Designing a SIP Trunk Network

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CIN Technology Workshop

Webinar Logistics

- Submit “Real time” Q&A in WebEx Q&A panel
- Presentation will be posted on My Cisco Community

https://www.myciscocommunity.com/docs
Session Overview

- CUBE Product Overview
- SIP Trunk Deployment Models
- Integrating Voice Gateway with Voice Policy
- Q&A
Cisco Unified Border Element Product Overview
Cisco Unified Border Element (CUBE)
Enabling Session Border Control (SBC) Features on Cisco Routers

SESSION CONTROL
- Call Admissions Control
- Ensuring QoS
- Statistics and Billing
- Redundancy/Scalability

SECURITY
- Encryption
- Authentication
- Registration
- SIP Protection
- Firewall Placement
- Toll Fraud

INTERWORKING
- SIP - SIP
- H.323 - SIP
- SIP Normalization
- DTMF Interworking
- Transcoding
- Codec Filtering

DEMARCATION
- Fault Isolation
- Topology Hiding
- Network Borders
- L5/L7 Protocol Demarcation

Enterprise 1
IP
CUBE
SIP
PSTN

Enterprise 2
IP
CUBE
SIP

Rich Media (real time Voice, Video, Screenshare)
Rich Media
Cisco Unified Border Element—
Integration of Routing and Session Border Controller Functions

Cisco Unified Border Element
- Address Hiding
- H.323 and SIP interworking
- DTMF interworking
- SIP security
- Transcoding

Note: An SBC appliance would have only these features

Note: Some features/components may require additional licensing
SBC Integration on Cisco Routers
Use L3 Network Visibility for Real Time Session Enhancement

**Savings**
- Reduced equipment footprint
- Common sparing
- Common technical knowledge base

**Flexibility**
- Simplified migration to SIP trunking
- Repurpose router as collaboration needs change

**User-Experience**
- Network Aware Call Admission Control (CAC)
- Integrated Media processing
- Enhanced Video Transport
- Enhance mobility integration
CUBE Integration with Cisco Collaboration Solutions

CUCM Integration with CUBE -

- **MOBILITY FEATURES:**
  - Single Number Reach requires separate MTP to terminate media, which can co-reside with CUBE on ISR-G2.
  - Dial Via Office requires translation of DTMF digits from inband to out of band.

- **NETWORK MANAGEMENT:**
  - CUOM provides capacity analysis for both CUCM lines and Cisco Voice GW (CUBE & TDM) trunks.
  - Serviceability with common call session numbers between CUBE and CUCM.
  - CUCM and Cisco Voice GW (CUBE & TDM) mutual discovery of dial peers via Service Advertisement Framework (SAF)

- **CONFERENCING**
  - DSP resources used for SBC transcoding or CUCM conferencing

Call Center Integration with CUBE

- vXML Server on CVP integrates with vXML client on ISR with CUBE
- MRCP interaction with ASR applications
- Media Forking on CUBE integrates with MediaSense

WEBEX Integration with CUBE

- WEBEX Connect is designed to work with CUBE as the SBC integration for WEBEX CCA
- Integrated SIP session control for video, voice and desk top sharing with specific Cisco end points.
CUBE Scalability
Scalable Voice Trunk Capacity for Small to Large Businesses

- **ASR 1004/6 RP2**
  - Active Voice Call (Session) Capacity: <5

- **ASR 1002**
  - 8 - 12 Calls Per Second
  - Scalable Voice Trunk Capacity for Small to Large Businesses

- **3900 ISG G2**
  - 50 - 150 Calls Per Second

- **2900 ISG G2**
  - 17 Calls Per Second

- **2801 ISG**
  - <5 Calls Per Second

- **ISG 3945**
  - Transcoding of ~800 calls
  - 600-800 Active Voice Call (Session) Capacity

- **800/1861 ISG**
  - <5 Calls Per Second

- **ASR 1001**
  - Highest density SBC: 10,000 sessions in 1RU

- **ASR 1006**
  - Highest SBC transcoding capacity: 9,000 G729 to G711 calls

- **ISR 3945**
  - Transcoding of ~800 calls
  - 600-800 Active Voice Call (Session) Capacity

- **ISR 3945**
  - 900-1000 Active Voice Call (Session) Capacity

- **3900E ISG G2**
  - 12 - 16K+ Active Voice Call (Session) Capacity

- **ASR 1001**
  - 12 - 16K+ Active Voice Call (Session) Capacity

- **ASR 1004/6 RP2**
  - 12-16K+ Active Voice Call (Session) Capacity

- **ASR 1001**
  - 20-35 Active Voice Call (Session) Capacity

- **ASR 1002**
  - 20-35 Active Voice Call (Session) Capacity
<table>
<thead>
<tr>
<th>Platform</th>
<th>CUBE Sessions</th>
</tr>
</thead>
<tbody>
<tr>
<td>C880/C890 SKUs</td>
<td>5-25</td>
</tr>
<tr>
<td>1861</td>
<td>5-15</td>
</tr>
<tr>
<td>2801</td>
<td>55</td>
</tr>
<tr>
<td>2811</td>
<td>110</td>
</tr>
<tr>
<td>2821</td>
<td>200</td>
</tr>
<tr>
<td>2851</td>
<td>225</td>
</tr>
<tr>
<td>3825</td>
<td>400</td>
</tr>
<tr>
<td>3845</td>
<td>500</td>
</tr>
<tr>
<td>AS5000XM</td>
<td>600</td>
</tr>
<tr>
<td>2901</td>
<td>100</td>
</tr>
<tr>
<td>2911</td>
<td>200</td>
</tr>
<tr>
<td>2921</td>
<td>400</td>
</tr>
<tr>
<td>2951</td>
<td>600</td>
</tr>
<tr>
<td>3925</td>
<td>800</td>
</tr>
<tr>
<td>3945</td>
<td>950</td>
</tr>
<tr>
<td>3925E</td>
<td>2100</td>
</tr>
<tr>
<td>3945E</td>
<td>2500</td>
</tr>
<tr>
<td>ASR1002/1004/1006 RP1</td>
<td>1750</td>
</tr>
<tr>
<td>ASR1001</td>
<td>10000</td>
</tr>
<tr>
<td>ASR1004/1006 RP2</td>
<td>16000</td>
</tr>
</tbody>
</table>

ASR1001 introduced in RLS 3.2 in Nov 2010

End of Life Platforms
Last IOS Release: 15.1.4M

AS5000XM

Introduced in March 2011
CUBE Versatility
Enterprise VOIP Interworking for Internal or External Use Cases

Network-based Media Recording

SIP
RTP

SP IP Network
PSTN

SIP

Communications

SIP

SIP

SIP

MediaSense

SP VOIP Services

SIP

SBC

SBC

SP IP Network

RTP

SIP

Traditional PSTN

SIP

ISR G2

H.323

ASR Server

vXML Client

vXML Server

Business to Business Telepresence

Integration for Call Center

Migration from TDM to SIP Trunks
CUBE’s Single Platform Reduces Costs & Complexity for SIP Migrations

Cisco UCM

SIP

H.323

TDM Gateway

SIP

TDM

Traditional PSTN

SP VOIP Services
CUBE High Availability
Alternative Deployment Strategies for Redundancy and High Availability

- **Inbox redundancy**
  - ASR 1001/2/4/6
  - ASR 2.5, 2.6, 3.1
    - Media preservation
  - ASR 3.2 (Nov 2010)
    - Stateful failover

- **Box-to-Box redundancy**
  - ISR G2 and ASR 1001/2/4
  - ISR G2 CUBE 8.5 (15.1.2T)
    - Media preservation
  - ASR 3.2 (Nov 2010)
    - Stateful failover
  - *Local redundancy only*

- **Clustering with load balancing**
  - All platforms
  - Load balancing by
    - SP call agent
    - Internal SIP proxy/load balancer
  - *Local or distributed redundancy*
Cisco SME & CUBE: Proven Interoperability
Use what you have, eliminate upgrades, deploy globally

- Industry-leading SIP interoperability (sophisticated header normalization)
- SME Industry-leading interop on any protocol (SIP, H.323, TDM, QSIG, PRI, CAS)
- Standards based
- Tested with 3rd party PBXs & IP PBXs
- Validated with SPs world-wide

Cisco Interop Portal: www.cisco.com/go/interoperability
SIP Trunking Deployment Models
Collaboration Technologies
Voice is Converging with Other Technologies

Expanding Collaboration to include Broader, Richer Interactions
Future Benefits of Collaboration will Require End to End IP

- TDM
- Hybrid
- IP

TCO Savings → Productivity

Basic Telephony
Toll bypass

IP and TDM PBX w/ mix of TDM and IP trunks
Unified Communications
Basic mobile integration

IP infrastructure w/ TDM survivability
Enterprise Communications and Collaboration for message, voice, screen share & HD video
B2B, Mobile, and Web 2.0 Collaboration
Policy-enabled control
Flexible SIP Trunk Deployments
Accommodating Centralized or Distributed SIP Trunk Deployments
WEBEX Optimization thru SIP Trunking
Webex CCA Solution using CUBE Enterprise

Requirements
- Replacement for TDM audio connection to WEBEX with VOIP using SIP signaling.
- High capacity SIP media connectivity for WEBEX cloud, including telepresence integration.

How
- CUBE Reduces SIP protocol “chatter” between CUCM and WEBEX cloud thru normalization.
- CUBE allows SIP sessions from ALL enterprise sites to WEBEX to avoid “hairpin” media flows.
- CUBE support on ASR provides high performance for signaling and media transport of WEBEX.

Benefit
- Best possible WEB conference experience for Enterprise users, with most efficient network usage.

Future Capabilities
- Additional Cloud services (e.g. QUAD) under same architecture and identity is possible.
- Integration with WEBEX One Touch for improved telepresence session set up (i.e. one touch)
CUBE Provides Phased Approach to SIP Trunking

Centralized Architectures Change BOTH Technology and Topology

- This distributed TDM model fits well with SP services like BEST from Verizon
- Centralized SIP Trunks
  - TDM Circuits to SP
  - SIP Trunks to SP
  - Internal SIP Trunks
# SIP Trunk Deployment Models
## Selection Criteria Summary

<table>
<thead>
<tr>
<th>SELECTION CRITERIA</th>
<th>CENTRALIZED</th>
<th>DISTRIBUTED</th>
<th>HYBRID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Limitations on Headend Bandwidth</td>
<td>Need strong QoS strategies on Ent WAN</td>
<td>Not affected by Headend BW availability</td>
<td>Adaptable to BW availability</td>
</tr>
<tr>
<td>Existing distributed PBX Architecture</td>
<td>Not Recommended</td>
<td>Preferred</td>
<td>Allowable</td>
</tr>
<tr>
<td>Variability of branch office capacity requirements</td>
<td>Low branch office capacity (&lt;20% of trunks)</td>
<td>High branch office capacity (&gt;50% of trunks)</td>
<td>High branch office capacity but varies from site to site.</td>
</tr>
<tr>
<td>Video conferencing /Video telephony requirements thru Service Provider</td>
<td>Requires strong QoS strategies on Ent.WAN</td>
<td>Requires adequate BW at each site.</td>
<td>Offers flexibility in phased deployment</td>
</tr>
<tr>
<td>Need to maintain branch IT functions</td>
<td>Allowable</td>
<td>Recommended</td>
<td>Allowable</td>
</tr>
<tr>
<td>Consistent latency across network</td>
<td>Inconsistent latency can occur</td>
<td>Recommended</td>
<td>Recommended</td>
</tr>
<tr>
<td>Gateway protocols for TDM access</td>
<td>If MGCP is used on TDM GW, then may be easiest transition.</td>
<td>H323 or SIP used on TDM GW = easiest transition</td>
<td>Preferred when both MGCP &amp; H323/SIP are used on TDM GWs</td>
</tr>
<tr>
<td>Centralized management Capabilities</td>
<td>Device management may be adequate</td>
<td>Requires strong centralized management</td>
<td>Same as distributed</td>
</tr>
<tr>
<td>Enterprise WAN Capabilities</td>
<td>Needs strong QoS &amp; CAC</td>
<td>Not a consideration</td>
<td>Needs QoS &amp; CAC</td>
</tr>
<tr>
<td>Survivability &amp; alternative path Strategy</td>
<td>Requires TDM backup in distributed branch offices</td>
<td>Multiple SIP connection points provide survivability</td>
<td>Multiple SIP connection points provide survivability</td>
</tr>
</tbody>
</table>
In-Depth Explanation of SIP Deployment Models

Educate your customer on SIP Deployment Models

SIP Trunking Deployment Models:
Choose the One That Is Right for Your Company

Executive Summary

In the past few years, innovations in collaboration services have delivered significant improvements in employee productivity. By every indicator, new collaboration services will continue to be introduced, providing additional benefits to the enterprise. Although no one can predict exactly what these new services will be, we expect them to incorporate existing fundamental elements of collaboration like voice, video, mobility, instant messaging, and virtual desktop technologies.

However, the problem facing most businesses is that their underlying network resources were not designed to support these new services. For example, most networks are not designed to support a significant volume of real-time video communication. However, real-time video is increasingly becoming a critical element in these new collaboration services. This problem will worsen as these collaboration use cases begin to move beyond the enterprise to suppliers and customers.

New White Paper will be posted by the end of January at the following URL:
www.cisco.com/go/cube
Integrating Voice Gateway with Voice Policy
(for real time visibility and control)
Voice Network Challenges For the CIO

No holistic view of network & no real-time control

• Fragmented call information and access
• “Un-explained” voice billing issues
• Unanswered questions about network behaviour
• Heterogeneous networks
• Risk & compliance discussions
• Cyber security issues - social media, Web 2.0
• Reactive versus proactive
• Cloud reliability
• Making the CIO office more business rele...
Using The Network Edge to Drive Innovative Applications to the Cloud and Premise

TDM Gateways & SBCs:

- Remain in the call path for the life of the call
- Demarcation point – can mitigate security threats before they hit the network
Examples of Recent Voice Threats

Chardon: Bomb threat forces Walmart evacuation
8:33 PM, Nov 26, 2011 | 0 comments

Long distance toll fraud on the rise: complaints commissioner
The Commissioner for Complaints for Telecommunications Services warned Canadian businesses and consumers yesterday that long distance fraudsters are stepping up their game. Here are some tips on how to prevent them from scoring.
11/7/2011 9:00:00 AM By: Nestor E. Arellano

Wayne County man charged with harassing Cleveland lawmaker
Published: Tuesday, November 22, 2011, 6:40 PM Updated: Tuesday, November 22, 2011, 6:40 PM

Identity Theft Call Centre Uncovered – UK Contact Centres Warned
2-News, Calls And Lines, Security
Investigators have uncovered a publicly listed rogue call-centre which offers services catered exclusively for cyber-criminals, credit and identity theft fraudsters.
Cisco UC Gateway Services API

Combines applications to:

• Remove latency issues (less hops)
• Allow 3rd parties to integrate applications onto Cisco ISR using UCS express
• Simplify management and architecture
  • Single platform using Cisco ISR
  • Enterprise-wide solutions (TDM+SIP)
  • Combine with data solutions
Integrated Voice Gateway / Voice Policy Solution
Real time Monitoring and Control of the Enterprise Voice Network

Application Layer Voice Policy:
- Centralized policy creation/distribution
- Protection from external harassing calls
- Service Abuse Control by Internal Users
- Enterprise-wide UC reporting & analytics
- Compliance & Data Leakage prevention
- Call recording archive

Voice Service Infrastructure:
- Protocol Normalization (SIP)
- Transcoding & Transrating
- Protocol fixes and interoperability
- Packet level encryption security
- NAT and topology protection
- IP Firewall & SP registration
- QoS & CAC
Voice Policy Solution Topology
Embedded API Enables Integrated Voice Policy

Voice Policy Appliance

Service Provider

TDM

SIP
MGCP
H323
RTP

Corporate Network

Call Control CUCM

LAN

Voice Policy Server

CUBE +
TDM GW

IP

API for Tone
Detection and
Media Forking

API for TDM & VOIP Signaling

Corporate Network

Presentation_ID

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# Voice Policy Use Cases

**Broad Range of Use Cases Provides Many Opportunities for ROI**

## Enterprise Wide Capacity Management
- Centralized reporting for the enterprise
- Baseline and inventory voice network infrastructure
- Recover capacity lost to unauthorized traffic
- Right-size trunk infrastructure
- Eliminate unused PBX bypass lines
- Identify orphaned or unused extensions
- Consolidate/reduce unused fax resources
- Absence of call activity on trunking resources
- Excessive unanswered/busy calls on trunking resources
- Optimize staffing based on call activity reports

## Enterprise Wide Security Management

**Centralized Security Policy Definition**
- TDOS (Telephony Denial of Service) Mitigation
- Reduce Toll Fraud Losses from external dial through
- Prevent network penetration via blocking modems
- Alert and control business disrupting bomb threats
- Identify and Manage harassing calls.
- Alert/log maintenance port access, and block unauthorized connections
- Service abuse/misuse/anomalies
- Prevent identity theft on voice lines

## Enterprise Wide Control of Service Abuse

**Centralized abuse prevention policy definition**
- Unauthorized Modem usage
- Voice Data Leakage Protection (DLP)
- Reduce toll fraud losses by blocking unauthorized calls
- 911 notification and response
- Managed calls to and from restricted

## Customer Service Monitoring

- Record inbound customer calls to audit and aid staff training through the entire call session.

## SLA Monitoring

- Log of service outages, disruptions, and errors
- Voice Usage uptime and performance reports
# Enterprise Wide Voice Policy Definition

## Firewall Policy - Dallas

<table>
<thead>
<tr>
<th>No.</th>
<th>Comments</th>
<th>Call Direction</th>
<th>Source</th>
<th>Destination</th>
<th>Call Type</th>
<th>Attributes</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>911 calls</td>
<td>Outbound</td>
<td>Any</td>
<td>Emergency Group</td>
<td>Any</td>
<td>None</td>
<td>Allow</td>
</tr>
<tr>
<td>2</td>
<td>No VoIP telemarketers</td>
<td>Inbound</td>
<td>Porn.com</td>
<td>Any</td>
<td>Any</td>
<td>None</td>
<td>Terminate</td>
</tr>
<tr>
<td>3</td>
<td>No CID restricted to modem pool</td>
<td>Inbound</td>
<td>Caller ID Restricted</td>
<td>Any</td>
<td>Any</td>
<td>None</td>
<td>Terminate</td>
</tr>
<tr>
<td>4</td>
<td>No ISP sessions</td>
<td>Outbound</td>
<td>Any</td>
<td>ISP Modems</td>
<td>Modem</td>
<td>None</td>
<td>Terminate</td>
</tr>
<tr>
<td>5</td>
<td>No 1-900 toll calls</td>
<td>Outbound</td>
<td>Any</td>
<td>1-900 Numbers</td>
<td>Any</td>
<td>None</td>
<td>Terminate</td>
</tr>
<tr>
<td>6</td>
<td>No voice or modem on Fax lines</td>
<td>Outbound</td>
<td>Any</td>
<td>Fax Numbers</td>
<td>Fax</td>
<td>None</td>
<td>Terminate</td>
</tr>
<tr>
<td>7</td>
<td>VoIP GoS issues</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Excessive Media Rate</td>
<td>Allow</td>
</tr>
<tr>
<td>8</td>
<td>VoIP CODEC issue</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Unknown Codec</td>
<td>Allow</td>
</tr>
<tr>
<td>9</td>
<td>VoIP Signaling issue</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Any</td>
<td>Signaling Anomaly</td>
<td>Allow</td>
</tr>
</tbody>
</table>

## Other Policies
- Firewall Policies
  - Dallas
  - Houston
  - Washington DC
- IPS Policies
  - Dallas
  - Houston
  - Washington DC
- Recording Policies
  - Dallas
- Span Groups
  - Dallas
  - Houston
  - Washington DC
  - DC-to-Dallas SIP
- Telco Configuration
  - Dallas DMS-100
  - Houston Avaya G3
  - Houston PRI-NFAS
  - Houston SS7
  - Washington DC NEC
  - DC-to-Dallas SIP
- Platform Configuration
  - Dallas 1012
  - DC-to-Dallas SIP
  - Dallas Fax Lines
  - Houston 3200
  - Houston 3270 SS7
Real-time Alerting With Custom Thresholds

**Infrastructure Health Alerts**
- ETM Card in status
- ETM Appliance Health
- Management Server status
- Data migration failures
- Power supply fail
- Thermal fail
- etc

**Policy Alerts**
- 911 call
- Modem calls & duration
- Inbound busy/unanswered
- Excessive Long Distance
- Harassing caller volume
- Specific Country call block
- Firewall Terminate rule firing
- etc

**Telecom & Data Center Alerts**
- Trunk D-Channel up/down
- Trunk frame slip
- Trunk bit-error/CRC fail
- SIP interface up/down
- SIP proxy offline
- QoS violation/Excessive Rate
- Unknown CODEC
- etc.
Enterprise Wide Voice Policy Reporting

REAL TIME CALL MONITORING

- Cost Allocation / Call Acct.
- Resource Utilization
- UC Diagnostics
- UC Network Audits
- UC Operations
- UC Security

HISTORICAL REPORTS
Enable Security, Visibility & Control with Voice Policy

- **Harassing / Threatening Callers**
  - Lowers productivity & safety

- **Voice Service Abuse & Theft**
  - Ensure employee use of voice network complies with business objectives

- **Capacity Monitoring**
  - Enables better network planning and staffing requirements

- **Toll Fraud**
  - Corporations lack real-time defense

- **Contact Center Fraud/ID Theft Schemes**
  - Legal risk and financial losses for corporations and customers

- **Unauthorized Fax or Modem Usage**
  - Most commonly found issue
BT Assure Cloud Security Services
See, Connect, Prevent

- **BT enhances service to include voice:**
  See unknown patterns: visualize converged threats in a single command center
  Connect - correlates threats for voice & data enterprise-wide
  Prevent: dynamically control the network to mitigate threats

- **Enterprise-wide on Cisco ISR G2**
  SIP+TDM trunking
  Use with any service provider - including BT