Configuring Cisco Unified Border Element for PSTN SIP trunks

Randy Wu, CCIE

Cisco Support Community – Expert Series Webcast

- Today's featured expert is Cisco Support Engineer **Randy Wu**
- Ask him questions now about Cisco Unified Border Element (CUBE)

Randy Wu

CCIE in Routing and Switching, Voice, and Service Provider

Thank You for Joining Us Today

Today's presentation will include audience polling questions

We encourage you to participate!

Thank You for Joining Us Today

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https://supportforums.cisco.com/community/netpro/collaboration-voice-video/ip-telephony

Polling Question 1

What is your Level of experience with Cisco Unified Border Element?

a) I am thinking about using it

b) I am playing with it in my lab

c) We are running it in production

d) We are using 3rd party SBC, might consider Cisco UBE.

Submit Your Questions Now

Use the Q&A text box to submit your questions

Configuring Cisco Unified Border Element for PSTN SIP trunks

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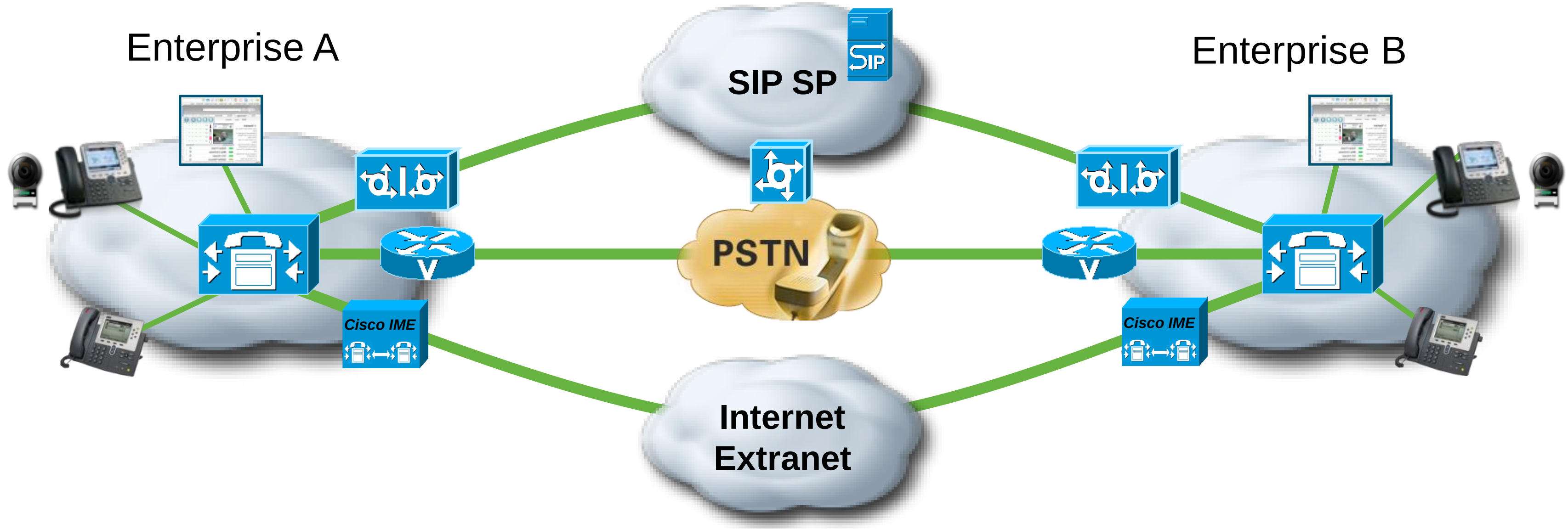
Agenda

- Selecting a SIP Trunk Service
- Migrating from PRI to SIP Trunks for PSTN Access
  - CLI Commands used to configure CUBE
- CUBE Implementation and Configuration and SP SIP Trunks
  - General UBE Configuration for SP SIP Trunks
  - Advanced Cisco UBE Feature Configuration Examples
  - Troubleshooting Configuration

Selecting a SIP Trunk Service

Evolution of Enterprise "PSTN" Connectivity
Choices in Trunking



▪ Traditional TDM PSTN Connectivity	Voice only
▪ Commercial Service Provider (SP) SIP Trunk offerings	Voice only today, but expected to evolve to video services
▪ Intercompany Media Engine (IME)	Rich collaboration

Current SP SIP Trunk Services
Compared to TDM Services

Consideration	SIP Trunk	TDM Trunk
Basic call completion	Well defined	Well defined
Suppl. services (Xfer, FWD, Hold, Conf)	Requires validation testing	Well defined
Fault Monitoring and Isolation	Options PING monitoring	Yellow/Red Alarms
Emergency Call (911) Handling	Special Handling per SP	Well defined
Malicious Call-ID (MCID) and Multi-level Priority and Preemption (MLPP)	Not defined	Well defined
Caller-ID delivery	Inconsistent	Consistent
Voice Band Data	Modems/Baudot TDD ill-defined or unsupported	Well defined
Fax Technology	Industry interop issues	Well defined
Deterministic traffic engineering. How are bursts handled? Who sends back equipment busy, enterprise or SP? Who provides announcements?	SP dependent	Well defined
Porting numbers	Within single SP control	Well defined
Geographic and legal dependencies of call routing	Independent of geography but not of legislation	Geographically dependent
Future rich media services	Great potential	No
Cost to enterprise for service	Inconsistent	Well defined
Flexibility of call routing; site aggregation	Very flexible	SP dependent
Security considerations	IP considerations; toll fraud	Toll fraud

Why Cisco Unified Border Element (CUBE) as CPE?

- Highly recommended for Enterprise SIP Trunk deployments
- Co-existence with other features such as MTP, SRST, TDM GW

Call Counting

Call Admission Control
IP SLA

Demarcation

Troubleshooting
Billing
Topology Hiding

Cisco Unified Border Element

H323 to SIP
SIP to SIP
SIP Profiles and Variants

Interworking

Integrated with IOS FW
RTP Media Validation
Signaling protection

Security

SIP Trunk Best Practises Summary

- The SIP trunk market is maturing
  - Plan and execute thorough testing before production
  - Most interop issues can be resolved with targeted configuration changes and protocol normalization
  - Test complex call flows thoroughly
    - Mobility, contact center, supplementary services
  - Tune failover timers
- Evaluate different providers
  - Offerings vary considerably
- CUCM recommendations
  - CUCM 5.x and older: H.323
  - CUCM 6.x and newer: SIP
  - Avoid MTPs if possible
- Use the SRNDs and Configuration App Notes

- Use CUBE as the onsite enterprise Border Element to
  - Resolve interop issues with SIP profiles
    - Normalize traffic – SP UNI
  - SIP DO-EO conversion
  - Interconnect/share a SIP trunk to different enterprise IP-PBXs
  - Security for CUCM/enterprise apps
  - QoS and troubleshooting demarc
- Use a G.711 SIP trunk
  - Avoid transcoding if possible
- Ensure these are addressed:
  - Redundancy – especially for large, centralized SIP trunk designs
  - Fax
  - Emergency Calls
  - DID porting
  - SIP trunk security (SIP ports, ACLs, CAC…)
  - SIP Trunk monitoring

Top Five Issues with SIP Trunks for PSTN Access

1. Interoperability with IP-PBX

   Getting calls to work

2. Fax Calls

   Long duration faxes, older fax machines

3. Voice Band Data and Media Issues (i.e. Early Media)

   Postal machines, early audio in a call

4. Supplementary Features

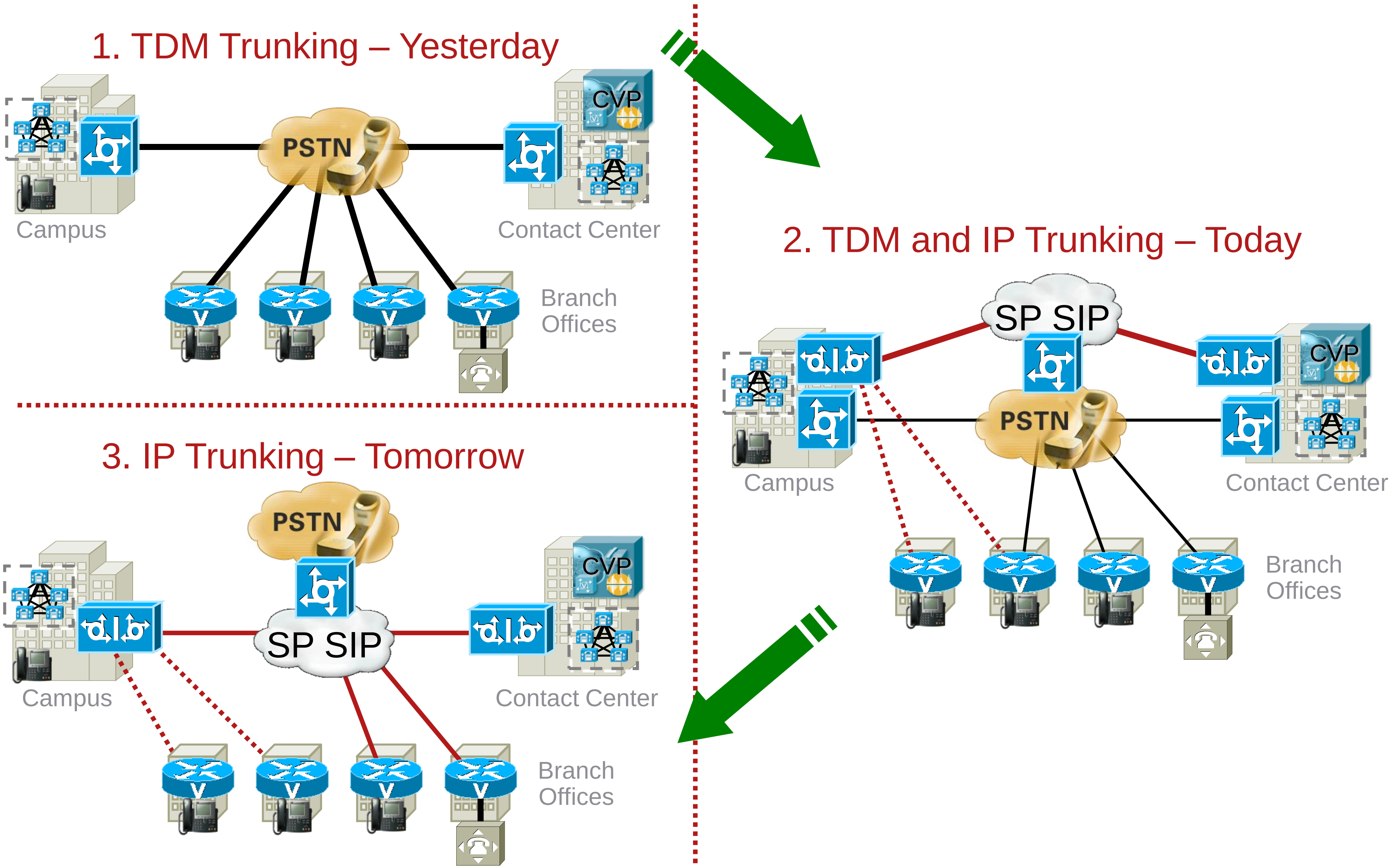
   Hold, resume, Music on Hold, etc.

5. Quality Control

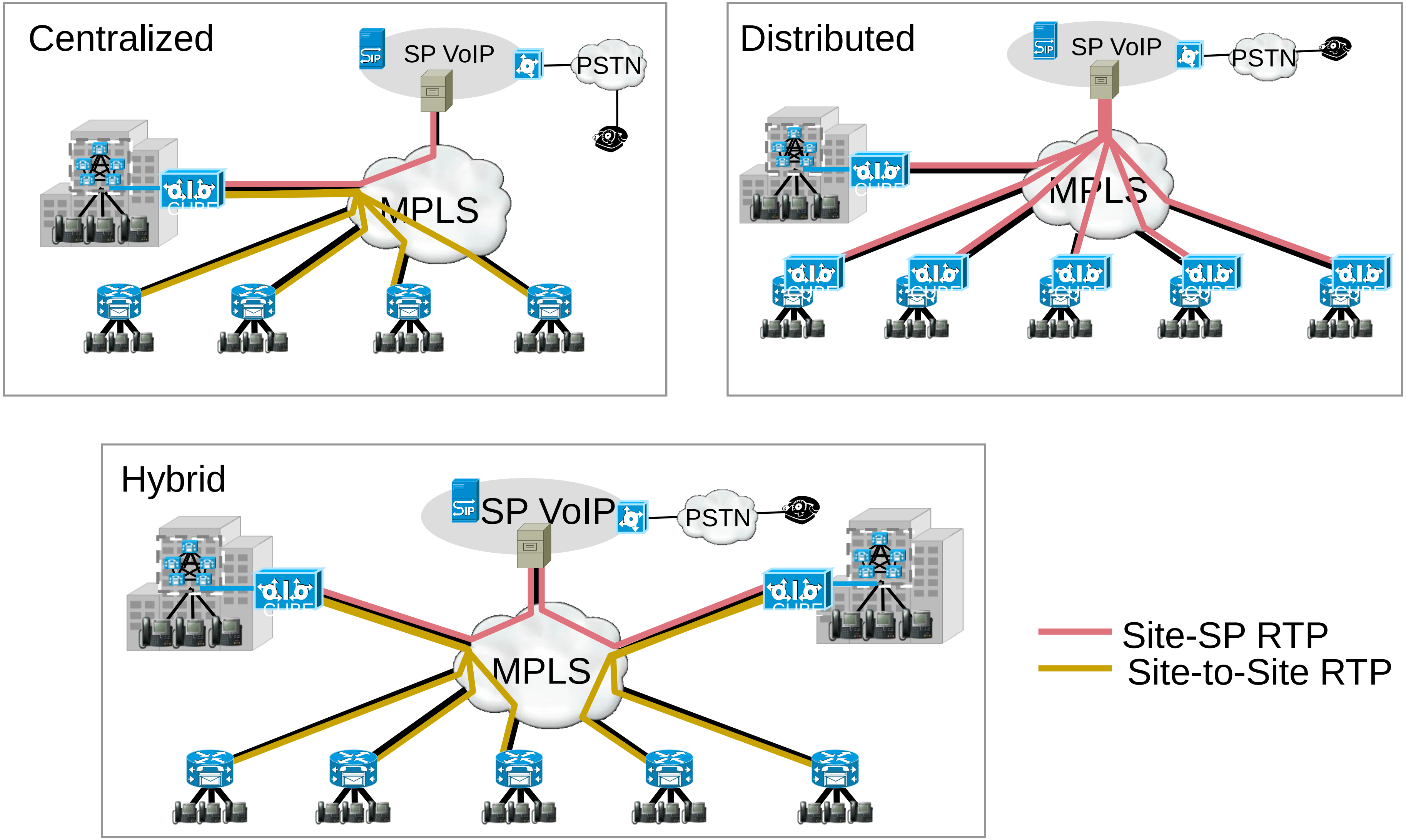
   Call quality and call experience different than PSTN

Migrating from PRI to SIP Trunks for PSTN Access

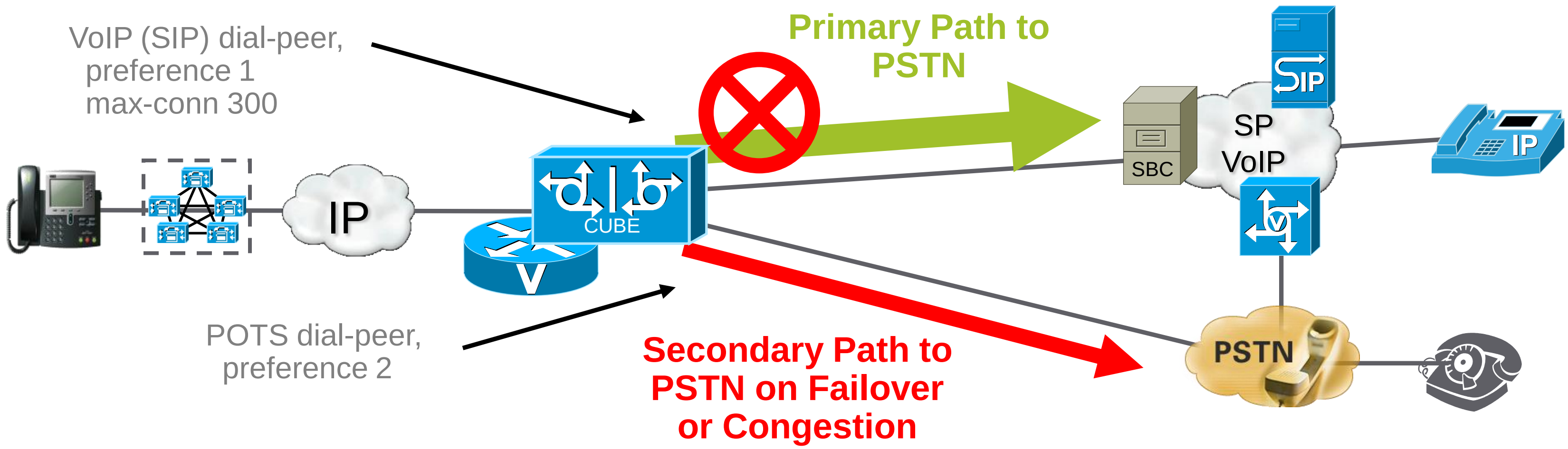
Migration to SP SIP Trunking for PSTN Access



Centralized and Distributed SIP Trunk Models



SIP Trunk to TDM PSTN Failover



- Collocated Cisco Unified Border Element (SIP trunks) and TDMGW (PSTN trunks)

  Easy SIP trunk migration—Cisco Unified Border Element platform can also host TDMPSTN trunks for alternate or failover call routing

- Integrated Cisco Unified Border Element and TDMGW platform can also provide many other integrated services to the site

  MTP, SRST, RSVP Agent, Routing and security

CUBE / CUCME SIP Trunk Interop Test Plan Outline

- Circuit Acceptance Test Cases
  - SP Layer 2 Connection
  - SP Layer 3 Connection
  - SP Reachability & Routing
- Connectivity Test Cases
  - Registration sequence
  - Session Refresh
  - Basic outbound/inbound call completion
  - Quality of Service
  - Call Admission Control
  - Management Access
  - Call Accounting
  - Voice Quality
  - Stability and Duration
  - Restart
- SIP Application (Call Flow) Test Cases
  - Caller ID
  - Codec Negotiation
  - Call Hold / Resume
  - Call Forward
  - Call Transfer
  - Ad-Hoc Conference
  - IVR Interaction
  - DTMF
  - FAX, Mode, TTY
  - Emergency / 911
  - Call types (Local, Long Distance, International)
- Failover Test Cases
  - Layer 1, 2, 3, 4 failover scenarios

SIP Trunk Validations on Cisco.com

- Cisco focuses on standards-compliance and participates in major IETF SIP Standards bodies
- Perform interoperability validations for SIP trunk providers and PBXs
- Completed validations posted to Cisco.com at:

  www.cisco.com/go/interoperability

- SP SIP Trunk examples:
  - ATT
    - FlexReach, TollFree, EVPN
  - Verizon
  - Allstream
  - PAETEC
  - Sprint
  - TelenorIPT
  - XO
  - Qwest
  - Many more…

- PBX Interop examples:
  - Aastra
  - Alcatel
  - Avaya
  - Nortel
  - Siemens

Polling Question 2

What is your deployment plan of PSTN access using Cisco UBE in the next 12-18 months?

a) Traditional PSTN access is working fine, no plans for deploying Cisco UBE yet

b) We are planning to migrate the PSTN access to SIP service using Cisco CUBE

c) We will implement a green field implementation of the PSTN access using Cisco CUBE

d) We will migrate our PSTN access to SIP service using 3rd party SBC products.

e) We will implement managed service for the enterprise users using Cisco UBE

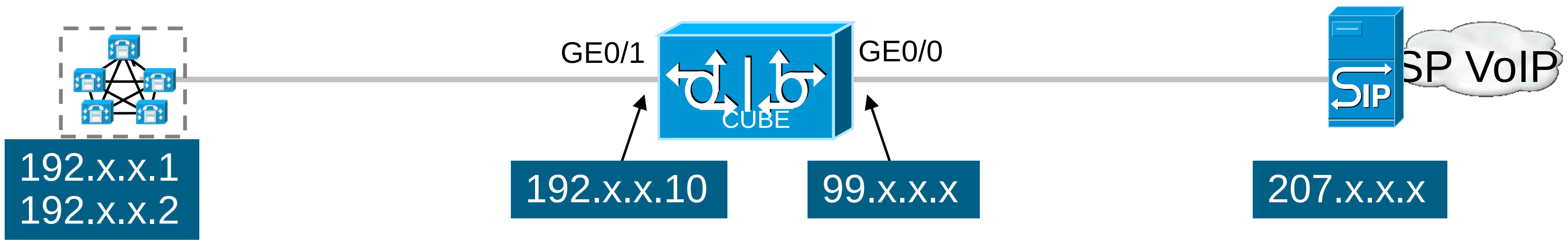
CUBE Implementation and Configuration and SP SIP Trunks

CUBE Implementation and Configuration and SP SIP Trunks - Topics

- General CUBE Configuration for SP SIP Trunks
- Advanced CUBE Feature Configuration Examples
- Troubleshooting Configuration

CUBE Configuration

- Global Configuration
  - Generic router capabilities
    - Required: Routing, IP connectivity, Interfaces, ACLs
    - Optional: DHCP, QoS, FW…
  - Global CUBE capabilities
    - Enable CUBE
    - CAC and SIP capabilities
    - Xcoding, codec classes and preferences
- Security Configuration
- SIP Configuration
  - Message handling and interpretation
  - SIP Normalization (all calls)
  - Fax (all calls)
  - Failover timers
- Dial-peer Configuration
  - Dial-plan; Digit Manipulation
  - SIP Normalization (per destination)
  - DTMF settings
  - Fax (per destination)
  - QoS Marking



Global CUBE Configuration

```
voice service voip
  address-hiding
allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
mode border-element
  sip
    early-offer forced
    header-passing
    error-passthru
midcall-signaling passthru
    privacy-policy passthru
    no update-callerid
g729annexb-all
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class sip-profiles 1
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732320\1@\2>"
  request REINVITEsdp-header Attribute modify "a=T38FaxFillBitRemoval:0" "“
!
call treatment on
call threshold global cpu-avg low 68 high 75
call threshold global total-mem low 75 high 85
call threshold global total-calls low 230 high 250
call spike 10 steps 6 size 250
!
sip-ua
 no remote-party-id
retry invite 2
 retry bye 2
 retry cancel 2
 sip-server dns:pcclv1n0005.pipiptrunksit2.gsiv.com
g729-annexb override
```

Enable CUBE (“mode” cmd required only on ISRG2s)

T.38 fax – can override per dial-peer

Global config on how to handle/interpret SIP messaging; specific CLI will depend on your specific SP’sUNI

Codec settings and negotiation lists; 15.1.2T enables more features with VCC; earlier, codecs often directly on dial-peer

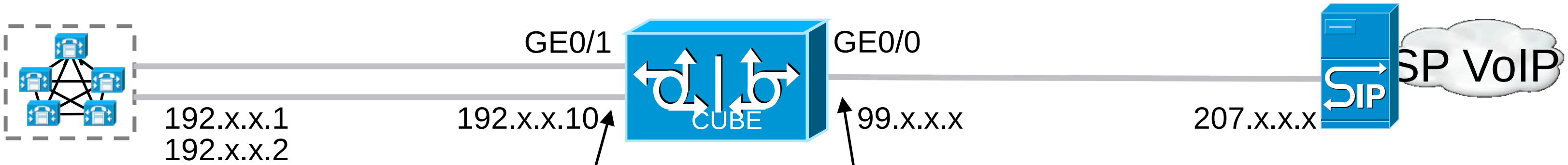
SIP Normalization; remove fax attribute

CAC settings

Tune failover timing (PDD)

Global SIP settings

CUBE Dial-Peer Configuration



DID range:
14085551000-5999

```
voice translation-rule 9
  rule 2 /^9\(.*\)/ /\1/
!
voice translation-profile DIGITSTRIP-9
  translate called 9
```

```
dial-peer voice 102 voip
  description to/from CUCM subscriber
preference 2
  destination-pattern 408555[1-5]...
  voice-class codec 1
  session protocol sipv2
  session target ipv4:192.168.0.4
  incoming called-number 408555[1-5]…
dtmf-relay rtp-nte
  no vad
```

```
dial-peer voice 1 voip
 description OUTBOUND SIP calls to SP
translation-profile outgoing DIGITSTRIP-9
 max-conn 50
 destination-pattern 91[2-9][1-9][1-9]…….
 voice-class codec 1
 session protocol sipv2
 session target ipv4:207.x.x.x
dtmf-relay rtp-nte
 no vad
```

```
dial-peer voice 103 voip
  description to/from CUCM publisher
preference 5
  destination-pattern 408555[1-5]...
  voice-class codec 1
  session protocol sipv2
  session target ipv4:192.168.0.6
  incoming called-number 408555[1-5]...
dtmf-relay rtp-nte
  no vad
```

```
dial-peer voice 2 voip
 description INBOUND SIP calls from SP
max-conn 50
 voice-class sip options-keepalive up-interval
                     10 down-interval 60 retry 2
 session protocol sipv2
 session target sip-server
 incoming called-number 408555[1-5]...
dtmf-relay rtp-nte
 no vad
```

CUBE Xcoding Configuration

```
voice-card 0
dspfarm
dsp services dspfarm
!
interface GigabitEthernet0/1
  description inside interface
ip address 192.x.x.10255.255.255.0
!
sccp local GigabitEthernet0/1
sccpccm192.x.x.10identifier 1 version 7.0
sccp
!
sccpccm group 1
 associate ccm 1 priority 1
 associate profile 1 register XCODE123456
!
dspfarm profile 11 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
maximum sessions 200
associate application SCCP
!
telephony-service
  max-dn 1
ip source-address 192.x.x.10 port 2000
sdspfarm units 1
sdspfarmtranscode sessions 200
sdspfarm tag 1 XCODE123456
```

Make DSPs available as media resources

Define the xcoding call agent, which is CUBE (its own IP address)

Register the Xcoding DSPs with the call agent (CUBE)

Configure the DSPs for xcoding (codecs, sessions…)

Configure CUBE as the controlling call agent (its own IP address)

More information at www.cisco.com/go/cube> Configure > Configuration Examples and TechNotes

CUBE Configuration – Security

```
voice service voip
   sip
      listen-port non-secure 2000 secure 2050
!
interface GigabitEthernet0/0
  description outside interface
ip address 99.x.x.x 255.255.255.0
ip access-group 101 in
!
interface GigabitEthernet0/1
  description inside interface
ip address 192.x.x.10 255.255.255.0
ip access-group 102 in
!
access-list 101 permit udp host 207.x.x.x any
access-list 101 deny udp any anyeq 5060
access-list 101 deny udp any anyeqtftp
access-list 101 deny tcp any anyeq www
access-list 101 deny tcp any anyeq telnet
access-list 101 deny tcp any anyeq ftp
access-list 101 permit ip any any
!
access-list 102 permit udp host 192.x.x.1 any
access-list 102 permit udp host 192.x.x.2 any
access-list 102 deny udp any anyeq 5060
access-list 102 permit ip any any
!
call treatment on
call threshold global cpu-avg low 68 high 75
call threshold global total-calls low 230 high 250
call spike 10 steps 6 size 250
!
sip-ua
  no transport tcp
  permit hostname dns:10.10.10.10
  permit hostname dns:example1.sip.com
```

Change SIP port away from default 5060

Define ACLs to talk only to SP SBC on outside and CUCM on inside

Define CAC to limit total calls, CPU use and spike detection

If only UDP is used, turn off TCP

Restrict hostnames/IP-addresses that can talk to CUBE

CUBE Implementation and Configuration and SP SIP Trunks - Topics

- General CUBE Configuration for SP SIP Trunks
- Advanced CUBE Feature Configuration Examples
- Troubleshooting Configuration

SIP Profile Use Cases: Add

Message: INVITE

Action: Add b=AS:4000sdp-header for video-media line

```
voice class sip-profiles 100
  request INVITE sdp-header Video-Bandwidth-Info add "b=AS:4000"
```

Message: 480 Temporarily Not Available

Action: Add Retry-After sip-header

add-value includes header-name and header-value

```
voice class sip-profiles 100
  response 480 sip-header Retry-After add “Retry-After: 60”
```

Message: INVITEs and REINVITEs

Action: Add “user=phone”

```
voice class sip-profiles 100
  request INVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone SIP/2.0"
  request REINVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone SIP/2.0"
```

Message: 200 response

Action: Add User-Agent header

```
voice class sip-profiles 100
  response 200 sip-header User-Agent add "User-Agent: CiscoSystems-SIP-GW-UA"
```

SIP Profile Use Cases: Remove

Message: All the requests and responses

Action: Remove Cisco-Guid sip-header

```
voice class sip-profiles 100
  request ANY sip-header Cisco-Guid remove
  response ANY sip-header Cisco-Guid remove
```

Message: BYE and CANCEL

Action: Remove Reason header

```
voice class sip-profiles 100
  request BYE sip-header Reason remove
  request CANCEL sip-header Reason remove
```

Message: 100 and 180 responses

Action: Remove Server header

```
voice class sip-profiles 100
  response 100 sip-header Server remove
  response 180 sip-header Server remove
```

SIP Profile Use Cases: Modify

Message: INVITE
Action: Modify From: header to “gateway@gw-ip-address” format,

e.g. change 2222000020@9.13.24.7 to gateway@9.13.24.7

```
voice class sip-profiles 100
  request INVITE sip-header From modify "(<.*:)(.*@)" "\1gateway@"
```

Message: INVITE
Action: replace "CiscoSystems-SIP-GW-UserAgent" with "-" in o= line of SDP

```
voice class sip-profiles 100
  request INVITE sdp-header Session-Owner modify "CiscoSystems-SIP-GW-UserAgent“ "-"
```

Message: INVITE
Action: Convert “sip” uri to “tel” uri in the Req-URI, From and

To headers, e.g. from “sip:2222000020@9.13.24.6:5060” to “tel:2222000020”

```
voice class sip-profiles 100
  request INVITE sip-header SIP-Req-URI modify "sip:(.*)@[^ ]+" "tel:\1"
  request INVITE sip-header From modify "<sip:(.*)@.*>" "<tel:\1>"
  request INVITE sip-header To modify "<sip:(.*)@.*>" "<tel:\1>"
```

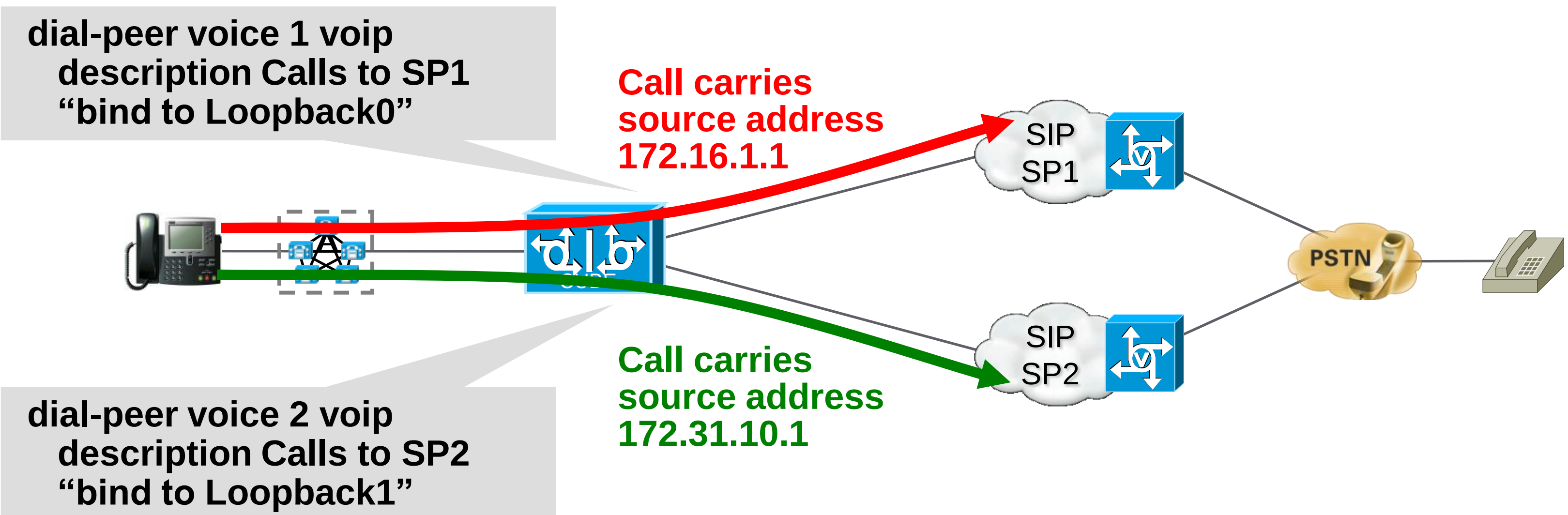
Message: All the requests and responses
Action: Convert “inactive” or “sendonly” to “sendrecv” for MOH

```
voice class sip-profiles 100
  request any sdp-header Audio-Attribute modify “inactive” “sendrecv”
  request any sdp-header Audio-Attribute modify “sendonly” “sendrecv”
  response any sdp-header Audio-Attribute modify “inactive” “sendrecv”
  response any sdp-header Audio-Attribute modify “sendonly” “sendrecv”
```

Dial-peer Bind

15.1.2T

- Enables CUBE to connect to two (or more) different SPs – each SP sees its own IP address for CUBE/SIP trunk
  - Provides a measure of "multi-tenancy" on CUBE
- Allows enterprise customers to connect to multiple SPs for redundancy, call type routing (internal, long distance, local to different providers), TOD routing or least-cost routing
- Can also be used on the CUBE inside interface if there is a need to distinguish source IP addresses for different enterprise network segments or applications



Dial-peer Bind
Configuration

15.1.2T

```
voice service voip
  sip
    bind control source-interface Loopback1
    bind media source-interface Loopback1
!
dial-peer voice 1 voip
  description SP1
  voice-class sip bind control source-interface GigabitEthernet0/0
  voice-class sip bind media source-interface GigabitEthernet0/0
!
dial-peer voice 2 voip
  description SP2
voice-class sip bind control source-interface Loopback0
  voice-class sip bind media source-interface Loopback0
!
dial-peer voice 3 voip
  description internal
```

Global level bind configuration – calls outgoing via dial-peer 3 use this

Calls outbound to SP1 use the IP address of interface GE0/0

Calls outbound to SP2 use the IP address of Loopback0

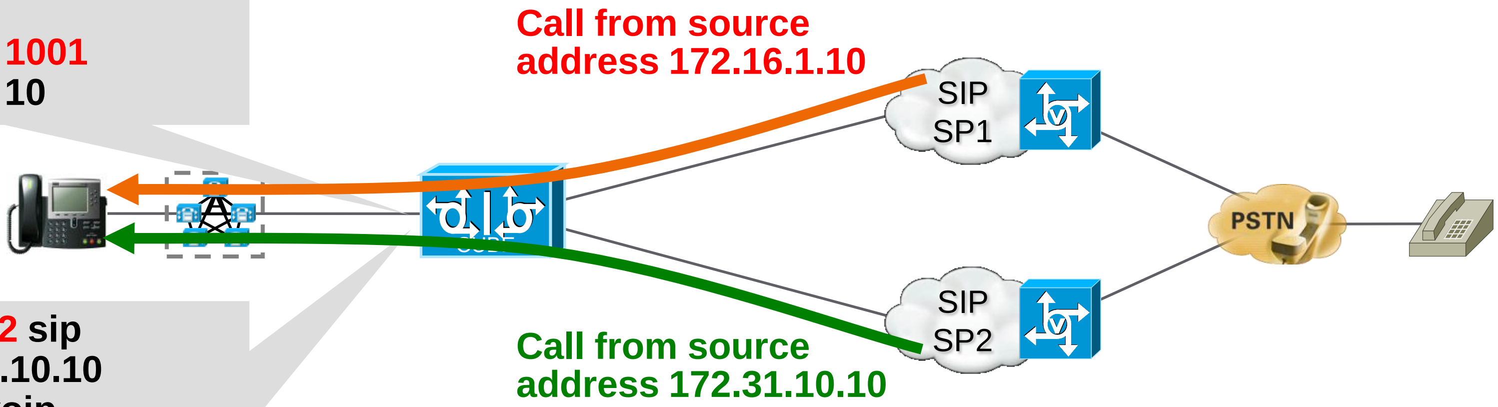
Calls outbound to the internal side of the network (CUCM or other IP PBX) will use the global configuration as there is no dial-peer level configuration

Incoming Dial-peer Match on IP Address

15.1.2T

- Enables CUBE to connect to two (or more) different SPs
- CUBE can apply specific call routing, codec selection, digit manipulation, CAC, QoS, fax treatment, or security policies to the dial-peers specific to SP1 or SP2

```
voice class uri1001 sip
  host ipv4:172.16.1.10
dial-peer voice 1 voip
  description Calls from SP1
  codec g711ulaw
  incoming uri via 1001
  max-connection 10
```




```
voice class uri1002 sip
  host ipv4:172.31.10.10
dial-peer voice 2 voip
  description Calls from SP2
  codec g729
  incoming uri via 1002
  max-connection 30
```

Incoming Dial-peer Match on IP Address
Configuration

15.1.2T

```
voice class uri 1 sip
  host ipv4:9.13.38.82
  host ipv4:9.13.38.84
  host ipv6:[2001:0db8:85a3:0000:0000:8a2e:0370:7334]
  host ipv6:[2001:0db8:85a3:0000:0000:8a2e:0370:7331]
!
voice class uri 2 sip
  host dns:yyy.cisco.com
  host xxx.cisco.com
!
dial-peer voice 1 voip
  description Calls from SP1
  destination-pattern 3328
  session protocol sipv2
  incoming urivia 1
  codec g711ulaw
!
dial-peer voice 2 voip
  description Calls from SP2
  session protocol sipv2
  codec g711ulaw
  incoming urifrom 2
  max-connection 10
```

Define the IPv4, IPv6 and/or DNS source address lists

Use the Via header to match the inbound call address against the list in "voice class uri 1"

Apply G.711 codec policy to calls from sources listed in "voice class uri 1"

Use the From header to match the inbound call address against the list in "voice class uri 2"

Apply CAC policy of max 10 calls from sources listed in "voice class uri 2"

CUBE Implementation and Configuration and SP SIP Trunks - Topics

- General CUBE Configuration for SP SIP Trunks
- Advanced CUBE Feature Configuration Examples
- Troubleshooting Configuration

General CUBE Show Commands

```
show call history ?
   fax              Show calls stored in the history table for fax
   media            Show calls stored in the history table for media
   video            Show calls stored in the history table for video
   voice            Show calls stored in the history table for voice

show call active voice ?
  brief             Show brief version of active voice calls
  compact           Show compact version of active voice calls
  stats             Show voice statistics for the call
  summary           Show voice call summary

show voice call ?
<0-0>               Voice interface slot #
   status           Show status for active calls
   summary          Summary of all voice calls

show voip ?
   debug            Show voip debug info
rtp                 Display Real Time Protocol (RTP) information

show voiprtp ?
   connections      Display all the active RTP connections

show voice statistics ?
csr                 Show Call Statistics Records information
iec                 Show Internal Error Code information
   interval-tag     Show Voice Statistics time-range intervals
   memory-usage     Show current memory utilization of voice statistics
```

General CUBE Debug Commands

Category	Command
SIP and H.323	debug voip ccapi all/input/inout
	debug voip dialpeer all
	debug voip ipipgw
SIP	debug ccsip all/info/messages/media/error
H.323	debug cch323 all
	debug h225 asn/events/q931
	debug h245 asn/events
Transcoding and MTP	debug sccp all/events/messages/errors
	debug dspfarm all
	debug voip xcodemsp
Media	debug rtpspi error
	debug voip rtp all/session/error
	debug voip app

General CUBE Debug Procedures

- If the issue can be reproduced with one call, collect this debugs/show output:

  Configure the logging buffer

  ```
  conf t
  logging console informational
  logging buffer 10000000 debug
  service sequence-number
  service timestamp debug date msec
  end
  debug iecsyslog
  ```

  Enable the debugs

  Clear the logging buffer

  ```
  clear logging
  ```

  Perform test

  Run the show commands

  Copy buffer content to a file

  ```
  term length 0
  show logging | redirect tftp://...
  ```

- If the issue can only be reproduced under call load conditions
  - Enable only the “error” debugs from the given list, repeat the test, then collect the logging buffer and show command output
  - Use PCD (Per-Call Debugging) to trigger debugs only for specific calls

TechWise TV Deep Dive

Troubleshooting
Protocol Ladder Diagrams

1) Configure capture profile

```
! create profile
ip traffic-export profile TAC mode capture
  bidirectional
  incoming access-list 123
  outgoing access-list 123
!
! access-list to filter only SIP messages (port 5060)
access-list 123 permit udp any anyeq 5060
access-list 123 permit tcp any anyeq 5060
!
! apply to an interface, default memory is 5M
interface fa0/0
ip traffic-export apply TAC [size <bytes>]
```

2) Capture traffic with these exec (enable) level commands

Note: The exec cmds don't appear until a profile has been configured

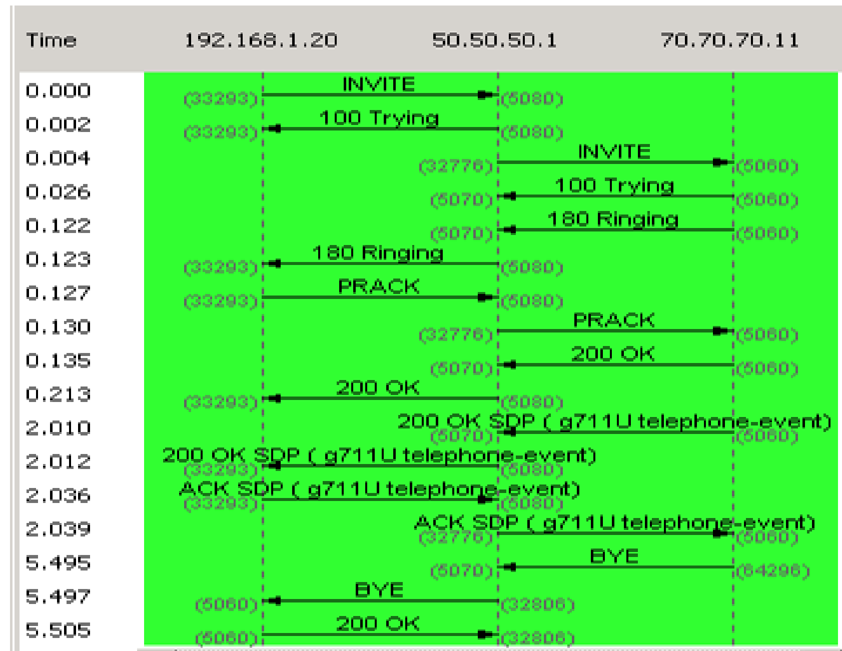
```
router#traffic-export interface fa0/0 clear
router#traffic-export interface fa0/0 start
<capture the problem>
router#traffic-export interface fa0/0 stop
```

3) Export the pcap file to a server

```
router#traffic-export interface fa0/0 copy
ftp://x.x.x.x/capture.pcap
```

4) Display ladder diagram (with Wireshark)

Note: Allows filtering of calling/called numbers when creating the flow graph



IP Traffic Capture: http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/ht_rawip.html

High-Traffic Troubleshooting
Per-Call Debugging (PCD)

- Capture per-call debug into circular memory buffers
- If a trigger-point is hit, export buffer content
  - SIP 4xx, 5xx and 6xx error messages
  - Q.850 cause codes
  - CAC limits
- Examine buffer content on router or on an offline system

1. Define buffers and buffer sizes

```
per-call num-buffer <num>
per-call buffer-size debug <num>
```

2. Turn per-call debugging on/off

```
per-call shutdown
per-call active debug
per-call inactive
```

3. Set trigger points

```
per-call trigger cause 1
per-call trigger cause 41
per-call trigger sip-message 404
per-call trigger sip-message 488
```

4. Export debug buffer content

```
per-call export primary [flash | ftp | http | pram | rcp |
tftp] secondary [flash | ftp | http | pram | rcp | tftp]
```

5. Show buffer content status

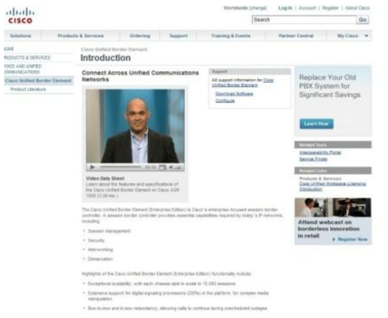
```
show per-call stat
show per-call buffer list
show per-call buffer content <buf-id>
show voice per-call trigger
```

6. Show buffer contents on console

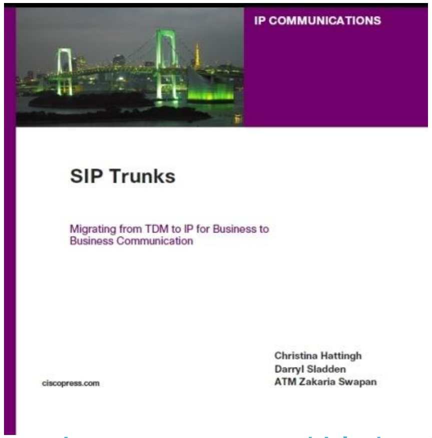
```
router#show per-call buffer content ?
<0-10000000>  Specify the buffer number

router#show per-call buffer content 1
```

Cisco.com SIP Trunk and CUBE Resources



- Cisco UBE on Cisco.com
  - http://www.cisco.com/go/cube
- Cisco Communications Transformations Whitepapers
  - Section on Whitepapers
- Cisco Support Community
  - https://supportforums.cisco.com/community/netpro/collaboration-voice-video/ip-telephony
- Cisco Interoperability Portal
  - Cisco UBE SP SIP Trunk Interoperability Reports
  - Cisco UBE PBX Interoperability Reports (Avaya/Nortel)
- Cisco SRND Portal
  - www.cisco.com/go/srnd
  - CUCM SIP Trunk Documentation
    - CUCM8.xSRND
    - CUCM7.xSRND
    - CUCM6.xSRND
  - CVP 7.0 SIP Trunk Integration
- Marketing Support: ask-cube@external.cisco.com
- TechWise TV: SIP, Session Management and Beyond
  http://www.youtube.com/watch?v=YFoLTsqEI0w



www.ciscopress.com/title/1587059444

Cisco.com SIP Trunk Design Documents

Document	Coverage	Location
CUCM 7.x SRND	CUCM Connectivity to SIP Trunks	cisco.com/go/srnd - Unified Communications - Unified Communications Manager - View Design Guide (CUCM7.x) - Cisco Unified CM Trunks - Cisco Unified Border Element
CVP 7.x SRND	Contact Center: CVP + CUBE	cisco.com/go/srnd - Unified Communications - Voice Portal - View Design Guide (CVP 7.x) - Gateway Options - Cisco Unified Border Element
CUBE in Contact Center Configuration Guide	Contact Center: CVP + CUBE	http://cisco.com/en/US/docs/voice_ip_comm/unified_communications/cubecc.html
SP SIP Trunk Interop	CUCM/CUBE Validation testing with specific SP Offerings: - AT&T TollFree, FlexReach, VoEVPN - Allstream - Verizon - Paetec	cisco.com/go/interoperability Cisco Unified Border Element

Polling Question 3

As a result of this presentation

a) We are able to understand the technical details of Cisco UBE, and broaden the knowledge in the industry.

b) We understand the function of Cisco UBE and might consider using it in a future plan

c) We are more confident about the Cisco UBE implementation, we will accelerate the deployment.

d) We understand how to integrate Cisco UBE with 3rd party SBC in our implementation.

e) We think Cisco UBE is great, we might need more detailed information,

Submit Your Questions Now

Use the Q&A text box to submit your questions

Q&A

We Appreciate Your Feedback!

The first 5 listeners
who fill out the Evaluation Survey
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Ask The Experts Event (with Randy Wu)

If you have additional questions, you can ask them to Randy here:

https://supportforums.cisco.com/community/netpro/ask-the-expert

He will be answering from July 26th to August 5.

Next CSC Expert Series Webcast

Topic: Introduction to MPLSVPN

Wednesday August 17, at

4:30 p.m. IST (India UTC +5:30 hrs)
1:00 p.m. CEST Brussels (UTC +2),
7:00 a.m. EDT New York (UTC -4).

Join double CCIE, Senior Cisco Support Engineer **Nagendra Kumar** from **Bangalore,** India.

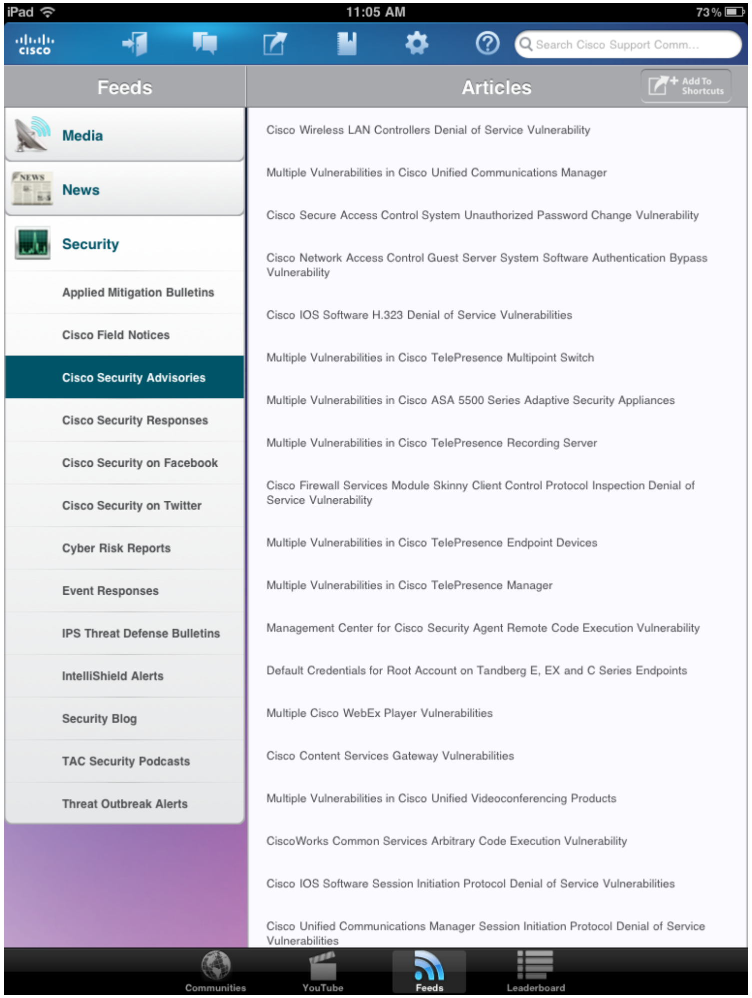
He will provide an Introduction to MPLSVPN and discuss common terminology, configuration, and best practices in setting up MPLSVPN networks.

During this interactive session you will be able ask all your questions related to this topic.

Register for this live Webcast at

www.CiscoLive.com/ATE

iPad support app available July 1st



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