MITEL

Examining the Value of SIP in the Enterprise

White Paper

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White Paper

Background

After years of intense debate and research in both the Internet and telecommunications communities, Session Initiation Protocol (SIP) is now starting to emerge in both public and enterprise networks. But what exactly is it, what applications will it support, and perhaps more importantly what will it mean for the future growth of the communications industry?

It's rare for a communications protocol to receive the wide attention that SIP is generating. For most of its history, telecommunications has kept its underlying complexity safely hidden from its users – in direct contrast to the approach taken by IT. The complex signalling systems that deliver advanced voice services such as conferencing and redial, or that support sophisticated enterprise applications like contact centers, have largely remained a specialist domain. As an important side note, telecommunications was also the industry that invented the concept of 99.999 per cent availability.

As the industry begins to deploy next generation packet-based networks coupled with advanced user applications it becomes clear that a new approach is necessary. The result is SIP, an open IP standard capable of doing for next generation networks what HTTP did for the Internet. Alternatively, SIP can be seen as an IP-oriented equivalent of DTMF (commonly known as 'touch-tone' dialling), which has supported numerous advanced features in the public telephone network of today. SIP will eventually become as ubiquitous in these new networks as DTMF is on the traditional Public Switched Telephone Network (PSTN), playing a similar role as a basic enabler behind many advanced services.

Like many aspects of telecommunications, perceptions of SIP's future role differ across different segments of the market. For the IP telephony sector – both enterprise and service provider – it represents an opportunity to extend new value through simplified integration with other applications and services.

For communications service providers it offers a chance to bridge the circuit and packet infrastructure divide. It has the potential to open up the field of telecommunications service creation to a new community of developers who won't need specialist industry knowledge. For stakeholders in those service providers, it will enable the generation of new revenue streams and protect their position in an increasingly competitive marketplace.

For the end user – in both business and private life – SIP holds the promise of bringing together disparate communications services and user devices to seamlessly integrate across multiple media. Recognition of its strategic role was recently highlighted in a March 2002 report from Analysis (www.analysys.com), which predicted European revenues from SIP-based services, particularly on mobile networks, rising from zero in 2002 to $3 billion US in 2007.

The benefits of SIP have not gone unnoticed in the software industry with market leaders like Microsoft and AOL announcing support for the standard. The 3G mobile community has also embraced the benefits of SIP, putting it at the heart of their network architecture. Service providers are also rolling numerous 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 enterprise communications, we are already starting to see SIP-based systems enter the market. Mitel Networks, for example, has already embraced the technology and is shipping the 5055 SIP Phone while SIP-enabling its full suite of IP phones. In addition, the company has introduced the Mitel Networks 3050 Integrated Communications Platform (ICP) in fall 2002. This is a native SIP-based system for small businesses and networked offices that will be interoperable with all SIP-enabled services currently being developed by organizations such as Microsoft and the 3G community's 3GPP standards body.

In an increasingly connected world, we need the proper technological tools to be able to easily create integrated services. SIP, to date, gives every indication of being the right protocol, in the right place at the right time for the next stage of industry growth and adoption.

Real World Deployment of SIP

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

Strategically – Supports the increased interoperability and convergence of public and enterprise, wired and wireless networks, applying Web concepts – rather than traditional public telephony approaches – to add more functionality and ease of use, without corresponding overhead in bandwidth or complexity.

Tactically – Supports the introduction of new messaging and conference services, linking information on a user's presence and location into these applications, and increasing the interoperability between the different media streams of voice, data and multimedia.

In the SIP world users are able to:

- Be reachable anywhere, through just one address
- Have the freedom to change communications parameters on the fly and even automatically, adjusting audio bandwidth, security policies and so on according to particular user needs
- Use a publicly available 'buddy list', like that currently in use by proprietary messaging systems such as AOL Instant Messenger, to check availability and make appointments with doctors, dentists, hair stylists 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 be able to see which media is most appropriate at the time – voice, Instant Messaging (IM), email, or multimedia.
- 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 club, an exhibition or an airport.
- Manage one's personal availability, with the system prioritizing calls based on current activities and caller identities.
- Use presence information to set up conference sessions automatically when all the required users show themselves 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.

How does SIP work?

SIP is often defined as a more user and network friendly way of achieving the goals that H.323 was intended to solve. H.323 came from the telecommunications world and betrayed those origins in terms of its highly defined breadth and rigour, as well as the message overhead that it placed on networks. SIP originated within the Internet Engineering Task Force and, as a result, is much more web-centric in nature.

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 .NET and similar web-based services. Applications such as conferencing can be created and made available to the network through SIP, with the SIP event framework allowing services to be managed and shared effectively between users. The presence and buddy list applications outlined above show how this event-based information sharing can increase the effectiveness of applications.

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. Members in a session can communicate via multicast or via a mesh of unicast relations. 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 more user-friendly terms, SIP provides a velcro approach to the problem of interoperability for disparate devices and applications, giving users the freedom to interact with communications in both circuit and packet-switched technologies.

In practice, SIP relies on four key elements:

SIP User Agents – these are the end points, residing in desktop SIP telephones, PCs, PDAs, 2.5/3G wireless handsets, or SIP gateways, which initiate and answer calls and are responsible for call features such as transfer, conference and hold.

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 requests to the correct end point user agent, possibly via a chain of other proxies in the 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 user location, redirecting the calling side to a different end point rather than doing the routing first hand. While some of these servers deal specifically with SIP, others – such as location and ENUM servers (effectively directories that translate between IP addresses and telephone numbers) provide support functions to the primary SIP servers in the network.

SIP Registration Service – this provides a means for a particular device to register to use a SIP address. SIP addresses use 'URLs' based on the same addressing scheme used in the web and similar in form to an email 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, people 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.

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

SIP Deployment Issues

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 IP telephony as a whole – is only now beginning to reach maturity. While SIP enables a wide range of new services using a networking approach that is more manageable than previous circuit-switched systems, it still has some issues to resolve. There are areas that SIP vendors and developers are currently addressing in order to ensure that SIP lives up to expectations of service providers and enterprise users. Many of these areas are not actually related to SIP directly, but are underpinning issues that will require resolution in the creation of viable SIP-based devices, applications and services.

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

- Improved network management
- e911
- Power over Ethernet
- Security

1. Improved Network Management

In a traditional, circuit-switched environment, moves, adds and changes (MACs) afflict telecom managers on a regular basis. Simply adding a new service to the network, or moving an employee from one line to another is an onerous task that most network managers would rather avoid.

In a SIP environment however, MACs are not a major concern. SIP devices on the network all carry an IP address. To move a user from one part of an office building to another, they simply plug their phone into the network connection at their new office in the same way they would plug in their laptop. The device will then tell network services via the SIP Registrar where it is by the IP address.

Adding a new user to the network simply involves issuing them with a SIP address (SIP URL) in a similar way to issuing an email address. The only difference is that the SIP phone they now use is also related to a telephone extension and is accessible from the PSTN. Changes for current users would either mean changing a user's SIP address or the telephone number associated with their SIP address, depending on the scenario. 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. Needless to say, the Move-Add-Change process under SIP represents a clearer and easier approach. However, in order to truly ensure that SIP makes management as centralized and "remote control" as possible for enterprise 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 to let them autonomously work with network-based SIP systems to carry out processes such as system upgrades, applications upgrades, policy changes etc. SIP phones will need to function as intelligent end devices in the same way network client software does on an end user PCs.

2. E911

One of the toughest obstacles in SIP's path towards wide scale deployment is accommodating the demands of Enhanced 211 (Europe) and 911 (North America) services which allow the 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) go 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 indicate from where in the building the 911 call originated. Typically, when someone dials 911 from their 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 incorrect information can create harmful delays. As a result, some local and state governments make incorporation of E911 services a legal obligation. To that end, many private phone system vendors have incorporated ways for PBXs to fulfil these E911 requirements.

The most straightforward way is to relate circuit terminations to an extension number. 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 packetized network, the circuit switched world's E911 fix doesn't apply to SIP (or any other VoIP protocol for that matter). 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 arisen from within the SIP community:

- Install GPS chipsets in SIP end user devices that can provide geographic location information.
- Simply have users "log on" to their devices when moving them 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 relate circuits terminations to DID lines in much the same way as a circuit-switched PBX.

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

3. Power over the LAN

Current public and private telephony systems provide their own power, delivered over the phone lines. This ensures that if the power for a neighbourhood or office building fails, users will still have the power needed 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.

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

- The IEEE's 802.3af task force is developing methods to send power in-band through the Ethernet media dependent interface (MDI) to equipment on the network.

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

4. Security

Security is also a major concern in the IP communications industry with IP phones open to attack if they are being carried on an open LAN. Security concerns range from unwanted parties packet-sniffing a user's voice communications 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 attacks pose a considerable threat to IP communications, especially for service providers.

However, the outlook for SIP, as well as other IP technologies, is positive. Within the SIP community, many security problems, such as “forking” and “reflection attacks” can be addressed by PGP encryption. However, while PGP presents a low-overhead approach to security, it is limited in what it encrypts and relies on public key infrastructure (PKI).

SIP's technology community is therefore striving to push 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. Needless to say, securing SIP – and all IP communications – is a work in progress, with no fixes currently standardized.

5. The Devices

Enterprises will need a range of high-quality, robust SIP telephony devices 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's a traditional phone or an IP phone? Some features include:

- Echo cancellation.
- Acoustic echo suppression in handsets.
- LCD screens
- 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 and reliability that enterprise users expect from their telephones.

Furthermore, businesses expect these capabilities to apply to a broad range of telephony devices. Quality doesn't stop at the desktop phone and users expect the same capabilities and reliability from conference stations, operator consoles, etc.

However, to date the main share of SIP devices manufactured have been desktop phones and most of these have been aimed at the high end of the market and are cost-prohibitive for mass use within an organization. This one-size-fits-all approach could seriously impede SIP's early adoption 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.

However, developing a broad SIP product line suitable for enterprise needs is not straightforward. 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.

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