SIP and IPLink™ in the Next Generation Network

An Overview

Executive Summary

Because of its roots in standard Internet protocols, Session Initiation Protocol (SIP) is quickly gaining popularity with application, communication, and network service providers (ASPs, CSPs, and NSPs) focused on offering their customers innovative new services. Another key to offering powerful next generation network services is IPLink, a comprehensive, standards-based software and hardware development platform for Internet Protocol (IP) telephony servers.

IPLink provides the architectural and programmatic link to other Intel telephony technologies, as well as to products from the entire Intel computer telephony community. SIP greatly expands the access of the PC and Web developer community to computer telephony resources, in both the enterprise and service provider spaces, by enabling developers to interact with telephony resources the same way they interact with other Internet resources.

The strengths of SIP are its simplicity and its synergy with other standard IP protocols including HTTP, DNS, and SDP. Because of its design, SIP can more efficiently set up and tear down connections than other protocols. Using standard IP infrastructure, SIP is much easier to adopt and integrate for the large body of IP-literate developers familiar with Web-based programming.

Like SIP, IPLink also offers simplicity. Compliant with all relevant IP protocols and specifications, this single PCI or cPCI board provides maximum flexibility in supporting standard IP call control and media gateway protocols, as well as a wide variety of vocoder algorithms.

Together, SIP and IPLink provide a streamlined and powerful foundation for developing the ground-breaking next generation network services that are essential to the success of today's ASPs, CSPs, and NSPs. SIP and IPLink combine to provide a major building block for killer apps in the next generation network.

To learn more about SIP, IPLink, and the open, next generation network, contact your technical sales representative, call 1-800-755-4444, visit http://www.dialogic.com on the Web, or send email to telecomsales@intel.com.

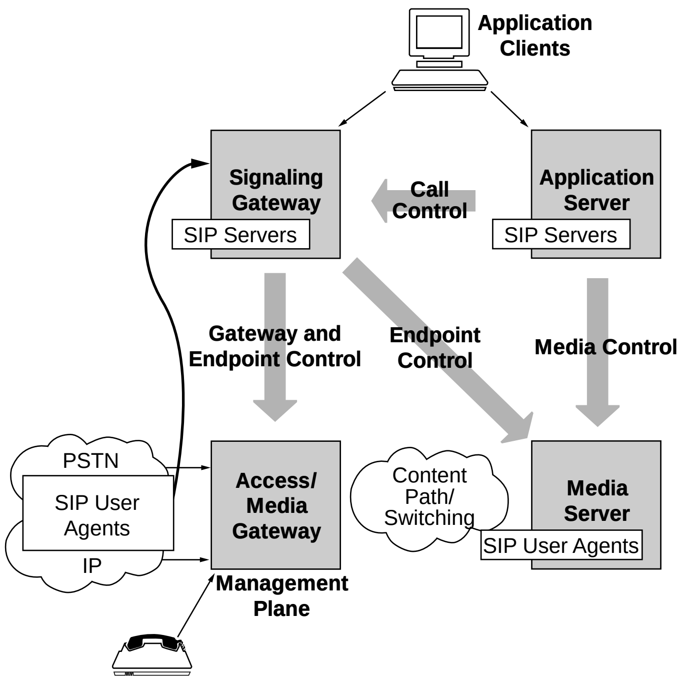

Figure 1: Call control paths in the next generation network

SIP Overview

Session Initiation Protocol (SIP) is used between communications devices and media gateways and media servers to establish various types of sessions on an IP network. The protocol establishes a session using a group of plain-text messages that carry IP address and port information, media capabilities, and encoding formats. Proposed as a standard (RFC 2543) by the Internet Engineering Task Force (IETF) in late 1999, SIP borrows heavily from two Web browsing and email protocols: Hyper Text Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP). SIP has its origins in the MMUSIC IETF working group, circa 1995, which focused on multimedia session control. This was defined as the advertisement, management, and coordination of multiple sessions and their multiple users in multiple media (like audio, video, and collaborative applications). MMUSIC was chartered to design and specify three protocols to perform these functions, and to ensure session-level interoperability between different teleconferencing implementations. The three protocols were:

1. SIP
2. Session Description Protocol (SDP), used today by SIP and media gateway control protocols
3. Session Announcement Protocol (SAP) defined in Remote Function Call (RFC 2974) but not widely used

Updates to SIP are continuing in the RFC 2543bis, which is a (nearly) backward-compatible version of SIP. Also, the IETF SIP Working Group has documented a method by which ISDN User Part (ISUP) signaling is encapsulated inside the body of a SIP message. This method is referred to as SIP for Telephony (SIP-T).

This paper does not cover all the details of SIP, providing only a general overview. For details on SIP, visit Henning Schulzrinne's SIP site (http://www.cs.columbia.edu/~hgs/sip), the SIP forum Web site (http://www.sipforum.org), or contact your technical sales representative at 1-800-755-4444.

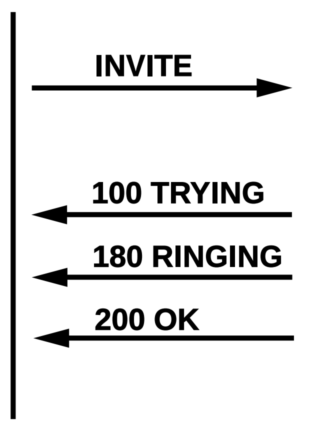

Figure 2: Typical SIP Message Flow

Comparing SIP and H.323

Both SIP and H.323 have distinct advantages and disadvantages.

H.323 STRENGTHS

Although H.323 was originally developed to support voice and video over IP, it has since enjoyed widespread deployment. H.323 is an all-encompassing standard for delivering traditional telephony service over IP, or packet, facilities. Designed to be independent of other standards, H.323 is also designed to support many telephony supplementary services — like conferencing and call forwarding — as standard features. The advantage of this approach is that it leaves very specific openings for a developer to add enhanced features and services, maintaining a high degree of compatibility between implementations. The industry has spent great efforts on H.323, focusing on adding capabilities and improving interoperability. H.323 is, and will continue to be, an important signaling protocol in the next generation network.

SIP STRENGTHS

In contrast, SIP does not support advanced functionality like conferencing and muting. The beauty of SIP is its simplicity. Less monolithic than H.323, SIP relies on many other protocols including RTSP and HTTP. SIP is more efficient than H.323 at setting up and tearing down calls, requiring fewer messages. Also, SIP provides no capabilities for keypad signaling during a call. DTMF digits are sent either in the media stream (when using G.711) or via specialized RTP packets (when using G.726 or G.729). A unique feature of the SIP protocol is that an INVITE message can be sent to multiple destinations at the same time. The first contact to respond with an OK message receives the RTP stream.

A significant event in the adoption of H.323 came when Microsoft* released NetMeeting client. This essentially H.323-enabled most or all Windows* users. Similarly, Microsoft recently announced that SIP capabilities will be included in Windows XP client and server editions. This translates to a rapid increase in the number of SIP clients.

Table 1 compares SIP and H.323.

Table 1: Comparing SIP and H.323

	SIP	H.323
Message Encoding Format	Plain Text	ASN.1
Minimum Call Setup Messages	2	2[1]
Maximum Call Setup Messages	4	8[1]
DTMF Handling	Handled by RTP Protocol as In-band Audio or Special Packet)	Out-of-band or In-band (as Configurable (Out-of-band RTP Recommended)
Call Transfer	Reinvite Message	H.450
Aliasing Capability	Proxy/Redirect Server	Gateway/Name Server

[1]Packet size is much larger than SIP message and total bytes exchanged depends on the Terminal Capabilities of H.323 terminals.

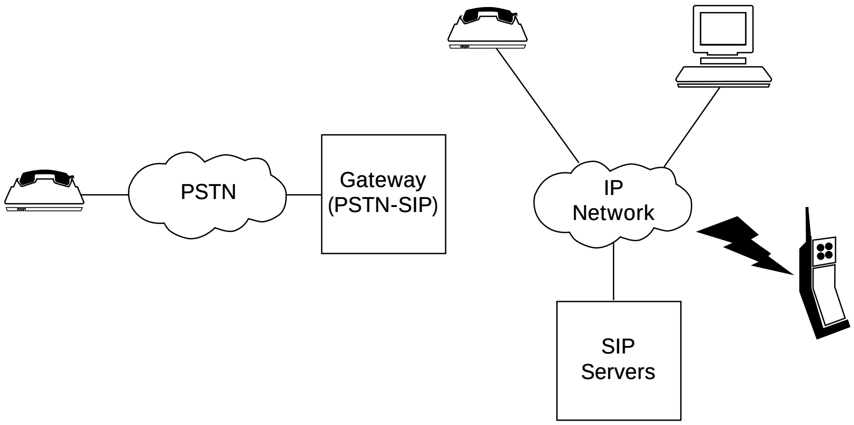

Figure 3: Elements of a SIP Network

Elements of a SIP Network

MEDIA GATEWAY

Bridging calls between the PSTN and IP networks requires a media gateway. Media gateways can be implemented in a variety of ways using a variety of signaling protocols. The example in Figure 3 shows a PSTN-SIP media gateway. This server includes interfaces to both the PSTN and IP networks and to DSP resources for processing the calls. The DSP resources perform two major functions. First, some PSTN protocols, like E-1 and T-1 CAS, require tone detection/generation media capabilities. Also, on the IP side of the gateway, it is often desirable to change from a high-bandwidth codec like G.711 to a codec with a lower bit rate, like G.729a.

MEDIA SERVER

Many voice service components are delivered by media servers. For example, a customer might call an e-commerce site and reach an interactive voice response system or automated attendant. These applications, using various voice, speech, and tone capabilities, run on media servers. In a next generation network implementation, media servers can appear like another phone (in other words, a SIP end point). Thus, they can gain the same advantage of SIP as users can.

The media server element delivers all the voice computing resource needed to interact with the caller. Since the network is entirely computing-driven and the embedded DSP resources allow programmable access to the audio stream, developers can deliver exactly the mix of voice, tone, and speech components they need for their unique service offerings. The media servers allow the technology developers to deliver and connect to up-to-date resources like text-to-speech engines, speech recognition, speaker identification, echo cancellation, noise reduction, and many others.

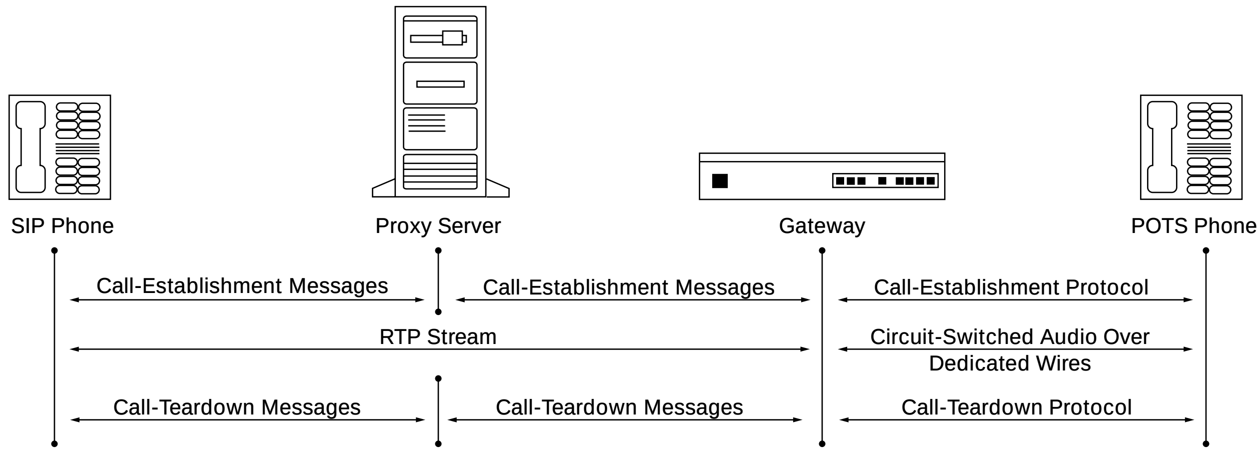

Figure 4: Proxy Server

PROXY, REDIRECT, AND LOCATION SERVERS

A key feature of SIP is the separation of a user's logical address from his or her current actual (physical) address. This enables the user to specify a mapping or alias from his (constant) logical address to one or more (changing) actual addresses. Proxy, redirect, and location servers provide this function. The combination of a proxy server and a registration/location server (Figure 4) allows users of the SIP network to identify themselves by one address while calls are actually sent to one or more different locations.

A Typical SIP Call with a Proxy Server

For example, Joe Smith may have the address jsmith@sip.org. A person wishing to contact Joe places a SIP call to that address. The proxy server then looks up information on the user jsmith and determines where to send the call. A SIP INVITE message is then sent to the address "jsmith" has set up. The destination then sends the response to the proxy server, which forwards the response to the caller. An RTP session is set up directly between the caller and the destination. The proxy server can continue to actively participate in call control messages if required, or it can step out of any further messaging. In some situations, eliminating the proxy server from the signaling path can lead to a much more scalable system.

A Typical SIP Call with a Redirect Server

A redirect server (Figure 5), on the other hand, only processes the first INVITE message to the destination by returning a special response to the caller. The caller extracts from the response a new address to submit an INVITE response, which could be the actual endpoint, a proxy server, or yet another redirect server. From that point forward, all messages go directly between the caller and the destination.

To determine where to send the call, the proxy or redirect server needs to know where the user is located. This requires a location server, which can be on the same machine as the proxy server (as in the case of a simple database) or could use a remote protocol like LDAP or whois.

In a typical scenario, a SIP terminal registers with a location server using the SIP REGISTER method, providing its contact information. For example, a person can register his home SIP phone IP address with the location server, which knows him as Joe.Smith@sip.org. When he is at the office, he registers the office SIP phone IP address with the location server.

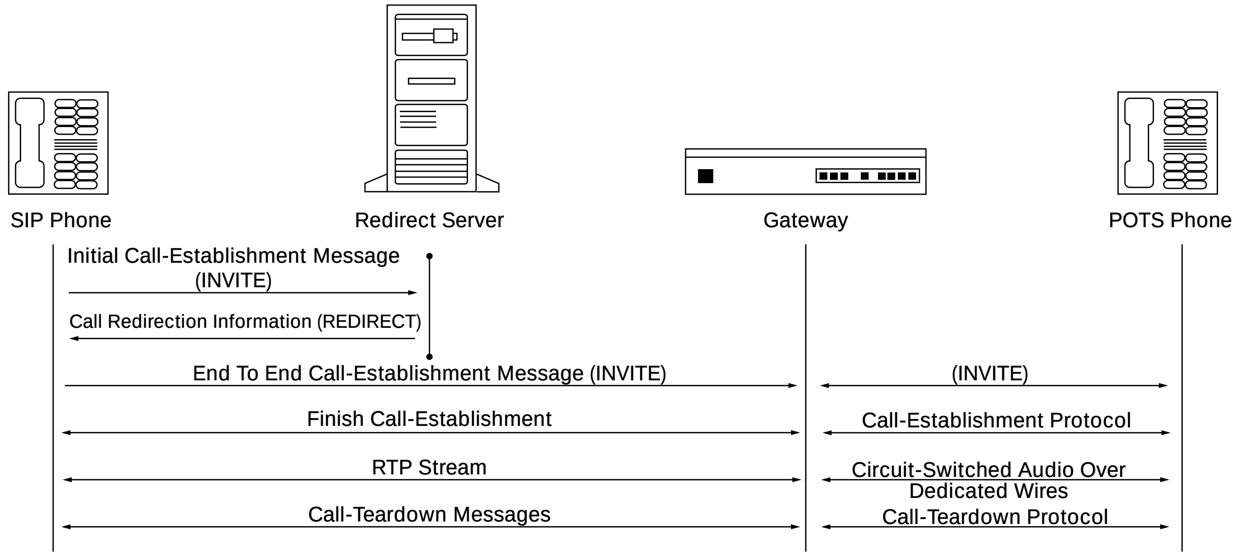

Figure 5: Redirect Server

A Typical SIP Call with a Location Server

When someone needs to contact Joe Smith, a SIP call is placed to a proxy or redirect server at sip.org using the user name Joe.Smith. The proxy server contacts the location server, which reads the database and determines where to send the call. In the case of a proxy server, the INVITE message is sent to the destination address and the proxy waits for the appropriate response. However, if the server is acting as a redirect server, the redirect server uses the location server to look up the desired address and returns that address in a redirect response to the caller. It then sends the message directly to the destination. In both cases the RTP media streaming is performed directly between the end points.

Many proxy server products offer the additional capability of running call processing applications when receiving or making calls. When a new call arrives at the proxy server, the application set up by the user can send the call to different addresses based on the time of day or the response to outbound calls. For example, the user can register three different locations, then the proxy server could try each until the call is answered. Alternatively, the server can try all three locations simultaneously, returning one or more calls as they are answered. These services could be used for a one-number “follow-me” solution.

IPLink

IPLink is an open, standards-based IP telephony platform for gateways and enhanced services applications. This single-card solution includes both Ethernet and PSTN interfaces. IPLink is ideal for OEMs, application developers, and integrators building next generation IP telephony gateway and IP media server systems for enterprise and public network deployments. Highly flexible and robust, the platform has proven, successful field experience.

LEADING CODER SUPPORT

IPLink supports all standard IP coders including G.723.1, G.729a, G.711, and enhanced coders like GSM-EFR. It also supports any IP call control or media gateway control protocol such as H.323, SIP, MGCP, and H.248 with its split call control feature. IPLink is also interoperable with VoIP solutions from leading providers including Cisco, Clarent, VocalTec, and DIGI.

PROTOCOL-AGNOSTIC ARCHITECTURE

IPLink has a protocol-agnostic design that lets the developer choose from embedded or host-based signaling protocols. This "split call control" feature allows unprecedented flexibility in implementing next generation network signaling protocols. This mode allows an application to directly control the Real Time Protocol (RTP) streaming to and from the IPLink board with call control performed in the host application. There are two benefits to host-based call control: First, all the IPLink boards in the system can be addressed using a single IP address for call control. (Separate IP addresses and interfaces are used for the media streaming.) Second, developers can choose from one or more implementations of standard protocols like SIP or MEGACO or choose a non-standard or proprietary protocol.

COMPREHENSIVE SUPPORT AND DEVELOPER TOOLS

Besides providing the open components developers need, like IPLink, Intel also pays attention to intangible requirements, supplying developer and integrator training, coordinating multi-vendor demonstrations and integration, providing a wide variety of example applications with source code, and taking ownership of national or international compliance testing and qualifications. You can explore the broad array of support for developers — and all parties in the value chain, including communications service users — at http://www.dialogic.com.

Also, Intel is working to help developers by defining a reference system for voice communication services architecture for the next generation network. Rooted in multiple legacy circuit-switched networks, voice services require the most effort to transition to the next generation network. The development process traditionally starts with selecting the right components. To do this, developers try different pieces of the solution one after another. Once they have the right pieces, they must begin to integrate them into a complete solution. Normally, this integration process adds no value; it is simply a time-consuming part of the process. By supplying reference integrations, Intel helps developers avoid months of trial and error in choosing the right components and integrating them into complete solutions. This frees developers to focus on their unique added value. Reference integrations also embody much of what Intel has learned over many years of searching for the best methods of developing solutions. The industry has evolved through the efforts of many small developers or development teams with brilliant ideas. Intel works to foster an environment of innovation by delivering the answers to many of the common challenges developers face.

Conclusion

SIP is becoming increasingly popular with ASPs and CSPs looking to stay competitive and increase revenues by offering their customers innovative new services. The open architecture of the IP world allows tremendous flexibility in creating new services. With the steady introduction of new types of end user devices, protocols like SIP allow creative developers almost limitless ability to innovate. They also allow the Internet to meet and exceed the demands of all users — those who are technologically aware as well as those who just want to communicate.

Another key to offering powerful next generation network services is IPLink, a comprehensive, standards-based software and hardware development platform for Internet Protocol (IP) telephony servers. Together, SIP and IPLink provide a streamlined and powerful building block for developing the ground-breaking, next generation network services that are essential to the success of today's ASPs, CSPs, and NSPs.

For More Information

To see an example of an integration with one SIP user agent, contact your technical sales representative at 1-800-755-4444 and ask for the application note "Integrating the dynamicsoft SIP UserAgent with Dialogic IPLink." This note explains how to integrate the dynamicsoft SIP UserAgent with the Dialogic IPLink board. You can also use the information as a starting point for integrating other call control protocols.

You can learn more about the Intel reference system voice communication services architecture for the next generation network effort in the white paper “Reference Systems for Next Generation Network Voice Services.” This paper analyzes next generation network service providers’ business objectives and how the elements for the next generation voice architecture, when based on an open computing model, best meet these objectives. You can download this white paper at http://www.dialogic.com/company/whitepap/7299web.htm.

To learn more about SIP, IPLink, and the open next generation network, contact your technical sales representative by calling 1-800-755-4444, visit http://www.dialogic.com on the Web, or send email to telecomsales@intel.com.