

SIP Trunking Workshop

SIP Trunking/SIPconnect Overview and Value Proposition

Marc Robins, SIP Forum President and Managing Director
Richard Shockey, SIP Forum Chairman of the Board



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 (@8000)

SIPFORUM Full Member Companies

(as of 1-19-10)



SIP Forum Academic/Institutional Members

Columbia University:



The Fu Foundation
School of Engineering & Applied Science



Fraunhofer
Institute for Open
Communication Systems



TEXAS A&M
UNIVERSITY



ILLINOIS INSTITUTE OF TECHNOLOGY



University of Glamorgan

because great minds don't think alike

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.”

Current Focus

- ❖ The battle for widespread adoption has been won (but there are other battles to win...)
- ❖ The new battle cry is “interoperability” -
- among end-point devices, enterprise IP-PBXs, and SIP-enabled Service Provider Networks – for all types of applications and services.

Current SIP Forum Activities

- ❖ Advances product/service interoperability
 - SIPit interoperability test events
 - Defines and create compliance tests
- ❖ Addresses key technical issues
 - Technical Working Group efforts include SIP Trunking (SIPconnect), User Agent Configuration, Fax-over-IP
 - SIP Security Special Interest Group (SIG)
 - Unified Communications, HD Video, Smart Grid task groups in planning stages
- ❖ Develops industry-wide technical recommendations and best-practice implementation guides (i.e., SIPconnect)

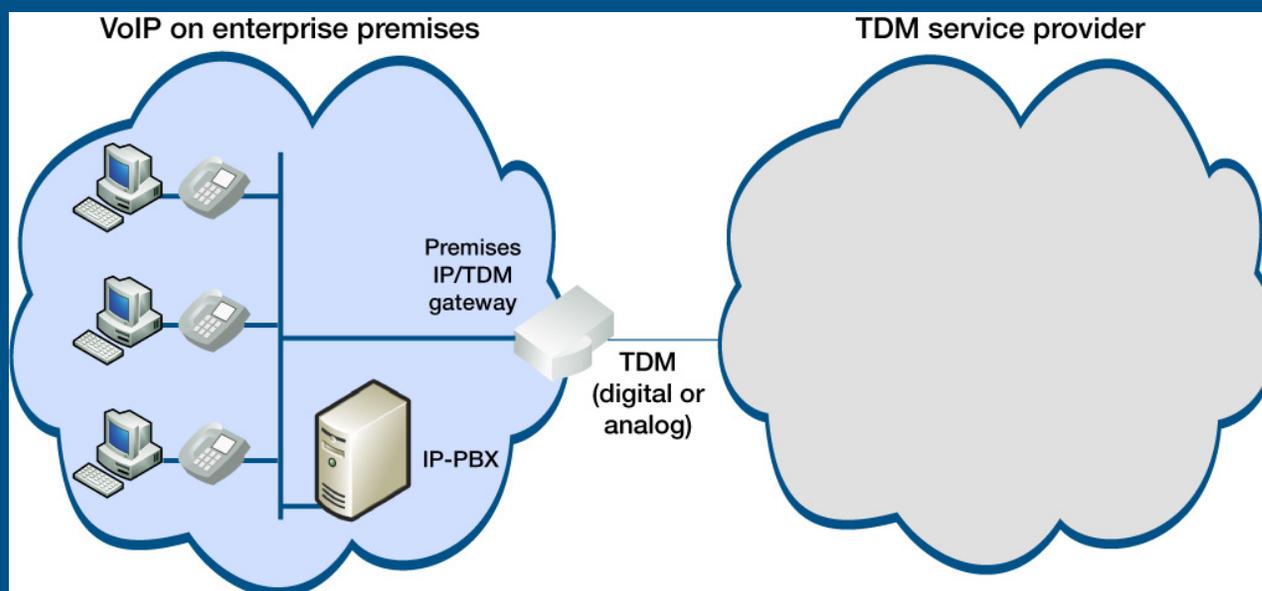
Current SIP Forum Activities, con't

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

Realizing The Promise of IP Communications

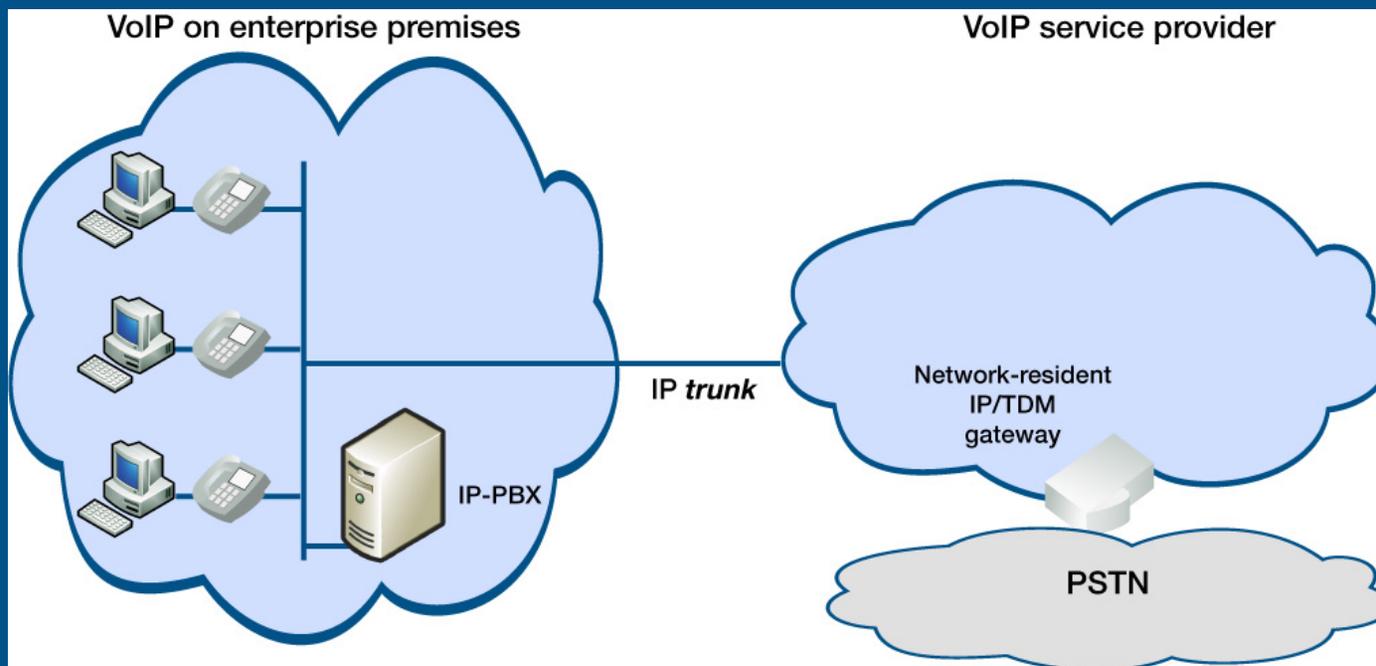
- ❖ **Problem:** IP-PBXs have successfully cut costs and delivered new features to customers, BUT...TDM Routing of VoIP Traffic is a Limited Approach to Achieving Next Generation Telephony
- ❖ **Opportunity:** Preserving and Extending Next-Generation IP Communications Capabilities Beyond the Enterprise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

The Old Way



- ❖ Enterprises must use a gateway to connect IP PBXs to PSTN
 - Increases investment required to move to VoIP, reduces ROI
- ❖ Limits ability to leverage VoIP's advantages
 - Adapting to the PSTN means least common denominator functionality
 - Enterprises cannot fully leverage low-cost VoIP termination providers

The New Way



- ❖ Connecting IP PBXs directly to VoIP service providers provides significant advantages
 - More features, less cost
- ❖ But, how to do it?

SIP Is Key, but SIP Alone is Not Enough

- ❖ SIP is the industry standard for VoIP, but...
 - There's a lot to SIP; but **what parts are relevant** for this?
 - e.g. How to handle addressing in the presence of multiple firewalls
 - When we have SIP options, **what choices do we make?**
 - e.g. Inter-domain authentication / registration policy
 - Some **solution elements lie "above" SIP**
 - e.g. OA&M around hierarchical logical identities
 - Users, customers, locations, DID blocks, ...
- ❖ What's needed?
 - An **industry accepted interconnection method** that uses SIP to build links between SIP-enabled PBXs and SIP-compliant service provider networks

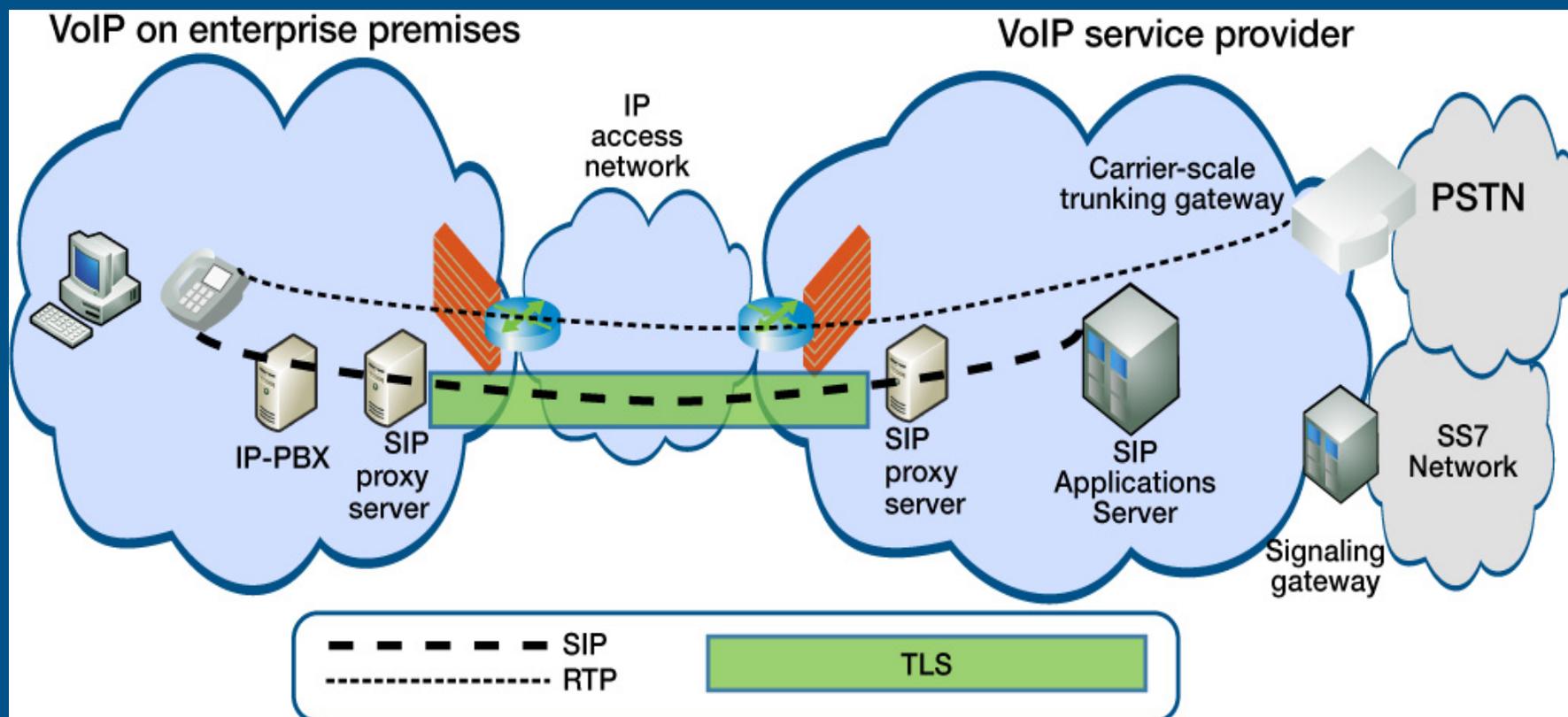
What is SIPconnect?

❖ SIPconnect specifies a reference architecture

- Minimum set of IETF and ITU-T standards that must be supported
- Provides precise implementation rules and guidelines where existing standards allow for multiple implementation options.
- Specifies a minimum set of capabilities that should be supported by service provider and enterprise networks

SIPconnect Reference Architecture

Common Functional Elements Required to Support SIPconnect



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

A Competitive Edge for IP PBX Manufacturers

- ❖ Why should IP PBX manufacturers care?
- ❖ Because direct IP peering is a huge value add for businesses and service providers alike – entities that purchase and interconnect with IP PBXs
 - Addresses QoS and security issues
 - Reduces equipment and transport costs
 - Increases features and functionality
 - Eliminates need to set up proprietary interfaces

Benefits for Service Providers

- ❖ Improved QoS and security via superior interconnection to the network
- ❖ Ability to offer higher quality services with advanced features tailored to IP PBX users
- ❖ Ability to forge strong relationships with IP PBX vendors
- ❖ Ability to establish new relationships with distribution channel: interconnects, system integrators and VARs.

Cost Savings and New Features for Business Customers

- ❖ Eliminates TDM gateways and increases efficiency of local access facilities
- ❖ Provides DID capabilities w/o requiring the recurring expense of analog lines or expensive digital circuits
- ❖ Improves voice quality by removing gateway latency and includes the attentive management of QoS, echo cancellation as well as fax and modem support
- ❖ Creates the right foundation for personalized applications and rich media services between customers and service providers as well as between customers and other IP-connected PBXs

Benefits for Distributors and Channel Partners

- ❖ Eliminates PSTN interconnection woes
 - No quality of service problems (i.e. latency and echo)
 - No need to perform custom configurations on a customer-by-customer basis
- ❖ Allows service providers to manage QoS
- ❖ Allows security-related functions to be “off-loaded” from customer premises to VoIP networks (incl. NAT traversal for seamless SIP connectivity) and other security concerns (i.e, denial of service attacks, etc.)