



## **A sip of the SIP.**

*The standard for presence awareness, instant messaging, telephony and more.*

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**Executive summary**

The Session Initiation Protocol (SIP) standard was created by the Internet Engineering Task Force (IETF) in 1996. Initially, this protocol was built to provide media capabilities (that is, audio or video) among users. Today, SIP has evolved significantly and is now able to perform a wide range of real-time collaboration functionalities such as presence awareness and instant messaging (IM), wireless applications, telephony and more.

To leverage the capabilities provided by SIP, Lotus® software from IBM, the leader in collaboration, is creating a SIP infrastructure as part of its newest platform known as IBM Lotus Workplace. With the help of SIP, Lotus Workplace will allow IBM to build a series of applications aimed to help integrate multiple collaborative capabilities into a single, easily managed platform.

Key points to remember why SIP is an important and powerful tool:

- *SIP is a standard. The SIP standard is a protocol that was created by the IETF. This is the same organization that created TCP/IP, HTTP and many other common protocols that have shaped the Internet as we know it today. A standard protocol offers many benefits that include, but are not limited to: interoperability, security, standard application programming interfaces (APIs), provider independence and more.*
- *SIP is one protocol. Compared to other standards that have limited functionality, SIP supports a very wide range of applications and features such as presence awareness and IM, multimedia conferences, telephony, wireless devices and more. Because SIP is one protocol that offers the most flexibility, implementation ease and power for integrating real-time collaboration capabilities within Lotus Workplace, it is the best choice.*
- *SIP is widely adopted. SIP is recognized as the de facto protocol for real-time collaboration by many technology authority groups, companies and markets. For example, it is being widely adopted to support the vast and ever-evolving Voice over IP (VoIP) and telephony markets.*

- *SIP evolves quickly. SIP-affiliated groups are the most active in the IETF. Many new features are being added to the protocol on an ongoing basis. Some of these features such as multiway conferencing, emergency preparedness, deaf people support and more, are crucial for supporting a real-time collaboration environment. SIP continues to provide the right level of functionality required to meet the collaborative demands of its users.*

### **One protocol**

Some of the most-common real-time collaboration solutions that are used by enterprises today include one or more of the following solutions:

- *Standard private branch exchange (PBX)*
- *Presence awareness, IM and conferencing such as IBM Lotus Instant Messaging and Web Conferencing (IBM Lotus Sametime®)*
- *E-mail, calendar and scheduling, for example, IBM Lotus Notes® software*
- *Mobile phones*

With the help of a single protocol, these solutions can be replaced or enhanced in two beneficial ways. First, the use of a standard protocol eliminates the need for proprietary protocols (see “Questions and answers” on page 13 for more details on the benefits of using a standard). For example, standards ensure interoperability and allow for companies to adopt this protocol more readily because it integrates easily with their existing infrastructures and more. Second, using a single protocol instead of a collection of protocols will ensure a more seamless and easier integration of the real-time collaboration solution.

A few common examples of how SIP-based technologies can replace existing real-time collaboration solutions include:

- *General notification and presence awareness – use SIP to determine the state of a user (for example, “Active,” “Away”) and get notifications on status changes.*
- *IM – use SIP messaging to communicate using IM with colleagues and talk securely with customers from other companies.*

- *Conferencing*—use SIP session initiation to create one-to-one or multiparty media conferences.
- *SIP telephony and VoIP*—use SIP phones and SIP PBXs for intra- and inter-company communications.
- *Mobile devices*—use SIP-based mobile architectures and services to enable mobile employees to be more connected to the company intranet when off-site.

#### *General notification and presence awareness*

Recently, subscription and notification capabilities have been incorporated with SIP. A protocol is now defined for enabling SIP users to subscribe to an object and get a notification when the status of that object changes. Using the concept of an event package, it is possible to define the protocol details of subscription and notifications for different objects.

The SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) working group at the IETF has defined an important event package known as the *Presence Event Package*. The Presence Event Package describes how to use SIP for presence awareness and generates a presence list (that is, the Buddy List as trademarked by AOL). To implement the presence list, a new SIP server component is introduced, called the *Presence Server*. The Presence Server is responsible for getting and serving subscription requests from users. It provides this service by forwarding status change notifications to all appropriate subscribers.

Figure 1 shows a Presence Server and two different user agents – Alice creates a subscription about the status of her colleague, Bob (User A). The Presence Server will remember to notify Alice (User B) when Bob's status changes (User A).

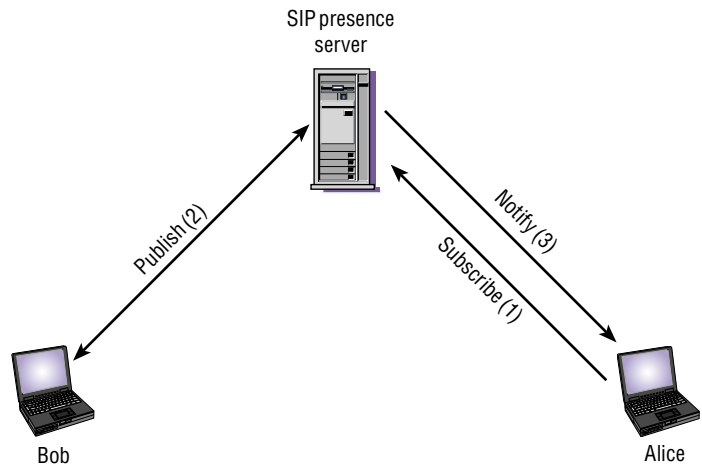


Figure 1. Subscription and notification

### Conferencing

The original purpose of SIP was to enable media sessions, and it is able to do this in the following way:

- *SIP Uniform Resource Identifier (URI) – the SIP URI is the logical identifier of a user (for example, bob@example.com).*
- *Registrar – A SIP user agent (client) can send a REGISTER message to the registrar to advertise its availability. The register operation associates the SIP URI with one or more access points known as contacts. When a caller wants to establish a session with the callee, the information in the registrar is used to find the callee.*
- *Session management – to place a call to another user, a SIP user agent sends an INVITE message to the server and will forward it to the target user. This message and those that follow allow for negotiation of media types, such as audio and video.*

After the session is established, the media can be exchanged directly (peer-to-peer) between the session participants. The session can be one-to-one or one-to-many. The following diagrams show several types of SIP conferences.

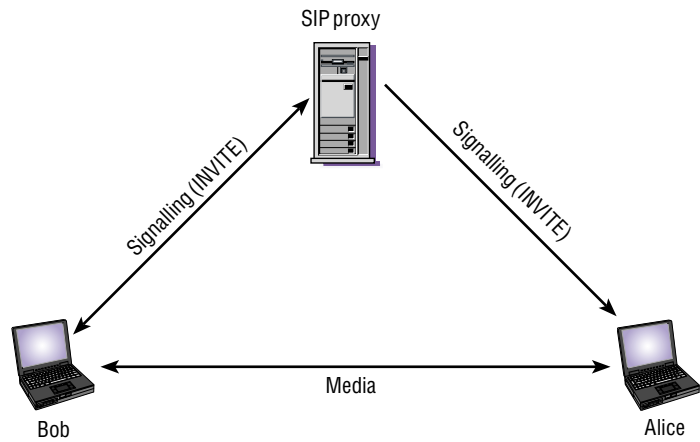


Figure 2. Two-way media session

Figure 2 shows a SIP session between two parties. The INVITE messages establish the session parameters through the server and allow for peer-to-peer interaction.

When a third party is added to the call, there is a need for merging the media streams of the participating users. One option, as shown in Figure 3, is for a server to act as the multipoint control unit (MCU). The central server is responsible for receiving media streams from all conference participants, mixing the streams and sending them to the other participants.

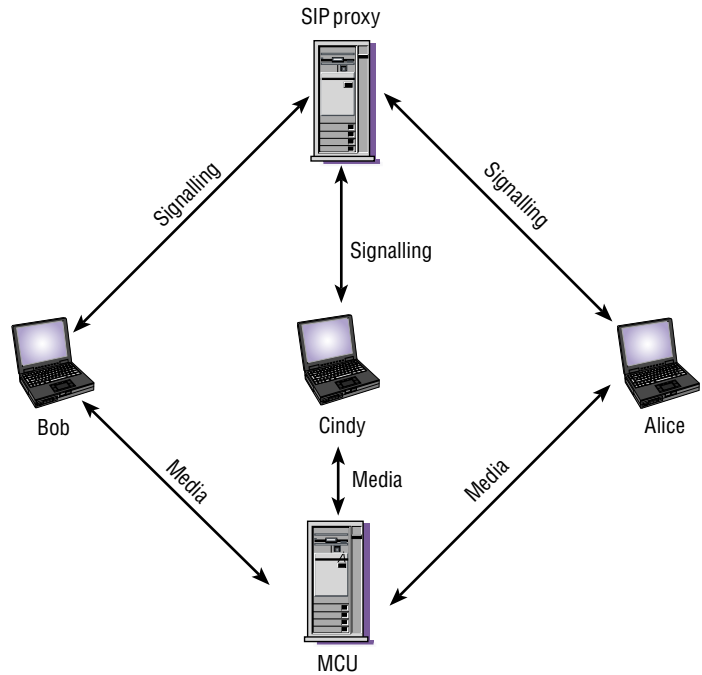


Figure 3. Three-way media session

#### *Awareness in conferences*

Having presence awareness incorporated into online conferences with SIP presence technology adds a whole new dimension to virtual business practices. Not only can users have audio and video capabilities, but also they can view the names of attendees, see who is speaking and much more. The experience of a virtual conference becomes richer, more efficient and closer to a real-life person-to-person conversation.

In addition to a presence list that can be part of an online conference, the ability to use conference policies is available. A conference policy determines the membership and media policy of a conference. For example, a conference policy would be who can attend the conference while a media policy would determine what media types can be used in the conference and by whom. This is a very complex issue and a new working group (XCON – Centralized Conference) has been established in the IETF to focus on this.

Note that there are existing solutions today, such as Lotus Instant Messaging and Web Conferencing (Sametime) meeting technology, that can enable a presence list in a conference and some form of conference control. However, these technologies use several proprietary protocols. Adding new features involves in-depth knowledge of these proprietary protocols and the relationships between them. The key benefit of SIP is that it enables all of these features using a single standard protocol, easily and quickly.

#### *IP telephony and VoIP*

SIP can be regarded as the enabler protocol for telephony and VoIP services. SIP plays a major role in the enablement of IP telephony and VoIP in the following ways:

- *Location and status of users – using the SIP registration and Presence Server, it is possible to locate other users. It is also possible to see the capabilities that a user may attach to each contact in order to select the best contact for a given operation. For example, if a user has several devices on which he or she runs a SIP client: one device is more capable for running video sessions (it has a better screen) and the other is more capable for running audio sessions (it has better speakers). Each device will have a different contact and each contact can indicate the capabilities of the device that the contact represents. The subscription and notification mechanism provided by the Presence Server allows for continuous notification on the status of users. This allows, for example, the ability to determine when a user is most likely able to take a call.*
- *“Who is on the call?” – in a phone conference call, SIP enables awareness of the call participants. Adding indicators such as “who is speaking now” eliminates the embarrassment of guessing a speaker’s identity.*
- *User-centric – with traditional phone systems, the logical unit of operation is the phone extension; whereas in the new SIP world, the logical unit is the user. Calls can be created and directed according to user registrations, not to a specific extension that is assigned to the user. The SIP structure allows for more flexible support with a wide range of applications that are simply not available with traditional systems.*



- *Media negotiation – the inherent SIP mechanisms that enable negotiation of the media used in a call enable selection of the appropriate codec for establishing a call among the various devices. This way, less-advanced devices can participate in the call, provided the appropriate codec is selected.*
- *Telephone – URIs and public-switched telephone network (PSTN) gateways – work done in the IP Telephony (IPTTEL) group of the IETF enables mapping among regular telephone numbers and SIP URIs. Additional work has been done in creating gateways between the IP world and the legacy PSTN world. All this work is an enabler for merging IP and the old telephone world. Now all users can be easily connected, regardless of the devices they use, simplifying communications among individuals, groups or entire organizations.*

#### *SIP phones*

Consider SIP phones an advanced version of the phone as we know it today. As an example, SIP phones allow users to view a presence list to see who is, or who is not, available. In addition, SIP phones have an “alert me” service notification when a user is available, they can initiate a call when the user is available and many other services that are bound only by our imagination (and ongoing technology advancements). In short, SIP phones open the possibilities to use a wide range of services that are simply not incorporated into standard or even the smartest phones available today.

#### *SIP PBXs*

The PBX is a telephone-switching center that is owned by a private business, as opposed to one that is owned by a common carrier or telephone company. Similar to SIP phones, a SIP PBX provides a significant amount of flexibility and functionality that is not possible with existing legacy PBXs. In addition, using a standard protocol for PBX enables a company to switch PBX providers without the need to replace all of the end devices as is done today. Consider all those “smart” phones that you need to replace when switching a PBX; a very costly and time-intensive effort that is virtually eliminated with SIP PBX.

Figure 4 illustrates a SIP PBX system. A SIP proxy that sits in front of the system routes calls to the appropriate devices according to several sources. One source can be a user's calendar specifying which room that user is occupying at specific times and whether he or she can be disturbed. Or companies may use an automatic tag system that identifies users and routes a call to the nearest phone system.

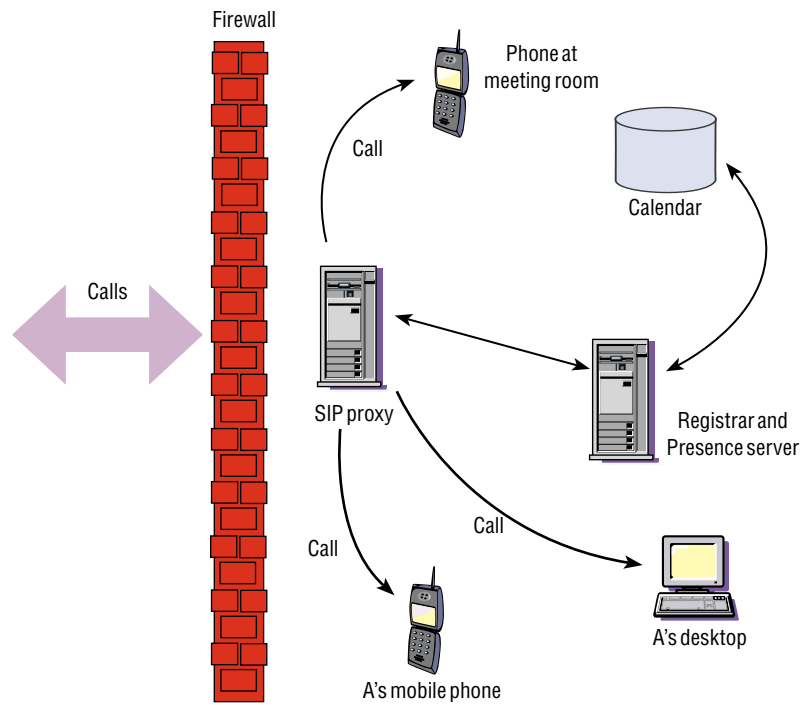


Figure 4. SIP PBX

*A SIP phone system*

Figure 5 shows more detail regarding a SIP-based phone system and a PSTN gateway. The PSTN gateway enables the routing of calls that are initiated in the PSTN system to the SIP system, and vice versa.

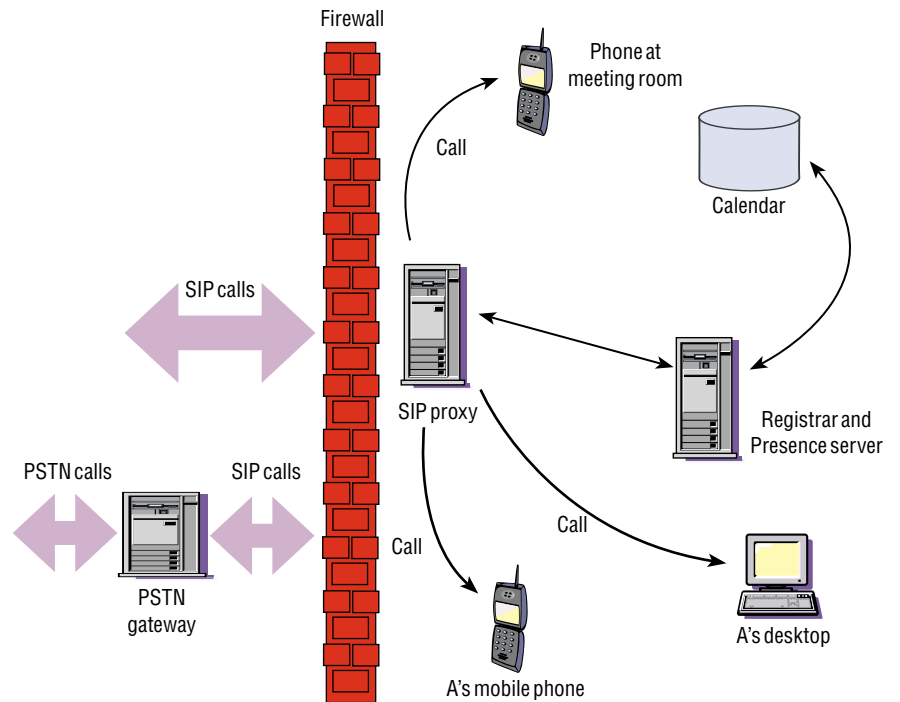


Figure 5. SIP phone system and a PSTN gateway

### Mobile devices

The 3rd Generation Partnership Project (3GPP) is an organization whose goal is to “produce globally applicable Technical Specifications and Technical Reports for a 3rd Generation Mobile System based on evolved GSM core networks and the radio access technologies that they support.”<sup>1</sup> This organization has adopted SIP as one of its major protocols to be used in its architecture. The 3GPP also has a liaison agreement with the IETF in order to be able to synchronize work on standards in both organizations.

The 3GPP organization has created the IP Multimedia System (IMS) architecture, which is based on a SIP standard. This document states:

*In order to achieve access independence and to maintain a smooth interoperation with wireline terminals across the Internet, the IP multimedia subsystem attempts to be conformant to IETF “Internet standards.” Therefore, the interfaces specified conform as far as possible to IETF “Internet standards” for the cases where an IETF protocol has been selected, e.g. SIP.*

The 3rd Generation Partnership Project 2 (3GPP2)<sup>2</sup> is another organization that focuses on mobile systems, but for a different set of network protocols – who said that the world of Internet standards is perfect? The 3GPP2 organization, like 3GPP, is basing its architecture on SIP. The dependencies list of 3GPP2 on the IETF work in SIP can be seen in [3GPP2-DEP].

Another standards organization that deals with mobile systems is the Open Mobile Alliance (OMA).<sup>3</sup> Its mission statement is as follows:

*The mission of the Open Mobile Alliance is to grow the market for the entire mobile industry by removing the barriers to global user adoption and by ensuring seamless application interoperability while allowing businesses to compete through innovation and differentiation.*

The OMA is also considering SIP as one of the major protocols for the services it defines.

The bottom line is that organizations that are defining architecture and services for next-generation mobile systems are looking at SIP as the major protocol to be used. Adopting SIP now enables better preparedness for new mobile systems.

### Questions and answers

Everything you wanted to know about SIP, but were afraid to ask ...

*(Q) Why is a standard protocol better than a proprietary protocol?*

*(A) Widespread and swift adoption of standards is very important for the technology industry for a myriad of reasons. Standards offer:*

- *More security – when a protocol undergoes standardization, it is scrutinized by the same community where it was originally developed. This is particularly true at the IETF. Hence, the security of a protocol can be verified only once and not for every proprietary protocol. In addition, a single company rarely has the expertise and resources to do an extensive security inspection for the protocol that is done by the standards community.*
- *More interoperability – standards prevent “walled gardens” (that is, the presence and IM communities of AOL, Yahoo, MSN and so on). Users of these services get very good service, but have no interoperability with their family, colleagues and friends who use other presence and IM communities. Companies justify this lack of interoperability with the lack of interoperability standards.*
- *More features – when a standard is available, groups that develop features (for example, third-party developers) only need to design and implement a feature once. This not only eliminates the need to design and implement a feature for every available proprietary protocol, but also prevents the need to design interoperability with other proprietary protocols.*
- *More freedom – when a company buys a solution based on a standard, the company can purchase additional features and solutions from third-party developers who support that standard (and not for a company-specific product). In addition, a company that buys a standards-based solution is not limited by the supplier of the solution. The company can always switch to another solution provider, without changing clients and thus minimizing redeployment costs.*

*(Q) Isn't a proprietary protocol more efficient?*

(A) It is true that proprietary protocols can be more efficient than standard protocols. However, the diminished efficiency of standards-based solutions is outweighed by the benefits that they provide. For example, the use of HTTP is so widely used today that using any other protocol would be considered greatly inefficient.

*(Q) There are already several widely accepted standards in the real-time collaboration area. Why do we need another one?*

(A) Let's review the status of real-time collaboration protocols prior to the acceptance of SIP:

- *Proprietary presence and IM solutions – current presence and IM communities like Sametime, AOL, Yahoo!, MSN and ICQ use proprietary protocols. Most of these communities are referred to as walled gardens because they do not allow users to use their presence and IM capabilities beyond their respective communities (or, gardenwall).*
- *ITU solutions – these are complicated solutions for multimedia and VoIP such as H.323 and T.120.*
- *Wireless Village – the Wireless Village protocol was created by the Wireless Village standard organization (now part of the OMA organization), and is a presence and IM protocol designed for mobile devices and wireless operators. However, no public deployment of Wireless Village by a mobile operator has yet been declared.*
- *Telephony solutions – the PSTN and the mobile world use several protocols for their operations. These protocols include Signaling System 7 (SS7), Global System for Mobile Communications (GSM) and so forth.*

The world of real-time collaboration protocols as it stands today is divided. There is no single base protocol that supplies a basic registration, locator and presence awareness service that can be used by all other protocols. A basic protocol allows to dramatically enhance collaboration services for activities such as conferences and IM. For starters, being able to see the status of a user when scheduling a call can be very helpful and time-efficient.

The primary intent for the development of SIP was to enable the initiation of a media session among two or more users. Since SIP's inception, many capabilities have been added that support most aspects of real-time collaboration functionality. Here is a partial list of current SIP capabilities:

- *Registration and locator* – enabling users to register and other users to locate these registered user online.
- *Session establishment* – using SIP for creating a media session, including media negotiation, call control, call forwarding and more.
- *Events notification* – subscription and notification of events, particularly with presence awareness (that is, being notified when someone becomes active).
- *Device capabilities* – IP enables registering multiple devices by the same user (for example, PC and mobile phone). In addition, SIP can define different capabilities for each device, thus enabling the selection of the appropriate device when a session is established.
- *Interfacing with traditional phone systems* – much work has been done towards creating gateways between SIP and PSTN, including mapping between phone numbers and SIP identifiers, and more. These enhancements to SIP are made to establish a simplified interface between SIP and PSTN.
- *Mobile world* – the standards organizations that support the mobile world have either adopted SIP as the de facto protocol (3GPP, 3GPP2), or use one of the basic protocols (OMA). The 3GPP and 3GPP2 organizations have defined an IP architecture that is based on the SIP protocol (IMS and MMD). The OMA is in the process of defining services with SIP.

*(Q) Are there standard APIs that are built on top of the SIP protocol?*

*(A) Yes. The Java™ Community Process (JCP)<sup>4</sup> has already defined several APIs for SIP. Two examples (Java Specification Requests [JSRs]) include:*

- *JSR 32* – a low-level Java SIP API.<sup>5</sup> This API specification provides a standard portable interface to share information between SIP clients and SIP servers, providing call control elements, enabling converged-network applications.
- *JSR 116* – An API that defines a high-level extension API for SIP servers.<sup>6</sup> It enables SIP applications to be deployed and managed based on the servlet model.

The importance of the APIs cannot be exaggerated. APIs add a new dimension to standards. An application that exposes the standard API can be easily extended by plugging in components that use the API. For example, JSR 116 enables extending the SIP server by writing servlets that will implement the required extensions.

*(Q) Should we invest in SIP now?*

*(A) Yes, the sooner the better.*

Food for thought: Did you know that the following companies are busy developing SIP infrastructures or SIP devices, or both?

- *IBM*
- *Microsoft*
- *Cisco*
- *DynamicSoft*
- *Sprint*
- *Nokia*

The SIP standardization efforts are tremendous. The core protocol is in a stable state and there are efforts to finalize the specifications for additional services and multiway conferences, emergency preparedness and much more. Dozens of companies are highly engaged in these efforts. SIP is responsible for the most active groups in the IETF (there are more than 150 active drafts at any given moment). In addition, as mentioned before, the organizations that are responsible for creating architectures and services for the mobile world are using SIP as the de facto standard for their work.

The SIP Forum,<sup>7</sup> which deals with SIP products and deployment, has 34 companies as full members, including IBM, Microsoft, Nokia, Ericsson, Hughes, Indigo, and many more.



The readiness of SIP enables the creation of many new applications that were previously too difficult to build using older protocols. Among these applications are:

- *SIP phones that have a presence list (for example, Buddy List).*
- *VoIP applications that use the SIP session management and the capability to interface with PSTN gateways.*
- *SIP PBXs that use SIP-affiliated capabilities such as session establishment and redirection, and SIP registrar.*

*(Q) Can SIP support multiple devices for a user?*

*(A) One of the primary design principles in SIP is its ability to support multiple devices. A user can register multiple contacts for his or her address-of-record (AOR). Another user can send a request to the server to locate a registered user and the server can fork (try all contacts simultaneously) the request or try the appropriate contact channels individually.*

Another important feature is the addition of user capabilities to user contact. For example, indicating the devices that can support video stream enables users to select only those contacts whose devices support video.

*(Q) Firewalls and Network Address Translations (NATs) do not provide a good experience when deploying real-time services. Does SIP handle NATs and firewalls?*

*(A) Yes.*

NAT is a network component that translates between external and internal IP addresses. For example, a company that wants to hide its IP addresses (for security or other reasons) can use a NAT that translates the IP of an outgoing connection to an external IP address. The problems occur when there are incoming connections that specify the internal IP address of a server behind a NAT; in these cases, the NAT does not know how to route the IP packets.

Firewalls are used for protecting a network from any incoming or outgoing packet connections, or both. Only those packets allowed by the firewall administrator will pass through the firewall. SIP messages, and audio and video User Datagram Protocol (UDP) packets in particular, will not pass through the firewall unless explicitly allowed by the firewall administrator.

The Middlebox Communication (MIDCOM) working group of the IETF is working on this issue. The emerging solution of the group is to enable applications behind a NAT or firewall (or any middlebox) to talk with

the middlebox and notify the middlebox of the application needs for communication. A SIP proxy, for example, tells the middlebox that it should accept SIP messages coming from the outside network. This is a very simplistic explanation of the solution, but the point is that solutions exist and will be part of NATs and firewalls. See also [www.checkpoint.com/press/2002/voip061102.htm](http://www.checkpoint.com/press/2002/voip061102.htm) as an example of an existing firewall that has a solution for SIP.

*(Q) What is the SIP security model?*

*(A) Each IETF protocol must have a section that explains the security threats to the protocol and the possible solutions. This section is revised by IETF security experts who provide feedback on any security holes. The core drafts of SIP have passed this review.*

The details of the SIP security model are beyond the scope of this white paper. However, note that:

- *SIP requires at least Digest Authentication. In Digest Authentication, the user name and password are passed encrypted using a challenge that is received from the server.*
- *The body of SIP messages can be encrypted using Secure Multipurpose Internet Mail Extensions (S/MIME). This way, only the two end points of the communication can see the message body.*
- *SIP enables secure communication between its components using a Transport Layer Security (TLS) connection. There is even a special SIP URI called SIPS (“S” for Secure). A SIPS URI indicates that it has traversed only secure connections (TLS) to the SIP component where it currently resides.*

*(Q) What about the XMPP (Jabber) protocol?*

(A) XMPP is an effort by the Jabber organization and company to standardize an existing presence awareness and IM system. Jabber is open-source protocol and based on XML. The IETF has agreed to form a working group with a charter to standardize the Jabber protocol. Generally, the IETF does not allow competing standards to be created under its umbrella. However, in this case, the IETF agreed due to Jabber being an open-source protocol and is widely adopted by a large community of users.<sup>8</sup>

XMPP is simpler than SIP because XMPP is only TCP-based and tailored for the purpose of presence awareness and IM using XML. However, it is important to note that XMPP is very different from SIP in that there is a need to use SIP or additional protocols, or both, for the following capabilities:

- *Session establishment*
- *Media negotiations*
- *Registration of multiple devices and capabilities*
- *Firewall solutions that were developed for SIP*

It may be possible to use XMPP as the presence awareness and IM solution and SIP as a solution for other capabilities. This solution has a fundamental disadvantage, though, because it does not use the same protocol for the registration of users and for the capabilities that are built above these registrations. For example, when implementing a SIP phone that has a list of users and their status, there will be a need to implement two protocols at the phone and two protocols at the servers. This adds the need to translate among the protocols.

When reviewing IETF activities, it is obvious how much more industry attention is being given to SIP protocols when compared to that of XMPP protocols.

*(Q) How much has IBM committed to SIP and in particular, what is IBM doing?*

(A) IBM is committed to SIP. A new Presence Server is being developed based on SIP rather than the proprietary protocol used in IBM Lotus Sametime.

The IBM SIP infrastructure will serve several purposes. It will serve as the infrastructure for Lotus Workplace, as well as provide SIP building blocks for other SIP applications in IBM, including a SIP stack, SIP proxy, SIP registrar, SIP presence server and more.

Of particular interest is the way that the SIP infrastructure is built in relation to IBM WebSphere® software, the IBM flagship applications server. One of the main components of the SIP infrastructure in IBM is the SIP container, which is built over an existing WebSphere Web container and uses the standard JCP API – JSR 116.<sup>9</sup> Building SIP this way is aimed to enable full integration of the SIP technology into WebSphere.

### **Summary**

In this white paper, we have outlined why SIP is an important protocol for real-time collaboration functionality. We have described how SIP, as one standard protocol, can be easily harnessed to do various tasks that are currently performed by multiple protocols, most of which are proprietary.

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*The IETF*

The following is a short background of the Internet Engineering Task Force (IETF). Further information is available in the IETF site at [www.ietf.org](http://www.ietf.org) and in the TAO of the IETF at [www.ietf.org/rfc/rfc3160.txt](http://www.ietf.org/rfc/rfc3160.txt).

The IETF is considered to be the most important standards organization for the Internet. This is the organization that created the IP protocols that enabled the creation of the Internet and the HTTP protocol which, in turn, enabled the creation of the World Wide Web (WWW).

The IETF is known for the strong security of its protocols which are rigorously inspected by IETF security experts.

The IETF is divided into several work areas:

- *Applications*
- *General*
- *Internet*
- *Operations and Management*
- *Routing*
- *Security*
- *Sub-IP*
- *Transport*

Each area has two area directors and several working groups. Each working group has at least two group chairs that are responsible for managing the group meetings and are acting as main contacts with the area directors.

The area directors of all working areas form a group that is called the Internet Engineering Steering Group (IESG). The role of the IESG is to “steer” the work of the IETF. An Internet-draft is created and is discussed in the mailing list and in working group meetings. After the group feels that it is ready, the draft passes a Working Group Last Call (WGLC). After it passes a WGLC, it is sent to the IESG for final approval. When a draft is approved by the IESG it becomes a Request For Comments (RFC). When a draft reaches the state of an RFC, it is considered to be mature enough for worldwide implementation.

### *SIP History*

In 1996, the IETF merged two proposals for session initiation protocols, one by Mark Handley on SIP and the other by Henning Schulzrinc that was named Simple Conference Invitation Protocol (SCIP).

SIP was approved as RFC 2543 (<http://www.ietf.org/rfc/rfc2543.txt>) in March 1999. In June 2002, RFCs 3261-3265 were approved by the IESG. These RFCs create the core protocol set for SIP and they were achieved after a tremendous amount of work by the authors and the IETF members.

### *SIP-related IETF Working Groups*

Following the publication of the core SIP drafts (RFCs 3261-3265), there continues to be very intensive activity in the IETF focused on SIP.

The following working groups at the IETF work on SIP directly and indirectly:

- *SIP* – the original SIP group, which works on extensions to the standard.
- *SIPPING* – Session Initiation ProPosal INvestiGation. This group is responsible for examining new requirements for the SIP protocol and for deciding if an extension to the protocol is required.
- *SIMPLE* – SIP for Instant Messaging and Presence Leveraging Extensions. This group is responsible for the IM and presence extensions for SIP.
- *XCON* – Centralized CONferencing. This group is responsible for creating a set of protocols for tightly coupled multimedia conferences. Most, if not all of the protocols of this group will be SIP.

### *SIP concepts*

Technical descriptions of SIP concepts and terms.

*Address-of-record. An AOR is a SIP or SIPS URI that points to a domain with a location service. This service then maps the URI to another URI where the user might be available. Typically, the location service is populated through registrations. An AOR is frequently thought of as the “public address” of the user.*

*AOR. See Address-of-service.*

*Contact. Provides a SIP URI that can be used to contact that specific instance of the UA for subsequent requests.*

Location service. *A location service is used by a SIP redirect or proxy server to obtain information about a callee's possible locations. It contains a list of bindings of address-of-record (AOR) keys to zero or more contact addresses. The bindings can be created and removed in many ways. This specification defines a REGISTER method that updates the bindings.*

Proxy. *See Proxy server.*

Proxy server. *An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it.*

URI. *Uniform Resource Identifier. For the purpose of this document, this is a unique string that identifies a resource and a user in particular.*

Registrar. *A registrar is a server that accepts REGISTER requests and places the information it receives into the location service for the domain it handles.*

UA. *See User Agent.*

User Agent. *A logical entity that acts on behalf of the user (also referred to as client).*

Figure 6, is a schematic view of a SIP server. Alice's SIP user agents are connected to a SIP server through a SIP proxy. In this case, the SIP server consists of a proxy, registrar and location service.

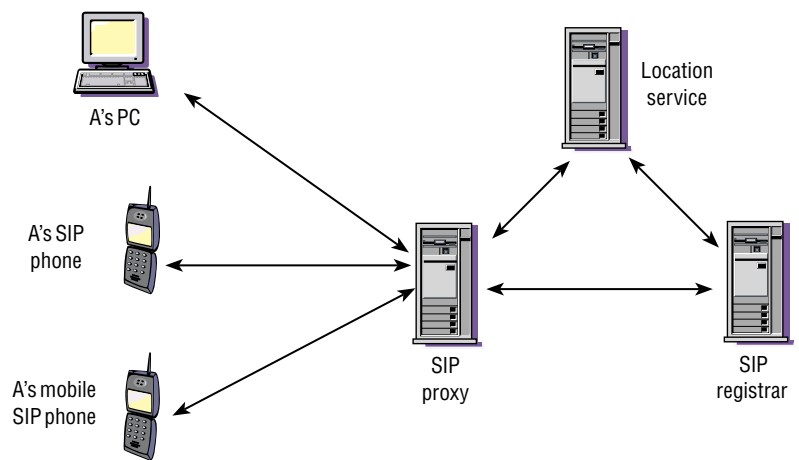


Figure 6. Basic SIP server

### **References**

Note: All Web addresses were current as of January 31, 2004.

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<sup>1</sup> [www.3gpp2.org](http://www.3gpp2.org)

<sup>2</sup> [www.3gpp2.org](http://www.3gpp2.org)

<sup>3</sup> [www.openmobilealliance.org](http://www.openmobilealliance.org)

<sup>4</sup> <http://jcp.org>

<sup>5</sup> <http://jcp.org/en/jsr/detail?id=32>

<sup>6</sup> <http://jcp.org/en/jsr/detail?id=116>

<sup>7</sup> [www.sipforum.org](http://www.sipforum.org)

<sup>8</sup> [www.jabber.org](http://www.jabber.org) and [www.jabber.com](http://www.jabber.com)

<sup>9</sup> <http://jcp.org/en/jsr/detail?id=116>