A SIP PRIMER

SIMPLE DEFINITION: SIP (session initiation protocol) is a simple, common language for real-time communications, including voice, video and instant messaging.

HISTORY OF TELECOM TECHNOLOGY

TIMEFRAME	***TECHNOLOGY***	***BENEFITS***
1970's	ANALOG	Make/Receive multiple phone calls, intercom
1980-1985	ELECTRONIC	Features such as speakerphone, forward, transfer, etc.
1985 - 1999	DIGITAL	Improved speakerphones, voice mail, easier-to-use
1999-2006	VOIP	Voice on data network, easy administration, unified messaging
2006 and on	***SIP (2nd generation VOIP)***	***Simplified architecture, a standard for all mfrs and carriers***

1ST GENERATION VOIP: The first generation of Voice over IP was all about putting voice on a data network. The manufacturers have succeeded and VOIP technology has replaced traditional TDM (time division multiplexing) telephone systems. Soon, VOIP will begin replacing standard business lines from AT&T and the other carriers. However, every manufacturer took a different approach and systems remained proprietary.

2ND GENERATION VOIP: SIP is the second generation of VOIP. Now the industry has a VOIP standard to which everyone will comply. Second, it has widened the scope from voice to include IM, video, wireless and carrier products.

TeleDynamic
COMMUNICATIONS

WHY DO I NEED A SIP-BASED PBX?

#1. NEXT GENERATION VOIP: SIP is the standard that all telephone systems, carriers and applications will use going forward. Everyone in the industry has a plan to go to SIP technology at some point.

#2. STANDARDS-BASED: SIP is the technology that will finally break the shackles of proprietary hardware equipment. For so long telecommunication equipment manufacturers have held their customers hostage to overpriced and limited-functionality telephones and other devices. SIP is the great equalizer, enabling various manufacturers' equipment to interoperate, allowing enterprises to choose the best equipment for their needs and budget. Now vendors can compete on performance and price in an open environment based on SIP.

#3. A MULTITUDE OF DEVICES: With SIP, PDA's, Instant Messaging, desktop phones, and dual-mode cell phones can all communicate to one another.

#4. RICH APPLICATIONS: SIP allows application developers to build solutions that will work on all phone systems, thus lowering the costs of applications dramatically.

#5. VOIP CALLING: SIP when used in place of the PSTN, will provide large cost savings in monthly phone bills and powerful new features unavailable from PSTN carriers.

#6. MICROSOFT: Microsoft is SIP-enabling its products so that your computer will be a SIP device, just like your phone.

#7. SECURITY: Generic Voice over IP has weak or no security standards. SIP provides a common security standard for providers and manufacturers.

SIP DEVICES

- IP Phones
- Laptops
- PC's
- Cell Phones
- IM Client
- Video Device
- PDA

Teledynamic Communications, Inc. · 3501 Breakwater Avenue · Hayward, CA 94545 · 510.785.2480 · info@teledynamic.com · www.teledynamic.com

IP PBX MANUFACTURER SIP IMPLEMENTATIONS

PRODUCT	SIP IMPLEMENTATION
Avaya IP Office	None
Nortel BCM 400	No station SIP, beta trunk SIP
Cisco Call Manager	SIP trunking and limited station support
Cisco CallManager Express	SIP trunking only
Mitel 3300	SIP trunking only
ShoreTel	SIP trunking only
3Com	SIP trunking, stations and applications

Out of the leading manufacturers, only 3Com and Cisco have made firm commitments to fully support SIP. The other manufacturers do not want to give up the profits on their expensive phones!

TeleDynamic COMMUNICATIONS

TOP 10 REASONS FOR GOING SIP

PRESENCE- BASED COMMUNICATIONS: SIP adds intelligence to communications by enabling users, as well as applications, to intelligently connect parties based on their presence (availability). SIP also has the ability to support intelligent forking — that is, the ability to route communications to the right person, using the right medium (voice, video, IM), on the right device, and at the right time.

PREFERENCE-BASED COMMUNICATIONS: Like SIP presence, SIP adds intelligence to communications through giving users control over the parameters of their communications (such as time of day, preferred medium, preferred callers, and so on). This concept is best exemplified through SIP's ability to enable you to control who contacts you on what devices.

NATIVE MOBILITY: As more devices become SIP-capable, users will be able to pick up and go at will, but still communicate as if they were in their office. Their presence and readiness to communicate will still be visible to their associates. For example, SIP's awareness of a user's communication capabilities will aid international travelers who have to use different cell phones and other messaging devices and protocols in different countries. A caller trying to locate such a traveler need not know the traveler's availability or location. SIP by nature will know how a person can be reached, and facilitate the connection.

UNIFIED ADDRESSING: A single SIP AOR (address of record) provides a unifying identifier that can be used for routing all communication to a user. Simply put, an AOR allows for a single user identity to be mapped across multiple devices so that people connect with people, without needing to know which devices they have and are presently using. This address eliminates the need for tracking users' multiple phone numbers, e-mail addresses, and IM contact names.

OPERATIONAL COST SAVINGS: SIP trunks are IP trunks from service providers that use SIP for call control and routing, enabling enterprises to create a single, pure IP connection to carrier clouds. Voice traverses the network just like other IP applications. SIP trunks reduce operational costs by enabling the enterprise to eliminate hardware, software, and recurring network charges associated with using traditional PSTN trunks for voice communications.

AN OPEN STANDARD: The SIP standard is defined in RFC 3261 by the Internet Engineering Task Force (IETF). The IETF is a large open international community of network designers, operators, vendors, and researchers who are all concerned with the evolution of the Internet architecture and the smooth operation of the Internet.

INTEROPERABILITY: Several working groups, including SIPit, SIP Foundry, and SIP Forum, arrange events where companies with SIP hardware and software products can test interoperability with other SIP products. This process helps to promote smoother integration of SIP products in enterprise networks. SIP's ability to work across a range of systems helps enterprises enjoy more seamless integrations between platforms, devices, and applications, so your company can get more done with less custom programming.

SIMPLIFIED COMMUNICATIONS ARCHITECTURE: At the foundation of SIP's philosophy is the concept that intelligence should reside in the endpoint. This concept is evident in SIP's native ability to support peer-to-peer communications. Peer-to-peer environments don't rely on communications servers, gateways, or other intermediate devices to support communications between users. Peer-to-peer SIP networks are easy to set up and administer yet can include features such as automated attendant, voicemail, and multi-party conferencing.

CREATION OF NEW SERVICES: SIP is a structured, text-based protocol that is modeled after HTTP, or HyperText Transport Protocol, the language that powers the World Wide Web. SIP opens the door to a much larger developer community than traditional CTI, and so offers your company the potential to create competitive advantages with intelligent communications. Because SIP is based on HTTP, application developers and system engineers will have an easier time developing and integrating applications into their communications environments.

EASE OF IMPLEMENTATION AND SUPPORT: Because SIP is modeled after HTTP as a text-based language, it is easy to learn, troubleshoot, and support. From analyzing network packets to application code, SIP's structured language stands out so that IT people can easily understand and interpret it.