MITEL

Examining the Value of SIP in the Enterprise

White Paper

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White Paper

Background

After a few years as the subject of intense debate and research and development work in both the Internet and telecommunications communities, Session Initiation Protocol (SIP) is now emerging in a commercial context in both public and enterprise networks. But what exactly is it, what applications will it support, and perhaps more importantly in today's context, what will it mean for the future revenues of the communications industry?

It's rare for a communications protocol to receive the wide attention that SIP generates. For most of its history, the telecommunications industry has protected its users from the underlying complexity of communication systems. The complex signalling systems that deliver advanced voice services such as conferencing and ring back, or that support more advanced enterprise applications such as contact centers, have largely remained a specialist domain.

As the communication industry moved to exploit the possibilities of next-generation, all-packet fixed and wireless communications networks – and to tie these directly in with the applications that are starting to appear on our desktops and mobile devices, such as web, multimedia, chat, presence, location and messaging services – it became clear to that a new approach was needed to globally unite these disparate applications and devices that support them. The result of research and development into the matter – already dubbed by some as potentially capable of doing for next generation networks what HTML did for the Internet – is SIP. SIP, a voice-over-IP (VoIP) networking protocol, can be understood as a sophisticated, IP-based equivalent of DTMF (a signalling protocol commonly known as ‘touch-tone’ dialling), which has supported and enabled numerous advanced features in the public telephone network of today. SIP is positioned to become as ubiquitous in today’s packet networks as DTMF is on the traditional PSTN, playing a similar role as enabler behind many advanced services.

Widespread Value

Like many aspects of telecommunications, perceptions of SIP's unfolding role largely depend on where you're standing in the industry. For the voice-over-IP sectors – both public and enterprise – it represents an opportunity to add value to the basic low cost connectivity benefits of the technology, growing functions, features and entirely new and easier ways of generating interactions with other services.

For communication service providers SIP bridges the circuit and packet infrastructure divide. It has can open up the field of telecommunications service creation to a new community of developers who won't need specialist industry knowledge to enable new services. And that translates into new revenue streams and the ability for a service provider to protect its position in an increasingly competitive marketplace.

For the end user – in both business and private life – SIP holds the promise of bringing together at the many different communications services and devices that we currently use, delivering appropriate services to the appropriate device and seamlessly integrating different media into an easily manageable, consistent and coherent whole.

Recognition of its strategic role was recently highlighted in a March 2002 report from Analysys (), which predicted European revenues from SIP-based services, particularly on mobile networks, rising from zero in 2002 to €2.9 billion in 2007.

"We believe SIP-based instant messaging on the fixed Internet could take off fast and create a significant market for SMS-SIP gateways, which allow seamless messaging across fixed and mobile networks," said Margaret Hopkins, author of the new report. "SIP is a disruptive technology that has the potential to fundamentally change the way telecoms services are delivered - great news for mobile operators, ASPs and ISPs."

Growing Adoption

SIP's potential has been recognised by Microsoft, which now bundles a SIP client with its desktop and PDA software; by the 3G mobile community, which has placed SIP at the heart of its network architecture; and by a number of service providers that are currently offering SIP-based services to business and residential customers. Examples of the latter include BT's recently announced PC2UK service and Sonera in Finland.

In the field of business communications, SIP-based systems are now entering the market. Mitel Networks, for example, has embraced the technology and is shipping the 5055 SIP Phone while also SIP-enabling its full suite of IP phones. In addition, the company introduced the Mitel Networks 3050 Integrated Communications Platform (ICP) in fall 2002. This native SIP-based system for small businesses and networked offices will be interoperable with all of the SIP-enabled services currently being developed by organisations such as Microsoft and the 3G wireless community's 3GPP standards body.

In an increasingly connected world, the right technological tools can create 'joined up' services, integrating end-user communications, enabling new revenue streams, and simplifying service delivery. SIPtoday appears to be the right protocol in the right place at the right time for the next stage of the communication industry's growth.

SIP in Action

SIP is currently well placed to carry out two extremely important roles:

1. Strategically to support the increased interworking and convergence occurring in public and enterprise networks, fixed and wireless networks; and, to leverage the web – rather than traditional public telephone networks – to increase system functionality and ease of use without corresponding overhead in either bandwidth or complexity.

2. Tactically to support the introduction of new messaging and conferencing services, delivering information on a user's presence and location into these applications, and increasing the interplay between the different media streams of voice, data and multimedia.

To understand the implications of SIP in action, imagine being able to:

- Be reached anywhere, through just one IP address
- Have the freedom to change communication parameters on-the-fly and even automatically, adjusting audio bandwidth, security policies and so on according to the particular needs of the communication event or to fit particular cost models.
- Use a publicly available 'buddy list', like that currently in use by proprietary messaging systems such as AOL’s Instant Messenger, to check availability of contacts and make appointments with doctors, dentists, plumbers and so on.
- Use buddy lists in a business context to check for the availability of friends, colleagues, or even the appropriate staff in a call centre – and to be able to see which media is most appropriate to at the time: voice, instant messaging (IM), e-mail, or multimedia.
- To share and enhance informal awareness of colleagues’ availability and activities, so supporting joint activities such as project teams.
- Link buddy list availability with presence and location information to get automatic alerts if friends or colleagues are in the same area, such as at a restaurant, an exhibition or an airport.
- Manage one’s personal availability, using the system to prioritize calls based on current activities and caller identities.
- Use presence information to set up conference sessions automatically when all the required display as available on the network.

- Link meeting rooms together across the Internet to enable ad hoc collaboration, using both installed conferencing equipment as well as personal devices such as PDAs, mobile phones and laptops to share applications, presentations and other media.

- Set up multi-party conferences instantly across different devices and types of network.

- Transform plain old telephone (POTS) calls into multimedia interactions that can involve messaging, gaming and other applications while still supporting quality-of-service voice traffic.

- Control, preview or order services or content from one device – such as a mobile phone or PDA – and direct them to another , such as a video-over-DSL-enabled PC.

How Does SIP Work?

SIP is often identified as being a more user-, network- and industry-friendly basis for achieving some of the same things that H.323, a specification that defines a set of protocols for video conferencing over the Internet, was intended to solve. While H.323 came from the telecommunications world,SIP originated within the Internet Engineering Task Force (IETF) and, as a result, is much more akin to the web.

Although SIP was originally designed to operate as an end-to-end, device-to-device protocol rather than a network-centric one, the development of a framework to manage events means that SIP is now becoming more like Microsoft's .NET and similar web-based application management services. Applications such as conferencing can be created and made available to the network through SIP, with the SIP event framework allows services to be managed and shared effectively between users. The presence and buddy list applications described above show how this event-based information sharing can increase the effectiveness of applications – and more importantly of the people and businesses using them.

SIP is formally described as a "control protocol for creating, modifying and terminating sessions with one or more participants, with sessions including multimedia conferences, internet or other IP network telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or via a combination of these. SIP supports session descriptions that allow participants to agree on a set of compatible media types and also supports user mobility by proxying and redirecting requests to the user's current location".

In lay terms, SIP provides a 'velcro' approach to the problem of interlinking different devices, applications, services and networks. That is, it gives users the freedom to interact with the evolving communications 'cloud' – including both circuit and packet switched technologies – simply and in ways appropriate to their particular situation.

In practice, SIP relies on four key elements:

1) SIP User Agents – these are the “end points” where communication is delivered, such as SIP telephones, PCs, PDAs, 2.5/3G wireless handsets, or SIP gateways, which initiate and answer calls and are responsible for most aspects of call features such as transfer, conference and hold.

2) SIP Proxy and Redirect Servers – these reside in the network and provide the necessary infrastructure for name resolution and user location. Proxy servers perform routing functions, directing communication requests to the correct end-point user agent, possibly via a chain of other proxies in the network path. Proxy servers may also perform a number of related functions, such ‘forking’ to attempt to contact multiple user locations simultaneously – or act as platforms for specific applications such as call filtering.

Redirect servers perform a similar function of locating users and redirecting the caller to a different end point rather than doing the routing first hand. While some redirect servers deal specifically with SIP, others – such as location and ENUM servers (directories that translate between IP addresses and telephone numbers) – provide support functions to the primary SIP servers in the network.

3) SIP Registration Service – this provides the means for a particular device or devices to register to use a SIP address. SIP addresses use ‘URIs’ based on the same addressing scheme used in the web and similar in form to an e-mail address – e.g.: SIP:johnsmith@mitel.com. The SIP address provides a single ‘address of record’ for the user that delivers a one number service for all communications applications.

Users can dynamically register the devices through which they may be contacted for all types of applications. As a result, users will no longer have to hand out multiple contact addresses as the system will automatically handle the distribution of all types of calls appropriately through the Proxy and Redirect Servers.

4) SIP Event and Presence Servers – these allow the effective sharing of information about and between users and/or applications.

SIP Deployment Challenges

Despite SIP's benefits and its widespread backing by many different parts of the communications industry, it is important to understand that SIP – as well as wider issues concerning VoIP in general – are still work-in-progress. While SIP enables a wide range of new services using a networking approach that is more manageable than previous circuit-switched or H.323 systems, it has some 'rough' edges.

SIP vendors and developers are currently leveraging areas of outstanding work in order to ensure that SIP lives up the expectations of service providers and enterprise and residential users. Many of these areas are not actually related to SIP directly, but are essential underpinnings that will require resolution as vendors create viable SIP-based devices, applications and services that are useful to the end user and generate revenues.

Some of these key elements that the SIP community is currently addressing include:

- Simplifying network management
- Enabling E911 emergency calling and response
- Addressing issues of power over the LAN
- Ensuring security of SIP-based communication
- Empowering the Desktop

Simplifying Network Management

In a traditional circuit-switched environment, moves, adds and changes (MACs) of end-points on the network perpetually plague telecom managers. Simply adding a new service to the network, or moving an employee from one line to another, is an onerous task that consumes network managers' time.

In a SIP environment, however, MACs are not much of a concern at all. SIP devices on the network all carry an IP address. So, to move a user from one part of an office building to another, a user simply plugs his or her phone into the network connection at the new office location in the same way one would plug in a laptop. Via the SIP Registrar, the device then tells network services where it is by providing its IP address.

Adding a new user to the network involves simply issuing them a SIP address (SIP URI) similar to the issue of an e-mail address. The only difference is that the SIP phone is also related to a telephone extension and is accessible from the PSTN. Making changes to current user profiles would either mean changing a user's SIP address or the telephone number associated with the SIP address, depending on the situation.

As SIP devices are inherently mobile, there is no move process in the traditional sense, and generally no need for a technician to visit a desk or update a device. Thus, the MAC process under SIP represents a clearer and easier approach that was the case in traditional networking. However, to ensure that SIP makes management as centralized and "remote control" as possible for enterprises and service providers, the phones themselves must incorporate a great deal of management capability.

SIP end-user devices incorporate a large amount of on-board intelligence that lets them autonomously work with network-based SIP systems to carry out processes such as system upgrades, applications upgrades, policy changes etc. SIP phones need to function as intelligent end devices in the same way network client software does on an end user PCs.

Enabling E911 emergency Calling and Response

One of the toughest obstacles on the path toward wide-spread SIP deployment is accommodating the demands of Enhanced 211 (Europe) and 911 (North America) services, which allow emergency services to instantly identify the location of a caller from a fixed line phone. While 211/911 services allow users to quickly request service from emergency personnel, Enhanced 211/911 (or E911) goes a step further by using automatic number identification (ANI) to relate location information to that number and determine which public service answering point (PSAP) should handle the call.

In an enterprise setting, it is not only important for E911 to direct emergency services to a particular building, but also to indicate from where in the building the 911 call originated. Typically, when someone dials 911 from a desk, the ANI information that reaches the PSAP is based on the company's general, seven-digit phone number associated with the PBX, not the user's extension. In a life-and-death circumstance, that vague information can create harmful delays. As a result, some local and state governments make incorporation of E911 services a legal obligation for network equipment providers. To that end, many private phone system vendors have incorporated methods for private branch exchanges (PBX) to fulfil E911 requirements.

The most direct method is to relate circuit terminations to extension numbers. In a traditional PBX environment, each phone jack is associated with a particular extension. When a 911 call is placed from a particular location, the PBX sends a seven-digit number associated with that extension to a central office router, which can then use ANI to route the call to the correct PSAP. The PSAP can then furnish emergency personnel with that special number's location information.

However, in a packet network, the circuit switched world's E911 fix doesn't apply to SIP (or any other VoIP protocol). Since SIP phones and other VoIP devices become ‘floating’ network appliances that can attach to the network from any point, their location isn't static. Just like a laptop, a SIP phone can connect anywhere on a company's network – all it needs is a valid IP address. If the phone can be anywhere, then how can a company's phone system serve the E911 network with the necessary information?

Several potential solutions to these issues have been presented by the SIP community:

- Install global positioning system (GPS) chipsets in SIP end-user devices that can provide geographic location information.
- Have users "log on" to their devices when moving from one location to another.
- If a SIP deployment makes use of pre-existing voice cabling to create a second, "VoIP only" network. The IP-PBX can then relate circuit terminations to direct inward dialing (DID) lines in much the same way as a circuit-switched PBX.

Whatever the final solution, "fixes" to the E911 issue will have to be standardized across SIP products from all vendors, as the E911 space is regulated.

Addressing Issues of Power Over the LAN

Current public and private telephony systems provide their own power, delivered over phone lines. This ensures that if the power for a neighbourhood or office building fails, users will still have the power required to operate their phones. However, the same cannot be said for devices connected to an enterprise data network. While a user may be able to plug a traditional telephone handset into a wall jack, pick up the handset and hear a dial tone, he or she must plug a computer into not only the company's LAN, but a power supply as well. This is, therefore, also true for SIP and other VoIP phones. Current devices require that users plug their phones into power sources as well as the network. Adding to the ever-growing tangle of cords under most enterprise users is not ideal.

Methods are therefore under development to provide power over the LAN so that SIP and other VoIP phones get power and network services from the same infrastructure – in the same way as traditional desktop telephones. These initiatives include:

- The Institute of Electrical and Electronic Engineers (IEEE) 802.3af task force is developing methods to send power in-band through the Ethernet media dependent interface (MDI) to equipment on the network.

- Cisco Delivery Protocol (CDP) has been proposed as a potential solution using free power-down-the-LAN technology to let switches detect network devices that need power and send it to them across the network.

Inevitably, the enterprises deploying SIP technologies will decide which "power down the LAN" system works right for their businesses. The key point is that standards bodies and vendors are responding to power needs, and that the buyers of SIP equipment will have choice.

Ensuring Security of SIP-Based Communication

Security is a major concern for voice and data communication of any type. Any any network and VoIP transmission – including SIP exchanges – are open to attack if they are being carried on an open LAN. Security concerns range from unwanted parties packet-sniffing a user's voice communication and then listening in on a corporate network, through to toll fraud and other theft-of-service attacks on a service provider's network. Additionally, denial of service (DoS) attacks pose a considerable threat to VoIP communications, especially for service providers.

Nevertheless, the picture for SIP and other VoIP technologies is positive. Within the SIP community, many security issues, such as "forking" and "reflection attacks" can be addressed by PGP encryption. However, while pretty good privacy (PGP) presents a low-overhead approach to security, it is limited in what it encrypts and relies on public key infrastructure (PKI).

The SIP technology community is therefore striving to push SIP security to the next level. Approaches include encrypting the SIP media stream, or hiding the identities of calling parties by encrypting their SIP exchange on a "hop by hop" basis.

Empowering the Desktop

Enterprises will need a range of high-quality, robust SIP telephony devices – desk sets, operator stations, conferencing units and more – that can live up to the day-to-day demands of business users. The key question is what makes for a reliable phone that delivers business-quality voice,regardless of whether it is a traditional phone or an IP phone? Some necessary features include:

- Echo cancellation.
- Acoustic echo suppression in handsets.
- Liquid crystal displays (LCD) that are clear enough and properly angled so users can read them.
- Buttons and switch-hooks that stand up to daily punishment.
- Ergonomics that increase telephone usability and comfort.
- Conformance with audio performance, safety, electromagnetic interference and other standards

While these features may seem mundane and somewhat obvious, they are not easily arrived at. Business telephone vendors have spent years perfecting their business appliances to deliver the quality of services, usability and reliability that users expect from their telephones. Indeed, most business users would be surprised if their phone wore out from continual use.

Furthermore, businesses expect these capabilities to apply to a broad range of telephony devices. Quality doesn't stop at the telephone, and users expect the same capabilities and reliability from conference stations, operator stations, etc.
However, to date the main share of SIP devices manufactured have been desktop phones. Most of these have been aimed at the high end of the market and are very expensive for mass use within an organisation. This one-size-fits-all approach could hamper the adoption of SIP into the enterprise. The responsibility is therefore upon SIP vendors to ensure that they produce a variety of SIP devices that suits all end user needs, while simultaneously providing dependable, business-quality telephony.

Many vendors have spent years perfecting their end-user telephony devices and a strong background in design and production, paired with expertise in networking, will be crucial for technology vendors to live up to this key SIP challenge.

In North America contact:
Mitel Networks
350 Leggett Drive
Ottawa, Ontario
K2K 2W7 Canada
(613) 592 2122
1 800 648 3579

In Europe, contact:
Mitel Networks Ltd.
Mitel Business Park
Portskewett, Caldicot
NP26 5YR UK
Sales: 0870 9093030
General: 0870 9092020
Int: +44 (0) 1291 430 000

In Latin America, contact:
Mitel Networks
350 Leggett Drive
Ottawa, Ontario
K2K 2W7 Canada
(613) 592 2122
1 800 648 3579

In Asia-Pacific, contact:
Mitel Networks Asia-Pacific Ltd.
88 Hing Fat Street,
2002-20 Floor
Causeway Bay, Hong Kong
China
Tel: +852 2508 9780
Fax: +852 2508 9232

www.mitel.com

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