Demystifying SIP

STC 2012
Tuesday, June 26 10:15AM
Presenters

• Molly Sawyer - TW Telecom

• Phil Birmingham – Annese

• John Green – System Administration
Some Acronyms to Start Your Day

• **VoIP** – Voice over Internet protocol
  • Sometimes Video sneaks in as well

• **SIP** – Session Initiation Protocol

• **IETF** – Internet Engineering Task Force
  • **RFC** – Request for Comments – a “manual” on how internet systems should work
What is SIP?

• A signaling protocol used to create, manage and terminate sessions in an IP based network

• Can be used for multiple purposes from something as simple as a 2 way phone call or as complex as a collaborative multi-media conference with voice, video, click to call and instant messaging
A (very) Brief History of SIP

- IETF began experimenting with voice transmissions over IP in the late 80s
- 1992 – MMUSIC – Multiparty Multimedia Session Control, the precursor to SIP is adopted by IETF
- 1996 – Henning Schulzrinne and Mark Handley release first SIP draft to IETF
- 1996 thru 1999 – SIP goes through 12 major updates and revisions
- 1999 – SIP accepted by the IETF as permanent element of IP multimedia subsystem architecture with RFC 2543
- 2002 – SIP RFC 3261 released with extensions RFC 3262 and RFC 3265 shortly afterward. These have become the standards in most widespread use.
- Over 290 RFCs are in existence that extend or modify SIP in some way
What does SIP do?

- Performs four main functions
  - Establishes user location
    - Translates names to network addresses
  - Manage Features
    - Enables only features all participants can use
  - Session Management
    - Adding, dropping, transferring participants
  - Feature Changes
    - Enables/Disables features as participants feature sets change
What are these features you mention?

Some common session features for a VoIP environment

- Forward/busy
- Transfer
- Call ID
- Multiparty Call
- Call Return (*69)

- Call Waiting
- Call Queuing
- Call Park
- Do-Not Disturb
- Hold
What doesn’t SIP do?

• Carry session data
  • Rather, it works with other protocols such as:
    • SDP – Session Description Protocol
    • RTP – Real-time Transport Protocol
    • HTTP – Hyper-Text Transfer Protocol
  • The protocols it works with detail items such as codecs to use and media type
How does SIP work?

- Magic
How does SIP work?

• Similar to HTTP, SMTP
• Client (Agents) -> Server
  • Uses a series of simple commands and status codes via request and responses
SIP Commands and Codes

- INVITE – invites a session
- ACK – acknowledge an invite
- BYE – terminate a session
- CANCEL – terminate a request
- OPTIONS – asks capabilities
- REGISTER – sends location (IP) to local network registration server
- INFO – used to pass other protocols mid session

- Provisional (1xx) – Request received, processing
- Success (2xx) – Request received, accepted
- Redirection (3xx) – Further action needed, usually by sender
- Client Error (4xx) – Request has bad syntax or server cannot fulfill
- Server Error (5xx) – Valid request not fulfilled by server
- Global Error (6xx) – Request cannot be fulfilled at any server
Example of a SIP Request

INVITE sip:user2@server2.com SIP/2.0 <- The request line
Via: SIP/2.0/UDP pc33.server1.com;branch=z9hG4bK776asdhds Max-Forwards: 70
<-Local address of requester, sip version, and max hops before disconnect
To: user2 sip:user2@server2.com <-Called
From: user1 <sip:user1@server1.com>;tag=1928301774 <- Caller, with SIP generated
tag identifier
Call-ID: a84b4c76e66710@pc33.server1.com <-Unique call identifier for system
CSeq: 314159 INVITE <- Used to detect a non delivery, the number will increment
every request/response cycle
Contact: sip:user1@pc33.server1.com <- Direct route to caller
Content-Type: application/sdp <- Information on protocol in body
Content-Length: 142 <-body size

---- User1 Message Body Not Shown ----
Example of a SIP Response

SIP/2.0 200 OK <- Status code response for request
Via: SIP/2.0/UDP site4.server2.com;branch=z9hG4bKnashds8;received=192.0.2.3
Via: SIP/2.0/UDP site3.server1.com;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.server1.com;branch=z9hG4bK776asdhds;received=192.0.2.1
<- Multiple Via lines as request had to bounce through multiple points to reach called, each leaves their info on way back
To: user2 <sip:user2@server2.com>;tag=a6c85cf <- SIP tag given to called for call
From: user1 <sip:user1@server1.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.server1.com
CSeq: 314159 INVITE
Contact: sip:user2@192.0.2.4 <- Direct IP of called so proxy servers removed for remaining request/responses of call
Content-Type: application/sdp
Content-Length: 131
A Full SIP Session Picture

Setup

Session

Breakdown
So...Why Use SIP?

- Open Source but Standards Based
  - Extensible to suite a wide variety of needs
- Interoperability with older revisions and other systems
- Reduction of Cost through unified communications solutions
Welcome to the Business of Connectology

Phil Birmingham
Virtualization Architect
CCNP, CCVP, VCP, CISSP

Annese & Associates, Inc.
Agenda

• Company Overview
• SIP and CUBE
• SIP Endpoints
• Q&A
Annese Profile:

- 100+ Employees
- $62M Revenue in 2011
- New York and New England Presence
- Women Owned Business Enterprise

Core Focus

- Professional and Managed Services
- Unified Communications
- Data Center
- Collaboration
- Safety and Security
- Infrastructure
- Energy Management

Recognition

- #292 on VAR500 of top technology integrators in North America
- Technology 200 list of fastest growing technology companies in the US
- #4191 on Inc. 500/5000 list of fastest growing small businesses in the country
- Providence Business News Fastest Growing Companies List
- Award Winning Customer Satisfaction Rating
- Cisco Customer Satisfaction Excellence

The Business of Connectology
What is CUBE?

- Cisco Unified Border Element
- It is a Session Border Controller in the “SIP world”
- It is a security boundary with many other features
- It exists as a SW feature on a Voice Gateway (Router)
- CUBE is licensed per simultaneous call
- CUBE licensing is required for a SIP connection if the voice gateway has a public address. It is also highly advisable….
- CUBE licensing is optional if the Voice Gateway has a private address …. but still recommended.
- It is currently a RTU license (Honor System)….. but that may change
Migration to SP SIP Trunking for PSTN Access

1. TDM Trunking – Yesterday

2. TDM and IP Trunking – Today

3. IP Trunking – Tomorrow
Protecting Your Network with an Enterprise Session Border Controller

**PRI Trunks for PSTN Access**

- Implicit Call Admission Control: Max 23/30 calls per T1/E1 PRI
- Implicit security: Cannot hack via PRI into your IP network; Your IP network is not visible via PRI to the outside; The IP characteristics of your endpoints are unknown to the PSTN endpoint
- Implicit demarcation: The PRI-to-IP point of conversion is used for SP hand-off, troubleshooting and statistics
- Implicit QoS and codec control: The Gateway assigns IP packet QoS markings and determines codecs used on your internal network
- DOS attacks against PRI are rare and difficult to perpetrate: A hacker cannot control individual PRI messages - can only launch calls to your DIDs
- PRI interoperability is standard and well-understood in the industry
- The GW is a toll-fraud control point

**SIP Trunks for PSTN Access**

- No CAC w/o an SBC: A SIP Trunk on Ethernet ingress can deliver hundreds of calls
- W/o an SBC, your IP network topology and endpoint characteristics are visible to the PSTN endpoint
- W/o an SBC, there is no demarcation point for SP hand-off and troubleshooting purposes
- W/o an SBC, QoS and codec selections are under control of the SP/PSTN network owner
- DOS attacks against SIP are easy and within reach of even very unsophisticated hackers
- SIP interoperability varies greatly and the industry is still maturing, and SBC can help greatly to normalize SIP
- Toll-fraud control point
SIP Trunk Validations on Cisco.com

- Cisco focuses on standards-compliance and participates in major IETF SIP Standards bodies
- Cisco performs interoperability validations with SIP trunk providers and PBXs
- Completed validations are posted to Cisco.com at www.cisco.com/go/interoperability
Enterprise Interconnect Internally and to Realms beyond the Enterprise

SIP Trunks for PSTN Access

Enterprise Networks in Transition

Business to Business Telepresence

H.323 Video Between Companies over Internet

The Business of Connectology
Cisco Unified Border Element (Enterprise Edition) Portfolio
Cisco Unified Border Element—More Than an SBC
An Integrated Network Infrastructure Service

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

WAN Interfaces

TDM Gateway
- Voice and Video TDM Interconnect
- PSTN Backup

Routing, FW, IPS, QoS

Unified CM Conferencing and Transcoding

RSVP Agent

VXML

GK

SRST

Note: Some features/components may require additional licensing

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Cisco Unified Border Element Key Features

Session Mgmt
- Real-time session Mgmt
- Call Admissions Control
- Ensuring QoS
- PSTN GW Fallback
- Statistics and Billing
- Redundancy/Scalability

Demarcation
- Fault isolation
- Topology Hiding
- Network Borders
- L5/L7 Protocol Demarc
- Statistics and Billing

Interworking
- H.323 and SIP
- SIP Normalization
- DTMF Interworking
- Transcoding
- Codec Filtering
- Fax/Modem Support

Security
- Encryption
- Authentication
- Registration
- SIP Protection
- FW Placement
- Toll fraud

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Call Admissions Control

- CUBE provides various different CAC mechanisms
  - Total calls, CPU, Memory, GK IP call capacity, max-connections, RSVP

Call Spike Detection

- Call spike call-number [steps number-of-steps size milliseconds]
  - call spike 10 steps 5 size 200

Max Calls per Destination

- Call #1
- Call #2
- Call #3

- Call #3 Rejected by CUBE
CUBE Security Protection Points

**DOS**
- B2BUA – L7 Inspection
- Call Volume/BW Limiting (CAC)
- Call Codec Limiting
- SIP Malformed Inspection
- SIP Listen Port Configuration
- RTP Malformed
- Topology Hiding
- Co-resident IOS: ACLs, FW, IPS

**Identity / Service Theft**
- SIP Digest Authentication
- SIP Hostname Validation
- SIP Trunk Register
- CDR
- Toll Fraud
- Co-resident IOS: ACLs, COR

**Privacy**
- SIP Header Manipulation
- Authentication and encryption (media) – SRTP
- Authentication and encryption (signaling) – TLS
- Co-resident IOS: All VPN features

**Voice Application Code**
- L7 Protocol-independent memory structures holding call state and attributes (CLID, Called #, Codec…)

**Dial-peer**
- SIP/H.323 Protocol Stack
- DTMF xlation Codec Filtering Xcoding Control

**RTP Library**
- TCP
- UDP
- TLS

**DSP API**
- DSP Hardware

**IOS Infrastructure (ACLs, FW, IPS, VPN)**

**HW LAN/WAN Interfaces**

**Ingress I/F**

**Egress I/F**

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CUBE Topology/Address Hiding

- CUBE provides complete topology hiding on signaling and media
  - Maintains security and operational independence of both networks
  - Provides implicit NAT service by substituting CUBE IP addresses on all traffic
- Allows for NAT and Firewall (FW) traversal
Centralized and Distributed SIP Trunk Models

Centralized

Distributed

Hybrid

Site-SP RTP
Site-to-Site RTP

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Cisco.com SIP Trunk and CUBE Resources

- **Cisco UBE** on Cisco.com

- **Cisco Communications Transformations Whitepapers**
  - Section on Whitepapers

- **Cisco Interoperability Portal**
  - [www.cisco.com/go/interoperability](http://www.cisco.com/go/interoperability)
    - Cisco Unified Border Element (CUBE)/SIP Trunking Solutions
    - Cisco UBE SP SIP Trunk Interoperability Reports
    - Cisco UBE PBX Interoperability Reports (Avaya/Nortel)

- **Cisco SRND Portal**
  - [www.cisco.com/go/srnd](http://www.cisco.com/go/srnd)
  - CUCM SIP Trunk Documentation
    - CUCM 8.x SRND
    - CUCM 7.x SRND
    - CUCM 6.x SRND
  - CVP 7.0 SIP Trunk Integration

- **Marketing Support:** [ask-cube@external.cisco.com](mailto:ask-cube@external.cisco.com)

- **Cisco Press: SIP Trunks**
  - SIP Trunking @ [www.ciscopress.com/title/1587059444](http://www.ciscopress.com/title/1587059444)

- **TechWise TV: SIP, Session Management and Beyond**
  - [http://www.youtube.com/watch?v=YFoLTsqEI0w](http://www.youtube.com/watch?v=YFoLTsqEI0w)
# Cisco Wireless Endpoints

<table>
<thead>
<tr>
<th>Model</th>
<th>Wireless 7921G</th>
<th>Wireless 7925G</th>
<th>7925G-EX</th>
<th>7926G</th>
</tr>
</thead>
<tbody>
<tr>
<td>License (RTU)</td>
<td>4 DLU</td>
<td>4 DLU</td>
<td>4 DLU</td>
<td>4 DLU</td>
</tr>
<tr>
<td>Integral Switch</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
</tr>
<tr>
<td>Display</td>
<td>2.0” (5.1 cm), graphical, backlit, 176 x 220, color</td>
<td>2” (5.1 cm) graphical, backlit, 176 x 220, color</td>
<td>Digital, 16-bit graphical backlit TFT Color, 2”</td>
<td>Digital, 16-bit graphical backlit TFT Color, 2”</td>
</tr>
<tr>
<td>Touch Screen</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Speakerphone</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Number of Line Keys</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
<td>N/A</td>
</tr>
<tr>
<td>Programmable (Soft) Keys</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Fixed Feature Keys</td>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Advanced Features</td>
<td>Push-to-Talk via XML</td>
<td>Bluetooth v2.0, Push-to-Talk via XML</td>
<td>Atmospheres Explosibles (ATEX) Zone 2/Class 22, IP64 rating</td>
<td>Integrated EA11 2D bar-code scanner, MIDlets</td>
</tr>
<tr>
<td>Handsfree</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Message Waiting Indication</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Third-Party XML Support</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Headset Port</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Signaling Protocol</td>
<td>SCCP</td>
<td>SCCP</td>
<td>SCCP</td>
<td>SCCP</td>
</tr>
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The Business of Connectology
# Cisco SCCP / SIP Endpoints

<table>
<thead>
<tr>
<th>Model</th>
<th>7931G</th>
<th>Conference Station 7937G</th>
<th>3905</th>
</tr>
</thead>
<tbody>
<tr>
<td>License (RTU)</td>
<td>4 DLUs</td>
<td>3 DLUs</td>
<td>N/A</td>
</tr>
<tr>
<td>Integral Switch</td>
<td>10/100</td>
<td>No</td>
<td>10/100</td>
</tr>
<tr>
<td>Display</td>
<td>4.3” (10.8 cm), graphical, backlit, 192 x 64, monochrome</td>
<td>Pixel-based</td>
<td>128 x 32 pixel-based, graphical monochrome LCD without backlit</td>
</tr>
<tr>
<td>Touch Screen</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Speakerphone</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Number of Line Keys</td>
<td>24</td>
<td>1 line (0 line keys)</td>
<td>1</td>
</tr>
<tr>
<td>Programmable (Soft) Keys</td>
<td>4 soft keys, 22 line keys for use as lines, speed dials or PLKs</td>
<td>4</td>
<td>0</td>
</tr>
<tr>
<td>Fixed Feature Keys</td>
<td>7</td>
<td>9</td>
<td>8</td>
</tr>
<tr>
<td>Advanced Features</td>
<td>Join across lines, Transfer across lines</td>
<td>Optional wired microphone kit</td>
<td></td>
</tr>
<tr>
<td>Handsfree</td>
<td>Yes</td>
<td>Yes Speakerphone</td>
<td>Yes</td>
</tr>
<tr>
<td>Message Waiting Indication</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Third-Party XML Support</td>
<td>Text XML only</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Headset Port</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Signaling Protocol</td>
<td>SCCP/SIP</td>
<td>SCCP</td>
<td>SIP</td>
</tr>
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The Business of Connectology
## Cisco SCCP / SIP Endpoints

<table>
<thead>
<tr>
<th>Model</th>
<th>7942G</th>
<th>7945G</th>
<th>7962G</th>
<th>7965G</th>
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<tbody>
<tr>
<td>License (RTU)</td>
<td>4 DLU</td>
<td>4 DLU</td>
<td>4 DLU</td>
<td>4 DLU</td>
</tr>
<tr>
<td>Integral Switch</td>
<td>10/100</td>
<td>10/100/1000</td>
<td>10/100</td>
<td>10/100/1000</td>
</tr>
<tr>
<td>Display</td>
<td>5.0” (12.7 cm), graphical, 320 x 222, 4-bit, gray scale</td>
<td>5.0” (12.7 cm), graphical, 320 x 240, backlit, 16-bit, color</td>
<td>5.0” (12.7 cm), graphical, 320 x 222, 4-bit, gray scale</td>
<td>5.0” (12.7 cm), graphical, 320 x 240, backlit, 16-bit, color</td>
</tr>
<tr>
<td>Touch Screen</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>Speakerphone</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Number of Line Keys</td>
<td>2 - lighted</td>
<td>2 - lighted</td>
<td>6 - lighted</td>
<td>6 - lighted</td>
</tr>
<tr>
<td>Programmable (Soft) Keys</td>
<td>4 soft keys, 2 line keys for use as lines, or as a line plus a speed dial or PLK</td>
<td>4 soft keys, 2 line keys for use as lines, or as a line plus a speed dial or PLK</td>
<td>4 soft keys, 6 line keys for use as lines, speed dials or PLKs</td>
<td>4 soft keys, 6 line keys for use as lines, speed dials or PLKs</td>
</tr>
<tr>
<td>Fixed Feature Keys</td>
<td>10</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Advanced Features</td>
<td>High-resolution and many infrastructure integration options; support of Headset Hookswitch Control</td>
<td>High-resolution and many infrastructure integration options; support of Headset Hookswitch Control</td>
<td>High-resolution and many infrastructure integration options; support of Headset Hookswitch Control; key Expansion Module support with up to 2 79145, 79155 or 79165</td>
<td>High-resolution and many infrastructure integration options; support of Headset Hookswitch Control; key Expansion Module support with up to 2 79145, 79155 or 79165</td>
</tr>
<tr>
<td>Handsfree</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
</tr>
<tr>
<td>Message Waiting Indication</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Third-Party XML Support</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Headset Port</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
<td>Yes, Wideband support</td>
</tr>
<tr>
<td>Signaling Protocol</td>
<td>SCCP/SIP</td>
<td>SCCP/SIP</td>
<td>SCCP/SIP</td>
<td>SCCP/SIP</td>
</tr>
</tbody>
</table>

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The Business of Connectology
# Cisco SCCP / SIP Endpoints

<table>
<thead>
<tr>
<th>Model</th>
<th>CUPC</th>
<th>9971</th>
<th>9951</th>
<th>8941</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>License (RTU)</strong></td>
<td>5 DLU</td>
<td>4 DLU</td>
<td>4 DLU</td>
<td>N/A</td>
</tr>
<tr>
<td><strong>Integral Switch</strong></td>
<td>N/A</td>
<td>10/100/1000 802.11 a/b/g Wi-Fi</td>
<td>10/100/1000</td>
<td>10/100</td>
</tr>
<tr>
<td><strong>Display</strong></td>
<td>PC Settings</td>
<td>VGA presentation, 5.6 in. (14 cm) graphical color touch screen, 24 bit color depth 640 x 480 effective pixel resolution</td>
<td>VGA presentation 5 in (10 cm) Graphical TFT color display, 24-bit color depth 640 x 480 effective pixel resolution</td>
<td>VGA video calling, and applications, 5-inch graphical TFT color display, 24-bit color depth, 640 x 480 pixel resolution</td>
</tr>
<tr>
<td><strong>Number of Line Keys</strong></td>
<td>N/A</td>
<td>6</td>
<td>5</td>
<td>4</td>
</tr>
<tr>
<td><strong>Programmable (Soft) Keys</strong></td>
<td>N/A</td>
<td>4 (via touch screen)</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td><strong>Fixed Feature Keys</strong></td>
<td>N/A</td>
<td>12</td>
<td>12</td>
<td>13</td>
</tr>
<tr>
<td><strong>Advanced Features</strong></td>
<td>Group Chat, Presence, Directory Pictures, One Click Email - Video Call - Regular Call</td>
<td>Desktop Wi-Fi Ethernet XML &amp; Midlet apps QoS reporting, KEM support/SDIO Card</td>
<td>XML and Midlet apps QoS reporting, KEM support</td>
<td>Integrated Camera</td>
</tr>
<tr>
<td><strong>Handset</strong></td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td><strong>Message Waiting Indication</strong></td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td><strong>Third-Party XML Support</strong></td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td><strong>Headset Port</strong></td>
<td>Required on PC</td>
<td>Yes HD Voice</td>
<td>Yes HD Voice</td>
<td>Yes</td>
</tr>
<tr>
<td><strong>Signaling Protocol</strong></td>
<td>SIP</td>
<td>SIP for signaling H.264 for video</td>
<td>SIP for signaling H.264 for video</td>
<td>SCCP or SIP for signaling, H.264 for video</td>
</tr>
</tbody>
</table>

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**The Business of Connectology**
## Cisco SIP / Video Endpoints

<table>
<thead>
<tr>
<th>Feature</th>
<th>E20</th>
<th>EX60</th>
<th>EX90</th>
<th>C20</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco TelePresence Endpoints</strong></td>
<td>![E20 Image]</td>
<td>![EX60 Image]</td>
<td>![EX90 Image]</td>
<td>![C20 Image]</td>
</tr>
<tr>
<td><strong>CUCM Version</strong></td>
<td>8.5(1)</td>
<td>8.6(1)</td>
<td>8.6(1)</td>
<td>8.6(1)</td>
</tr>
<tr>
<td><strong>Video Quality</strong></td>
<td>W448p30 (16:9)</td>
<td>1080p 30/720p 60</td>
<td>1080p 30/720p 60</td>
<td>1080p 30/720p 60</td>
</tr>
<tr>
<td><strong>Screen Size/Resolution</strong></td>
<td>10.6” WXGA (1280 x 768)</td>
<td>21.5”/1920 x 1080</td>
<td>24”/1920 x 1200</td>
<td>16:9 Widescreen, default resolution for HDMI is 1280 x 720@60Hz.</td>
</tr>
<tr>
<td><strong>Camera Included</strong></td>
<td>Yes, 35° total view +5°/-5° digital tilt</td>
<td>Yes, 1080p30, 50° view Doc cam</td>
<td>1080p30, 45° - 65° view (zoom) Doc cam</td>
<td>1080p12 x, 12 x zoom, +15°/-25° tilt, +/- 90° pan, 43.5° vertical, 72° horizontal</td>
</tr>
<tr>
<td><strong>MultiSite</strong></td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td><strong>Bandwidth</strong></td>
<td>SIP up to 1152 kbps</td>
<td>SIP up to 6 Mbps point-to-point</td>
<td>SIP up to 6 Mbps point-to-point</td>
<td>SIP up to 6 Mbps point-to-point</td>
</tr>
</tbody>
</table>

*The Business of Connectology*
Thank you
SIP Trunking – Carrier Perspective

Version 1.2
Agenda

• Company Overview
• Basic Architecture / Delivery Methods
• Common Misconceptions
• Q&A
Who is tw telecom?

- About **28,000 local and regional fiber route miles** across 75 markets
- Nearly **16,000 buildings** with fiber based services and connectivity
- National footprint interconnected with fiber and **10 Gig IP backbone**
Basic SIP Trunking Architecture

- **tw telecom** provides SIP Trunking using a consistent architecture regardless of the customer’s voice equipment.
- Managed eSBC ensures reliable interoperability between networks.
Fundamentals

- Quality of the Network
  - Jitter
  - Latency
  - Packet Delivery
- Codec

Performance Metrics for Albany

<table>
<thead>
<tr>
<th>Destination City</th>
<th>Packet Delivery</th>
<th>Latency (ms)</th>
<th>Jitter (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Colorado Springs</td>
<td>100%</td>
<td>46 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Portland</td>
<td>100%</td>
<td>73 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Greensboro</td>
<td>100%</td>
<td>23 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Birmingham</td>
<td>100%</td>
<td>32 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Charlotte</td>
<td>100%</td>
<td>22 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Dallas</td>
<td>100%</td>
<td>39 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Las Vegas</td>
<td>100%</td>
<td>66 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>San Antonio</td>
<td>100%</td>
<td>46 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Ft. Worth</td>
<td>100%</td>
<td>40 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Chattanooga</td>
<td>100%</td>
<td>32 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Jackson MS</td>
<td>100%</td>
<td>42 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Boise</td>
<td>100%</td>
<td>64 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Denver</td>
<td>100%</td>
<td>44 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Lake Charles</td>
<td>100%</td>
<td>45 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Santa Barbara</td>
<td>100%</td>
<td>73 ms</td>
<td>0 ms</td>
</tr>
<tr>
<td>Columbus GA</td>
<td>100%</td>
<td>32 ms</td>
<td>0 ms</td>
</tr>
</tbody>
</table>
SIP Mi$conceptions

VOIP and SIP calls are free from 800 charges? NOT

VOIP and SIP calls are free from LD charges? NOT

SIP will save me hardware cost with Softphone usage? NOT

SIP call quality is not up to par and could cost my company’s image? NOT

SIP will save me hardware cost with less Voice TDM cards to buy for my legacy TDM PBX? TRUE

SIP will save me DR downtime cost with phone mobility? TRUE

SIP is Cheaper

- Internet based SIP may be cheaper, but…….
  - No Class of Service.
  - Little control over voice priority.
  - Voice traffic traverses multiple carrier networks/ISP’s.
  - Voice quality affected by internet performance.
  - DDoS attacks can take down voice service.
- Is it really less expensive in the long run?
  - How much does it cost the customer to lose a single call? What is the cost of a choppy voice conversation with a customer?
What about 911 Service?

Companies like Vonage and residential type Vendors and Providers really hurt 911 and VOIP reputation early on. Today E911 issues are solved with advances in 911 service and PS/ALI (private switch/automatic location identifier) with the PSAP itself to give the ability for multiple emergency response locations per trunk group.
How about SIP & Fax Machines

Three forms of fax over IP networking:

• **Realtime fax** using the T.38 protocol and T.38 based fax gateway devices installed on the IP network.

• **Internet fax** - Also known as T.37. The ITU standard for sending a fax-image file via e-mail to the intended recipient of a fax.

• **VoIP based fax** - Also known as G.711 pass through - This is where the fax call is carried in a VoIP call encoded as audio. **Most Vendors only support this type of fax.**
After all that – WHY?

Key Service Benefits

- Cost Savings
- Interoperability
- Security
- Scalability
- Reliability

Application Versatility

- Local and Long Distance calling
- Disaster Recovery
- Virtual Telephone Numbers
- Unified Communications
- Interactive Video
- Organizational Collaboration
- Interactive Conferencing

Cost Savings

- Native SIP hand-off eliminates protocol conversion on LAN
- Avoids costly TDM infrastructure expenses
- Managed eSBC minimizes capital and operating expenses
- Supports centralized application management
- Provides free on-net calling
- Includes bundle of free long distance minutes
- Enables more efficient bandwidth utilization
PSTN Sunset Coming! SIP will Grow!

- A Technical Advisory Council (TAC) recommended on June 29, 2011 to the FCC they set a “date certain” for the sunset of the PSTN.
- When will the PSTN “end”? A recent study by the National Center for Health Statistics says it all.

As of My 2010:
- 23% of respondents lived in a mobile-only household
  - 37% of adults in the 18-24 and 30-34 age groups
  - Only 6% of the US population will still be served by the PSTN (defined as TDM access line service) by the end of 2018

- What will replace the PSTN?
  - Some future technology?
  - Cell (mobile)
  - VOIP/SIP has the lead
Questions