What's SIP Got To Do With It?

Five Compelling Reasons Why SIP Will Dominate Enterprise IP Telephony

WHITE PAPER

Introduction

Recent LAN and WAN technologies make voice sound good.

Since 1998 when 3Com brought to market the first IP PBX, the enterprise Voice over IP (VoIP) phenomenon has changed not just telecommunications, but also enterprise networking. The 3Com® NBX® system ushered in a new era of communications. For the first time, companies were able to leverage their investments in high performance local area networks (LANs) to deliver real-time services such as IP telephony.

In fact, the major standardized innovations in Layer 2 switches and enterprise routers in the past five years—virtual LAN (VLAN) segmentation, packet prioritization, Power over Ethernet (PoE), differentiated services, multi-protocol label switching—enhance converged networking and its delivery of high-quality audio.

- VLAN segmentation separates voice traffic from e-mail uploads, web surfing, and database interactions
- Packet prioritization techniques enable higher priority for voice packets than other applications so that jitter and delay are minimized or eliminated
- PoE enables a combination power-and-communications cable to minimize the required number of 120V power ports
- Differentiated services and Multi-Protocol Label Switching (MPLS) enable prioritization for voice packet streams and rapid transit across IP wide area networks (WANs)

Enhancements to the wireless LAN environment through the auspices of the IEEE 802.11 committees also facilitate high quality IP telephony services. Recently ratified specifications for Quality of Service over the air (802.11e) and privacy services (802.11i) address many of the same issues being addressed by other 802.1 committees.

Technology advances are driving significant investment in the development and deployment of more reliable and high-performance LANs. This momentum helps maximize the reliability and audio quality of IP telephony implementations and create solutions to match the cost expectations of enterprises.

2005—the Year of VoIP

2005 is a fundamental turning point for the IP telephony industry. For the first time, more IP-PBX ports will ship in the United States than legacy digital ports . This is a major milestone, indicative of the reliability and audio quality now available in IP telephony services. The discussion of "if" users will migrate to IP telephony is now "when" and with what strategic urgency.

Classically, the year of inflection represents an acceleration of the pace of growth in the market. In the case of IP telephony, this accelerated growth is occurring now for two primary reasons:

- Depreciation schedules—equipment purchased in preparation for Y2K is being retired now, enabling new purchases
- Large enterprise needs—emerging business requirements necessitate enhanced, cost-effectives services, already enjoyed by small businesses with IP telephony deployments

The value proposition for IP telephony is moving beyond cost reduction-oriented business cases, to a framework of "convergence applications" that offer advantages difficult to accomplish with legacy telephone implementations. Applications—such as IP messaging—can broadcast voice mail and deliver voice mail as e-mail. IP conferencing can provide audio, video, and data conferencing services. IP contact centers offer freedom from the geographic constraints of legacy solutions. IP mobility solutions that provide on-campus voice and data services via wireless LANs are proving themselves critical for business productivity and customer interactions.

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The Importance of Session Initiation Protocol in Enterprise IP Telephony

If indeed applications are the driving force behind IP telephony implementations, what will be the key enabler of enterprise applications, enterprise applications developers, and integrations into existing enterprise applications?

It is the premise of this white paper that Session Initiation Protocol (SIP) will fill that need.

SIP is simple.

Using the Internet Engineering Task Force (IETF) fundamental of technology reuse and the proven value of simplicity, the SIP message set is quite an elegant construction—six messages that appear in clear text to facilitate call setup. Clear text allows for easy troubleshooting and avoids complex software interactions and other processing that affect interoperability. The six messages are Invite, Trying, Ringing, OK, ACK, Bye.

SIP is both elegant and practical. Its base assumption is that all SIP endpoints and elements exist in the IP environment, an arena already equipped with standard mechanisms to handle packet transport priorities, privacy, and other required services. These value-added services do not require specification in the SIP framework. Whereas, in legacy environments such as the public telephone network the use of hyper-text transport protocol (http) is neither integral or readily available.

This "building block" approach on an IP base is unique to IETF initiatives. The results are nearly trivial protocol definitions such as SIP in contrast with older session control or interface protocols such as H.323 or Q.SIG from the International Telecommunications Union (ITU). These older, telecom-centric protocols have considerably more complex definitions as shown in Table 1.

In civilization and in nature, simple has been proven to be both useful and enduring. For example, the sphere and its derivatives are frequently occurring shapes in fruit and celestial objects as contrasted to the more complex pyramid. The steering wheel, rather than the handle bar, is used in automobile control designs. The flat CD replaced the more complex floppy disk cartridge design as the definitive media storage framework.

Years ago, a wise man once explained why Asynchronous Transfer Mode (ATM) was going to destroy a carrier service of interest to the phone companies at the time—Switched Multi-megabit Data Service (SMDS)—that was a connectionless ATM-type service. Two arguments were given for his assertion. First, ATM was simpler (notice three-letter acronym versus the four-letter acronym). Second, ATM didn't require a phone company to implement the technology, resulting in ATM having a larger addressable market that could provide ample opportunity for solving business problems (relevance).

SIP is secure.

There are tradeoffs between interoperability and cost when considering security. Simple standards make it easier to interoperate within a multi-vendor network, but they also expose the network to potential abuse. This vulnerability, however, can be effectively addressed by strong authentication and privacy services. that do not interfere with the primary business benefit of standard-based solutions and also enable a long investment life through interchangeable vendors, services, applications, and devices.

The IETF framework for SIP offers a rich set of standards for authentication and privacy, including secure SIP, secure RTCP, and secure RTP. These capabilities leverage IETF proposals for the use of standard implementations such as Transport Layer Security (TLS) for robust session privacy service and Secure/Multipart Internet Mail Extensions (S/MIME) for session control packet privacy.

Session Initiation Protocol (SIP)

SIP was developed in 1999 as a media and application independent session control protocol by the Internet Engineering Task Force, the same people and process developers that manage all important Internet protocols like IP, FTP, TLS, and HTTP. SIP is denoted as RFC 3261, which defines the messages, options, and protocols at the heart of SIP.

Learn more at:

www.3com.com/voip/sip.html or **www.ietf.org/rfc3261.html**

TABLE 1: Contrast between key SIP attributes and H.323 signaling protocols. [Source: Business Communications Review, October 2004]

ATTRIBUTE	H.323	SIP
Design	Complex	Simple
Standard	736 pages	128 pages
Number of elements	100s	37
Messages	Based on ASN.1	HTTP and RTP
Call Setup	23 steps	5 steps
Extensibility	Designed for LAN	Designed for IP
Large Phone number domains	Limited	Open to all sizes
Firewall Support	Difficult	Easier
Interoperability among vendors	Poor	Good

Most vendors implementing H.323 offer no such options. Instead, they choose to implement proprietary derivatives to facilitate rudimentary forms of privacy in first generation IP-PBX devices. This approach is typical of the PBX vendor solutions brought to market during the past two decades in which proprietary digital signaling protocols were implemented on endpoints and PBX fabric. The result of these decisions was nominally-better security, but considerably more vendor-specific lock-in that failed to provide many useful features—such as call forward and hold—at a standard service level.

SIP is a standard.

Despite being a standard, SIP implementations do vary. Some vendors, most notably legacy TDM-based PBX vendors and first generation IP telephony vendors who built their products around H.323 or around proprietary protocols, offer SIP interoperability as an interface into their otherwise non-SIP telephony system. This approach helps the vendor maintain account and feature control. They can charge extra for SIP support for only the most rudimentary features while offering advanced premium-priced feature options with their proprietary call control protocol and implementation.

Alone among the leading IP-PBX vendors, 3Com has implemented a different approach. Using standard SIP among endpoints such as IP phones, video cameras, call controllers and gateways to the public telephone network, 3Com SIP-based solutions enable a robust and inexpensive, vendor-neutral environment, capable of rapidly delivering a portfolio of cutting-edge applications for presence, conferencing, contact center, messaging, and mobility services. Furthermore, 3Com is a founding member of the SIP Forum and promotes the 3Com Voice Solutions Provider Program as an initiative to encourage SIP-based interoperability. A list of more than 30 vendors, service providers, and solutions that are proven to interoperate with 3Com IP telephony modules is available at: **www.3com.com/voip/interoperability.html**

SIP applications abound.

Internet search engines can be an amazing proxy for markets and issues. Their objective is to scour the Internet, capture prose, process whatever they find, and quickly present it to users with some rank ordering that approximates users' views of relevancy. Google, in particular, is an effective measure of "buzz" because of its contentpresentation in terms of users' views of relevancy.

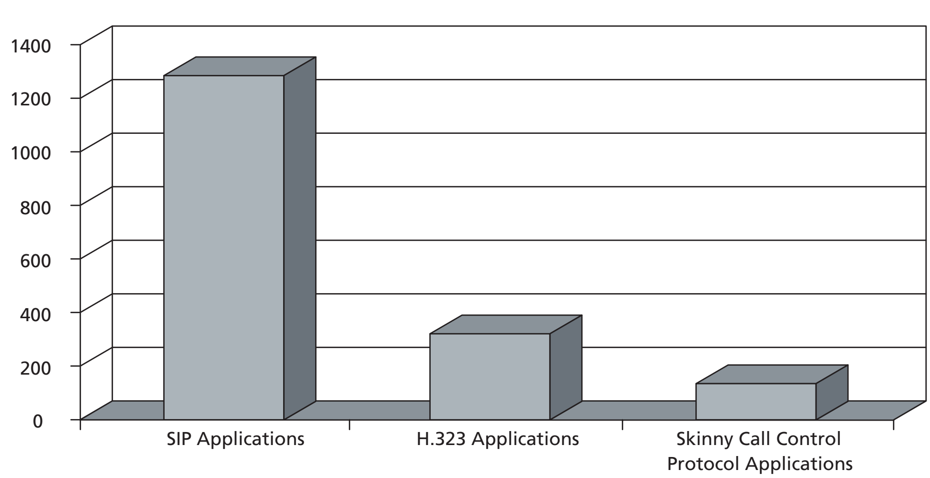
Figure 2 shows the contrast between the number of references found by the Google search engine for "SIP applications", "H.323 applications", and "Cisco Skinny Call Control Protocol." In this analysis, the presence of open, simple (SIP) references clearly outpaces references for open, complex (H.323) and proprietary applications. End users are recognizing the enormous flexibility and value of SIP.

SIP is used by enterprises and carriers.

Since SIP transcends carrier/enterprise boundaries, it can attract applications developers able to deliver their core applications technologies in a variety of value packages and to address a range of financing models and markets. For example:

- The **stand-alone enterprise market**, addressed by direct sales or resellers
- The **stand-alone consumer market**, addressed by retailers, online or direct marketers
- The **hosted service model**, sold to carriers but billed to enterprises in monthly charges

FIGURE 2: Thousands of references on Google.com, January 2005.



3Com has been a major contributor to the convergence industry since it brought to market the NBX IP-PBX in 1998. It developed an architecture for a distributed softswitch for AT&T and introduced it in the first commercially deployed carrier softswitch in 1999. Then in 2003, the 3Com VCX™ solution became a key component of the world's first convergence applications suite. Maintaining its leadership position, the company is also a founding member of the VoIP Security Alliance.

Transforming business through innovation is not new to 3Com. It holds over 1,600 patents and is the market leader in IP telephony for small to medium enterprises. It operates in over 45 countries with approximately 1,900 employees.

3Com: we're changing the way business speaks.

- The **hosted service model**, sold to carriers but billed to consumers in monthly charges
- The **stand-alone enterprise market**, addressed by carrier enterprise equipment sales organizations

This multi-market model provides ample space for initial focus. Depending on vendor strengths and skills, the model supports considerable opportunity for both startup and established developers.

Since SIP is used by enterprises and carriers across multiple markets, carriers can offer innovative services aimed at enterprises. For example, SIP trunking services allow two SIP-empowered enterprises to communicate using RTP from endpoint to endpoint without the need for a gateway between them. The growth of this category of services is helping reduce costs by lowering the load on packet-circuit gateways and reducing regulatory and tax burdens.

SIP trunking also improves audio quality since bandwidth allocation can be negotiated end-to-end instead of end-to-gateway-over-digital-to-gateway-to-end. Fewer digital hops with more bandwidth enables wide-band audio quality, as well as SIP-initiated video conferencing.

Summary

As demand for IP telephony and SIP services and implementations continues to accelerate, the public telephone network faces change. A mainstay of the global economy for the past ten decades, it is losing ground to a network of world-wide IP communications integrated with simple, secure, standards-based, applications-rich implementations and services.

There can be little doubt of the important role that SIP will play in facilitating this transition and enabling powerful enterprise applications that reduce cost, improve user productivity, and strengthen customer interactions. Some vendors will be slow to appreciate and take advantage of this opportunity. They may focus their energies on attempting to retrain the transition rather than exploring its possibilities.

3Com embraces innovation and standards. They are a foundation on which the company builds business-enhancing solutions. The question is no longer why, but when will SIP be integral to every business' future?

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