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The Business Value of SIP VoIP and Trunking

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The Transition of Trunking

This article will take a look at the evolution of telecommunications trunking technologies from traditional time division multiplexed (TDM) trunks and tie lines based on T-1 circuit infrastructure to IP-based trunking and built-in Session Initiation Protocol. SIP adoption has become a vital success factor on the road to unified communications and is critical to comprehensive integration with enterprise business applications to reach the Communications Enabled Business Processes (CEBP) of tomorrow. More than another standards-based protocol, SIP has become the *de facto* standard for unified communications that bring voice, video, and data together.

Traditional Trunking Technology Evolution

In the evolution of traditional telephony, trunking technology dates back to very early in the maturation of the Public Switched Telephone Network (PSTN). In this article, we can't begin to address the complexity of the evolution of the PSTN, but the outline in Figure 1 highlights major milestones from early in the history of telecommunications to the present.

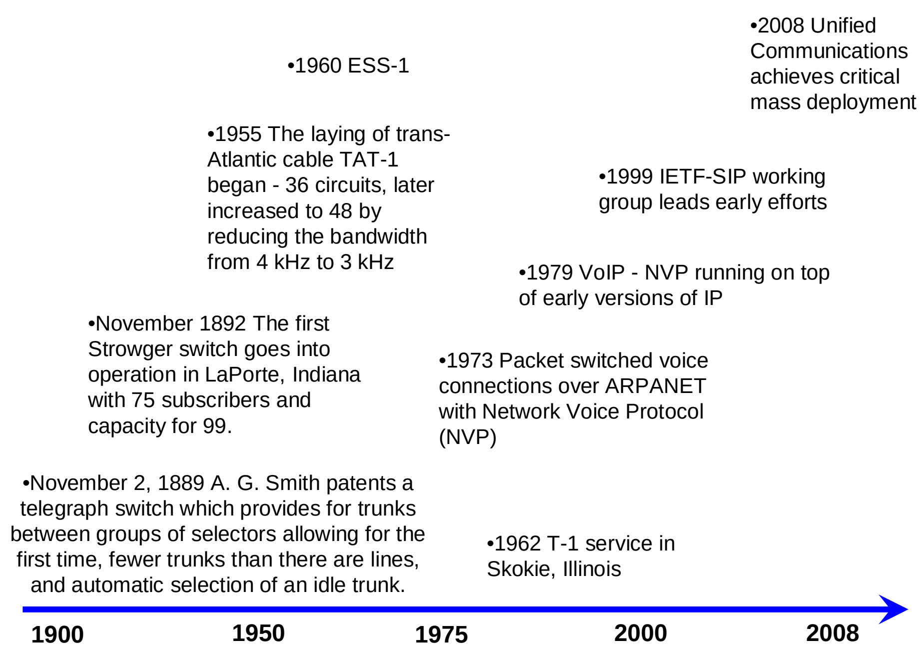

Figure 1: Telecommunications trunking evolution timeline.

Trunking circuits have provided several functions through the years as telecommunications networking became more complex, adding features and services. Some trunk circuits were used for outgoing calls. Others handled the supervision of incoming trunks. Traffic Supervision Position System (TSPS) trunks were used to provide a connection to the telephone operator. TSPS, and other trunking technologies, were based on *analog* circuit technologies. TSPS was later replaced by Operator Service Position System (OSPS) to incorporate the features of the Class 5 Electric Switching System (5ESS) telco switch that was introduced in 1982.

For long-distance phone calls, trunks could be connected from one to another (called *tandem trunking*) to create voice paths that traversed the continent and circled the globe. Trunking circuits also provided tie lines between enterprise Private Branch Exchange (PBX) systems.

The biggest change to trunking technology before SIP was the digitization of the PSTN. After the first T-1 service in Skokie, IL went live in 1962, the Bell System—the dominant provider in North America—embarked on an aggressive effort to digitize the entire PSTN. This conversion took many years as *digital* circuits replaced their analog predecessors.

> 🕮 For a detailed explanation of digital versus analog technologies, see *The Fundamentals of Packetized Voice*, Chapter 3 of *IP Telephony Demystified* (McGraw-Hill, ISBN 0071406700).

The Role of Multi-Protocol Label Switching in the Network

Multi-Protocol Label Switching (MPLS) is a widely used mechanism for delivering Quality of Service (QoS) in packet switched networks. In IP networks, MPLS eliminates the normal hop-by-hop routing in IP from the equation. MPLS adds a *tag* to each packet. This tag shortcuts the delivery path by sending packets to the best available route for a given traffic type. MPLS has been widely adopted in both enterprise and service provider networks. As business networks integrate Voice over IP (VoIP) service, these QoS enhancements are frequently needed to support the growing volume of VoIP and now video traffic.

MPLS has often been referred to as a bypass or "shim" protocol. The insertion of a tag into the packet stream adds minimal overhead that is easily offset by the enhancements of service. MPLS is often referred to as a *Layer 2½ protocol* because it straddles Layers 2 and 3 of the OSI Model.

> 🕮 See http://computer.howstuffworks.com/osi.htm for a simple, online explanation of the OSI Model.

Using MPLS for QoS

Implementing VoIP makes many enterprises reassess network capabilities. The readiness assessment performed in the planning stage of VoIP deployment typically presents findings that require this evolution. MPLS can further introduce fundamental changes to the way these IP networks operate. VoIP service brings a change in thinking about the expectations for the network. MPLS allows IP networks to mimic some of the behavioral characteristics of a circuit-switched network like the PSTN.

MPLS Structure

MPLS works by inserting an MPLS shim header, or *tag*, into the beginning of each packet. This shim header contains one or more *labels* and is often called a label stack.

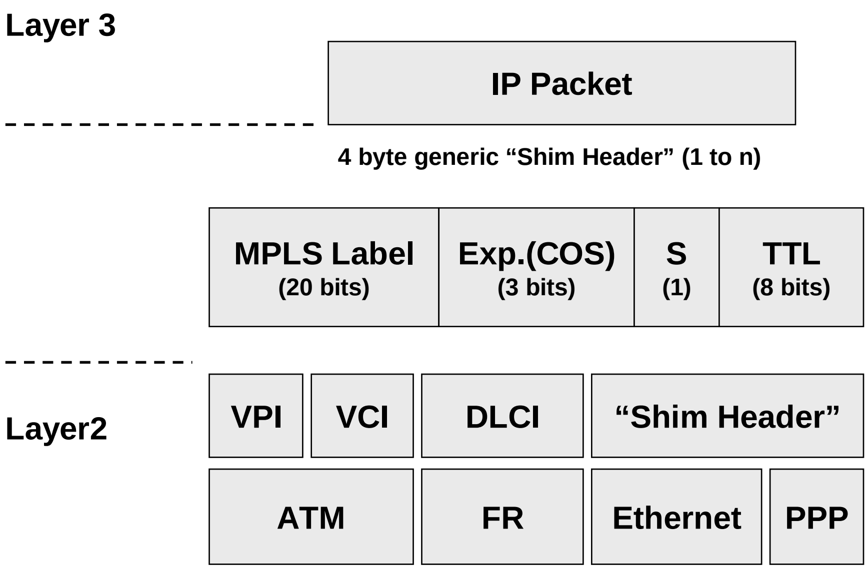

Figure 2: MPLS packet labels and encapsulation.

The MPLS label stack entry contains four fields:

- A label value
- Class of Service (COS)
- A flag to indicate whether this label is the last label in a stack
- A time to live (TTL) value

MPLS performs a label lookup/switch instead of a lookup into an IP routing table. Label switching can be performed within the switching fabric of the hardware, so it's faster than typical routing.

QoS vs. QoE

When implementing VoIP services, it's prudent to separate thinking about QoS and Quality of Experience (QoE). QoS parameters are typically systems-based metrics, including delay, jitter, and packet loss. QoE metrics are directly related to services such as VoIP. VoIP QoE metrics address areas such as call completion rate, call setup time, and audio quality—the factors that a person on the phone notices.

Traffic Aggregation for Similar Traffic Classes

The COS field designates the assignments for different classes of service. The common and simplest view is to assign three service classes for user traffic and a fourth for management traffic. Typical MPLS classes of service are

- Real-time traffic—Voice and interactive video, for example, are almost always assigned the highest priority. This assignment not only guarantees adequate bandwidth but also supports providing the delay, packet loss, and jitter characteristics needed for delivery of real-time traffic such as VoIP and video.
- Mission-critical data traffic—Traffic such as that from a legacy mainframe is usually placed in a call by itself that guarantees delivery. Delivery timing is often a more stringent requirement for this type of data than other factors.
- All remaining traffic—The leftover traffic is usually aggregated into a best efforts service class that simply mirrors how IP normally handles traffic delivery.
- Management traffic—This traffic is commonly aggregated into a management class by itself because it requires assurances that it can still be delivered even during periods of heavy network congestion.

MPLS Virtual Private Networks for Privacy and Aggregation by Business Class

Using these techniques, not only can similar traffic types be aggregated into the same class of service but it can also be separated into what are typically called Virtual Routing Forwarding (VRF) tables. This aggregation creates a forwarding equivalency class (FEC) for switching throughout the network. These VRFs are used for separation of traffic for privacy or for traffic engineering.

This approach creates a virtual private network (VPN) environment. It all takes place at the MPLS header level, so traffic destined for one destination—whether a customer or a business division in an enterprise—can never be seen by another. Service providers commonly use MPLS-based VPN aggregation; they provide QoS markings within each VPN and at the egress points to other networks.

Using MPLS encapsulation techniques, layers of MPLS labels are built, enabling hierarchical switching of MPLS packets. This approach enables a carrier to deliver the privacy of a dedicated network coupled with the QoS guarantees required to support real-time traffic such as VoIP.

> MPLS Resources
>
> MPLS Resource Center at http://www.mplsrc.com/index.shtml
>
> MPLS MFA Forum at http://www.mplsforum.org/
>
> IETF MPLS Working Group at http://www.ietf.org/html.charters/mpls-charter.html
>
> IETF RFC 3031 at http://www.ietf.org/rfc/rfc3031.txt

MPLS can't be compared directly to IP. It's a complimentary protocol. MPLS works in conjunction with IP and IP's interior gateway (IGP) routing protocols. MPLS introduces the capability of traffic engineering to the IP network.

MPLS and IP's routing protocols are the glue that bind a network together and ensure it works as designed. Border Gateway Protocol (BGP) enables the concept of peering between organizations for passing or sharing traffic routes. This BGP peering conceptually provides a nice way of thinking about SIP trunking. Just as BGP is the glue that binds the Internet together as a *network of networks*, SIP is the glue that binds unified communications systems together for VoIP and other communications services.

Introduction to SIP

Intense work on SIP really began in earnest with the Internet Engineering Task Force (IETF) in 1999. It was one of many efforts and has been led by the IETF-SIP working group. Their charter states that SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users. Such sessions include voice, video, chat, interactive games, and virtual reality. This group has worked long and hard to help SIP mature. What began as a series of proposed drafts and standards, including numerous extensions, has become a foundation for VoIP and unified communications.

The basic model and architecture defined for SIP sets out some specific characteristics:

- Wherever possible, SIP services and features are provided end-to-end
- Extensions and new features must be generally applicable; they cannot apply only to some specific set of session types
- Simplicity is key
- Existing IP protocols and architectures are re-used and integrated tightly

SIP uses an addressing structure similar to email addresses. Users may log in anywhere and be dynamically assigned an IP address, so there has to be a way to resolve some of the common conventions in the active and current IP address.

SIP is text based, so the addresses, which are SIP URLs or URIs (Uniform Resource Locaters or Indicators), can be imbedded in email messages or Web pages. Additionally, as SIP is a text protocol, SIP URLs and URIs are network-neutral. Thus, a URL might point to an email-like address, using SIP, an H.323 address or even a telephone number on the PSTN.

SIP operates independently of the IP network layer. It requires only unreliable packet delivery and provides its own reliability mechanism. Although it's widely used in IP networks today (usually over UDP to avoid the overhead of TCP), SIP can run over IPX, Frame Relay, ATM, AAL5, or X.25 with no changes.

> There are hundreds of references on the Internet for readers who want to learn more about SIP. For example, there is an excellent set of resources maintained by Columbia University at http://www.cs.columbia.edu/sip/. Wikipedia also has a good starting point article at http://en.wikipedia.org/wiki/Session_Initiation_Protocol.

Evolution of SIP Beyond Phone Calls to Trunking

Why does trunking matter to the enterprise? One of the most obvious reasons is that SIP trunks reduce telecommunications expenses. For a recent explanation, Scott Lowe reviewed the situation at Westminster College on the Tech Republic Web site (http://blogs.techrepublic.com.com/tech-manager/?p=501). First, let's look at why the interest in change:

> *Between T1 costs, a "billing charge" and local and long distance usage, Westminster College spends quite a bit of money on communications costs each month. The billing charge is a several-hundred-dollars charge that we incur for the privilege of receiving an itemized statement at the end of each month. We're a relatively small place, so we have just a single T1, which is dedicated for outgoing calls. For incoming calls, we have a pair of PRI circuits in place and, because of rather strict contract termination terms, will stay in place for the next couple of years. Our outgoing T-1, however, is up for grabs.*

Their primary driver was expense. Cost is a big driver for change in any enterprise, and VoIP as a cost-reduction tool has often provided the impetus for change. For many organizations, that is the first step toward unified communications.

As Lowe describes the situation, SIP trunks to replace the existing T-1 resulted in a significant cost savings, and the inclusion of all local and long-distance calling within the US. According to their calculations, a move to SIP trunks would cut usage charges by 95 percent.

An added incentive for many organizations is that SIP trunks can be delivered over existing Internet connections. Thus, when they aren't in use, the bandwidth is still available for other business activity. When you compare this broad use of SIP trunks against existing T-1 lines that mostly sit idle, it is a measurably more efficient use of resources.

A SIP trunk can be connected to any IP-PBX that supports SIP. Most manufacturers of IP-PBXs support SIP today. At Westminster College, they're looking at the Avaya IP Office PBX as a solution.

If you use existing Internet connections, keep QoS and QoE issues in mind. If you have enough bandwidth and your Internet provider has appropriate peering agreements, quality is likely to be quite good. But remember that SIP runs over IP. Your solution, if not properly designed, might be more prone to dropped calls or echo than with legacy voice circuits to a traditional PBX.

SIP has been widely adopted by many of the fixed-line carriers. They are all migrating their core networks to VoIP. SIP is widely used in the gateways for the connections between these carriers.

SIP itself has evolved. Its early focus was on the idea of a peer-to-peer communications protocol. Today. SIP has become the centerpiece of the family of protocols delivering our next-generation networks (NGNs). SIP's biggest near-term growth will be in interworking, or SIP trunking, to connect IP-PBXs and communication systems to IP service providers.

Summary

SIP will continue to extend the communications market. IP-based PBXs are increasing in use as traditional TDM systems reach the end of usable life, with applications now taking center-stage. The adoption rate of VoIP gateways, handsets, and other components is on the rise.

As enterprises adopt unified communications strategies as a business tool that integrates voice, video, and data services onto a single, converged network, SIP is truly the unifying protocol that brings all communications together. SIP sets the foundation for reduced cost, improved efficiency, and the next generation of CEBP.