The State of SIP and the Role of the SIP Forum

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SIP Forum President and Managing Director

Today's Presentations

- Review of the SIP Forum and the current State of SIP and IP Communications (Marc Robins)
- Latest from the IETF Real-Time Applications (RAI) WG and the new UA-Config Common Data Profile TG (Mary Barnes, SIP Forum Director, new UA-Config TG Chair and Chair of DISPATCH)
- SIP Forum FoIP TG Update (Mike Coffee, TG Co-Chair and Max Schroeder, TG Whip)
- SIP and IPv6 (Dan York, Renaissance Man and Director, ISOC IPv6 Effort)

SIP Forum Background

- Founded in 2000 in Sweden
- Leading Non-Profit IP Communications Industry Association
- Membership ranks comprised of Corporate "Full Members" that pay annual dues and support the work of the Forum, Academic Institutions and Individual "Participant" Members (10K+)

Founding SIP Forum Mission

- “Advance the development and deployment of innovative IP communications solutions that comply with, and properly interoperate with, other products and services that use the Session Initiation Protocol (SIP) protocol.”

We Won!

- Interoperability is the new battle cry -- among end-point devices, enterprise IP-PBXs, and SIP-enabled Service Provider Networks – for all types of applications and services.

SIP FORUM Full Member Companies

(as of 9-14-2011)

4psa.com
acme packet
Alcatel·Lucent
Aspect
AudioCodes
AVAYA
Bell
BROADSOFT
Innovation calling.
CableLabs

Optimum
Lightpath
CBEYOND
Cedar Point
COMMUNICATIONS
Simplifying The Network
cisco
comcast
cox
ERICSSON
GENBAND

HUAWEI
inGate
JUNIPER
NETWORKS
Metaswitch
Networks
Microsoft
MITEL
net
Network Equipment Technologies
NOKIA
CONNECTING PEOPLE

POLYCOM
REDWOOD
TECHNOLOGIES
ROGERS
SAMSUNG
SAGEMCOM
ShoreTel
SIEMENS

snom
VoIP phones
Sonus
NETWORKS
Sorenson
Communications
TECNOTREE
TEKELEC
TELE2
Telsis
iol

SIP Forum Academic/Institutional Members

Columbia University:

The Fu Foundation
School of Engineering & Applied Science

Fraunhofer Institute for Open Communication Systems
FOKUS

TEXAS A&M UNIVERSITY

ILLINOIS INSTITUTE OF TECHNOLOGY

University of Glamorgan
because great minds don't think alike

Georgetown UNIVERSITY est. 1789

Gradiant

Telecom's Evolutionary Phases

- First : Replace the RJ-11
  - Immediate gains in CAPEX as single wiring harness simplifies campus management.
  - Greenfield ROI – NO Brainer
- Second : Replace the TIE Lines
  - Integrate Enterprise wide Dial Plan Management into single IP Network. Immediate OPEX gains.
- Third : Replace the PRI (Today) -- SIP Trunking
  - All IP E2E
- Fourth : SIP Federations (Direct Peering with Business Partners?)
- Fifth : Seamless Campus/Mobility Integration?

Growth of SIP Adoption

- 2009 FCC estimates were that 20% of all US carrier-delivered voice was running on a SIP-based infrastructure. Today…35%?

- With the introduction of VoLTE in mobile networks, SIP will quickly become the dominant protocol for real-time voice communications and eventually video.

More Statistics

		Sources
Establishments (business sites) in the US, 2007	7.7 million	*US Census, 2011*
Under 20 employees	6.6 million	*US Census, 2011*
20 to 99 employees	892,000	*US Census, 2011*
100-499 employees	161,300	*US Census, 2011*
Over 500 employees	19,200	*US Census, 2011*
Business Lines in the US, 2007	45 million	*US Census, 2011*
PBX lines in the US, 2011	104.9 million	*TIA, 2010*
IP PBX lines in the US	80.3 million	*TIA, 2010*
TDM PBX lines in the US	24.6 million	*TIA, 2010*
% of installed base on TDM, 2012	<20%	*TIA, 2010*
SIP Trunk penetration, US, 2011	8%	*Wall Street estimates*
SIP endpoints in use, US, 2016	46 million	*Gartner, 2010*
Unified IPT/ IM/ presence clients in use, 2015	48 million	*Frost & Sullivan, 2010*
Average monthly cost per TDM business line	$48	*US Census, 2011*
Average monthly cost per VOIP business line	$35	*Current Analysis estimate*

Current SIP Forum Activities

- Advances product/service interoperability
  - SIPit interoperability test events (SIPit 29 in Monte Carlo October 24-28, 2011)
  - Technical Working Group efforts include SIP Trunking (SIPconnect 1.1), Fax-over-IP, UA Config, IPv6 – Security, Video and ??? To Come?
- Develops industry-wide technical recommendations and best-practice implementation guides (i.e., SIPconnect)
- Produces Technical Conferences (SIPNOC)
- Contribute to IETF and ITU-T, liaise with industry groups (i3 Forum, UCIF, IPv6 Forum, etc)

Current SIP Forum Activities, con't

- Provides Industry Licensing Programs (i.e., SIPconnect Compliant Program)
- Creates educational content
  - White papers, Informational RFCs and other reference documentation
- Builds awareness about SIP and IP Communications Technology
  - Educational seminars and other events
  - Articles and other editorial in industry online newsletters and blogs, trade magazines and journals
- Maintains growing community of IP Communications industry professionals

SIPit

- Week-long engineering test events
  - Held twice a year
  - Moves around the globe
- Averages around 100 implementations
  - 60 to 90 companies
  - 16 to 20 countries
- Issues are corrected in real-time
  - Allows immediate retesting
- Very high-yield testing
  - Participants claim 4 to 6 months of results from the week at SIPit compared to what they would achieve testing separately

Goal: Find and Fix What Doesn't Work

- The Standards Improve
  - If two teams have to argue about the specification, the specification needs to be corrected
  - Testing identifies errors and omissions in the Standards
  - Reports from the SIPit events allow the IETF to remain focused on real industry needs

SIPNOC – SIP Network Operators Conference

Two-day educational conference for service providers to focus on how to “make SIP work in the network” and address the key operational issues facing SIP in today’s telecom world

SIPNOC Con't

- SIPNOC EU (March/April 2012 Timeframe), Cologne or Amsterdam
- SIPNOC US 2012 -- June 25-27, Hyatt Dulles, Herndon VA

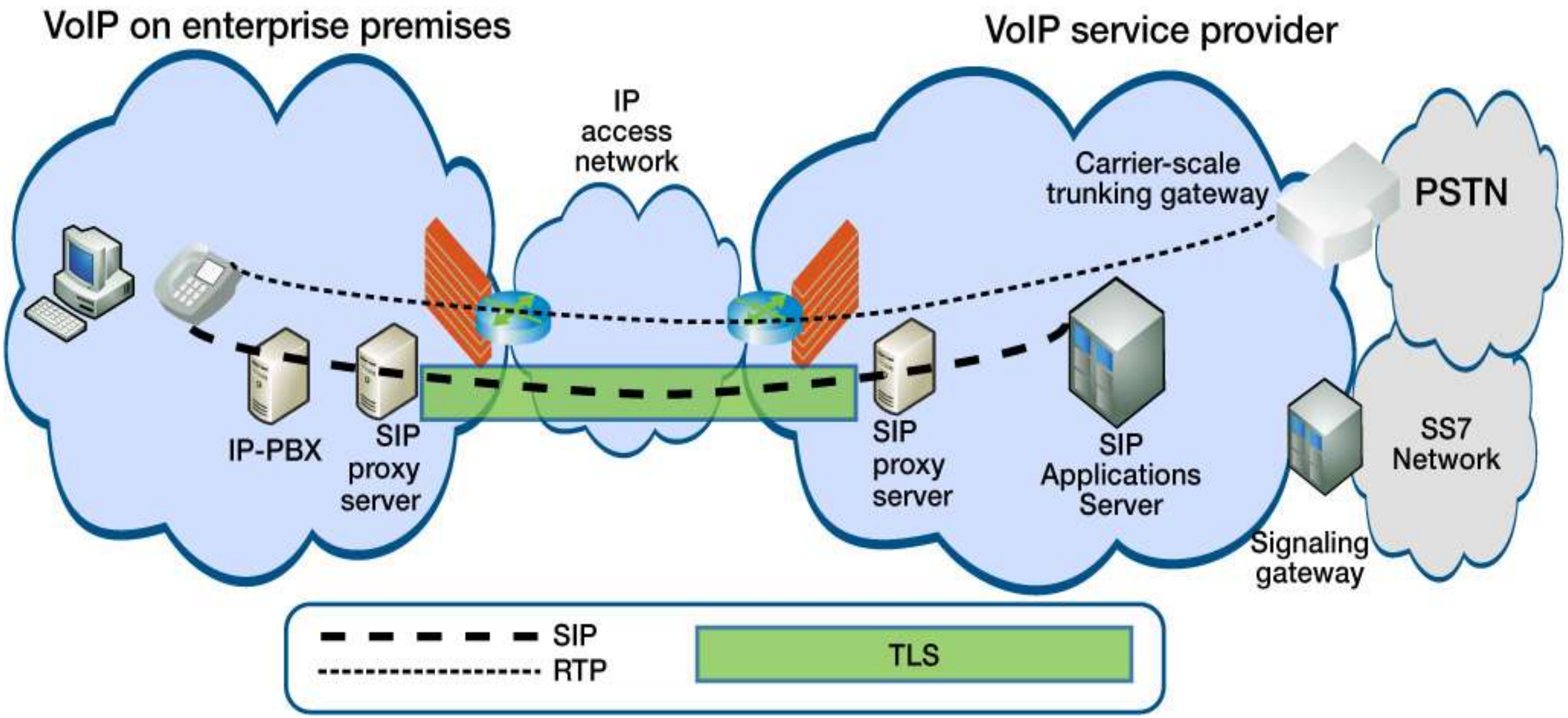
SIPconnect

- The **SIPconnect Technical Recommendation** is a consensus-driven, vendor and service provider agnostic, standards-based "Best-Practices" specification that provides detailed rules and guidelines for achieving direct IP peering and interoperability (SIP Trunking) between IP PBXs and VoIP service provider networks.

Realizing The Promise of IP Communications

- Problem: IP-PBXs have successfully cut costs and delivered new features to customers, BUT…TDM Routing of VoIP Traffic is a Limited Approach to Achieving Next Generation Telephony
- Opportunity: Preserving and Extending Next-Generation IP Communications Capabilities Beyond the Enterprise
- Solution: Direct IP Peering, or Creating a Seamless, End-to-End Connection between SIP-enabled IP-PBXs and SIP-enabled VoIP Service Provider Networks

SIPconnect Reference Architecture



The SIPconnect Compliant Program

- The **SIPconnect Compliant Certification Program** allows eligible companies to license the use of the SIP Forum's 'SIPconnect Compliant' certification mark -- the official brand of the leading standard for SIP Trunking products and services.

The SIPconnect Value Proposition

- Offers a Universal Approach to SIP Trunking
- Delivers Customer Cost Savings
  - eliminates gateways and extends VoIP’s benefits (DID, conferencing, etc.)
- Enables Transparent Feature Transport
  - end-user info can be passed from IP-PBX to network enabling presence and other apps to travel from point-to-point
- Optimizes Quality of Service
  - transport layer issues are defined – i.e., QoS configuration, echo cancellation, method for DTMF relay, packetization rates, codec support and fax/modem traffic
- Provides Security
  - well-defined approaches to identity and authentication provide a secure model for direct IP peering

SIPconnect 1.1 Milestones

- Ratified March 2011.
- Notable changes and improvements from SIPconnect v 1.0 to Version 1.1 include:
- Standards-based support for both Static (DNS-based) and Registration (SIP REGISTER-based) modes of operation
- Description of SIP Endpoint functionality required for interworking, with detailed discussion of various error conditions and appropriate responses to those errors
- Description of Media Endpoint functionality required for interworking
- Focus on E.164-based SIP AoRs - the common case for deployments
- Additional voice services (Call Forward, Call Transfer, etc.) using SIP techniques with the widest deployment, with simplified call flows
- A detailed description of TLS usage
- A roadmap on what implementers can expect in subsequent SIPconnect revisions (IPv6, Emergency Services, etc.)
- Use of RFC 6140 to achieve bulk Registration of E.164-based AoRs in registration mode

FoIP TG Milestones

- Brought together researchers, engineers, and service providers to exchange ideas, share experiences, and propose approaches to address FoIP problems. .
- Performed Tandem Network T.38 FoIP Call Testing
- Logged results and are now in the process of analyzing the data.
- The End-Game – Deliver a “Best Practices” Recommendation

Active FoIP Task Group Participants

- Representatives from Alcatel-Lucent, AEMcom, AT&T, AudioCodes, Biscom, Dialogic, Digium, C4U Solutions, Cisco, Commetrex, Comunycarse, DevFoundry, emFAST, Faxback, Faxcore, Lexmark, LSI, NeuStar, Omnitor, Orange, Packetizer, Sagemcom, Siemens Enterprise, Sonus, Teridian, Telecom Poland
- SIP Forum Leadership
  - Eric Burger, CTO, NeuStar
  - Marc Robins, Managing Director and President, SIP Forum LLC
  - Richard Shockey, Chairman of the Board
- Task Group Leadership
  - Mike Coffee, Commetrex and Gonzalo Saguiero, Cisco, Task Group Co-Chairs
  - Max Schroeder, Faxcore, Task Group Whip

User Agent Configuration Task Group

- Formed as a result of SFSIW follow-up meeting during IETF 72
- Needed to address SIP UA configuration as a barrier to successful deployment of SIP
- Working method: mailing list discussion, conference calls, face-to-face meetings
- In conjunction with IETF meetings and SIPits
- Mailing list subscription:
  - http://sipforum.org/mailman/listinfo/ua-config
- @65 participants

Milestones of the UA-Config Effort

- Developed simple boot mechanism suitable for a variety of UA types.
- Handed off to IETF and was ratified as RFC 6011 -- Sets a standard procedure for how a SIP User Agent locates, retrieves and maintains current configuration information for a given SIP Service Provider.
- Phase 2 of the work now active: Development of a Common Data Profile for the Configuration File (Chaired by Mary Barnes)

Join the SIP Forum! -- Get Involved!

- Visit the SIP Forum Website: www.sipforum.org
  - Register for participant membership – it's FREE!
  - Join the discussion list
  - Learn more about SIP Forum events and initiatives
- Attend a SIPit: www.sipit.net
- Attend SIPNOC: www.sipnoc.org
- Get Serious -- Participate in a Forum Task Group!
- Join the IP Communications Industry Thought Leadership!

THANK YOU!