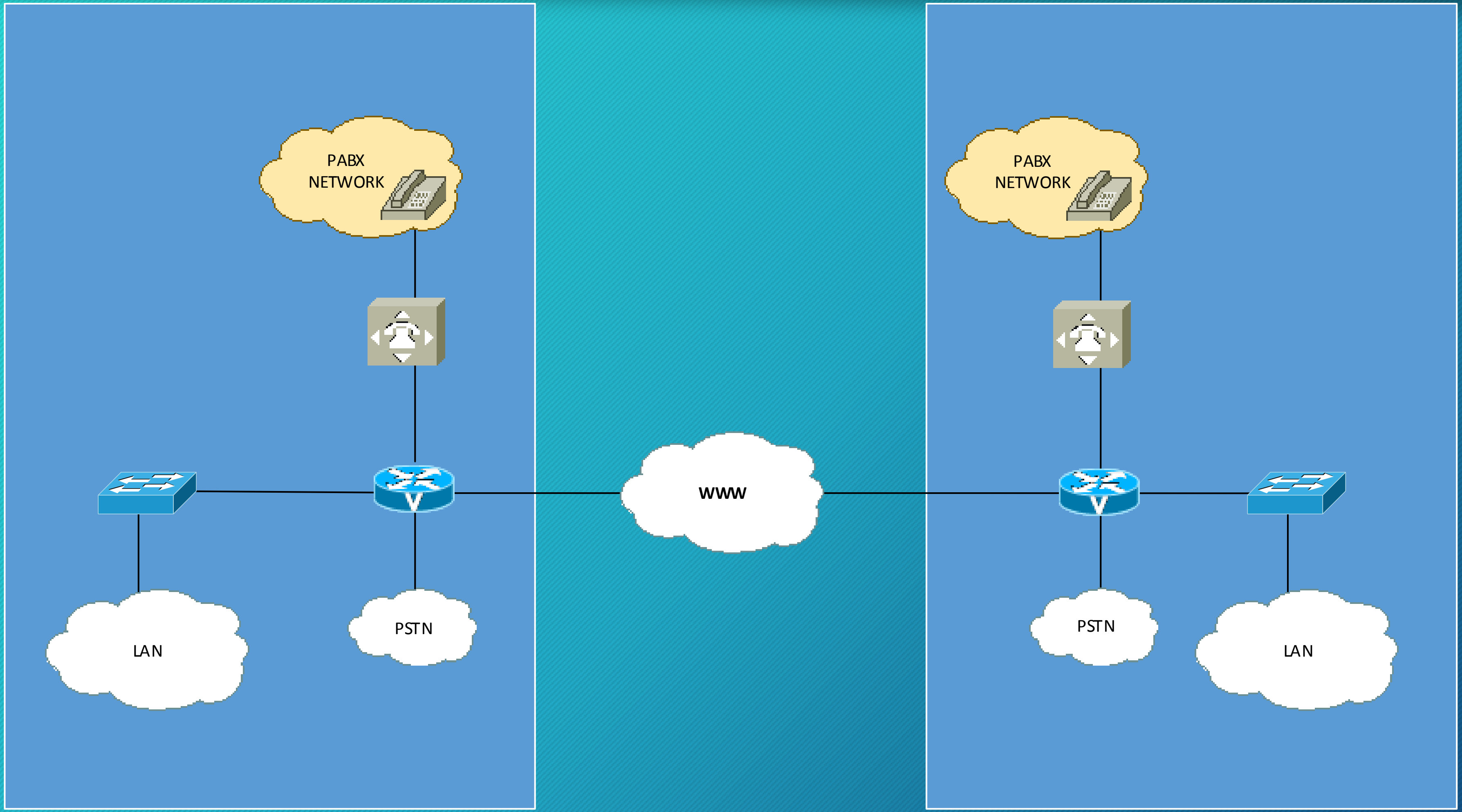
Common Organization Scenario



Transformation Scenario 1

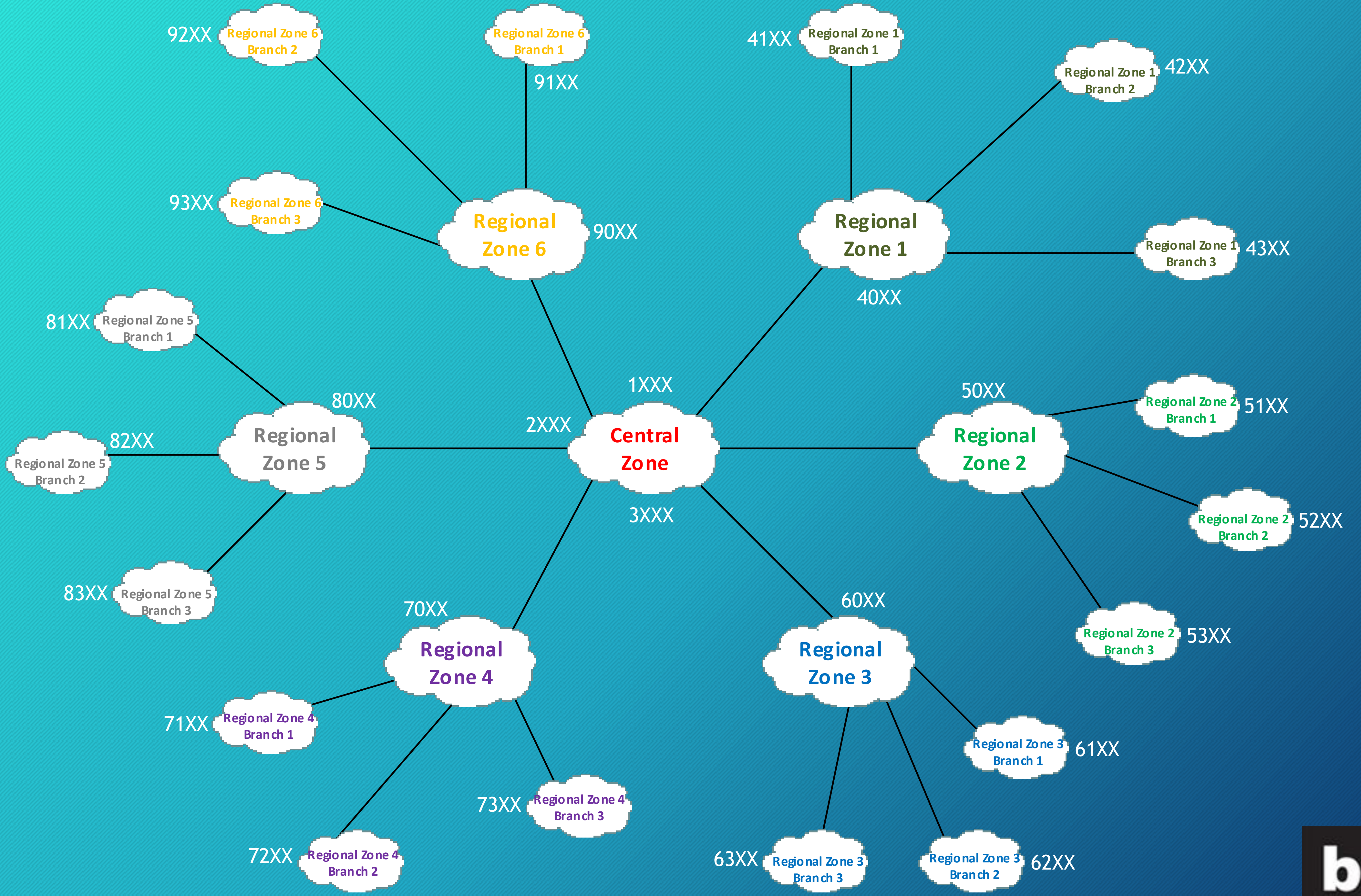


Transformation Scenario 2



CASE SCENARIO:

A Nationwide Spread Organization



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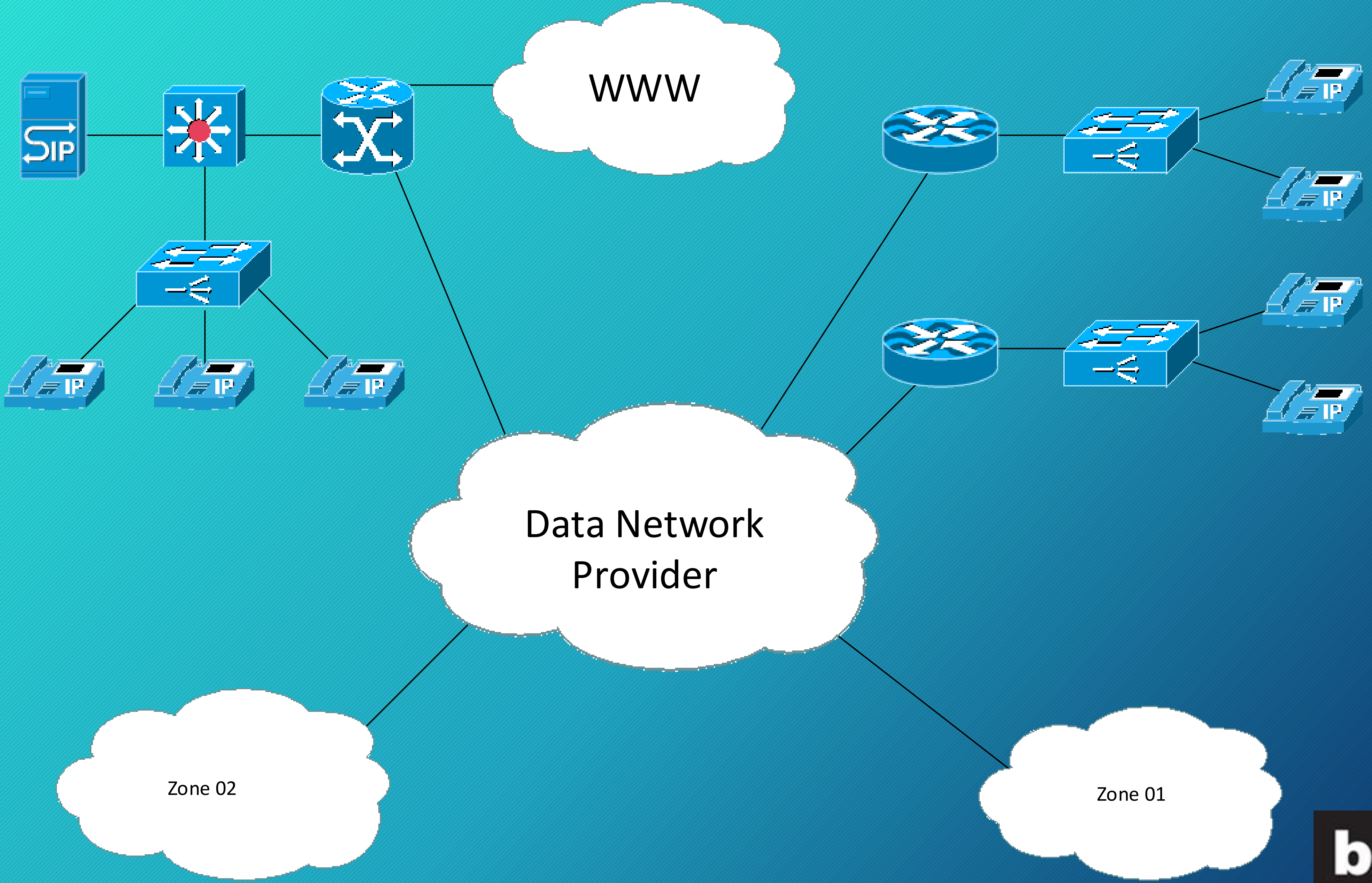
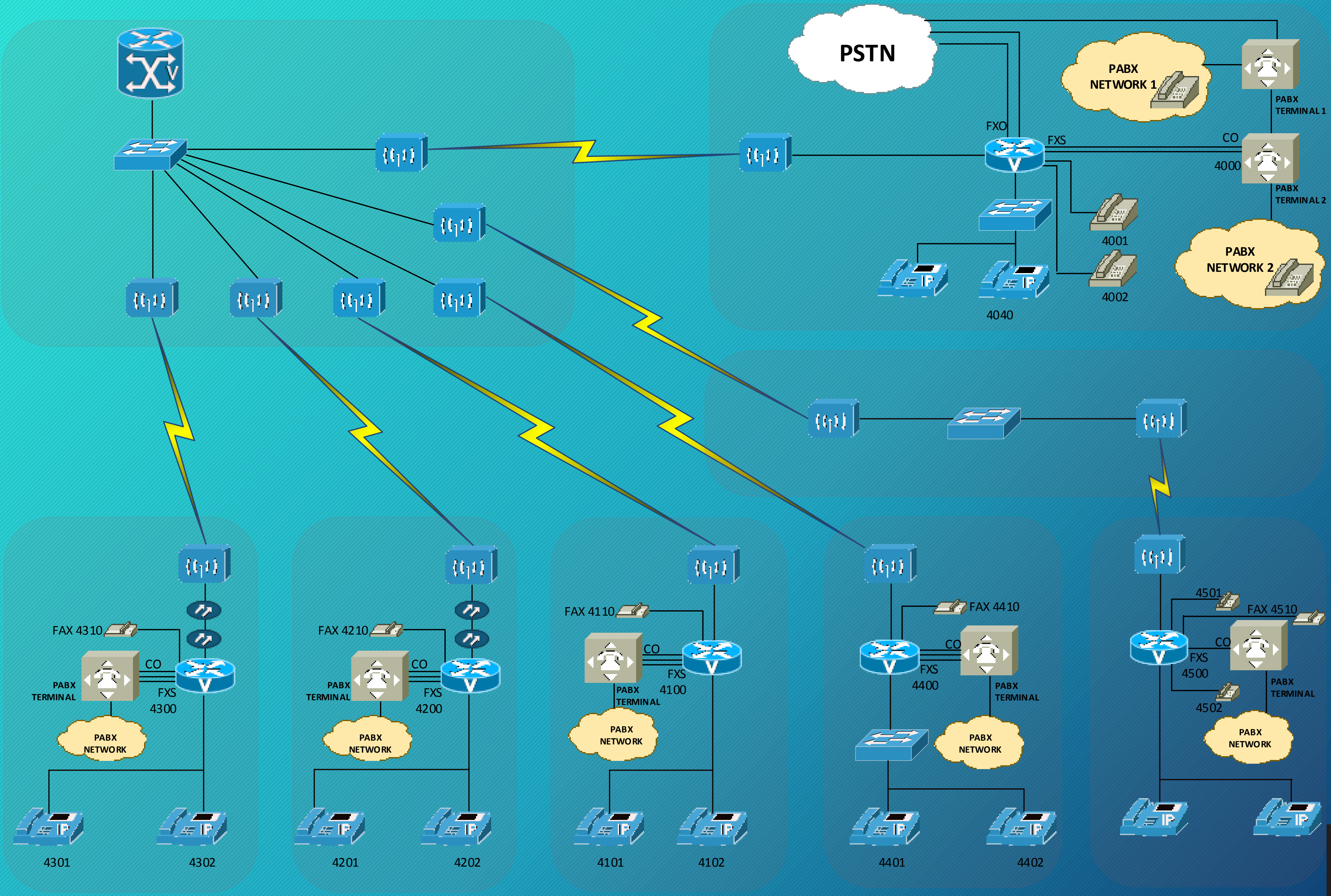
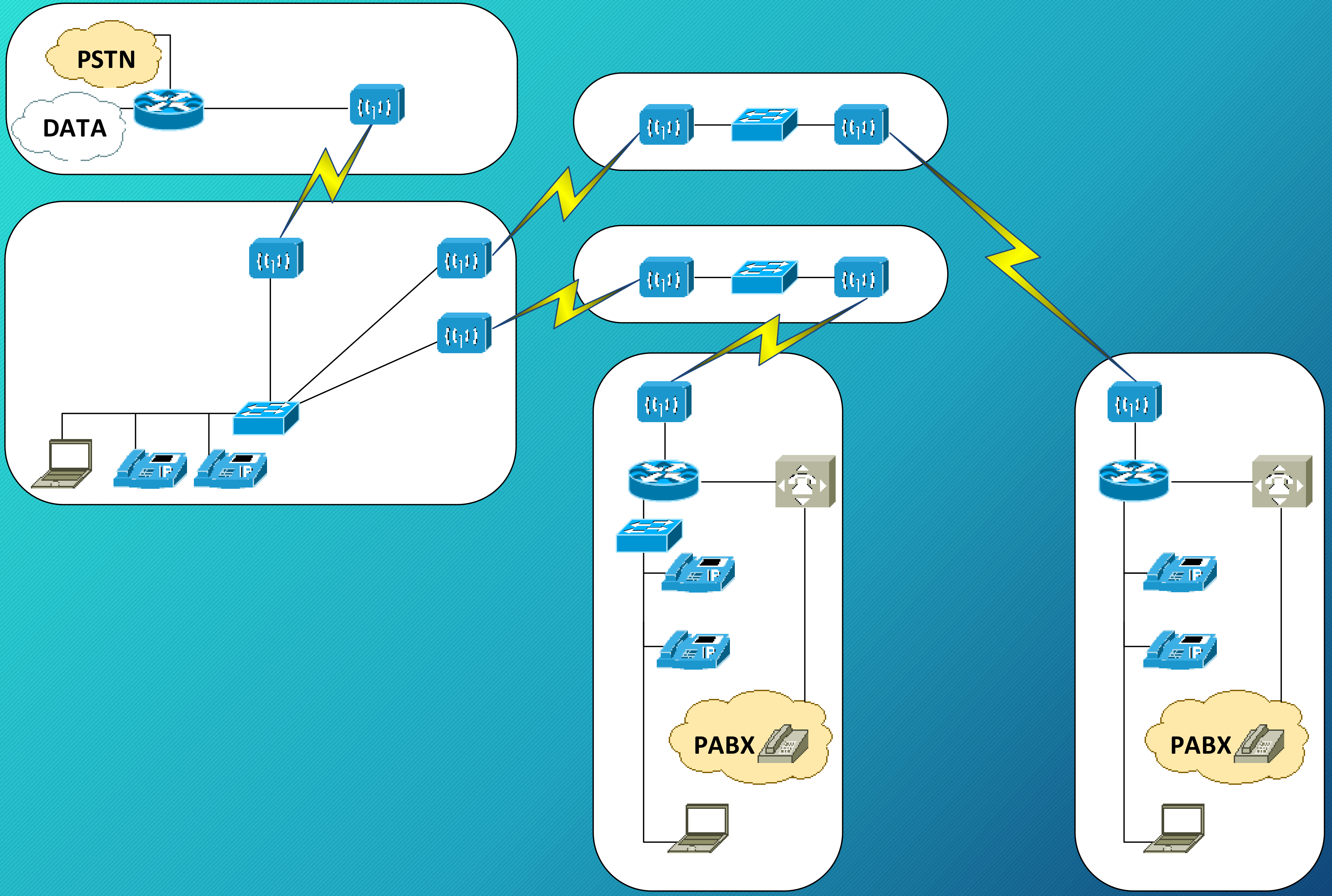


CASE SCENARIO: Actual Network Diagram



CASE SCENARIO:

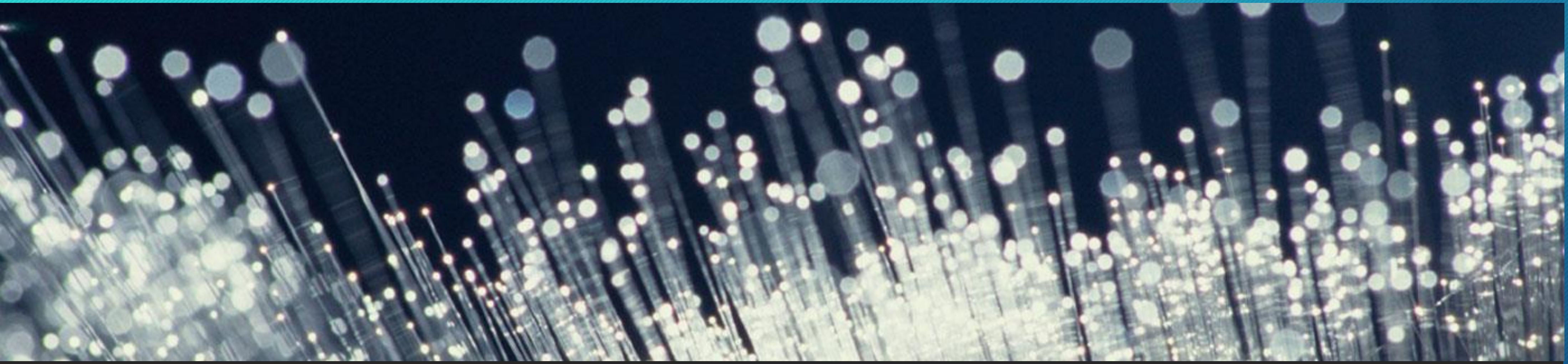
Zone 1 Network Diagram



CASE SCENARIO:
Zone 2 Network Diagram

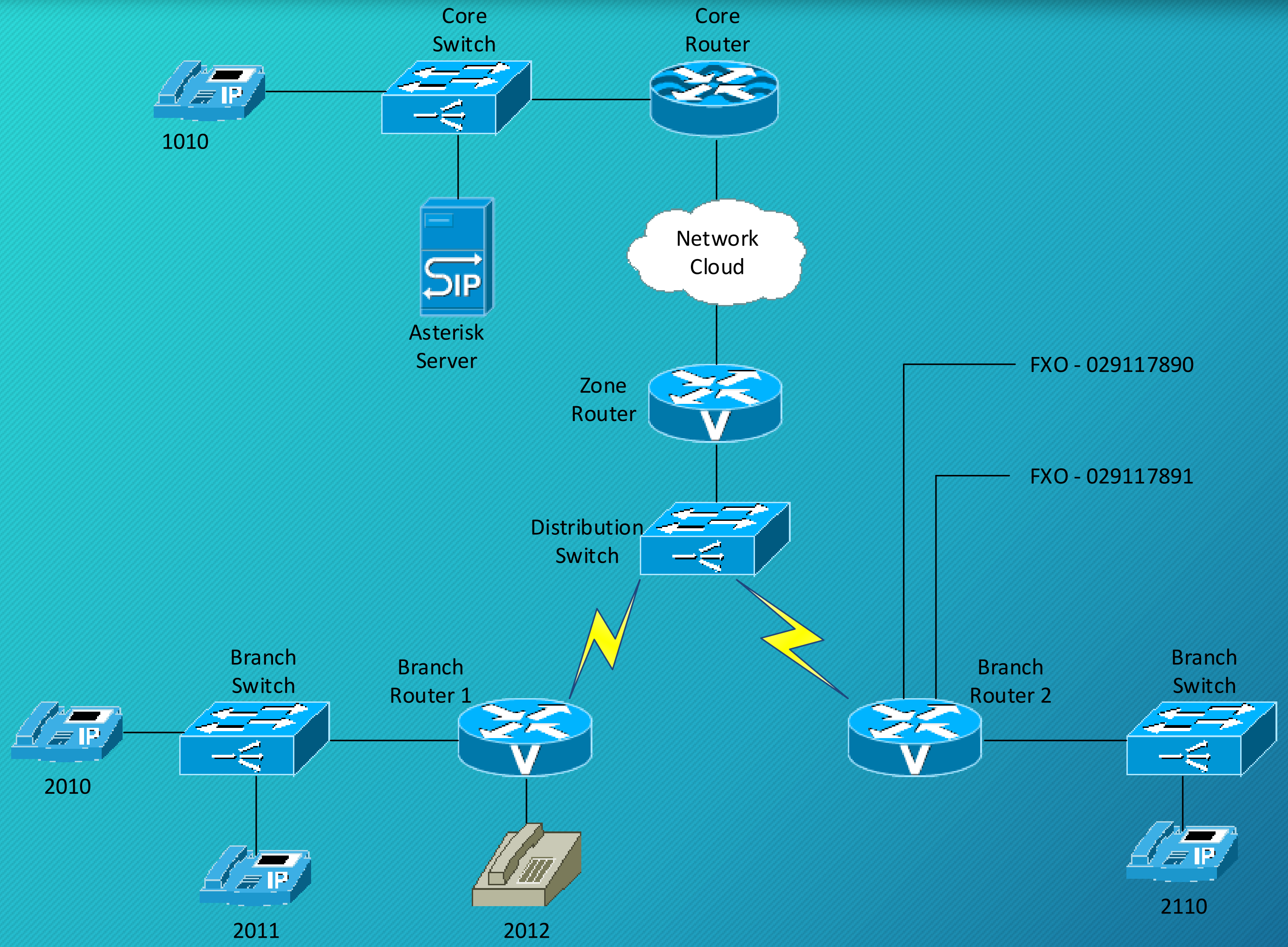
Why Such integration?

1. Using the customers existing circuit setup PABX or PSTN.
2. Using cross site communication.
3. Using packet communication for connectivity.
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

```
[cme-trunk]
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()
```

Context “cme-trunk”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

vi /etc/asterisk/sip.conf

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

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CASE STUDY: Cisco Router Configuration (Basic Configuration for voice)

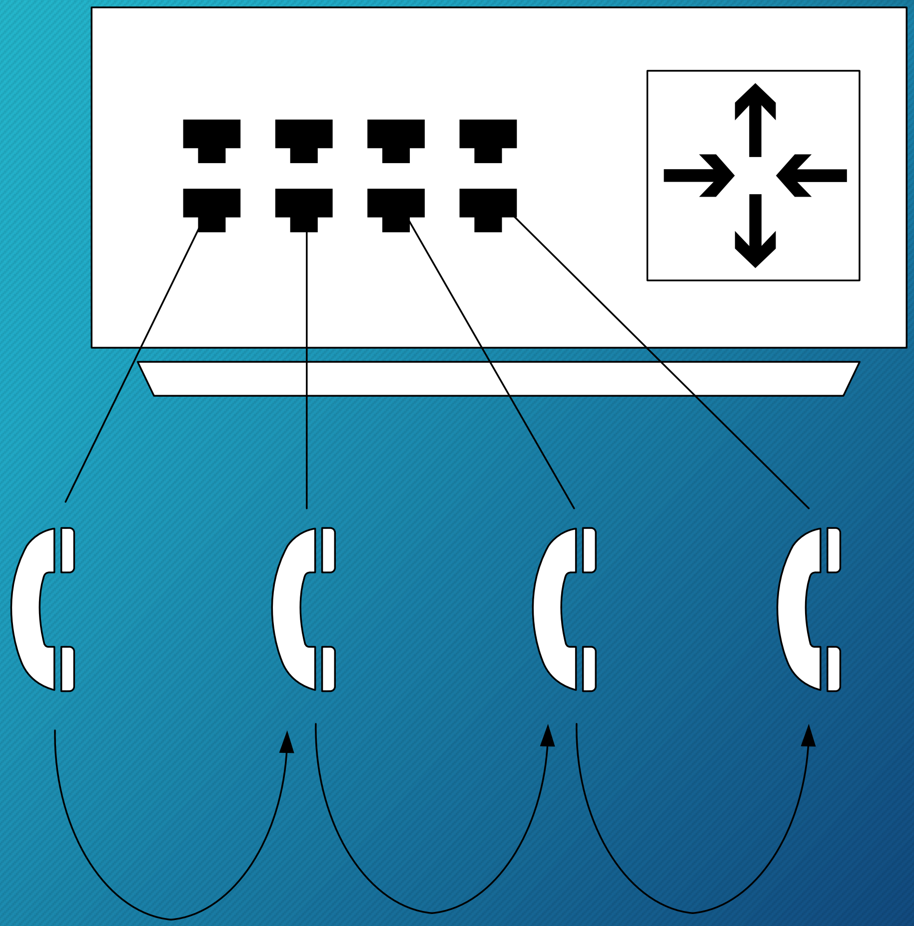
```
voice service voip // Declaring the voice service over which mode. In our case its IP.
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12 advertise-only // Common Information Additional Network Feature for H.323
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback 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
  rel1xx 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 g711alaw
!
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 transcode 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 expires 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
!
```

CASE STUDY: Cisco Router Configuration (Registering IP Phone / Pots Phone / Voice peer)

```
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)

```
!
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
!
```

An incoming call to the PSTN numbers are forwarded to 2000

Call out numbers beginning with 0
Through PSTN ports

```
!
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
!
```

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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
!
```

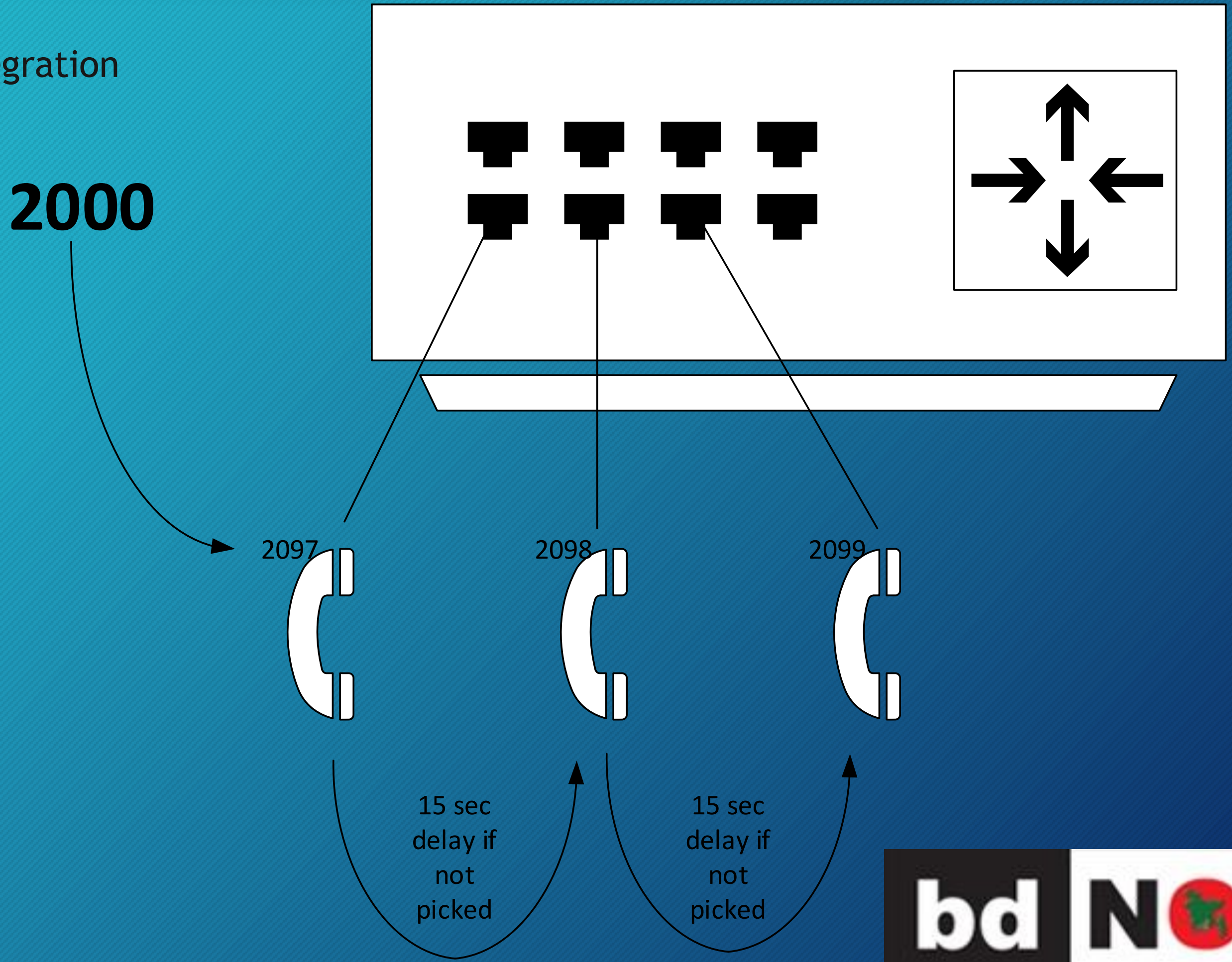
This will allow all to use outgoing through PSTN

```
!
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
!
```

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
!
```



CASE 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 Router2)

```
dial-peer voice 1 pots
 description Call-Out-For-2010
 destination-pattern A2010T
 direct-inward-dial
 port 0/3/0
!
dial-peer voice 2 pots
 description Call-Out-For-2012
 destination-pattern A4202T
 direct-inward-dial
 port 0/3/1
```

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
!
```

```
!
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
!
```

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CASE STUDY: PSTN Connectivity Integration (Dedicated PSTN and using it for remote router)

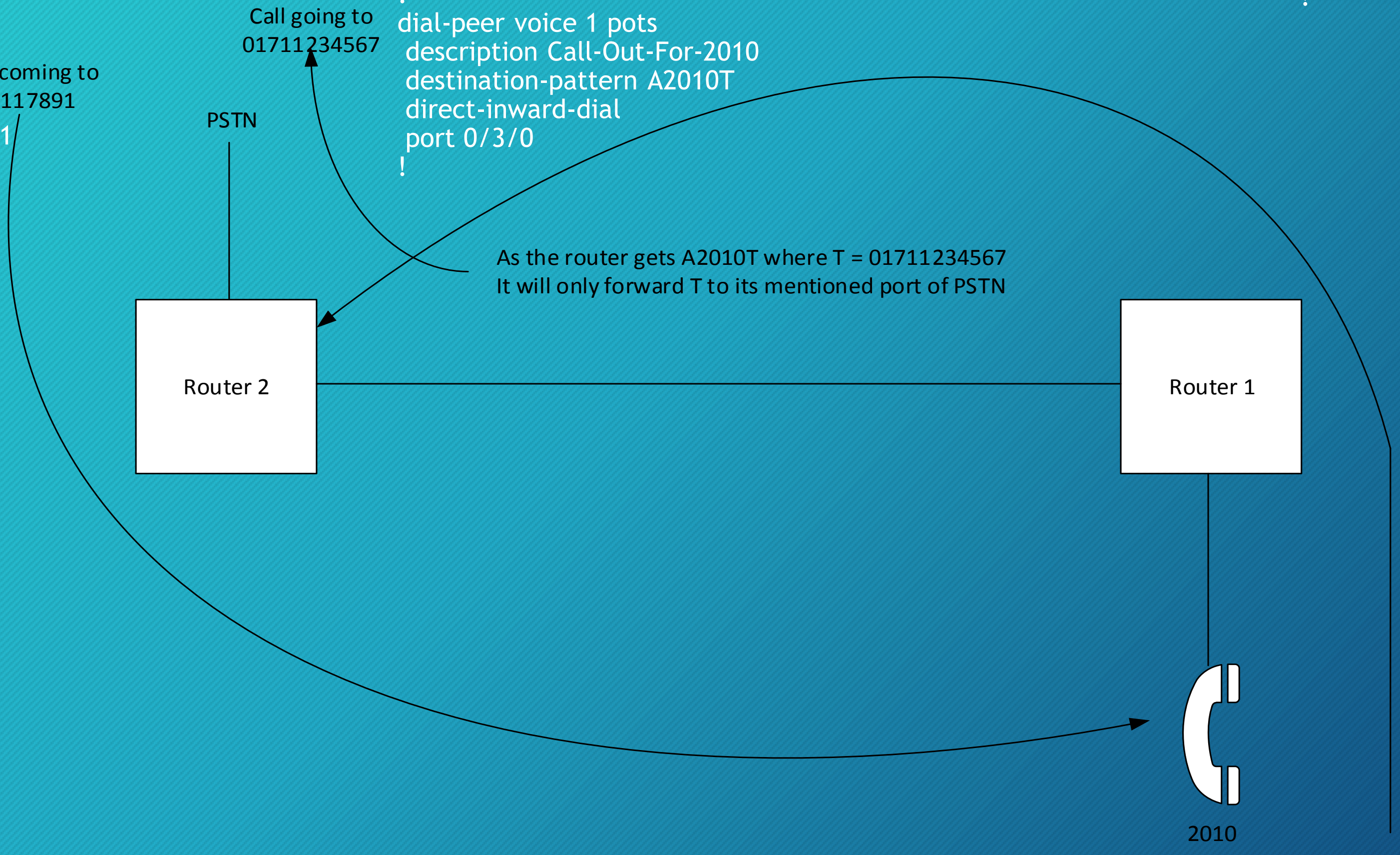
Call in for Router1 (Resided in Router2)

```
!
voice-port 0/3/1
 connection plar opx 2010
 description PSTN-FXO-Router1-PABX 029117891
 caller-id enable
!
```

Call out for Router1 (Resided in Router2)

```
!
dial-peer voice 1 pots
 description Call-Out-For-2010
 destination-pattern A2010T
 direct-inward-dial
 port 0/3/0
!
```

!



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
!
```

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

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CASE STUDY: Remote FXO activity due to local device or data network failure

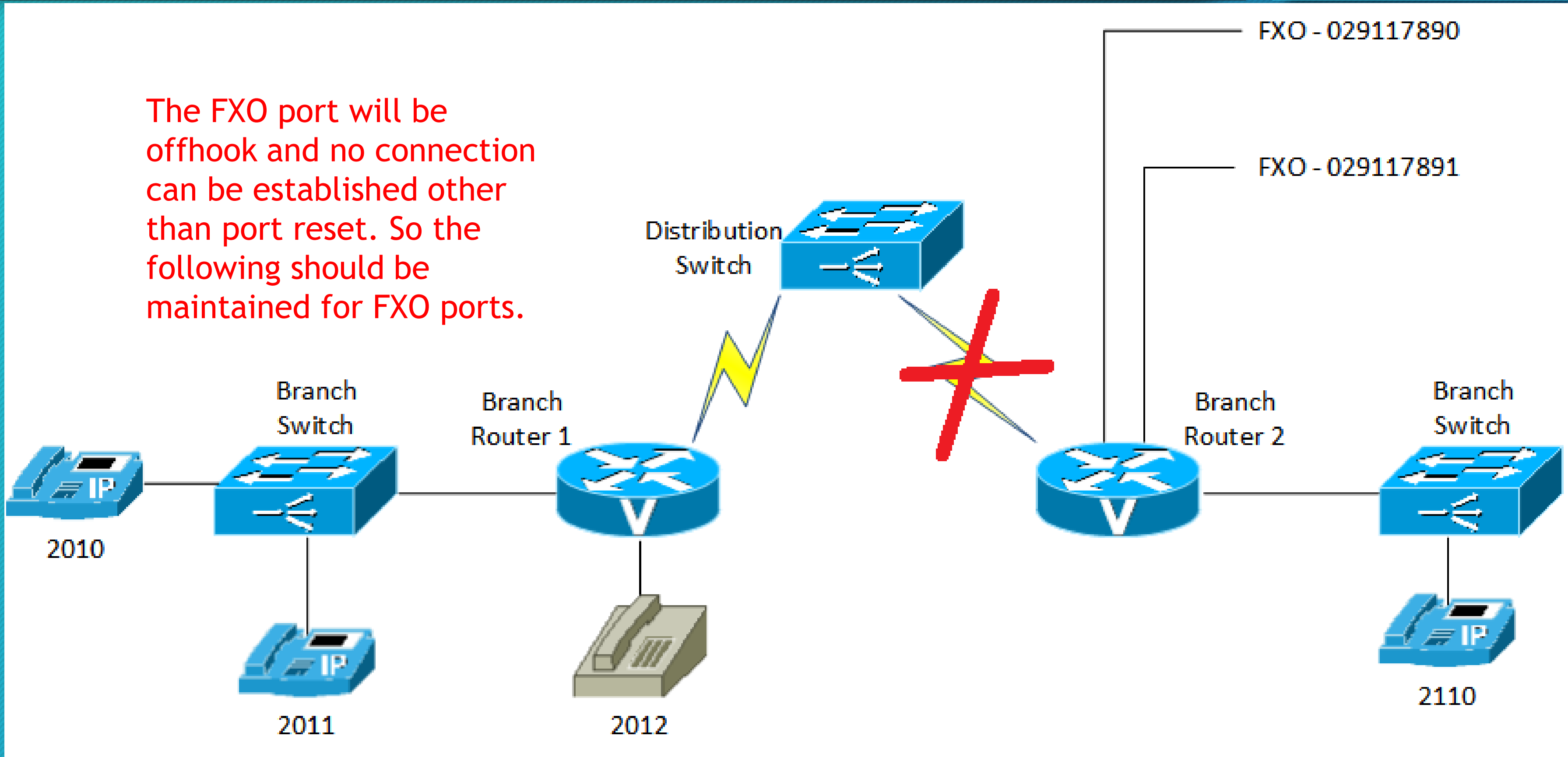
Reasons:

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 you 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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