The Basics of TDM to VoIP Interconnection

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Attendee Dashboard

- Type and submit your questions in the Questions Pane
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Control Panel
- Click plus [+ ] icon to expand menus
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Questions Panel
- Type and submit your questions in the Questions Pane
- Located on the right hand side of your screen
About Sangoma

• Industry pioneer with over 25 years of experience is communications hardware and software
• Publicly traded company since 2000
  – TSXV: STC
• One of the most financially healthy companies in our industry
  – Growing, Profitable, Cash on the Balance Sheet, No Debt
• Mid-market sized firm with just under 100 staff in all global territories
  – Offices in Canada (Toronto), US (NJ), EU (UK & Holland), APAC (India), CALA (Miami)
• World Wide Customer base
  – Selling direct to Carriers and OEMs
  – Selling to the Enterprise through a network of distribution partners
Broad Line of Great Products

- **Voice Telephony Boards**
  - Analog/digital/hybrid, WAN, ADSL

- **Software Applications**
  - NetBorder Express
  - Call Progress Analyzer
  - Lyra AMD for Asterisk

- **VoIP Gateways**
  - SIP-to-TDM
  - TDM-to-SIP
  - SS7-to-SIP

- **Session Border Controllers**
- **Microsoft Lync**
- **Wireless Products**
- **Cloud based monitoring**
- **Fiber connectivity (STM1)**
- **Transcoding (boards/appliances)**
Vibrant Ecosystem of Clients & Partners

Open Source Telephony
Ready to use drivers for Sangoma boards

- Asterisk
- FreeSWITCH
- caliweaver
- elastix®
- yate
- trixbx
- The Open Platform for Business Telephony

Proprietary PBX and IVR
Plug-in to major soft-PBX and IVRs

- Microsoft Lync
- BARRACUDA NETWORKS
- snom
- AASTRA
- 3CX
- Solutions by Fonality
- INTECH
- Nucleum

Contact Center
OEM Integration with major software suites

- ORACLE
- VICidial
- Genesys
- Orecx
- inConcert

Carriers, Cloud, Data Ntwks
Proven Infrastructure Technology

- BT
- verizon
- CISCO
- ERICSSON
- SIEMENS
- MTT
Innovation and Interoperability

Indian Army

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Key Points of this Presentation

• Common issues in IP-to-TDM and IP-to-IP interconnections, and mitigation strategies to address them

• Key technology solutions to bridge disparate networking technologies and applications common today and coming in the future

• A practical migration strategy to seamlessly and cost-effectively connect the service provider to the customer, even in the face of disparate technologies
Challenges in a TDM-to-VoIP Migration

• Service Provider View Point
  – Capital Expenditure
  – Learning Curve
  – Potential Service Disruption
  – Security

• Customer View Point
  – Capital Expenditure
  – Learning Curve
  – Potential Service Disruption
  – Security
Finger Pointing

You Go First

WE Go No Where

No, you Go First

NO! you Go First
Disparate Technology Islands

THE REST OF THE WORLD’S TDM NETWORK

Customer’s IP Network

Carrier IP Network

THE REST OF THE WORLD’S IP NETWORK

Customer’s TDM Network

Carrier TDM Network
All IP Doesn’t Solve All Issues

THE REST OF THE WORLD’S IP NETWORK

Customer’s IP Network

Your IP Network
SIP Interop Challenges

• SIP (RFC3261) and Interoperability challenges
  – Largest RFC
  – Not a ‘super tight’ spec:
    • Should: 344 times
    • Can: 475 times
    • May 381 times
    • Option: 144 times
  – Lots of room for interpretation
  – SIP Endpoints end up with slight differences that make it hard to interconnect

• End point could use different codecs
LOOKING AT THE CUSTOMER PREMISE
Applications Enabled

• SIP Trunking (SIP-SIP)
• VoIP Enablement of Existing TDM Equipment (TDM-SIP)
• Legacy Connectivity for the IP-PBX (SIP-TDM)
• Connecting Legacy Equipment to an IP-PBX
Over The Top Services

• SIP and Other VoIP Services Can Be Delivered Two Ways
  – Dedicated Facility
    • Broadband and SIP Delivered By Single Entity
  – Over The Top
    • Broadband and SIP Delivered By Different Providers
Defining SIP Trunking

• “The use of SIP service bundled in a familiar business construct to connect a PBX or gateway to the PSTN over the Internet”
• An alternative to T1s, PRIs, and POTS lines
• Pure SIP will allow you to make and receive calls worldwide and should include:
  – E911
  – Directory Assistance
  – CNAM (Caller ID/Name)
SBC used to provide security between the two IP Networks
May also handle signaling interoperability
May also handle basic transcoding
Connection from PBX could be digital or analog
SIP Trunking Benefits

• SIP Trunking allows companies to replace physical PSTN trunks with virtual, broadband trunks, deployed over data connections

• Can be
  – Dedicated lines or shared connections
  – Internet connections
  – Burstable

• Introduces more flexibility, more efficiency, reduces operational costs
VOIP ENABLEMENT OF EXISTING TDM EQUIPMENT
VoIP Enablement Diagram

- Gateway used to enable SIP service on legacy PBX
- Connection from PBX could be digital or analog
VoIP Enablement Benefits

- Enables Service Provider to retain customers who have moved not yet moved to an IP Infrastructure
- Mitigates the threat of encroachment of the Over-The-Top SIP Provider in the Future
- All Integration takes place at the customer premise. No need to change Service Provider network, equipment or operations.
- Allows customer to take advantage of cost savings and additional flexibility while still leveraging their existing investment in their legacy PBX
LEGACY CONNECTIVITY FOR IP-PBX
VoIP Legacy Connectivity

- Service Provider Delivers TDM
- IP-PBX or Other VoIP Services at Premise
Legacy Connectivity Benefits

Carrier Benefits

• Enables Service Provider to retain customers who have moved to VoIP before the Service Provider is Prepared
• Mitigates the threat of encroachment of the Over-The-Top SIP Provider
• All Integration takes place at the customer premise. No need to change Service Provider network, equipment or operations.

Customer Benefits

• No issues with 911 or other emergency services at customer premise
• Maintain connectivity even in the event of an Internet or WAN failure
• Alarm systems, fax machines, etc. can still be connected directly to the legacy PSTN connections
CONNECTING LEGACY EQUIPMENT TO AN IP NETWORK
Legacy Gear Connectivity

- FXS Devices look like IP Devices to the IP-PBX
Legacy Gear Connectivity Benefits

• No need to redevelop TDM-based systems (IVR/Contact Center) before moving to an IP Communications Platform
• Preserve investment in endpoints and structured wiring
LOOKING AT CARRIER-TO-CARRIER CONNECTIVITY
Technology Touch Points

• Extending investments in TDM equipment
• Accessing SS7 from the VoIP Network
• Managing Codec Requirements when integrating with other Providers
• Network Security in an all IP Environment
SS7 GATEWAY
SS7 Explained

• Signalling System number 7 (SS7) is a legacy PSTN protocol that is in use in carrier networks
• Some markets use SS7 to connect to large enterprises (such as a call center)
• It is a very vast protocol set
  – It is the base for the PTSN
  – All GSM networks run on SS7
• Network Nodes are identified with Point Codes
• SS7 protocol has several layers
  – MTP 1, 2, 3 for routing messages
  – ISUP for call set-up
  – TCAP for transactions (800, CNAM, LNP)
SS7 VoIP Gateway Components

- As networks are moving to VoIP, there is a need for Media Gateways to convert PSTN Media and Protocols to IP based Media and Protocols
- An SS7 Media Gateway needs
  - Telecom Interface (typically T1/E1)
  - IP Network Interface (typically Ethernet)
  - Protocol interpreters (or stacks) – SS7 ISUP/SIP
  - Lots of software to transform and translate how things are done on each side of the gateway
Audio Codecs

- Codecs encode and decode voice for network transmission
- **Sampling rate**: Number of samples per seconds
  - The higher rate the better the quality
- **Quantization**: Granularity of the representative data
  - The more granularity the better the quality
- **Bit rate** = [Sampling rate] X [Quantization bits]
  - The more quality is desired, the highest the bit rate will be
- **Sampling time**: How long do I collect samples before “feeding” the algorithm
  - 10 ms, 20 ms, 30 ms, etc.
- **Algorithm** is the core of the codec
  - Compression formulas
  - It’s is digital signal processing and a lot of math
Sample Audio Codecs

• G.711 (PCM/PCMA, mu-law / A-law) – Narrowband
  – Sampling rate: 8 KHz (8000 times per second)
  – Bit Rate: 64 Kbps
  – Sample Time: 10ms, 20ms, 30ms, 40ms, etc.

• G.722 – Wideband
  – Sampling rate: 16 KHz (16000 times per second)
  – Bit Rate: 48kbps, 56kbps and 64kbps
  – Sample Time: 10ms, 20ms, 30ms, 40ms, etc.

• G.722.1 Annex C – Ultra-wideband
  – Sampling Rate: 32 KHz (32000 times per second)
  – Bit Rate: 48kbps
  – Sample Time: 20ms, 40ms, 60ms
Need for Transcoding Servers/Proxy

- I have a call for you
- What do you support?

1. I’m calling the phone on the other side
   - I can do G.711

2. SIP Voice Call

3. OK. I will broker the call on both ends
   - I can do G.729

4. SIP Voice Call
POLL QUESTIONS
SESSION BORDER CONTROLLER
Benefits of Deploying an Enterprise SBC

• Network Security
  – Toll Fraud Protection
  – Protection against Denial of Service (DoS) attacks
  – Topology Hiding
  – Encrypting signaling and media (SIP/TLS & SRTP)

• Connectivity
  – NAT Traversal
  – IPv4 to IPv6 Interworking
  – Protocol Normalization (SIP-SIP, SIP-H.323)

• QoS/QoE
  – Quality of Service
  – Quality of Experience

• Media Services
  – Transcoding
  – DTMF Interworking
  – Call Recording

• BSS Support
  – Billing
  – Logging
  – Statistics

• Regulatory Compliance
  – Lawful Intercept
Enterprise Security Threats

• Denial of Services
  – Call/registration overload
  – Malformed messages (fuzzing)

• Configuration errors
  – Mis-configured devices
  – Operator and application errors

• Theft of service/Fraud
  – Unauthorized users
  – Unauthorized media types

• BYOD
  – Smartphones running unauthorized apps
  – Viruses and Malware attacking your VoIP network
Service Provider/Carrier Security Threats

• Denial of Services
  – Call/registration overload
  – Malformed messages (fuzzing)

• Configuration errors
  – Misconfigured devices
  – Operator and application errors

• Theft of service/Fraud
  – Unauthorized users
  – Unauthorized media types
Typical VoIP Service Provider Network

- ITSP (Internet Telephony Service Provider)
- PSTN (Public Switched Telephone Network)
- SS7 (Signaling System 7)
- VoIP (Voice over Internet Protocol)
- Legacy PBX
- Sangoma NetBorder Transcoding Gateway
- Sangoma NetBorder SS7 to VoIP
- IP PBX
- IP Network
- Softswitch
- CPE
- Broadband Router
- Vega SBC
- VoIP Peering
- NetBorder SBC
- Vega GW
- Legacy PBX

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Conclusions

• Both the carrier and the enterprise share common goals in migrating from TDM-to-IP
• “Integration at the Edge” can enable TDM-To-IP migration incrementally, preserving capital and resources while delivering advanced functionality to your user base.
• Sangoma can be your partner in managing this transition for you or your customers
Upcoming Webinars

Sangoma Education Series:

• **Cost Effective Tapping and Call Recording using the Sangoma T116**
  – April 22, 2013 @ 11:00am EST

• **The SBC – The Critical Component**
  – May 16, 2013 @ 2:00pm EST

• **Adding Telephony to Microsoft Lync with Office 365 and Other Use Cases**
  – June 11, 2013 @ 2:00pm EST

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