# SIP: The Global Standard that Enables Businesses to Leverage the Power of the Internet

**Session Initiation Protocol (SIP) Primer**

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## Introduction

Traditionally, businesses around the world exchanged electronic information in two ways: speaking via telephones, and exchanging data via communications networks. Voice and data system architectures were radically different, with voice offering two-way, realtime communications, and with data using a store-and-forward approach. The emergence of the Internet as a communications infrastructure provides a new architecture that offers the best of both worlds, and lays the foundation for simpler, more powerful, and less costly integration of voice, data and multimedia communications using common infrastructures. With the emergence of wireless infrastructure and the rapid pace of change in business, relocation and mobility are also key aspects of today's reality.

Initially a text-based medium, the Internet has moved to become a pervasive medium that has fundamentally changed our lives. HTTP set the stage for the World Wide Web by establishing how computers accessed and displayed Web pages stored on central servers. More recently, the focus has been on using Internet infrastructure to integrate voice, using Voice over Internet Protocol (VoIP).

VoIP strives to deliver the full-range of applications wanted by end users and service providers while increasing manageability and simplifying fundamental design to ease development and deployment. One protocol that has fit that bill is Session Initiation Protocol (SIP).

SIP is recognized as an excellent way to achieve fully integrated communications over Internet infrastructure. It promises to have the same impact as HTTP on how we communicate in realtime: on mobile or standard phones, using computers, Personal Digital Assistants (PDAs) or over any type of IP-based device. Not only does SIP enable VoIP communications, it can also be used for Instant Messaging and multimedia conferencing.

Many vendors and service providers have lined up behind SIP. Like VoIP in general, SIP is evolving and the technology community is working to ensure that as VoIP services and networks increase their deployment, SIP will reside at the core of that deployment.

SIP signals a new era of increased capability and flexibility for businesses around the world, offering a host of benefits and setting the stage for the emergence of even more capabilities in the future. This document provides an overview of the business and technical aspects of SIP, with examples of how SIP will enable a whole new generation of business services and advantages.

## What is SIP?

Session Initiation Protocol, or SIP, is a peer-to-peer, standards-based protocol that facilitates openness, connectivity, simplicity, choice and personalization. It is a standard that is being advanced by the Internet Engineering Task Force (IETF), the global non-profit Internet standards body behind Internet Protocol (IP) and HTTP. The IETF began standardization of SIP in 1996 to support multicast applications. Its simplicity, power and extensibility, however, led to SIP's rapid adoption for other uses across the IETF, notably for use with Voice Over Internet Protocol (VoIP), and for Instant Messaging.

Communications service providers saw VoIP as a way to merge their voice and data networks, reducing costs and opening up prospects for new and powerful applications. SIP offers a new degree of scalability and interoperability, and provides an easier way to build new services than earlier VoIP protocols.

From its inception, SIP was modeled closely after HTTP. Like HTTP, it was designed to work over IP networks. Also like HTTP, it significantly lowered the barriers to developing and deploying rich, innovative services by moving control of applications to the endpoints such as telephone handsets, mobile devices, and personal computers.

One of the most powerful concepts of the Internet is the fact that applications can operate between a web server and a browser with no dependence on the underlying IP network. The same is true for SIP-based sessions (a session begins when you connect with other parties and ends when connections are terminated). A SIP server (such as a Mitel Networks 3050 Integrated Communications Platform) and client (such as a Mitel Networks 5055 SIP phone) have complete control over their session (voice, video, conferencing, Instant Messaging, etc.). This is in direct contrast to the model for service control in the traditional circuit-switched telecom world, where endpoints like phones lack call control capabilities and all services are controlled by a central switching element. This delivers maximum flexibility to the business owner and system users.

SIP is being adopted by major telecommunications service providers around the world. It has also been specified as the call control for the 3GPP next generation cellular network has been adopted by Microsoft™ for use in with Xbox™ and Windows Messenger™, by AOL and many others for Instant Messaging.

## Business Implications of SIP

The implications of SIP to businesses of all sizes is significant. Today's business environment increasingly relies on effective communication between people and locations. In addition, pressure to reduce operating costs while innovating and increasing productivity has never been more intense. Add the desire for all businesses to offer better customer service, and the benefits of using the power of the Internet become all too clear.

Businesses are looking for simple ways to achieve all of these things, and employees at all levels of the organization contribute best when technology works behind the scenes to make their jobs easier. From the customer to employees to the supply chain, the smooth and timely flow of information is critical for success. Further, new multimedia capabilities allow your business to create and nurture its own communities that keep customers coming back, leverage supply channels, and contribute to growth over time.

By moving from older paradigms that have limited potential, SIP unlocks a whole new generation of business processes and potential, and delivers a host of benefits to your business today, with even more to come in the future. And SIP's inherent interoperability means that you have a full range of vendor choices now, and in the future.

## SIP's Business Benefits

SIP offers one of the first truly converged systems for business communications. In addition to a growing number of emerging capabilities, businesses which adopt SIP will enjoy a host of immediate benefits, including:

- Open standard, with multiple vendors introducing many new features, products and services that all work together, so you have lots of freedom and choice
- Simplified communications network, with one set of cables to handle voice, data and video to every desk
- Easy set-up, so you don't have to be a technical expert to install or modify your SIP-based system and preferences
- Maximum flexibility, by allowing multiple dialing methods, choice of devices, use of wireless devices, and ultra-flexible configurations
- Supports remote workers, by extending your company's network to them, wherever they are
- Scalable, making it easy to build on your systems as your company grows
- Futureproof, protecting your original investment by adding new services and capabilities via downloadable software, and by using open system architectures that are compatible with equipment from multiple vendors
- Reduced system costs, by having one system handle voice and data, meaning you have fewer components and one supplier instead of many
- Reduced long-distance bills, by placing long-distance calls and conference sessions over the Internet infrastructure instead of traditional communications infrastructures
- Potential for new competitive business advantages as new SIP-based services are offered over time
- Consistency across distributed organizations, by having the same dialing and access procedures to reach branch offices, home offices, teleworkers, and mobile employees
- Fast ramp-up, when adding new branch offices or new employees, or when reassigning or relocating staff
- Strengthens corporate culture inside distributed organizations, by dialing internally across distances, such as dialing by extension numbers – it feels like everyone is truly "inside" the organization
- Ability to dial locally, by connecting to regular public telephone systems

For communications service providers, SIP also offers a number of important benefits such as easy development and deployment of new services, faster response times for customer service, reduced need to send technicians to customer sites (saving time and money), and reduced network maintenance costs. For many service providers, SIP allows traditional voice suppliers to support data services, while data-based providers can enter into telephony areas, adding new revenue opportunities.

### Enhanced functionality via applications

The SIP model is that of the intelligent edge device. The SIP phone will become the desktop portal, whereby future applications and services can easily be delivered across the enterprise, enhancing the user experience. Facilities like Presence and Instant Messaging will be the basis for a new breed of pervasive connectivity solutions targeted at vertical market segments. With this in mind, many vendors and service providers are readying advanced new applications and services to take advantage of this emerging technology.

## Mitel is Committed to SIP

Mitel Networks recognizes the promise and potential of SIP, the emerging protocol of choice for setting up telephony, multimedia, conferencing, and other kinds of advanced communications sessions via the Internet. Our portfolio of IP-based phones and peripherals, integrated communications platforms and IP-based applications is already among the most advanced and comprehensive on the market. And we're committed to leveraging our leadership in VoIP to provide service providers and their customers with an equally sophisticated and complete SIP desktop portfolio to support the next generation of converged broadband services.

**A comprehensive SIP desktop portfolio**
The Mitel Networks SIP desktop portfolio provides business users with easy, intuitive access to sophisticated SIP-based services without compromising voice quality, reliability, or functionality. Beginning with the Mitel Networks 5055 SIP Phone and 3050 Integrated Communications Platform, and including both 5305 and 5310 conference units, our SIP desktop portfolio will increasingly extend to serve the different needs of users across your entire enterprise – from affordable, single line models for occasional users and multi-line browser-equipped Webphones for communications-intensive users, to powerful networking capabilities for networks of small offices and available wireless units, attendant devices, and more.

**The answer for accessing integrated SIP services**
Mitel Networks SIP phones, Integrated Communications Platforms, devices and peripherals open the doors to new and emerging SIP-based services available from major service providers – offerings such as Internet voice services, visual unified messaging, phone-based Web browsing, collaborative working services, speech recognition, location-based services, and more. As addressable devices on the IP network, SIP-based devices can download new applications right over the Internet, and can be remotely managed by you, your staff, your company's technical specialists, or by your service provider – whichever is easiest for you. SIP devices feature easy user-preference programming via the phone or PC, and integrate with PC, PDA and wireless applications and services.

**The successful delivery of SIP**
Making telephones – whether analog, digital, IP or SIP – isn't something that just any company can perfect overnight. Mitel Networks has a 30-year history of building high-quality voice-based products from the PBX switch to the desktop handset. User ergonomics and interface design are key aspects for ease of use and functionality, while choice of price point and degree of functionality are important business considerations. In addition, Digital Signal Processing (DSP) technology, echo cancellation, acoustic echo suppression in handsets, hands-free capabilities – all essential to meeting user voice quality expectations – are technologies that we have spent years refining and perfecting.

Mitel Networks is proud to be an active member of the SIP Forum and have been involved in IETF's SIP working groups for several years, advancing open global standards for IP communications such as SIP.

 

## The Evolution of SIP

SIP originated in the mid-'90s as a simple method for inviting people to view multicast sessions such as space shuttle launches over the M-Bone, the Internet's multimedia channel. Because of its simplicity, power and extensibility, SIP was rapidly adopted by international standards organizations and communications service providers for other uses, most notably as a Voice Over Internet Protocol (VoIP) communication standard.

Now, many service providers offer SIP-based communication services and device manufacturers also are offering an increasing number of SIP-based devices and capabilities such as downloadable applications.

Over time, SIP will enable a host of new services and capabilities that will provide easy, personalized communications and excellent cost efficiency. Complete control will be in the hands of each user. You will create your own individual profiles to instruct communications networks on your preferences for how and where you can be reached, by whom, and when. For example, dialing your phone number will reach you wherever you are in the world on whichever means of communication is available to you at the time (cell phone, PDA, PC, tablet, or even a wearable communicator). Different rules can be set for the system to find you inside, or outside office hours, and you will be able to set different options for each person on your buddy list, such as your spouse.

Setting up a conference call will be as simple as programming your SIP device to recognize when all of your selected buddies are available to talk to you, and the call will be dialed automatically. Voice recognition will mean that you don't have to use a keyboard or numeric pad for any of these steps. If you're on a video conference at your desk and have to leave early, you can transfer the video conference to your wireless PDA to continue participating while you are in transit.

Location-based services, such as making dinner reservations while on holidays, will be available, and you can program your profile to alert your London office when your plane lands. Your personal profile tells the hotel what newspaper you normally have – you can change this on the fly and from country to country. You may be renting a car, in which case the PDA slots into the dash and becomes your navigation tool – it knows where you are going and uses local map and travel info to guide you there. When you get into your hotel room you will have a broadband connection via either your 3G handset/PDA or the high-speed connection at the desk. You can choose which form of connectivity to use based on cost and/or convenience. "Presence" is an inherent feature of SIP-based systems, allowing the system to recognize your location and availability, routing communications to the right locations and devices as your schedule changes. And all this will be globally interoperable – you will be able to go anywhere in the world and access all of the services you get at home. You will be in complete control.

Once you have put SIP at the core of your communications network, all these capabilities become possible for you without changing your base equipment – you will simply download new capabilities as software becomes available. Or, your service provider will customize the right package of software to satisfy your needs.

## Technical Overview

SIP is a peer-to-peer signaling protocol that can set up and manage any type of communication session, regardless of the media type (phone call, Instant Messaging, gaming, or even live video). In fact, SIP's power stems from its simplicity and flexibility. SIP is a standard that is being advanced by the Internet Engineering Task Force (IETF).

SIP communication uses the same control, addressing, protocols, security and other mechanisms commonly found on IP networks and on the Web. In addition to voice communication features, SIP enables new services that are very difficult or impossible to provide in traditional telephony-centric systems, such as:

- Presence
- Mobility
- User preferences
- Instant multimedia communications: text, voice/video/data
- Advanced multimedia conferencing
- Multiple media: text, voice, video, shared data
- Multiple devices: phones, PC/laptop, handheld computers, pagers

SIP heralds a new generation of converged communications equipment and services.

**SIP characteristics**

Control over services is moved out to the endpoints
In the traditional telecom environment, centralized switching elements control voice and other services, which significantly increase the time and cost required to build new services. By moving service control out to the endpoints (such as SIP-based mobile phones or PC clients), SIP eliminates the need for a central switching element. Because of this, it promises to bring the low development costs and fast development cycles of Web-based services to realtime communications.

Flexibility
As a signaling protocol, SIP is session and message agnostic. While it can set up any session (voice, video, messaging, games) it carries no predefined rules for what that session should be. While it can transport messages, it supports any MIME (Multipurpose Internet Mail Extensions) type in the messages. This gives SIP the ability to support the broadest set of subscriber services — including the creation of new applications never envisioned by standards groups.

Futureproofing
SIP can easily be extended to support new messages and even new types of services. Like HTTP, SIP's capabilities can be augmented as new requirements emerge. For example, SIP for Instant Messaging and Presence (SIMPLE) is a SIP extension to support interoperable Instant Messaging and Presence systems. SIP is designed to remain interoperable and "backwards compatible." If two SIP endpoints do not jointly support a set of SIP extensions, they can agree to ignore those extensions and use the base protocol for their communication.

Integration with Internet standards
SIP provides full integration with open Internet standards and technologies. It uses URIs (Universal Resource Indicators), DNS (Domain Name Server), and MIME (Multipurpose Internet Mail Extensions) in ways that are compatible with other IP applications. This allows SIP to easily interoperate with Web applications, a critical capability in building compelling mobile services. With SIP, mobile carriers can deploy services that seamlessly integrate voice, Presence, messaging and Instant Messaging with Web interactions, providing the foundation for a virtually limitless set of service possibilities.

Because of its extensive capabilities, SIP has enjoyed phenomenal market success. SIP is being used extensively today to support a broad range of voice, Instant Messaging and Presence-based services over mobile, wireline and IP-based networks.

**SIP IP communications**
IP communication includes Voice over IP, but provides much more than just telephony. An example is using Instant Messaging applications, which can support the following types of communications:

- Presence (showing user availability)
- Text messaging
- Voice
- Video
- Whiteboard sharing
- Desktop application sharing
- File transfer

Similar rich communications can also be enabled on handheld devices, such as those running Windows CE with SIP User Agent. Depending on wireless bandwidth, a range of communication options will be possible from text-based Instant Messaging to full multimedia realtime communication.

**Voice communications**
Normal PBX-style voice communications are provided by SIP phones and devices. SIP-based IP PBX functions can be supported both in SIP end devices and through SIP enterprise servers. SIP phones, interactive voice response (IVR) systems, media and dialog servers, conferencing servers, etc. can support all typical PBX and Centrex functions. While not all of the traditional telephony features are fully standardized yet, most of the commonly used features are in place, and others are progressing rapidly.

An extremely rich portfolio of call features can be supported using a simple control endpoint such as a SIP Application Server or desktop PC acting as a SIP User Agent. Enterprise users can enjoy a rich communications portfolio provided from a desktop or Web-based application, customized to their individual needs. Special communication needs such as for secretaries, receptionists, agent consoles, customer service desks and many others, can be easily accommodated.

The remarkable fact is SIP can support all applications known from Computer Telephony Integration (CTI) and it is the only interface required between a SIP server or desktop computer, SIP Phone devices and the telephone system.

With the introduction of richer SIP-based systems, some PBX-based voice features may change or no longer be necessary. For example, the PBX function of attended call transfer may change radically by the use of Instant Messaging, where the receptionist may consult with employees using text chat before transferring incoming calls.

**Bandwidth requirements and QoS**
Using SIP for voice may save bandwidth on access links to the Internet on services such as frame relay and T1 lines if compressed codecs and header compression is used. Codecs of type G.723.1 and G.729 with header compression can reduce data transmission bandwidth requirements as low as 7-9 kb/s respectively per active voice channel, with acceptable voice quality between enterprise locations. Where commercial

 

voice quality is required under all circumstances, the G.711 codec using roughly 90 kb/s bandwidth per active caller is preferable. Better-than-PSTN voice quality for advanced needs such as business conferences can also be supported using the G.722 codec and other emerging codecs for the Internet, providing close to 8 KHz voice bandwidth, as compared to the 3.1 KHz of the PSTN. SIP is designed to carry out negotiation of the most desirable combination of these codecs between endpoints, on a per-call basis, to optimize use of bandwidth and voice quality based on user needs and enterprise policies.

Management of end-to-end Quality of Service (QoS) for enterprise communications requires the prioritization of voice and possibly video data transmission across enterprise LANs and on access links to the service provider which may offer advanced access technologies that feature both QoS management and header compression for low bandwidth links.

**Enterprise and carrier voice services**

Enterprise voice communication features can be provided entirely within the enterprise, managed internally, or outsourced to an ISP supporting IP-based communications, or a combination of the two approaches. SIP is inherently flexible with regard to where services are provided.

Some services are more appropriately thought of as enterprise-based, although it is also possible to support these as outsourced ISP services, such as:

- Enterprise integrated messaging
- Corporate directory
- Local Presence services
- In-enterprise personal mobility and roaming
- Call routing and filtering, based on personal and corporate preferences and policies
- Corporate vertical applications and access to corporate data/media
- SIP gateways to existing traditional equipment (analog and legacy digital phones, traditional PBX, etc).

Inter-enterprise communications on a global scale and certain network-based services are often more effectively provided by the ISP, such as:

- Global ENUM service for phone number to Internet address conversion
- Global presence on IP, PSTN and mobile networks
- Global roaming
- Global call routing
- Gateways to the global PSTN
- Inter-enterprise conferencing

“Presence” information, which alerts the system when a person is available, can be used for a wide variety of exciting new services, such as:

- Establish an Instant Messaging text chat session
- Make a phone call when the called party becomes available
- Invoke an instant conference when all the desired parties are online
- Get a notification when a cell phone is on the air
- Get a notification if an agent becomes available
- Specific vertical applications, such as monitoring delivery vehicles or employees on the move.

Due to the small GUI footprint and low computational requirements, SIP-based Presence and Instant Messaging can be supported on PC/laptop computers, intelligent SIP phones, palm computers and even mobile phones.

**Mobility and roaming**

Campus: Mobile communications within the enterprise and on a global basis can be fully supported with SIP. Wireless 802.11b LANs, or other wireless IP networking, can support rich communications from the laptop PC, portable phones, hand-held computers, and other mobile appliances. Full conferencing as well as private sidebar Instant Messaging and video communications can potentially be supported during meetings on the campus and between campuses. A combination of mobile IP and SIP mobility can provide mobility at the network level as well as personal mobility when changing devices - moving from PC/laptop to palmtop or SIP phone while maintaining services, such as the dialing plan and directory, and also personal user preferences on how to place calls and how/where to receive calls from what parties at what times and locations.

Telecommuting: Employees working from home, on the road or a telecommuting site can enjoy the full communication features available in the office, depending on access bandwidth to the Internet. Secure access to the enterprise network makes use of VPN and/or SIP-enabled firewall thus the SIP phones at home can have the same dialing plan and other services as in the office.

Dial-up: Travelling employees can use the global Internet access facilities of multinational service providers along with laptop-based VPN clients to access the SIP facilities of their home enterprise network while on the road. Narrow band codecs such as G.723.1 can even support phone calls over dial-up lines from hotel rooms, thus avoiding high international hotel telephone charges. Narrowband Presence and Instant Messaging can also be used over dial-up connections.

**Wireless LAN and 3G mobile systems**

Wireless LAN: Wireless 802.11b and emerging 802.11a "hot points" in locations such as airline lounges, hotels and convention centers equipped with wireless LANs can support the full range of IP communications for roaming employees equipped with VPN and SIP devices. Support of SIP in mobile appliances will enable user access to IP communications using personal devices as well.

3G Mobile Networks: The adoption of SIP by 3GPP mobile operators will allow seamless IP communications in countries where 3G wireless is deployed. 3GPP engineers are participating in the SIP, SIPPING, SIMPLE and other IETF working groups work to maximize the interoperability of IP communications with these emerging 3GPP services. Various mobile devices for 3G networks will feature SIP User Agents for realtime communications and interworking with non-3GPP devices and services.

**Operations and support systems**

Network administrators are now looking at the services provided by new enterprise communication systems, and also considering the associated costs of operations and support systems, such as:

- User management
- Service provisioning
- Server management

 

- SIP phones and PC/laptop User Agent management
- Security
- Traffic reporting
- Trouble management
- Carrier service management

Enterprise and carrier IP communications are built from similar building blocks as the IP network and the Web. HTML- and XML-based Web services supported across the board by most software providers allow a complete re-engineering of the operations and support systems as components in the Web services architecture.

**Migration from IP PBXs to IP communications**

Circuit-switched PBXs and their successors, IP PBXs, suffer from several limitations when compared to enterprise IP communication systems. Each enterprise manager will have their own migration issues, priorities and criteria according to their specific business needs and install base. The key is to achieve smooth migration from traditional telephony-style communications to the richer converged world of IP communications.

<u>Circuit-switched PBXs</u>

Except for rare "greenfield" cases where networks are designed and established for the first time, most network and IT managers are faced with the problem of migration from existing PBX systems, most of which are circuit-switched systems that likely still run well. A transition plan towards IP communications could be undertaken in several phased steps, as follows:

- Introduce plain IP trunking between PBXs using SIP signaling over IP, for various Internet access services, private lines (T1-T3) and frame relay. This requires only voice enterprise gateways between the PBXs and the IP network.
- Use shared network gateways for global voice access to the PSTN.
- Add SIP phones to gradually replace PBX desktop phones for new expansions or when moving offices. Network-based SIP servers can maintain consistent dialing plans between the legacy network part with PBX phones only and the new SIP phones.
- Add/replace PBX functionality with SIP-based enterprise servers or with "IP Centrex" ISP-based solutions.
- Add IP Communication capabilities using enterprise features such as SIP voice mail, SIP conferencing, SIP Presence and Instant Messaging, SIP dialog servers, 3rd party call control, user preferences, etc. A special place in this portfolio is taken by firewall traversal for SIP signaling, voice and video. The availability of many such services using enterprise SIP servers, SIP PC User Agents and firewall/NATs has made such features possible earlier in the market than previously believed feasible.

<u>H.323, IP PBX and softswitch systems</u>

Various IP voice systems on the market can be classified as H.323 systems (though no interoperability with other H.323 systems may generally be assumed), proprietary IP PBX systems and softswitch systems using one of the master-slave protocols such as MEGACO/H.248, MGCP, or some of their variants. The transition scenario from such systems to IP communications involves the following steps:

- H.323 PBX: Deploy H.323-SIP signaling converters for Gatekeeper-to-SIP proxy signaling. Dual mode (H.323 and SIP) VoIP gateways can be deployed in the transition for large installed H.323 systems.
- Miscellaneous IP PBXs and softswitch: Use the SIP signaling capability where available as the service/feature creation infrastructure (Megaco/H.248 is fully compatible underneath this at the gateway level, but is not involved in service/feature interactions). Explore the use of PBX TDM trunks to connect IP PBXs that do not support SIP to the TDM side of VoIP gateways.

## Conclusion

These are exciting times in the IP communications world. The emergence of SIP as the standard for converged communication services across a wide variety of platforms offers new capabilities to the users, new efficiencies and competitive advantages to enterprises, and new revenue generating opportunities for service providers. Businesses and consumers alike will benefit from increased productivity and a rich choice of alternatives for their communications needs.

Mitel Networks is an active member of the SIP Forum, a trade association that promotes the use of SIP technology.

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