Possibilities

#CiscoLive

Design and Scale Cisco ISR 4000 & VG Series for Unified Communications

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DGTL-BRKARC-2100

About me

Know your speaker

- Joined Cisco in 2016
- Technical Marketing Engineer, Voice & Unified Communications on Enterprise Routing Platform
- Previous roles at Cisco:
  - Technical Marketing Engineer, Content Security (Cisco Web Security Appliance)
  - Technical Consulting Engineer, Cloud Collaboration (Cisco Webex Portfolio)
- 10+ years of experience in networking industry (Collaboration & Security)
- Based out of Bangalore, India

Lokesh Kumar Lal

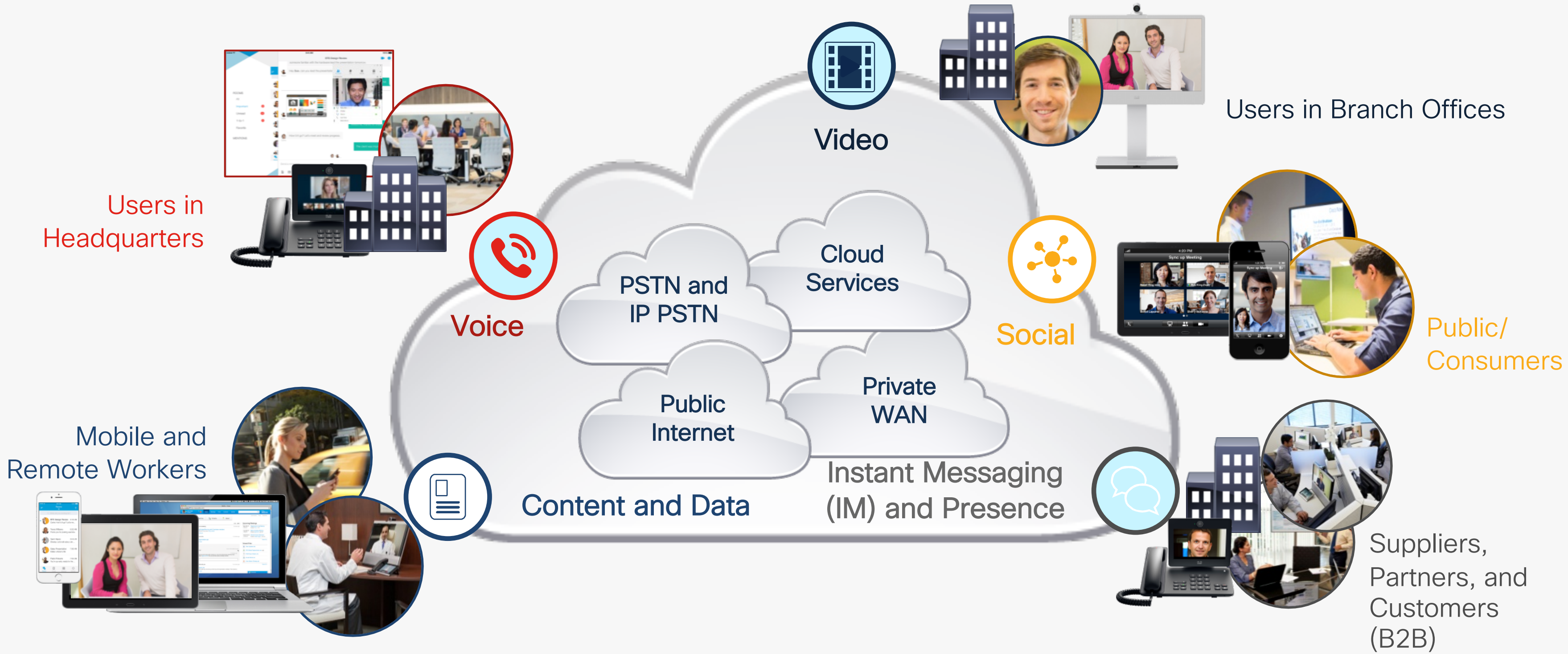
Agenda

Design and Scale ISR 4000 & VG Series for Unified Communications
DGTL-BRKARC-2100

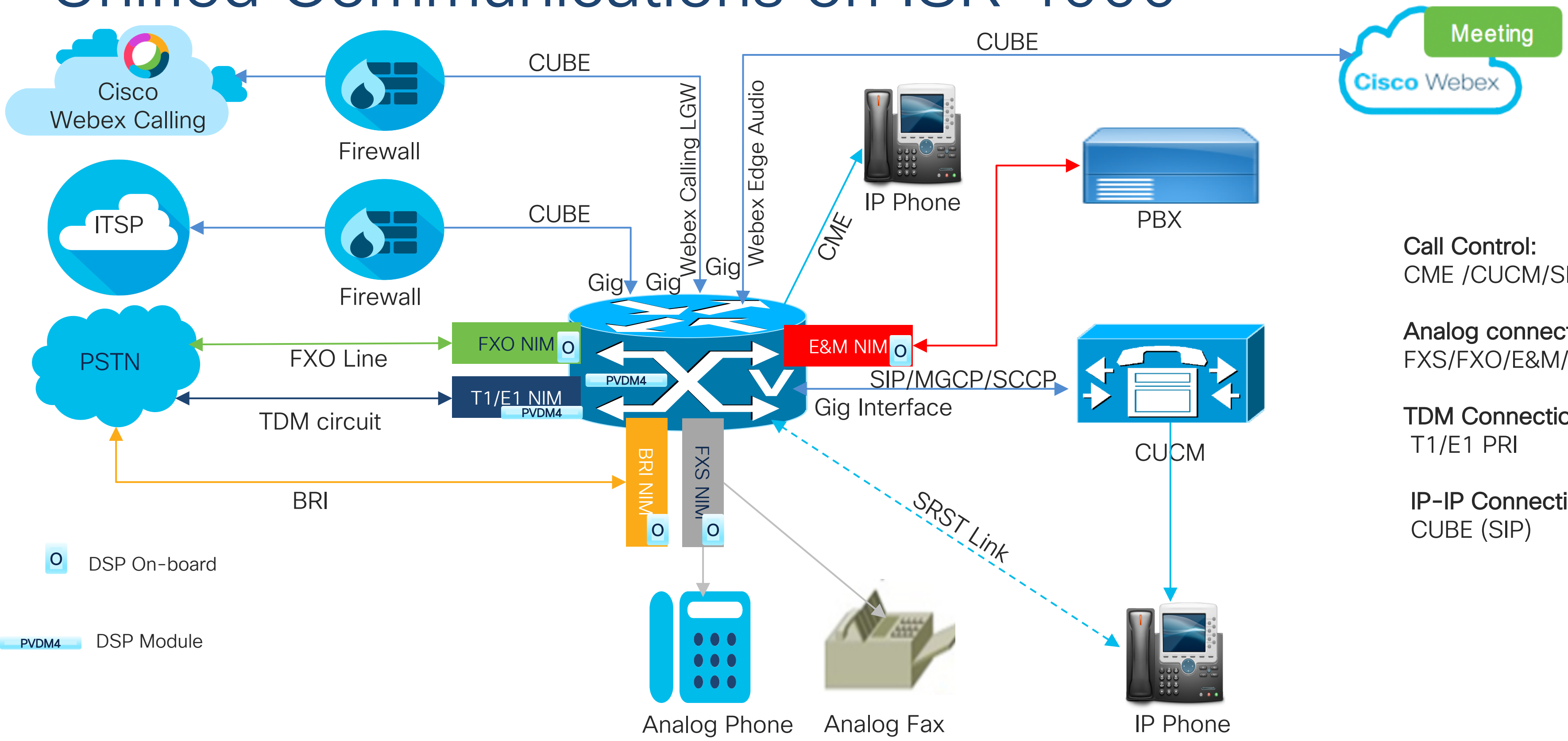
- UC Portfolio on Cisco 4000 Series ISR & VG Series Gateway
- Added Capabilities with Next Gen Voice Gateways
- BYoPSTN for Cloud Calling
- Voice Platform Sizing & Positioning
- Summary

UC Portfolio on Cisco 4000 Series ISR & Cisco VG Series Gateway

The Collaboration Landscape



Unified Communications on ISR 4000



Call Control:
CME /CUCM/SRST

Analog connection:
FXS/FXO/E&M/BRI

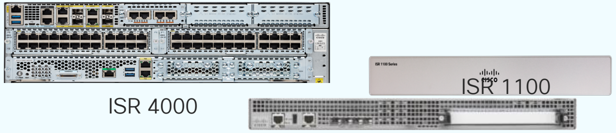
TDM Connection:
T1/E1 PRI

IP-IP Connection:
CUBE (SIP)

Unified Communication Offerings

Voice Router

- **ISR 4000** UC router for TDM and IP Voice services
- **ASR 1000, ISR 1100** support UC IP services
- **CSR1000v** for virtual UC IP services



ISR 4000 ISR 1100 ASR 1000

Voice Gateways

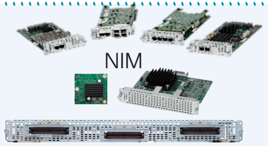
- Supports traditional devices (analog phones, fax machines, paging solution)
- **Fixed port** analog voice gateway **(VG202XM, VG204XM, VG400)**
- **Low to ultra high-density** gateways **(VG310,VG320,VG450)**

VG Series

Platform

Voice Modules

- **NIM modules** for Digital and Analog connections
- **4th Gen Packet Voice DSP Module (PVDM4)** for IP and TDM services
- **SM modules** for high density PVDM (768 to 3080 channels), high density FXS modules(up to 144 ports), **NIM adaptors** for SM to NIM conversion



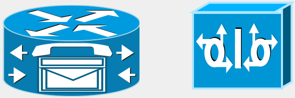
NIM

SM

Module

Call Control

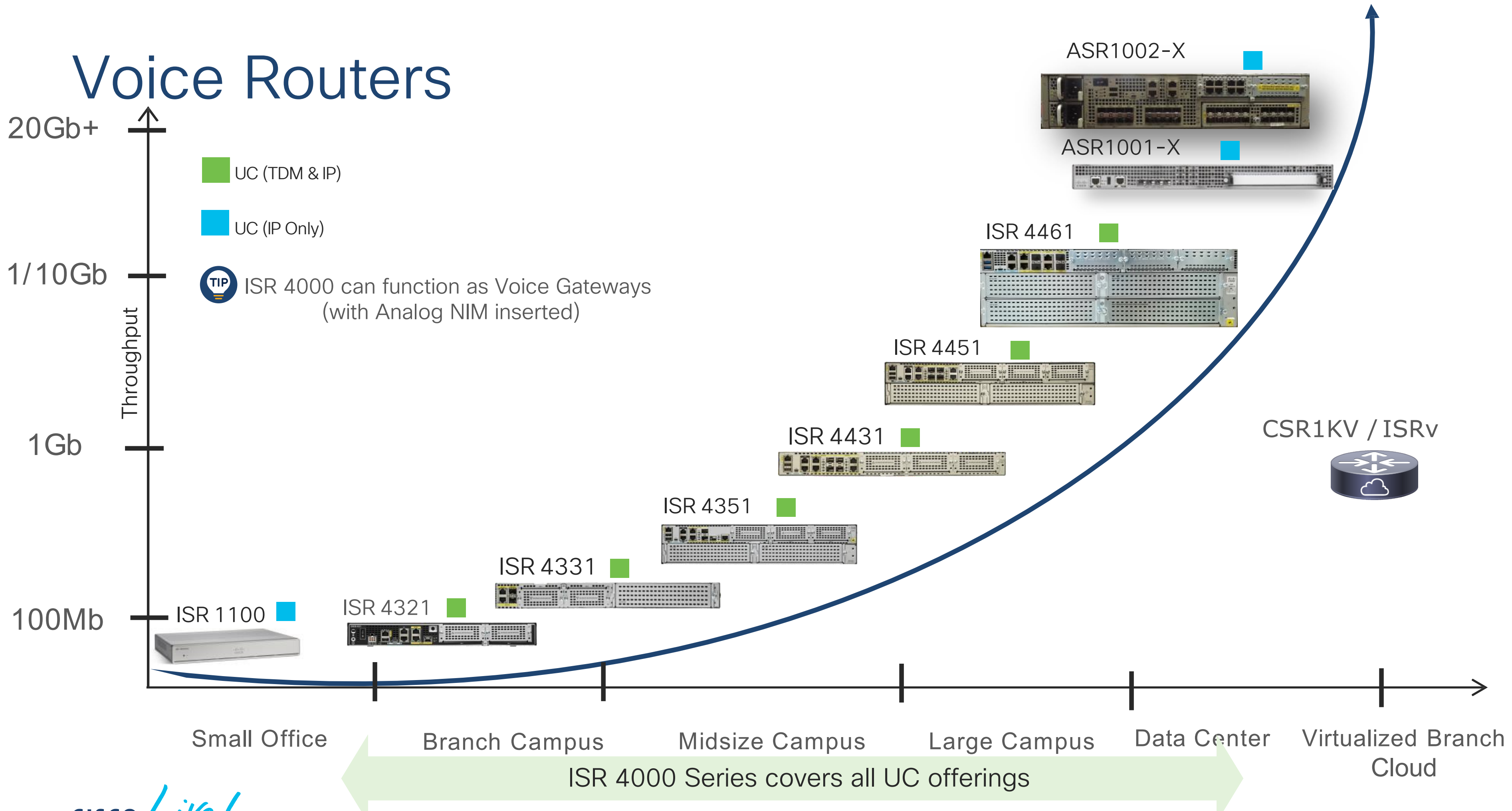
- **Communications Manager Express (CME)** for Call control
- **Cisco Unified Border Element (CUBE)** SBC for SIP calls
- **Survivable Remote Site Telephony (SRST)** for backup
- Secure voice Calling



CME/SRST CUBE

Software

Voice Routers



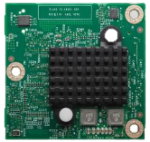
Voice Modules

Digital Voice

T1/E1 Multiflex Trunk NIM

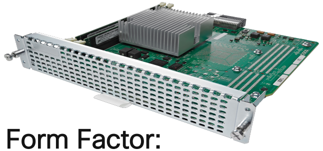
Form Factor:
NIM-1MFT-T1E1
NIM-2MFT-T1E1
NIM-4MFT-T1E1
NIM-8MFT-T1E1

Packet Voice DSP Module



Form Factor:
PVDM4 – 32
PVDM4 – 64
PVDM4 – 128
PVDM4 – 256

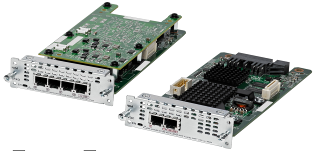
High Density DSP SM



Form Factor:
SM-X-PVDM-500
SM-X-PVDM-1000
SM-X-PVDM-2000
SM-X-PVDM-3000

Analog Voice

FXS/FXO NIM



Form Factor:
NIM-2FXO
NIM-4FXO
NIM-2FXS/P*
NIM-4FXS/P*
NIM-2FXS/4FXO/P*

BRI NIM

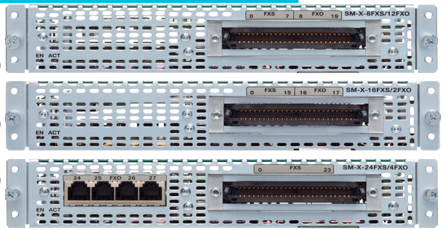
Form Factor:
NIM-2BRI-NT/TE
NIM-4BRI-NT/TE

E/M NIM

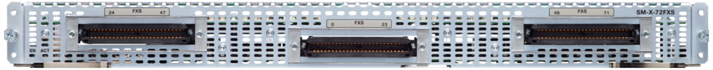
Form Factor:
NIM-4E/M

High Density Analog SM

Form Factor:
SM-X-8FXS/12FXO
SM-X-16FXS/2FXO
SM-X-24FXS/4FXO



Single Wide Module

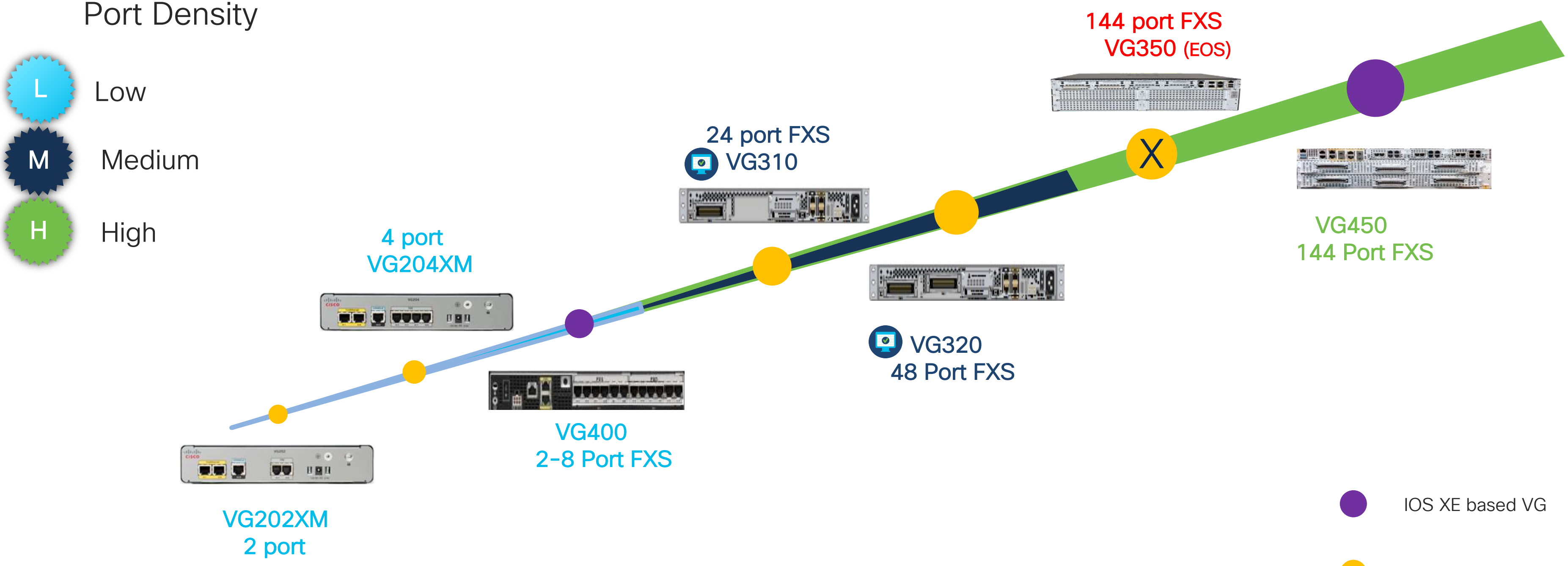


SM-X-72FXS

Double Wide Module

Cisco Analog Voice Gateway Series

Port Density



IOS XE based VG

IOS based VG

X End of Sale VG

TIP IOS XE based Voice gateways are **recommended**

IOS 15.9(3)M released

IOS XE Voice Gateways: VG400

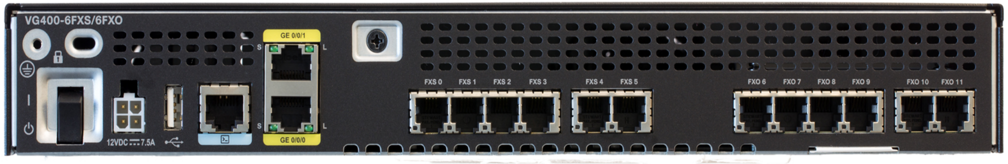
Fixed Port Models

VG400-2FXS/2FXO

VG400-4FXS/4FXO

VG400-6FXS/6FXO

VG400-8FXS



Internal: AC PSU	1 RU Form Factor
DRAM – 4 GB	Flash – 8GB
Fixed Analog Ports (4-12)	2 x Gig Ethernet Ports

CUCM:
11.5.1SU7 or higher
12.0.1SU2 or higher

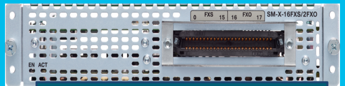
IOS XE:
IOS XE 16.10.1 or later

Call Control:
No CME/SRST/CUBE Support

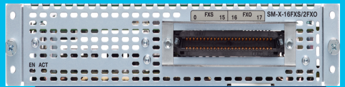
DSP:
On Board DSP
No DSPFARM Support

IOS XE Voice Gateways: VG450

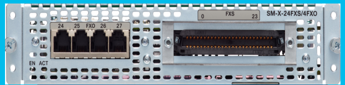
HW Configuration Options



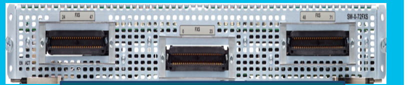
SM-X-8FXS/12FXO



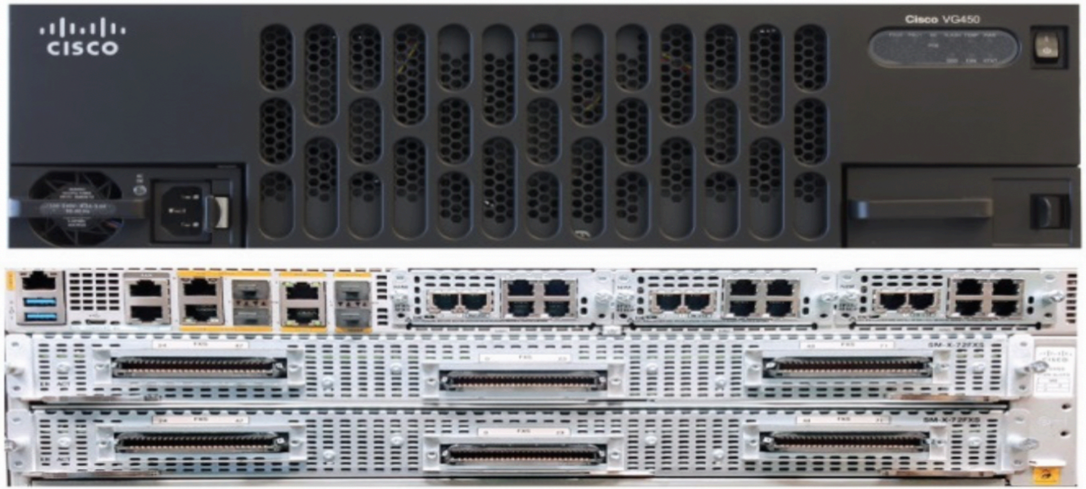
SM-X-16FXS/2FXO



SM-X-24FXS/4FXO



SM-X-72FXS



Internal:
Dual AC/DC PSU

3 RU Form Factor

DRAM – 8/16/32 GB

Flash – 8/16/32 GB

3 x NIMs, 3 x SMs

4 x Gig Ethernet Ports

CUCM:
10.5.2 (SU8),
11.5.1SU6 or higher
12.0.1SU2 or higher

IOS XE:
IOS XE 16.9.2 or later

Call Control:
No CME/SRST/CUBE support

DSP:
On Board DSP
No DSPFARM Support

TIP

Support for Online Insertion and Removal **(OIR)**

Added Capabilities with Next Gen Voice Gateways

Next Gen VG's

- ISR 4000 with Analog NIM
- VG450
- VG400

New Capabilities
Next Generation Voice Gateways

Next Gen VGs

Hardware capabilities:

- Inbuilt DSP capability
- FXS Extended loop length with up to 18000 ft. (24 AWG)
- Support for online insertion & removal of analog NIM
- FXO failover bypass

Software capabilities:

- SCCP Auto registration & Self provisioning
- SIP line side registration
- Native Media recording

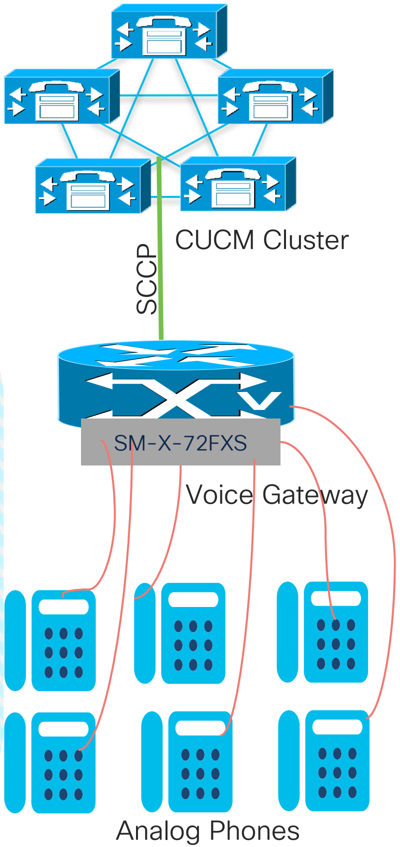
FXS Auto-registration & IVR Self-provisioning
Use Case

A Retail corporation with a huge install-base of Class IOS VGs spread over 5000 Stores looking to migrate to newer Voice Gateway platform with high density analog modules and CUCM for centralized management.

Challenges:
- Seamless migration with reduced port-level configuration
- Easy Day 2 add/remove changes without admin intervention

Solution:
- Add VG as SCCP gateway & enable auto registration & DN auto-assignment for FXS ports on CUCM
- IVR self-provisioning for day-2 changes

Benefits:
- No port-level configuration.
- Self-provisioning IVR for easy onboarding and DN changes without admin intervention.
- No additional configuration on CUCM in gateway replacement scenario. (Use virtual mac address)



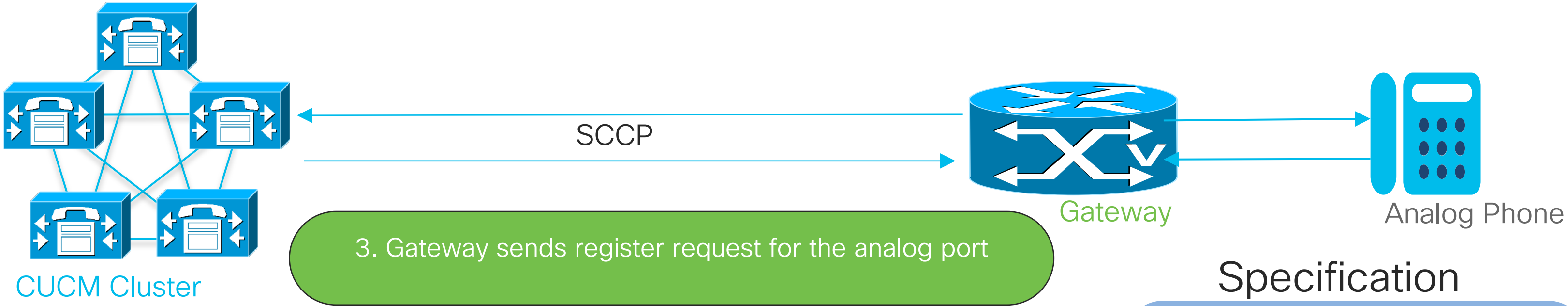
FXS Auto-registration & Self-provisioning
Configuration Checklist

1a. Configure Gateway in CUCM & assign Auto-Reg enabled DP

2a. Add gateway slots and modules

1b. Configure CUCM IP in gateway as SCCP & enable auto-config

2b. Gateway downloads TFTP config file from CUCM.



3. Gateway sends register request for the analog port

4. CUCM maps slot/module/port number to gateway based on MAC address in incoming Register request

5. CUCM assigns DN to port from Auto-Reg DN Pool

6 .User can start IVR self-Provisioning

TIP

- Device Pool assigned to the voice Gateway should have CM with Auto Registration enabled

Specification

Platform
ISR 4000 , VG450, VG400

IOS-XE
17.1 and above

CUCM
CUCM 12.5 (SU2) and above

Analog SIP-Line Registration
Use Case

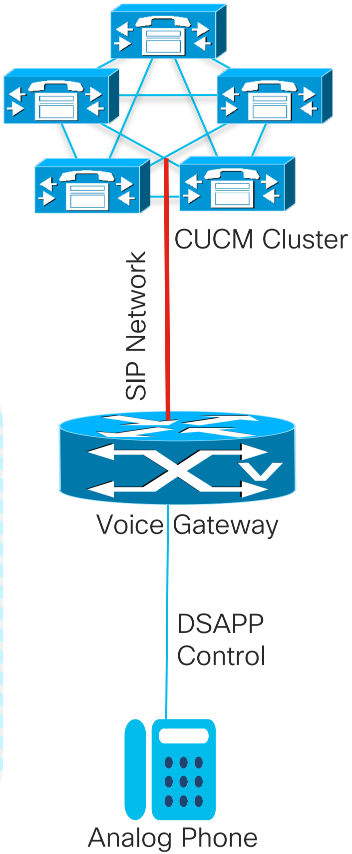
An Existing government customer with over 230 voice gateways deployed in SIP trunk mode and pointing to CUCM for analog phones to communicate with IP Phones.

Challenges:
- No Supplementary services in SIP trunk mode
- Cannot move away from SIP standard to Cisco proprietary SCCP

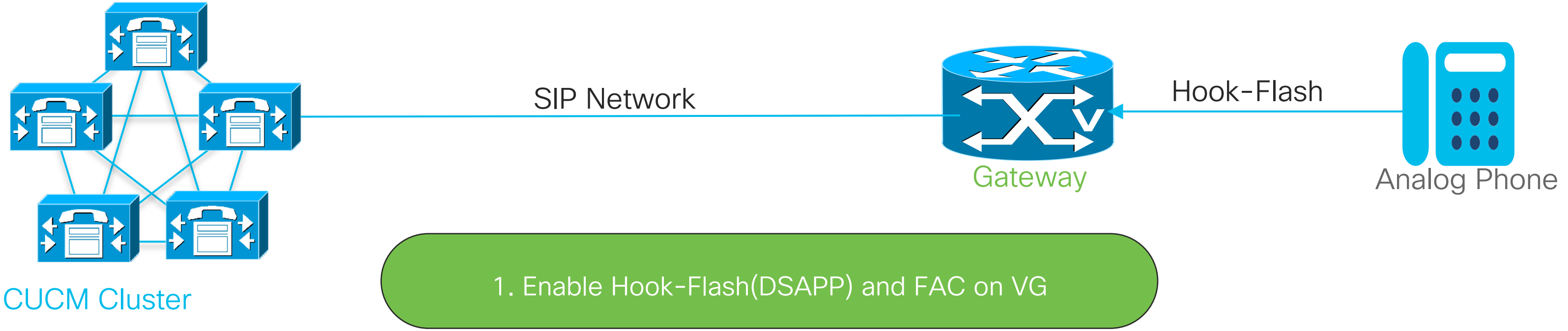
Solution:
- Add VG as SIP gateway on CUCM to register FXS ports as SIP endpoints
- Enable DSAPP& FAC on VG for CUCM to subscribe to HF & softkey events

Benefits:
- SIP supplementary services for analog phones
- Uniform SIP based network
- Future scope for add-ons & third-party call control

Note: DSAPP is **Device control Session Application**
FAC is **Feature Access Code**



Analog SIP-Line Registration
Configuration Checklist



1. Enable Hook-Flash(DSAPP) and FAC on VG
2. Configure Voice-Card and enable SIP-UA
3. Add dial-peers (SIP / POTS) and enable DSAPP control
4. Add VG on CUCM and configure module/slot/subslot
5. Assigns DN to port and configure call fwd/pickup settings

TIP

- SIP line VG registration supports Auto-configuration
- Only line-side POTS dial-peers are automatically added on VG

Specification

Platform
ISR 4461 , VG450

IOS-XE
16.12.1 and above

CUCM
CUCM 12.5 (SU1) and above

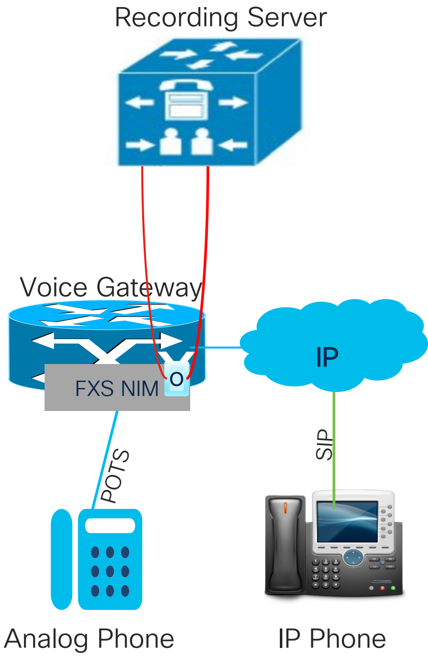
Native Media Recording
Use Case

Financial trading customer with media recording enabled as a two-box solution with a voice gateway and a separate box for CUBE on ISR.

Challenges:
- VG doesn't support CUBE
- Fragmented solution with complex configuration
- Added licensing & device cost (Opex/Capex)

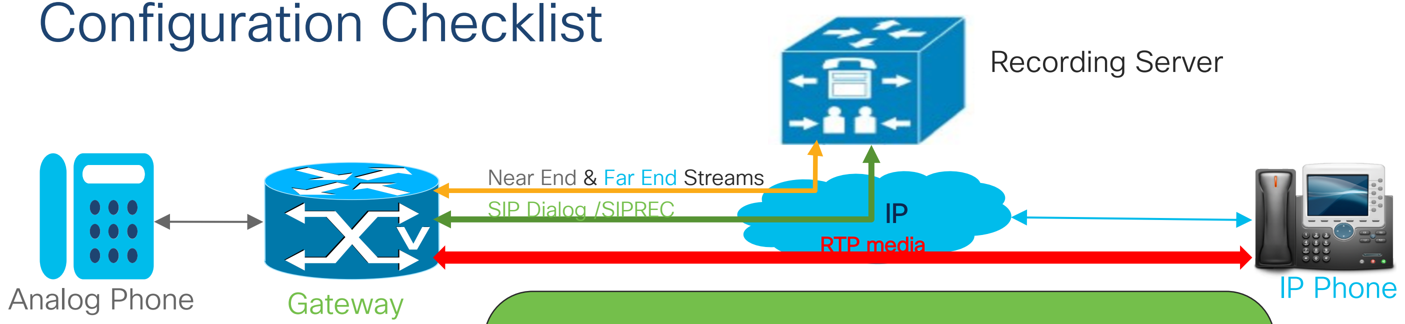
Solution:
- Enable Analog(FXS/FXO) port based SIPREC standard media recording
- DSP on-board(NIM) used to fork near-end/far-end media streams
- Cover TDM Flows: POTS-to-POTS, POTS-to-VoIP, VoIP-to-POTS

Benefits:
- Meet Compliance & regulatory requirements
- Simple implementation with a single box recording solution
- Eliminate the need for additional CUBE licensing or separate hardware to save cost



Native Media Recording

Configuration Checklist



1. Configure media-recording feature license count (Voice service pots)

2. Add SIP-Recorder VoIP dial-peer

3. Add POTS dial-peer and enable media recording under it & map it with SIP-Recorder dial-peer

4. Add dial-peer pointing to far-end device (depending on the call flow)

TIP

- Only G711 & G729 codes are supported for VoIP flows
- In POTS-to-POTS flow enable media recording on only one POTS dial-peer
- All analog NIMs & SM are supported

Specification

Platform
ISR 4000, VG450, VG400

IOS-XE
16.10 and above

Recording Solution
Verint, Call Recording Center(CrC)

BYoPSTN for Cloud Calling

Cisco Cloud Calling Portfolio

Cisco Cloud

Webex Calling

- Mid-market and large enterprises
- Multi-tenant cloud
- Proven cloud PBX

BYoPSTN

UCM Cloud

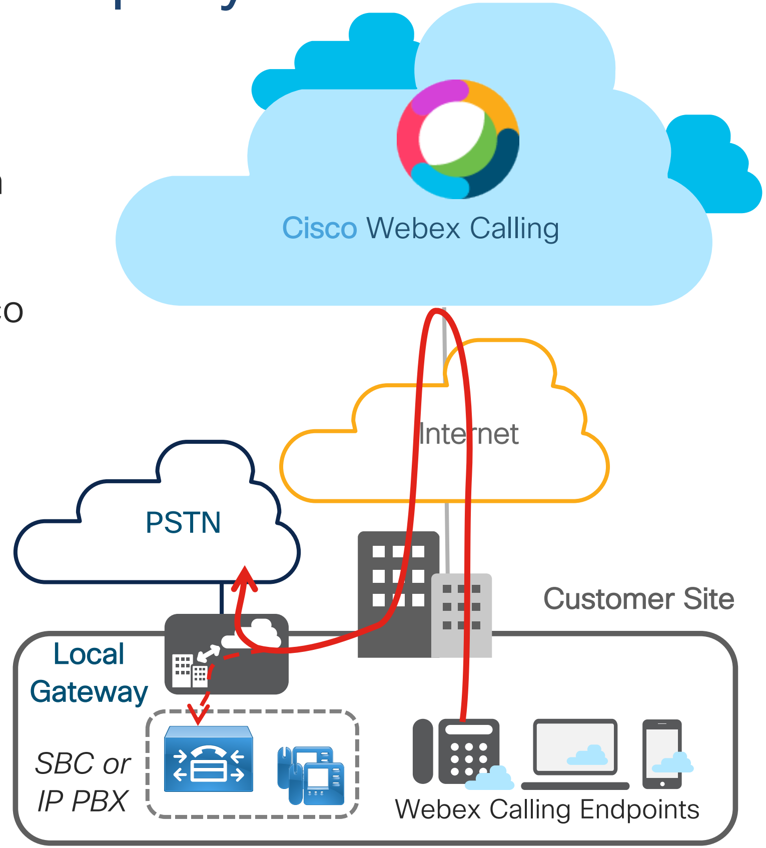
- Brownfield (on-prem UCM) migrations
- Large enterprises and gov't agencies
- UCM/Jabber® features

ByoPSTN

Webex Calling

Webex Calling Local Gateway Deployment

- Enables BYoPSTN option for Webex Calling
- Provide connectivity to a dedicated SBC/PSTN GW or an on-premises IP PBX
- Supports Cisco CUBE (for IP-based connectivity) or Cisco IOS Gateway (for TDM-based connectivity)
- CUBE calling licenses included in Webex Calling Flex License



TIP

Note:
Support for **ISR 4461** needs IOS-XE 17.2 and above

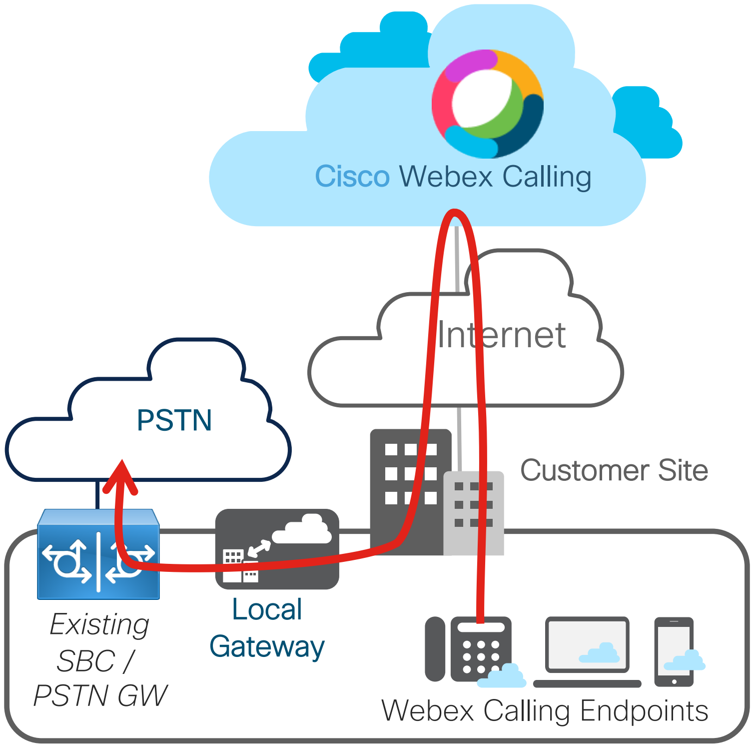
Specifications:

Platform	IOS-XE
ISR 4000 , CSR1000v, ISR 1000	16.12.2 and above

Customer Deployments – Scenario 1

Dedicated PSTN GW/CUBE Variant
(Preferred Option)

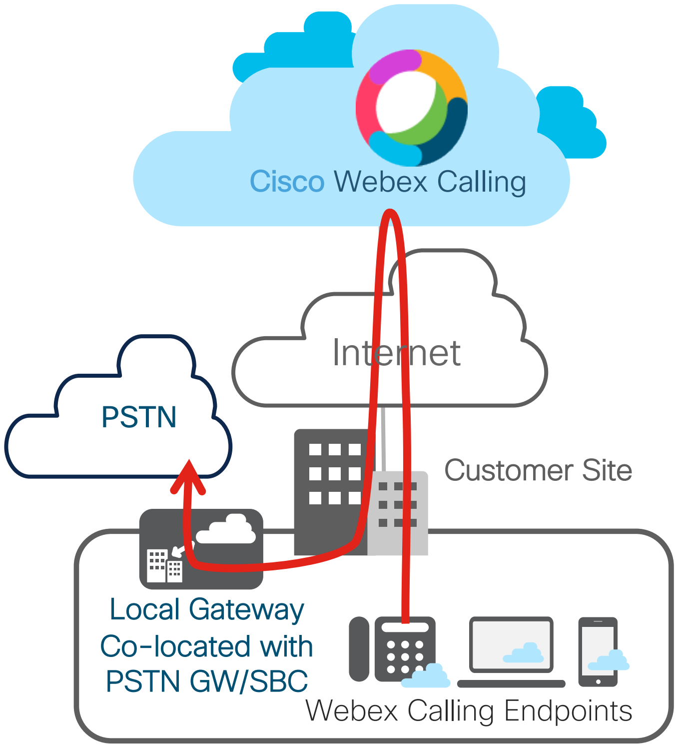
- PSTN GW/SBC and Webex Calling Local Gateway on different boxes
- Needed if existing PSTN GW is not a CUBE or managed by a different SP
- Applicable to both single site & multisite
- Local GW routes all calls coming from BroadCloud to PSTN GW/SBC (and vice versa)



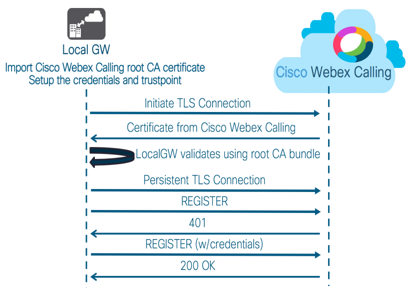
Customer Deployments – Scenario 2

Single Site with Local GW and PSTN GW/SBC (Co-located)

- PSTN GW/SBC and Webex Calling Local Gateway collocated
- Deployment suited for local gateway supported platforms
- Routes all calls including PSTN destinations & on-net calls towards CUCM
- Local Gateway PSTN connectivity may be IP-based (CUBE) or TDM-based (IOS-XE GW)



Webex Calling - Local Gateway Deployment
Configuration Checklist



TIP

Platform Configuration

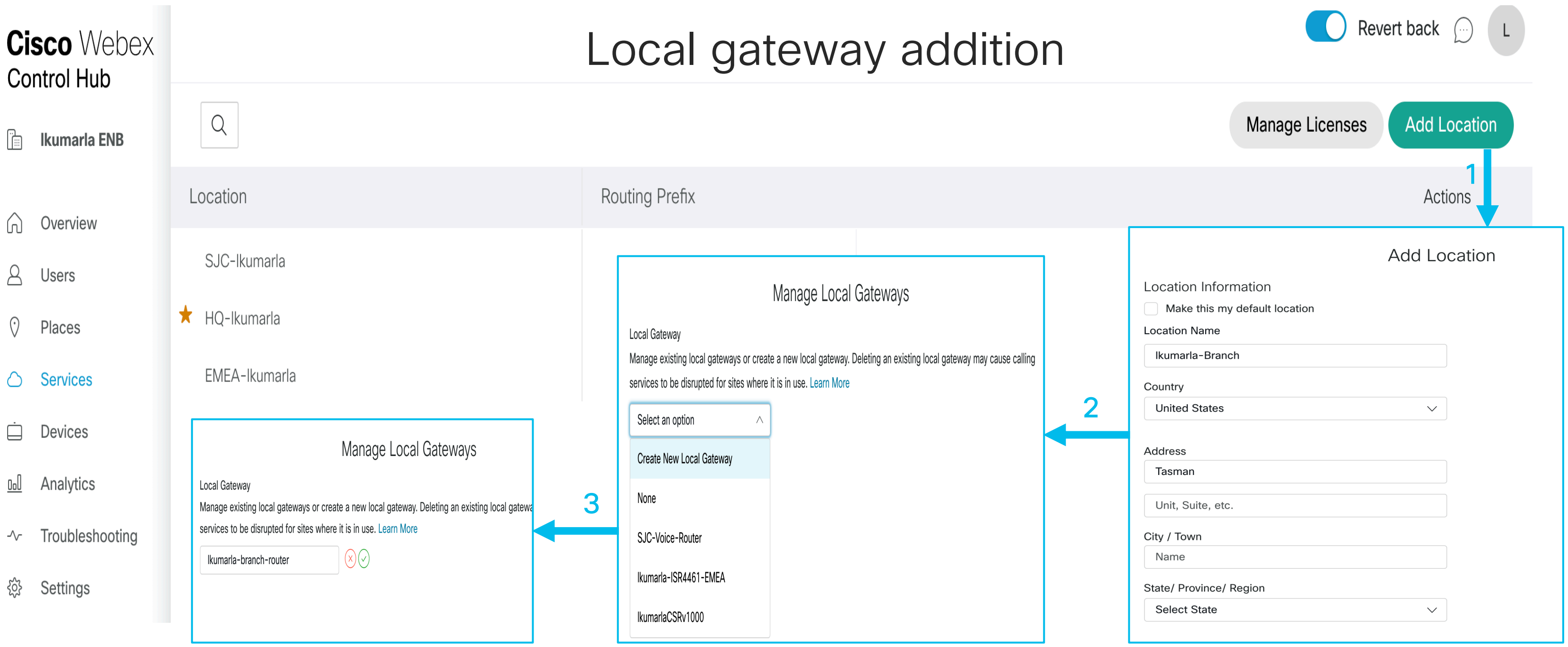
- Meet Webex local gateway network requirements prior to configuration
- Install DigiCert Root CA & enable TLS 1.2
- for Webex calling cert validation & TLS connection with LGW
- Enable STUN for firewall traversal (UDP)
- Configure/disable IP address Trusted list configuration

1. Create Location on Webex Control Hub
2. Add Local gateway and retrieve credentials
3. Create Voice Class Tenant
4. Add Voice Class URI configuration
5. Add Voice dial-peers (Outbound/Inbound)
6. Add Voice class Dial-peer group

Configuration Logic:

- Voice class tenant
  Tenant config for Cisco Webex calling facing dial-peers
- Voice class uri
  Patterns defining host IP addresses/ports for various trunks terminating on Local Gateway
- Outbound dial-peers
  Route outbound call legs from LGW to ITSP SIP trunk and Webex Calling
- Inbound dial-peers
  Accept inbound call legs from ITSP and Webex Calling
- Voice class dpg
  Target outbound dial-peer(s) invoked from an inbound dial-peer

Webex Control Hub Webex Calling - Local Gateway Deployment



Tenant Configuration

Manage Local Gateways

Manage existing local gateways or create a new local gateway. Deleting an existing local gateway may cause calling services to be disrupted for sites where it is in use. Learn More

SJC-Voice-Router

SJC-Voice-Router Info

Status

- Online

Registrar Domain
81213435.cisco-bcld.com

Trunk Group OTG/DTG
sjc-voice-router8902_lgu

Line/Port
SJC-Voice-Router1252_LGU@81213435.cisco-bcld.com

Outbound Proxy Address
la01.sipconnect-us10.cisco-bcld.com

Authentication Information

Retrieve the username and password for SJC-Voice-Router. Each time authentication information is retrieved, a new password is generated for this location. During the password generation, PSTN is disrupted until the new password is saved.

Retrieve Username and Reset Password

Locations using SJC-Voice-Router 1

SJC-lkumarla

Local Gateway CLI Configuration

```
voice class tenant 1
  registrar dns:81213435.cisco-bcld.com scheme sips expires 240 refresh-ratio 50 tcp tls
  credentials number SJC-Voice-Router1252_LGU username SJC-Voice-Router8902_LGU password 0 G}i1NsI55[ realm BroadWorks
  authentication username SJC-Voice-Router8902_LGU password 0 G}i1NsI55[ realm BroadWorks
  authentication username SJC-Voice-Router8902_LGU password 0 G}i1NsI55[ realm 81213435.cisco-bcld.com
  sip-server dns:81213435.cisco-bcld.com
  connection-reuse
  srtp-crypto 1
  session transport tcp tls
  url sips
  error-passthru
  asserted-id pai
  bind control source-interface GigabitEthernet1
  bind media source-interface GigabitEthernet1
  no pass-thru content custom-sdp
  sip-profiles 1
  outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
```

```
voice class sip-profiles 1
 rule 1 request ANY sip-header SIP-Req-URI modify "sips:" "sip:"
 rule 9 request ANY sip-header SIP-Req-URI modify "sips:(.*)" "sip:\1"
 rule 10 request ANY sip-header To modify "<sips:" "<sip:"
 rule 11 request ANY sip-header From modify "<sips:" "<sip:"
 rule 12 request ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
 rule 13 response ANY sip-header To modify "<sips:" "<sip:"
 rule 14 response ANY sip-header From modify "<sips:" "<sip:"
 rule 15 response ANY sip-header Contact modify "<sips:" "<sip:"
 rule 16 request ANY sip-header From modify ">" ";otg=SJC-Voice-Router8902_LGU>"
 rule 17 request ANY sip-header P-Asserted-Identity modify "<sips:" "<sip:"
```

SJC-Voice-Router Authentication Information

Record the username and password below. If you lose this information, you will need to reset the password again.

Username
SJC-Voice-Router8902_LGU

Password
G}i1NsI55[

Done

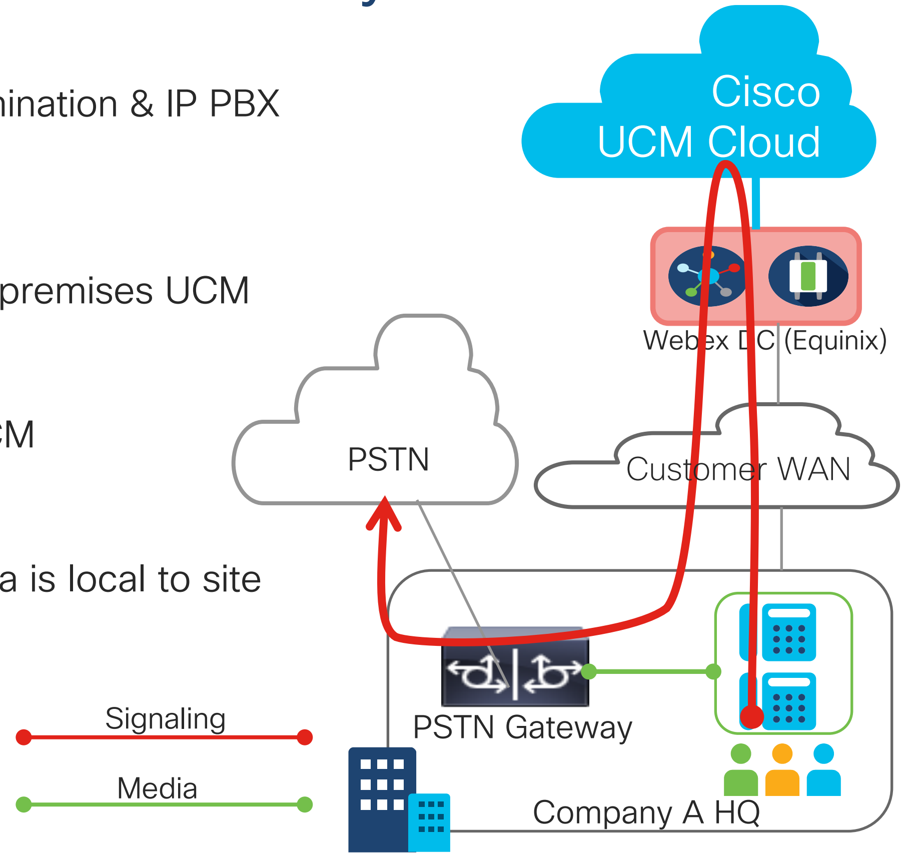
```
SJC-lkumarla-V1#show sip-ua register status

Tenant:  1
-------------------- Registrar-Index  1 --------------------
Line                             peer       expires(sec) reg survival
============================== ========= ============ === ========
SJC-Voice-Router1252_LGU          -1         52             yes normal
```

ByoPSTN

UCM Cloud

UCM Cloud Local Breakout Gateway

- Customer managed edge device for PSTN termination & IP PBX internetworking
- Voice gateway platform support parity with on-premises UCM
- Gateway configuration same as on-premise UCM
- Signaling traverses through UCM Cloud & media is local to site
- SIP Gateway recommended



Voice Platform Sizing & Positioning

Platform Positioning

Platform

- CSRv
- ISR
- ASR

Factors

- Features
- Functionality
- Module Capacity
- Specifications
- Cost

Platform Support Summary

Platform	CME	SRST	CUBE	TDM	DSP Farm	Analog NIM	UC SD-WAN
ISR 4000	Supported	Supported	Supported	Supported	Supported	Supported	Supported
ISR 800	Supported	Supported	Supported	Not Supported	Not Supported	Not Supported	Not Supported
CSR1000v	Supported	Not Supported	Supported	Not Supported	Not Supported	Not Supported	Not Supported
ASR 1000	Not Supported	Not Supported	Supported	Not Supported	Supported (SPA-DSP EOS)	Not Supported	Not Supported
ISR 1000	Not Supported	Not Supported	Supported	Not Supported	Not Supported	Not Supported	Not Supported
ISR 900	Not Supported	Not Supported	Not Supported	Not Supported	Not Supported	Not Supported	Not Supported

Sizing Baseline
ISR 4000 Voice Routers

Platform	NIM	SW	DW	MB
ISR 4321	2	0	0	1
ISR 4331	2	1	0	1
ISR 4351	3	2	1*	1
ISR 4431	3	0	0	1
ISR 4451	3	2	1*	1
ISR 4461	3	3	2*	0

NIM: Network Interface Module
SW: Single-wide Module
DW: Double-Wide Module
MB: Motherboard PVDM4 slot

*Assumes no singlewide **SM-X modules** installed

TIP The number of module slots define the T1/E1 port count, DSP capacity for packetization & IP service capability

TIP Type of module selected define the service type : T1/E1/FXO/FXS/Transcoding/Conferencing

TIP For Call control Features (CME/SRST/CUBE) , core platform attributes (DRAM/Flash/CPU) defines the max sessions

DSP Channel Considerations
ISR 4000 Voice Routers

- Use DSP Calculator to estimate the DSP resources needed for VoIP calls on ISR 4000.

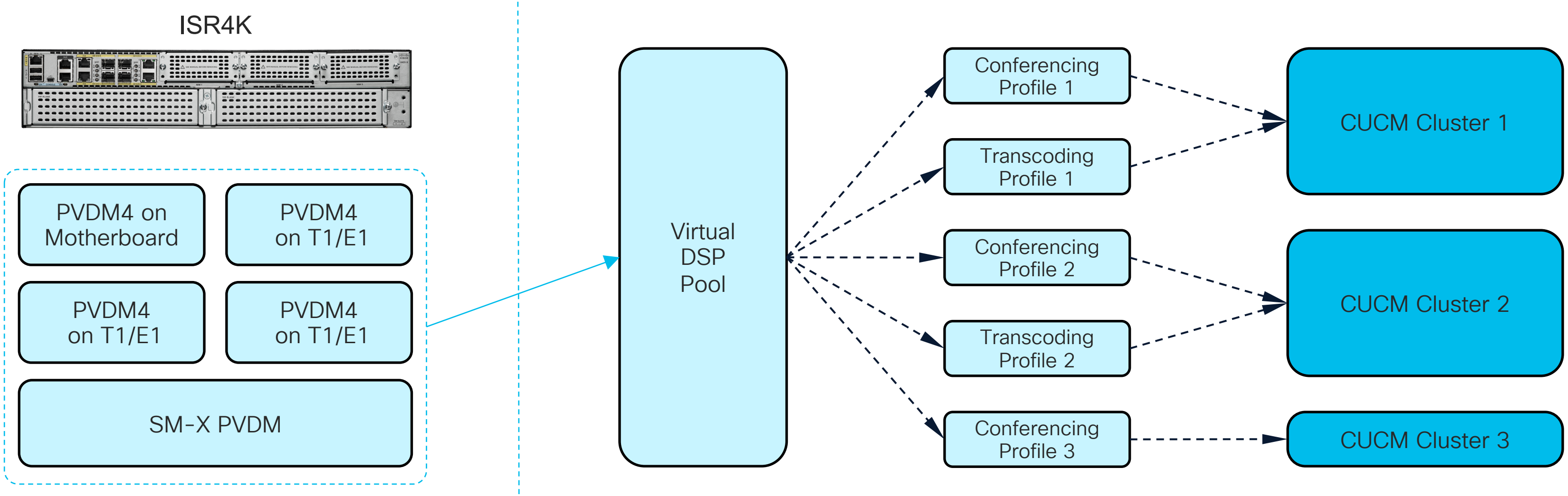
- No DSP motherboard slot on ISR 4461.

DSP Module	TDM Voice Services	Media Services Transcoding, Conferencing etc.
DSPs on Analog NIM (FXO/FXS)	Fixed on NIM Module	No
PVDM4 DSPs on T1/E1 NIM	Yes	Yes*
PVDM4 DSPs on SM/Motherboard Slot	No	Yes

*Enable **dsp services dspfarm** on NIM modules for excess DSP channels to be reused toward IP services

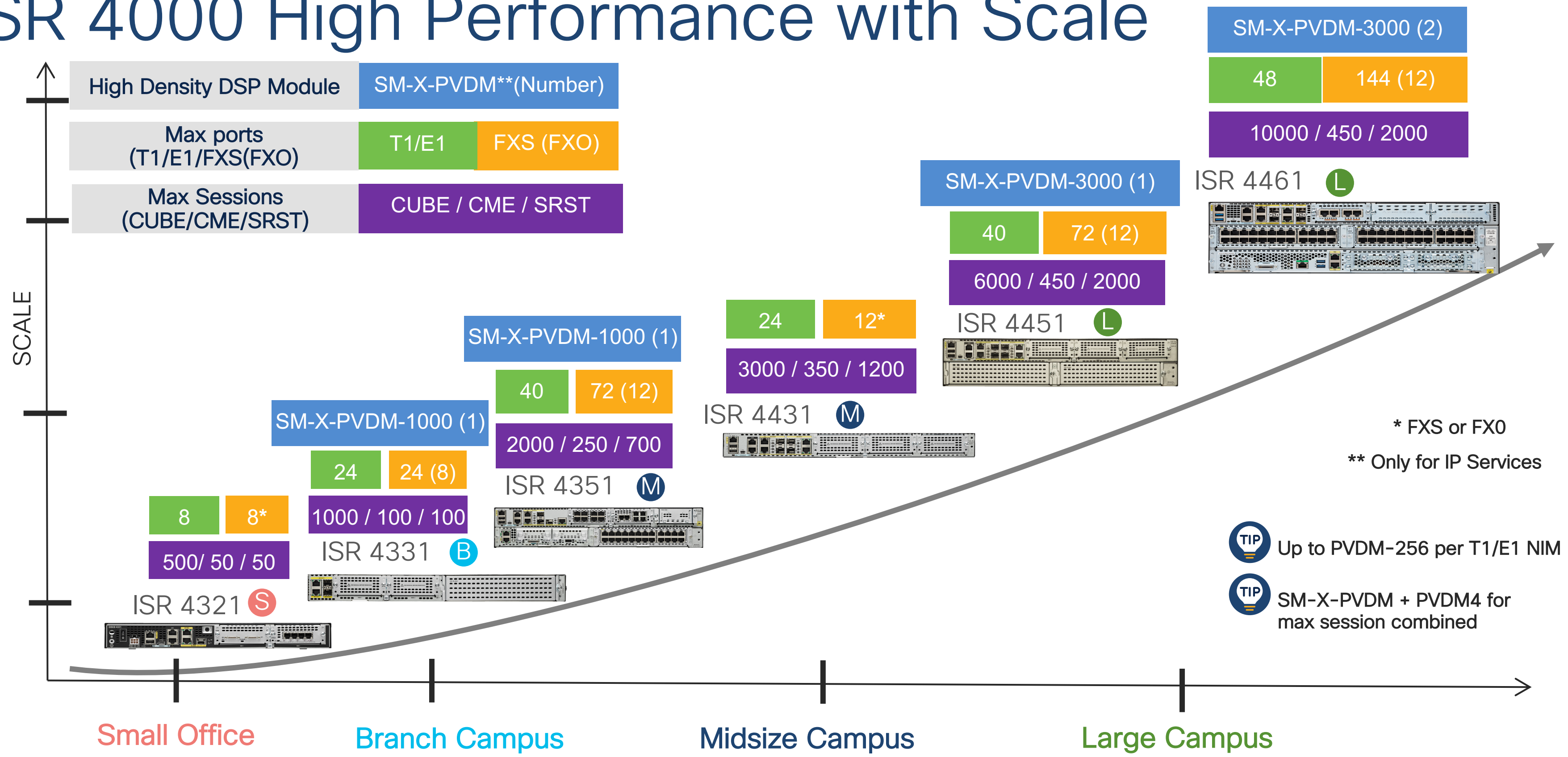
⚠ PVDM2 and PVDM3 DSP modules are *NOT* supported on the ISR 4000 series platforms

DSP Resource Pool Sharing



- Sharing enabled with **dsp services dspfarm** under NIM module with PVDM4 inserted
- Unallocated DSPs are pooled into one virtual group
- Can be shared across multiple CUCM clusters for conferencing and transcoding

ISR 4000 High Performance with Scale



Use Case Category

Scenario

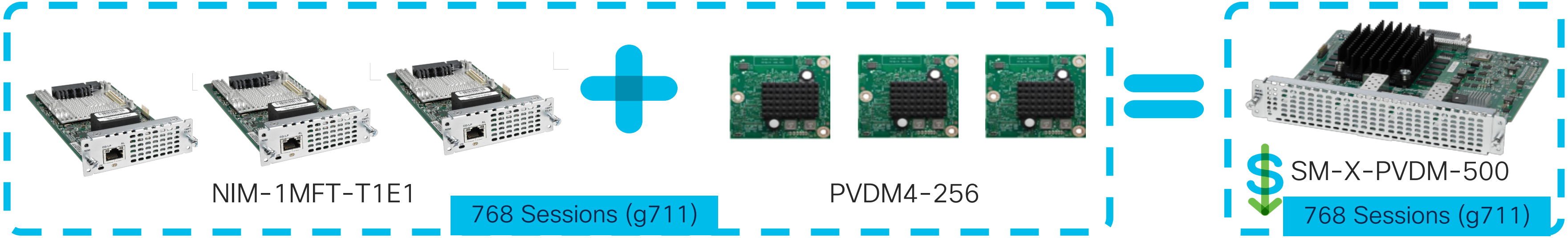
- Sizing
- Scale
- Cost

UC Services

- DSP Capacity
- PSTN Calls
- Analog Phones
- Call Control Services
- Emergency Calling

High DSP Channel Density

Scenario:
ISR 4000 for IP-IP CUBE Calls with 5000 DSP channel requirement to support dspfarm IP services (Call Progress Analysis , MTP etc.)



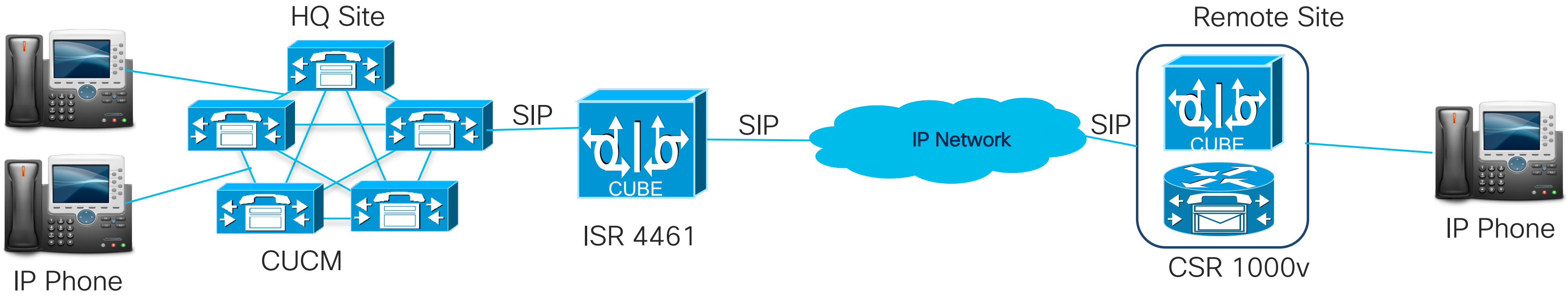
Facts:
- High channel density requirements can not be met with PVDM4 DSP beyond a point
- SM-X-PVDMs can scale up to 3 times the session count that can be achieved with PVDM4
- Optimized Cost and less space consumption to achieve equivalent channel density

Note: Performance depends on codec complexity & Support

Solution:
ISR 4461 with two SM-X-PVDM-3000 can meet this requirement

Virtual UC Router

Scenario:
Enable IP phone registration (200 Phones) and outbound PSTN calls (20 calls/second) at the remote site without increasing datacenter footprints



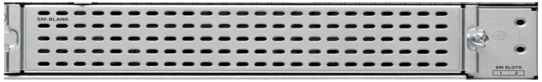
Facts:
- CSR 1000v with 4 vCPU & 8GB RAM can run vCUBE with max 6000 IP-IP sessions at 30 CPS
- Support for vCME with max 450 IP Phone registrations and 120 Active calls for same specifications

Note: Max session & CPS depends on vCPU & RAM

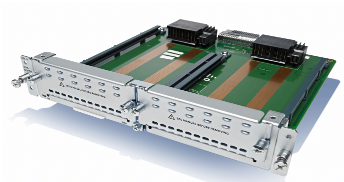
Solution:
CSR 1000v with 2 vCPU & 4GB RAM can meet this requirement

Augment T1/E1 PSTN Connections

Scenario:
ISR 4000 for midsize deployment with support for up to 28 T1 PRI ports



SM-X Slot



SM-X-NIM-ADPTR

NIM Slot

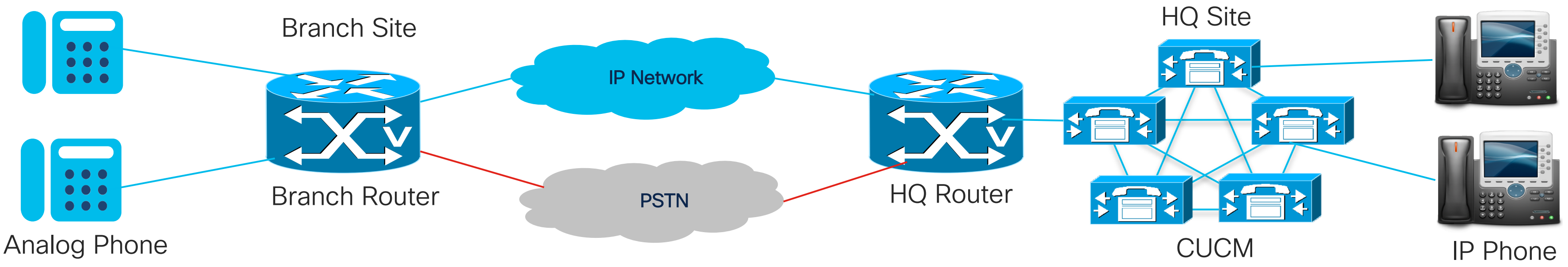
Facts:
- SM-X-NIM-ADPTR to convert SM-X slot to NIM slot
- Expand T1/E1 port capacity beyond what standard NIM slots can accommodate
- ISR 4000 platform can scale up to 48 T1/E1 ports with ISR 4461 & three SM-X-NIM-ADPTR

Note: T1/E1 NIMs and PVDM4 DSPs are to be purchased separately.

Solution:
ISR 4351 with three **NIM-8MFT-T1E1** & one **NIM-4MFT-T1E1** inserted in SM-X slot can meet this requirement

Analog Devices at Branch

Scenario:
WAN edge to support 100 analog phones at the branch site as a single box solution with local PSTN calling & at least 10 emergency lines



Facts:
- ISR 4000 platform supports analog voice NIM and high-density analog Voice modules
- ISR 4461 can scale up to 144 FXS ports with two double wide **SM-X-72FXS** or 36 FXO/24 FXS ports with three **SM-X-8FXS/12FXO**
- Max 8 FXO failover bypass ports with **SM-X-8FXS/12FXO** for emergency calls even during power outage

Note: ISR 4000 is SD-WAN ready and supports UC on SD-WAN (IOS XE 17.2.1r) covering FXS-FXO , SIP-FXS , SIP-FXO ,SIP SRST

Solution:
ISR 4461 with one **SM-X-72FXS** , one **SM-X-24FXS/4FXO** & one **SM-X-8FXS/12FXO** can meet this requirement

Summary

Recap

UC Portfolio on ISR 4000 & VG series

- PSTN termination options
- SIP Trunking to IP PBX
- Call Control features
- Analog telephony solution
- Emergency Calling
- Survivable telephony
- Audio conferencing
- Media termination
- Bandwidth saving

Added capabilities with Next Gen Voice Gateways

- Auto directory number assignment to analog phones
- Self service capability
- SIP based control of analog phones
- Media recording
- Emergency calling during power outage
- Module replacement in gateway powered on stage

ByoPSTN for Cloud Calling

- Decouple PSTN connections from cloud calling
- Support for Cisco Webex local gateway
- Support for UCM Cloud local breakout gateway
- Signaling through cloud calling solution
- Local media termination at the edge PSTN gateway

Voice Platform sizing & Positioning

- ISR 4000 support all UC capabilities such as SIP , TDM , analog telephony, voice module
- Derive IP/TDM voice sizing based on platform specifications such as NIM slots , SM Slots
- SM-X-PVDM for high density voice deployments
- SM-X-NIM-ADPTR to increase T1/E1 ports

Key Takeaways

- ISR4000 is a feature rich platform for unified communications
- Next Gen Voice Gateways comes with additional benefits
- Local PSTN termination option for cloud calling powered by enterprise voice routers.
- Know the platform to make the right choice for your network

Additional Sessions

BRKENT-2555 — Optimizing and Deploying Unified Communications on Cisco SD-WAN

BRKCOL-2112 — Utilizing Cloud Collaboration Services with CUBE

BRKCOL-2762 — Demystifying Cisco UCM Cloud, A Brand-new Offer in Cisco's Cloud Calling Portfolio

For Your Reference

- http://cs.co/ISR4000SeriesDatasheet
- http://cs.co/VG450datasheet
- http://cs.co/VG400datasheet
- http://cs.co/ISR4000InterfacesAndModules
- http://cs.co/HighDensityAnalogVoiceModules
- http://cs.co/PVDM4DSPModules
- http://cs.co/ISR4000NIMModules
- http://cs.co/AnalogSIPLineRegistration
- http://cs.co/NativeMediaRecording
- http://cs.co/WebexLocalGatewayOrderingGuide
- http://cs.co/WebexLocalGatewayConfiguration
- http://cs.co/UCMCloudCustomerOnboarding

Thank you

Possibilities

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