Agilent Technologies

Next Generation Telephony: A Look at Session Initiation Protocol

White Paper

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Introduction

Internet Protocol[1] or IP is rapidly gaining ground as an alternative to the traditional audio and video transport methods used for telephony today. "Voice over IP" or VoIP has emerged as the tag line for transmission of voice or video over IP-based data networks. In addition, the world of VoIP promises to take users beyond the telephone call of today by adding multimedia conference calling, personal mobility, WWW based "click to call" and other such advanced applications.

The ITU-T H.323[2] protocol suite is the dominant VoIP protocol suite as measured by the number of commercially available products. The H.323 protocol suite is also the dominant VoIP protocol suite as measured by the size and complexity of the specifications. The use of H.323 results in a steep learning curve, high cost of implementation, high connection setup latency, and difficulty in achieving interoperability in heterogeneous networks.

Although the ITU-T H.323 protocol suite currently dominates the VoIP world, there exists a lightweight contender for call signaling that avoids all the complexity, high connection setup latency, and implementation difficulties of H.323. The Session Initiation Protocol[3] or SIP brings simplicity, familiarity, and clarity of purpose to IP telephony that Internet savvy network professionals will appreciate.

Whether you are a next generation telephony service provider, a network or IT manager, or an established telephony carrier breaking into VoIP, you are likely to encounter SIP. SIP-based products are on the market now and more are under development. As they encounter it, network professionals should welcome SIP as a text-based, call signaling protocol that takes advantage of the power of the Internet by leveraging such common elements as the format of HTTP[4], Domain Name System[5] (DNS), web-like scripting and email style addressing.

Next generation telephony and the role of Session Initiation Protocol is more easily understood with a little background on the telephone industry and the Internet. The following sections present a brief overview of the current telephone system and the Internet.

Historical Perspective

The original public switched telephone system (PSTN) dates back to 1876 when the Bell Company was formed. Over the years the telephone system has evolved to a complex network that provides advanced services in addition to voice calling. These services include direct dialing, billing options (credit card, calling card, collect, and prepaid minutes), privacy options (caller ID, caller ID block, and selective call blocking), convenience options (call waiting, voice mail, and selective ringing) and directory information. The fundamental circuit switched architecture remains, while most everything else has changed. Until recently, the telephone system in the United States was a monopoly. As is common with monopolies, innovation is introduced slowly and costs are high. Along with the monopoly came strict governmental regulation of the quality and availability of service.

The telephone system of the United States had great influence on the telephone systems of other countries. Although differences in protocols and physical interfaces exist, the principles are the same. Many telephone systems outside of the United States are still monopolies under governmental control.

PSTN vs. the Internet

The PSTN architecture makes use of two distinct functional layers, the circuit switched transport layer and the control layer. The transport layer consists of end-office (Class 5) switches for end-user connections and tandem (Class 4) switches for inter-switch connections. The control network consists of computers (Signal Transfer Points or STPs), databases (Service Control Point or SCP), and service nodes. This network of components controls the behavior of circuit switches of the transport layer, and provides all the services of the PSTN.

The separation of the control from the transport layer frees the transport resources from the burden of carrying signaling traffic. All signaling traffic travels on the Signaling System 7, or SS7 control network. Figure 1 presents a diagram of the telephone system that highlights the architecture.

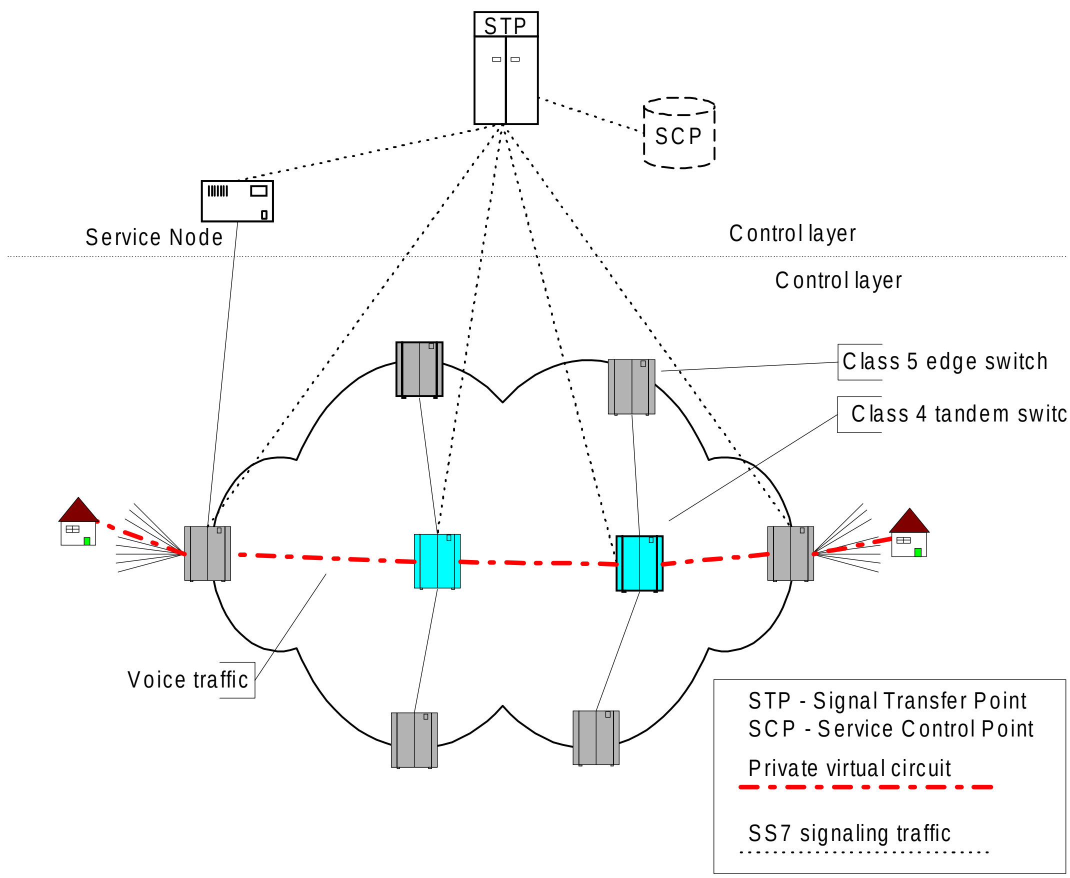

Figure 1: PSTN architecture

The Internet, on the other hand, grew out of United States Department of Defense projects with the goal of developing a communications network that could survive catastrophes. The Internet does not have nor need a separate control layer. The reason for this is that, unlike the telephone system, the Internet employs a packet switched transport layer. There is no signaling because circuits do not need to be established in advance of data transfer as in circuit switched transport. Each packet contains sufficient information so that packet switches (routers) can forward it to its ultimate destination.

The basis of the Internet is a suite of standards that are ubiquitous. Indeed TCP/IP[6] equipment can be and is used in corporations and over WANs. Over the years this has fostered fierce competition yielding network equipment that is relatively inexpensive when compared to the proprietary equipment and national protocol variations that make up the telephone network. Another difference with respect to the telephone system concerns regulations. Unlike the telephone system, Internet service is largely unregulated. Entry into Internet-based businesses is possible with a relatively low investment. Thus the Internet provides and environment conducive to rapidly evolving, market-based businesses.

The widespread applicability of the standards of the Internet has resulted in a large number of independent software developers producing innovative applications, including those applied to telephony. A great amount of research effort focusing on real-time applications on the Internet has produced an exciting opportunity for a next generation telephone system. This next generation telephone system will be based on the core Internet technologies of IP, TCP and UDP. There is an important distinction between building the next generation telephone system on the Internet and building it on Internet technologies. There are too many technical and management problems to overcome to use the actual Internet. Instead next generation telephone companies will build private data networks based on Internet or IP technology.

The monopolies of long distance telephony are gone, Internet technologies have matured, and it is time for the next generation telephone system to be built as a Voice over IP (VoIP) system.

IP Telephony Beneficiaries

A VoIP telephone system can provide basic telephony services at a lower cost. A VoIP telephone system can provide profits and growth for the companies that provide the networks and equipment for the networks. Almost over night new companies such as Qwest[7] , and Level3[8] have formed to provide integrated data and telephony services over their own managed IP networks. The advantages that these companies seek to exploit are inexpensive packet switching equipment, rapidly increasing performance, and lower cost software development.

Furthermore, there will be new opportunities for software development companies to provide innovative telephony applications that take advantage of the World Wide Web and computer/telephony integration. For example product support call centers are looking towards “click to call”. This “click to call” also has applications in electronic commerce as well.

IP Telephony Challenges

There are three main challenges in the path from the Internet technologies of today to a VoIP telephone system. The first is that call-signaling capability needs to be brought to packet switching. The second is that quality of service must be controlled. The third challenge is building a converged PSTN/VoIP network. The transition from PSTN to VoIP will be a gradual process because of significant technical and business issues to be solved. Since VoIP and PSTN networks will coexist for many years, a converged network will need to be built in order to bridge the gap between the two. This converged network will allow calls to originate on a VoIP network and terminate on a PSTN network (and vice versa). This converged network will make use of VoIP gateway devices to bridge the two networks. Figure 2 presents a conceptual diagram of a converged network.

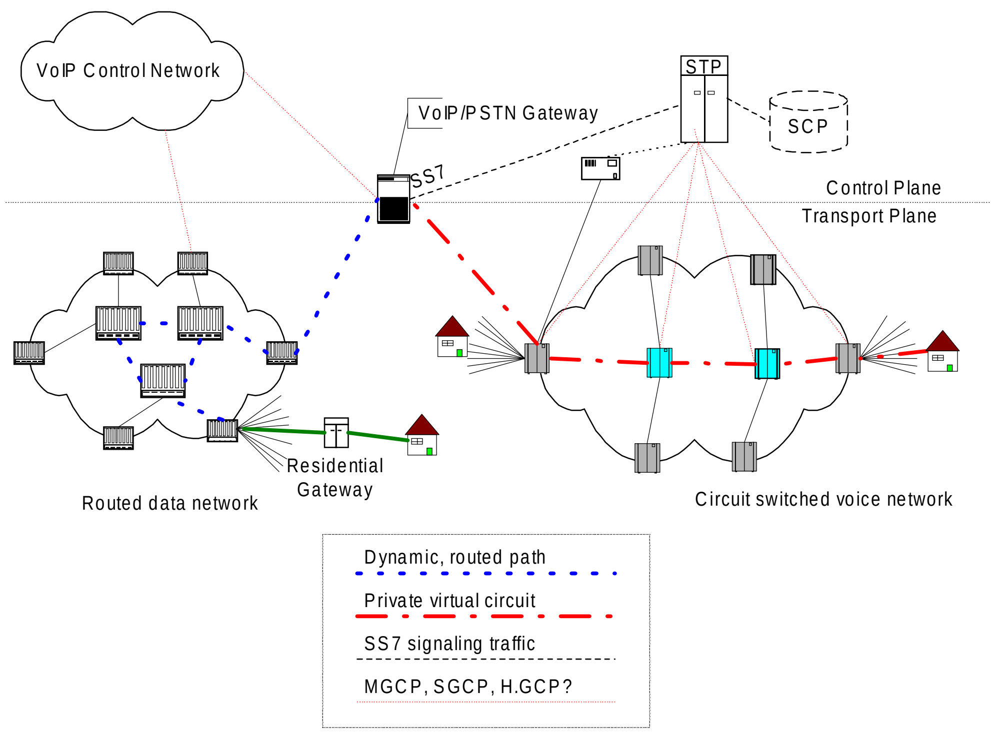

Figure 2: Converged architecture

There are efforts underway all over the world focused on these problems. This paper explores the topic of call signaling in the next generation telephone network.

IP Telephony Signaling

As described previously, IP networks do not need nor use call signaling. Telephony applications, on the other hand, introduce the requirement for signaling into IP networks because operating parameters for the call must be established prior to data transfer. For example, called and calling parties must establish the following:

- Encoding mechanism for the audio or video data
- Transport addresses to be used to transfer voice/video data
- Bandwidth requirements
- Authorization for initiating and accepting a call
- Call transfer and call diversion
- Location of the called party

In addition, call signaling must provide for an interface between the existing telephone system and the IP telephony system. To provide this signaling functionality into a network not inherently set up for it, Session Initiation Protocol (SIP) can be used. The remainder of this paper focuses on the operation and characteristics of SIP.

Introducing SIP

SIP (Session Initiation Protocol, a "Proposed Standard" described in IETF RFC 2543) is a text-based protocol that leverages the power of the Internet by borrowing such common elements as the format of HTTP, Domain Name System (DNS), and email style addressing. Further, SIP commonly employs the Session Description Protocol[9] or SDP for specification of the session parameters (although this is not a requirement). SIP provides the necessary protocol elements to provide services such as call forwarding, call diversion, personal mobility, calling and called party authentication, terminal capabilities negotiation, and multicast conferencing.

The most fundamental SIP operating model is one in which two SIP user agents (UA) communicate directly. User agents may be LAN telephones, computer based end-user applications, or gateways interfacing to the PSTN. The called UA may accept the invitation, acknowledging it with a response of "OK". Finally, the calling UA will "close the loop" with the called UA by sending an acknowledgment back to the called UA. Figure 3 depicts this call setup process. Audio and/or video data exchange follows the call setup process.

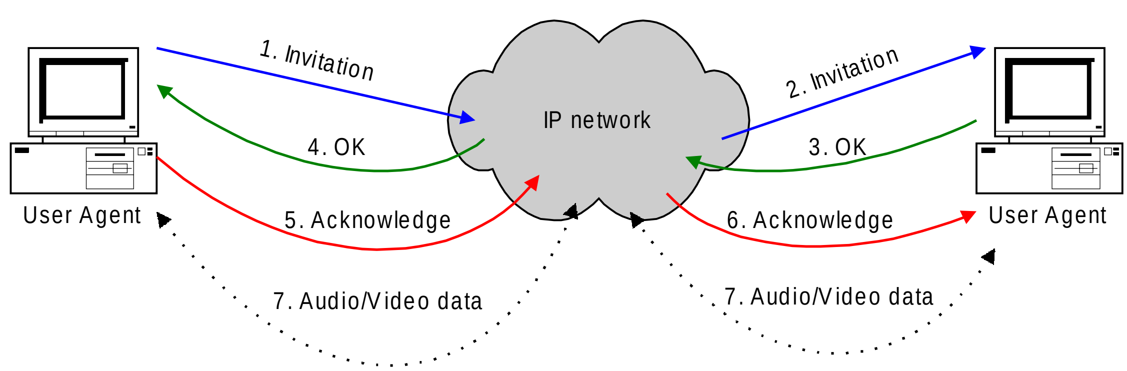

Figure 3: SIP operation

The addition of separate SIP servers to the IP network yields a more scalable architecture. The SIP server supports SIP-based telephony by providing a single access point for locating clients, mapping friendly names to addresses, routing signaling messages between user agents, and redirecting requests. There are two types of SIP servers, the *proxy server* and the *redirect server*. SIP operation is dependent upon which type of server is used. In the proxy model, the SIP proxy server is the only point of contact that the UAs have for signaling messages. In the redirect model, the SIP redirect server lets the calling UA know the location of the called UA and then gets out of the way of subsequent signaling messages. The following sections will describe in greater detail the operation of SIP using the proxy and redirect models.

SIP Protocols

SIP provides the basic elements of telephony: call setup and termination, call configuration, and data transfer. This is accomplished using SIP for call setup and termination portion, SDP to describe call configuration, and RTP for data transfer. RTCP is also used for data stream management.

SIP can run over any datagram or stream protocol such as UDP[10] , TCP, ATM, and frame relay. SIP is commonly run over TCP/IP because of inexpensive widespread connectivity, directory services, naming services, and a widely known development environment.

The audio and video data streams are transported using the Real-time Transport Protocol[11] (RTP) over UDP. SIP call signaling messages can be carried over UDP or TCP, with UDP being the preferred method because of its better performance and scaleability. One important consideration when using SIP over UDP is that the entire message should fit within a single packet. If a SIP message is fragmented into multiple datagrams, the probability of losing the entire message increases with the number of fragments. When SIP messages are being transmitted over a WAN, the retransmissions that result due to lost fragments can seriously degrade call signaling performance. The default port for SIP is 5060 although any available user port may be used. The port to be used for RTP/RTCP is specified in SIP call signaling messages. Figure 4 shows SIP over TCP/IP.

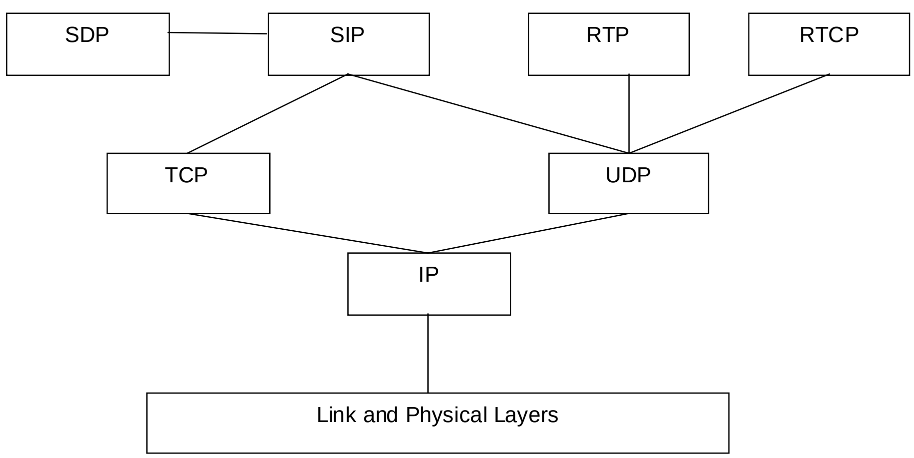

Figure 4: SIP protocol stack.

SDP

SIP commonly makes use of the Session Description Protocol (SDP) to describe the attributes of SIP sessions. SDP parameters are encapsulated as the message body of a SIP request. SDP plays a similar role as that of H.245 in the H.323 world. Like SIP, SDP headers are encoded with ASCII text. The SDP headers are of the simple form <type>=<value>. The <type> is always a single character and <value> is a text string whose format is dependent on <type>.

SDP is not really a protocol as much as it is a format for describing multimedia sessions. SDP headers specify:

- Session name and purpose
- Time(s) the session is active
- The media comprising the session
- Transport address and media format of the session
- Bandwidth to be used by the session
- Contact information for the person responsible for the session

A key component of SDP is the description of the media of the session. SDP media descriptions include:

- Media type (audio, video)
- Transport protocol (UDP, TCP, RTP)
- Media format (H.261, MPEG, etc.)
- Multicast address for IP multicast sessions
- Transport port for IP multicast sessions
- Remote address for IP unicast sessions
- Transport port for IP unicast sessions
- Session start and stop times

RTP

SIP sessions use RTP for end-to-end transport of audio and/or video information. The main features of RTP are the sequence number, the timestamp, and the payload type:

- Sequence number – used by the receiving client to detect lost packets and to "play" the audio or video packets in the correct order. This is important since, with UDP, there is no guarantee that the packets will arrive at the receiving client in the same order as they were transmitted (if they arrive at all).

- Timestamp - used by the receiving client to "play" the packet stream using the same timing that was used during transmission. This is critical as the packets may experience varying amounts of delay as they are forwarded through the network. Delay variations, or jitter, result in decreased audio or video quality.

- Payload type - indicates the encoding technique that was used to encode the audio or video information. The encoding technique is chosen to optimize quality or bandwidth usage.

RTCP

The functionality of RTP is augmented with the Real-time Transport Control Protocol, or RTCP. The purpose of RTCP is to provide feedback to all participants in a session about the quality of the data transmission. RTCP accomplishes this with periodic transmission of reports containing reception statistics. Reception statistics include the fraction of packets lost since the last report, the total number of packets lost since the last report, and the inter-arrival delay variation (jitter). Clients may use the information provided by RTCP to control adaptive encoding algorithms. In addition, the information is useful to network technicians for fault diagnosis.

SIP Operation

Earlier in this paper, SIP's basic operation was described. This section describes this in more detail. There are two SIP call models, the proxy model and the redirect call model. The calling user agent sends an invitation to the called user agent directly or through the SIP server. SIP user agents locate the SIP server by a configuration parameter similar to the proxy-server parameter of Internet browsers.

Proxy Server Operation

The proxy call model makes use of a SIP proxy server. This proxy server plays a role similar to that of the HTTP proxy server in an HTTP system. The proxy server routes signaling messages between the called and calling user agents. RTP audio or video packets are sent directly between the user agents after the call has been established. Figure 5 presents a typical call signaling procedure using a SIP proxy server. Appendix A presents additional details on the SIP messages used in this process.

Redirect Server Operation

The redirect server model makes use of a SIP redirect server. The SIP redirect server informs the calling UA of the SIP URL for the called UA. The calling UA then proceeds to set up the call directly with the called UA. Figure 6 shows the steps in a typical call. Appendix A presents additional details on the SIP messages used in this process.

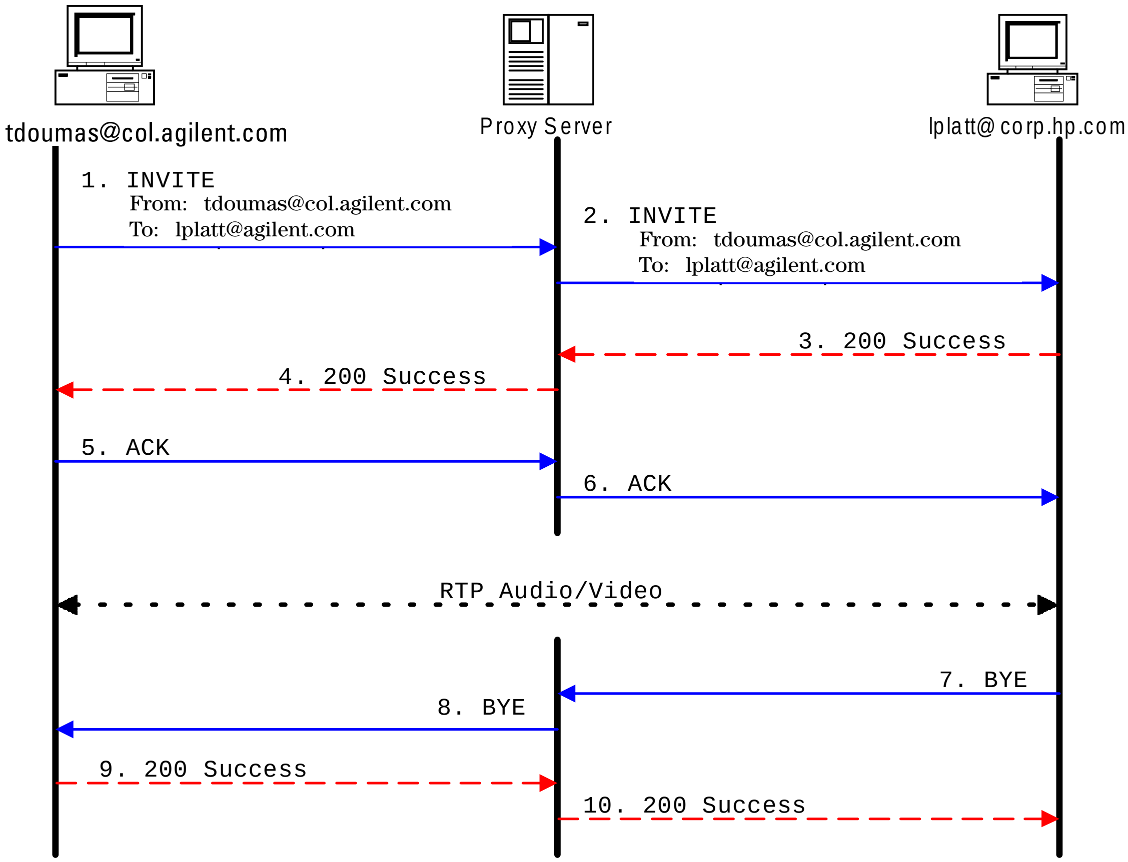

Figure 5: Proxy server operation.

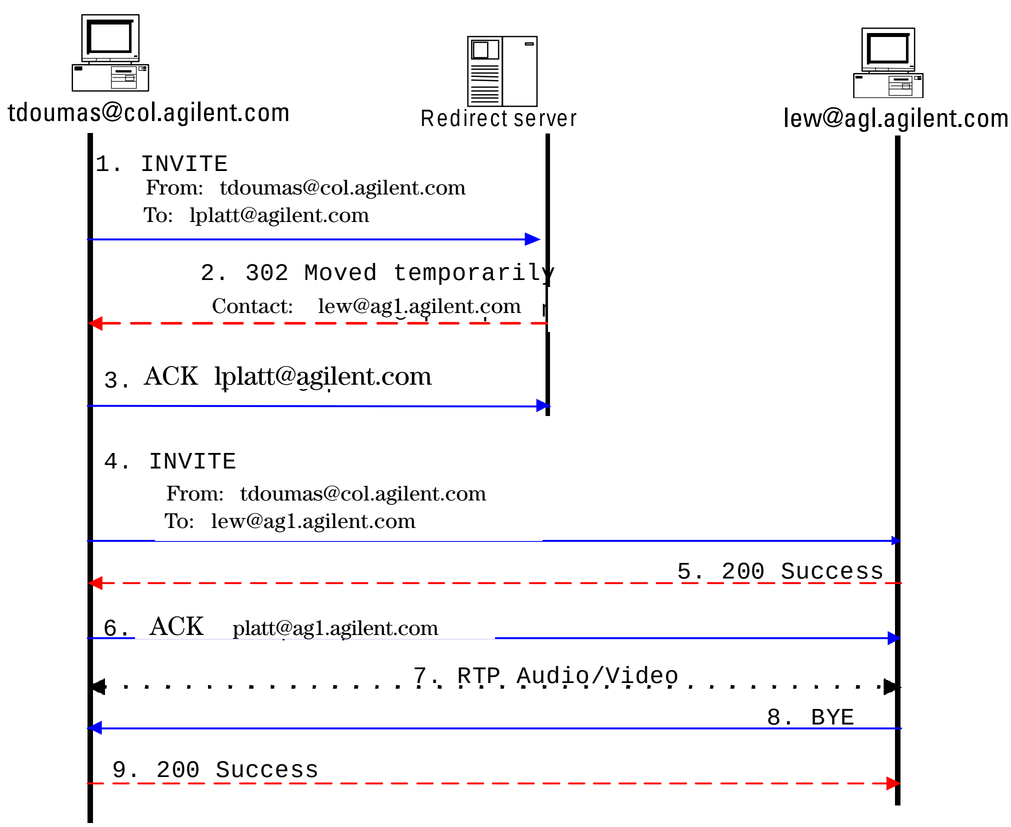

Figure 6: SIP redirect operation

SIP Addressing

The preceding figures presented familiar looking addresses for SIP clients. SIP clients are identified by a SIP URL which follows the "user@host" form of email addresses. The user part may be a user name or a telephone number. The host part may be a domain name, a host name or a numeric network address. Some typical SIP URLs are:

• sip:graceland@ixlmemphis.com (host independent)
• sip:bill@whitehouse.gov (host specific)
• sip:+1-800-555-1212@information.att.net

A common user need might be to call a person for whom the SIP URL is not known. It is easy to imagine a process of using World Wide Web search engines and directory services to resolve a name. For example, let's say that a user wants to call the governor of Minnesota. A WWW search engine might turn up the common name "Jesse Ventura". A directory service would resolve the name to a host-independent URL such as "jesse.ventura@minnesota.gov". A SIP server would resolve the host-independent URL to a host-specific URL such as "sip:jventura@governorsmansion.minnesota.gov". Ultimately, a DNS server would resolve the address to a specific numeric IP address.

SIP URLs may also specify the specific port on the host, the transport protocol (e.g. UDP) and a multicast address.

SIP vs. H.323

This paper could not assert that SIP is a viable alternative to H.323 unless a comparison is made between the two. While SIP, together with SDP, provides the same basic call signaling, call control services, and supplementary services as H.323, there are significant differences which can affect when one VoIP standard is chosen over the other.

Table 1 presents the various telephony services and their associated protocols for H.323 and SIP. It is clear from the table that H.323 makes use of more protocols and therefore must define the interactions between multiple protocols. The net result is that more effort is required to implement an H.323 device. In addition, the computing resources for an H.323 device are greater than that of a SIP device. In fact, a fully functional SIP client can be implemented with only two months of engineering effort[12] .

Service/Element	H.323 protocol	SIP protocol
Friendly name mapping	RAS	Existing directory service
Server discovery	RAS	Domain Name System
Authentication, security	RAS, H.235	SIP/SDP, web infrastructure
Call signaling	Q.931	SIP
Terminal capabilities exchange	H.245	SIP/SDP
Supplementary services	H.450.1, 2, and 3	SIP/SDP
Audio/Video transport	RTP	RTP
Codecs	ITU-T only	Any IANA register.
Session descriptions	H.245, ASN.1 PER	SDP

Table 1 SIP/H.323 protocols

A comparison of specification quantity demonstrates another measure of the difference between SIP and H.323. Table 2 presents a comparison of the size of specifications of SIP and H.323. Again, the scope and size of H.323's specification suggests complexity and difficulty of implementation.

H.32		SIP	
Specification	Length (pages)	Specification	Length (pages)
H.323 Version 2	117	SIP	~100
H.225.0 Version 2	171	SDP	42
H.245 Version 3	354		
X.691 (ASN.1 PER)	70		
H.450.1	22		
H.450.2	47		
H.450.3	65		
Totals	846		150

Table 2: SIP/SDP Complexity

Because it is text-based, SIP simplifies the implementation task. For example, message headers and SDP parameters may be typed into a file and read in by Perl or Tcl scripts to quickly create telephony services. On the other, hand implementation of an H.323 based device requires an ASN.1 PER encoder/ decoder. This is a formidable undertaking requiring not only the development of the encoding and decoding software but also the external specification of the semantics. The code space requirements will restrict the use of H.323 in inexpensive client devices. In addition, simple scripts cannot easily manipulate binary ASN.1 PER encoded data. Finally, SIP's adaptability allows interoperability between newer and older versions of the protocol. User agents of different versions can agree to use the simpler features of the older protocol. Since H.323 versions are fully backward compatible with older versions, implementations increase in size with each new release.

The use of TCP for the transport of call signaling in H.323 leads to high connection set up latency and scaleability problems. H.323 gateways are required to maintain state information for every connection (thousands or tens of thousands of TCP connections). The user of UDP in SIP allows not only faster call set up but also stateless gateways. Version 3 of H.323 is designed to address the call set up latency issue by allowing UDP for call signaling (H.225.0 and H.245).

Testing SIP

Even though SIP will likely provide easier implementation, problems will still occur. This section presents some testing issues related to SIP systems, with the focus on the problems that might occur during the call setup procedure. The call setup procedure is the primary point of failure in all IP telephony systems because call setup procedures involve multiple messages with each message containing multiple parameters. Call setup fails if the messages are not transmitted in the proper order with all of the mandatory parameters and compatible optional parameters present. Implementations from vendors often embody unique interpretations of the specification because the specification never perfectly and unambiguously defines the protocol. SIP is no exception.

SIP Proxy Address

In either of the SIP operating models, SIP user agents must be properly configured with the name or address of the appropriate SIP server. SIP relies on the Internet Control Message Protocol (ICMP) to return error information to the client in the event of an unreachable server. In analyzing a SIP sequence, one might see the INVITE message transmitted and an ICMP "Host unreachable" or "Network unreachable" message returned. This might indicate that the IP address configured for the SIP server is incorrect.

Troubleshooting such a problem can easily be accomplished with a protocol analyzer that has the capability to search for and decode ICMP messages.

Called UA URL

If the IP address of the SIP URL for the called user agent is incorrect, then the INVITE message that is forwarded from the SIP proxy server would result in an ICMP message similar to that in the preceding example of an incorrect SIP proxy address.

The SIP URL may include a UDP port number for the called UA. If the UDP port number is not included, then the assumption is that the default UDP port of 5060 is to be used. In the event that the called UA is not using the port number expected by the calling party, an ICMP message will result. In this case, the ICMP message would indicate "Port unreachable".

Proxy-Specific Problems

There are specific rules for the behavior of SIP proxies. One key rule is that proxies must not reorder or modify fields in the SIP header. This includes the restriction that proxies must not change how fields are split across multiple lines.

A simple comparison of the fields transmitted to the proxy with the fields that the proxy forwards to the called client would illuminate such a problem with the implementation of a proxy. This comparison could be easily accomplished with a protocol analyzer that can decode and present the fields of SIP in a user-friendly manner.

Message Size Problems

When running over UDP, SIP messages will generally be small enough to fit into a single packet. If they do not, then fragmentation at the IP layer will break the message up into multiple datagrams. Although SIP supports fragmentation, the probability of a lost datagram increases with the number of datagrams. Since the entire message must be retransmitted if a single datagram is lost, fragmentation would negatively impact performance and reliability. Packet sizes that are at the limit of what Ethernet networks are set up to handle would be indicators of fragmentation and possible SIP problems. Further analysis of the IP header using a protocol analyzer would show whether problems were being caused by fragmentation when using SIP over UDP. Timing information provided by a protocol analyzer can be used to indicate whether retransmissions occurred in a timely manner.

DNS Errors

A normal part of SIP operation is the DNS query that is used to resolve a host name to a numeric IP address. If the query fails, the DNS server returns the value –1 which, when converted to an IP address, yields 255.255.255.255 or a broadcast address. If the SIP implementation does not test for the DNS query failure value of –1, and simply accepts the return value as the valid IP address for the host, INVITE or REGISTER messages are broadcast instead of the preferred uni-cast. This then can have an adverse affect on overall network performance. A protocol analyzer with measurements designed to identify excessive broadcast traffic may be used to easily resolve these types of problems.

Cseq Handling

The Cseq header in SIP is used to determine the sequence of messages on different call legs. The number space for Cseq is unique for each end of a call leg. In addition, the header needs to be incremented even on BYE messages. Proper implementation of the Cseq handling state machines requires care and is essential for interoperability. The Cseq header number is easily seen with a protocol analyzer.

Compact Notation

Standard SIP headers are fairly descriptive. Decreased packet sizes may be achieved by using the compact form of the headers. For example, the From header may be shortened to f. Interoperability relies on all equipment properly accepting the compact form notation. Compact form notation is easily identified with a protocol analyzer.

Agilent Technologies Solutions

Agilent Technologies' Advisor provides support for SIP troubleshooting over LAN, WAN, and ATM interfaces. Protocol decodes provide a detailed view of the fields of each protocol. Figure 7 presents a typical SIP message sequence. The summary view shows a high level view of the sequence, while the detailed view shows each SIP message header. Note that in this example, the RTP messages that were in order transmitted between the ACK and the BYE messages were filtered out to highlight the signaling messages.

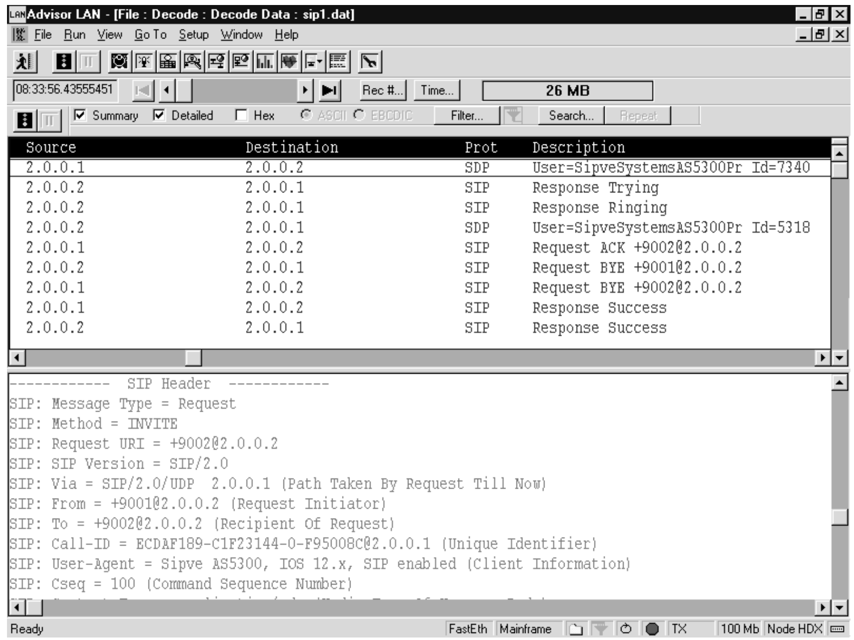

Figure 7: SIP message sequence

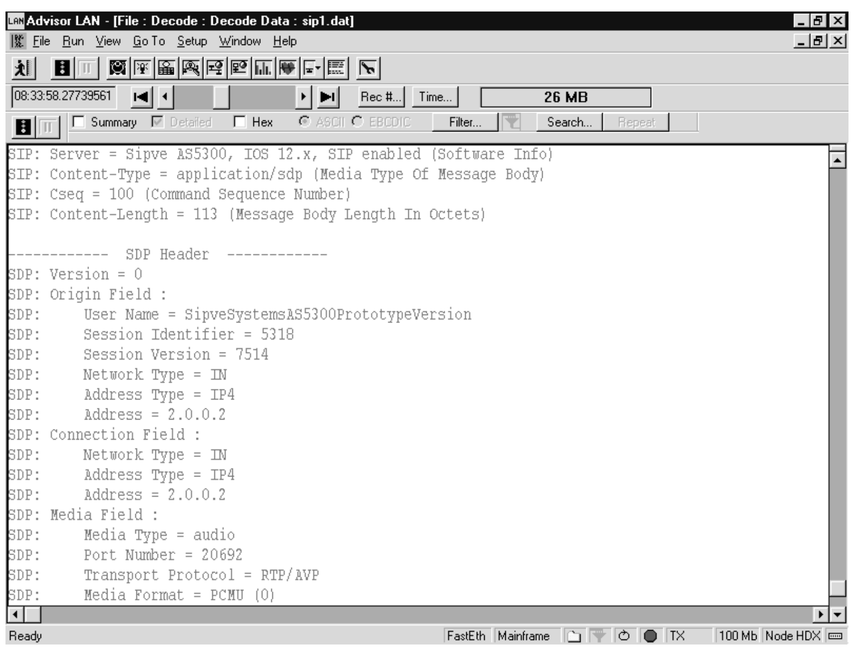

Figure 8: shows details of the SDP headers.

In addition, the Advisor provides filtering and searching to allow the network troubleshooter the ability to display the desired frames. Statistical analysis may be used to monitor the health of the network and to “drill down” to the connections or protocols that are consuming bandwidth.

Conclusion

The future of VoIP has many facets, and there are many markets with varied value propositions. The small business, the Fortune 500 company, and the residential consumer have different needs, and consequently VoIP solutions may take different forms in each case. For example, the telephony market for the small business requires inexpensive, turnkey solutions for VoIP. A new service from an Internet Service Provider (ISP) might present the implementation of VoIP for the small business. In the case of the Fortune 500 company that already has a worldwide, managed IP data network and a separate voice network, VoIP technology may allow this company to eliminate the voice network altogether.

Traditional carriers will face competition from next generation telephony carriers, and each company will be attempting to provide the highest value service package including data, voice, and Internet access. The networks that will be built to address these needs will interface to traditional networks such that calls can be originated and terminated regardless of the technology used. This ‘converged network’ concept will need to be implemented as the evolution towards IP based telephony progresses.

The real demand for SIP is being fueled by concerns about H.323. These concerns (call set up latency, scaleability, and complexity) have opened a window of opportunity for SIP. Vendors of VoIP products currently involved with H.323 are expanding their development efforts to take advantage of this opportunity. Early deployments of SIP based telephony have begun at the enterprise level. A SIP interoperability event held during April 1999, where major equipment vendors gathered for two days of intensive testing, demonstrated that interest in SIP is translating into commercial product development.

SIP is currently an IETF “Proposed Standard” described in RFC 2543. The next milestone in the IETF standardization process is the designation of “Draft Standard”. At this time, SIP is probably one year away from being a Draft Standard. The challenges ahead include resolving ambiguities, and filling in omissions in the specification. Progress requires additional analysis and review of the specification as well as additional interoperability events.

Product development progress is, of course, tightly coupled to the standardization process. Real progress is first defined by basic interoperability (call set up and clear), followed by advanced capability (authentication, encryption, and call transfer) and finally scaleability, reliability, and performance testing. Somewhere in all of this is the need to interface to the PSTN. Stable and robust standards are the basis for the success of this entire process. The results from the interoperability event mentioned earlier indicate product development is just short of achieving the first step.

Future interoperability events will probably place product interoperability just short of the second step - that is, solid basic interoperability and preliminary advanced capability interoperability.

Unless major breakthroughs occur that conclusively eliminate all fear, uncertainty, and doubt in the viability of H.323, SIP will take hold and capture a solid position in the VoIP world. SIP has the capability to satisfy the range of needs from the inexpensive IP telephony appliance to the high demands of major telephony carriers.

Appendix A: SIP Messages

SIP defines two basic message types, the request and the response. Request messages are used to initiate, confirm, modify and terminate calls. Response messages are used to convey either provisional information such as “ringing”, or “moved temporarily” response, or final information such as “busy” or “does not exist”. Table 2 presents the SIP request methods, and select messages are described in subservent sections.

Request method	Purpose
INVITE	Initiate a session
ACK	Confirm the final response to an INVITE
BYE	Terminate a session
CANCEL	Cancel searches and “ringing’
OPTIONS	Communicate features supported
REGISTER	Register a client with a location service

Table 3: SIP Request methods

The INVITE Message

The INVITE message is sent by the calling client to initiate a call with another client. There are 5 mandatory parameters for the INVITE message. Table 3 presents these mandatory parameters.

Parameter	Description
Call-ID	Uniquely identifies a particular session.
CSeq	A monotonically increasing sequence number used to identify the sequence of requests associated with a given Call-ID.
From	A SIP URL that identifies the initiator of the request. May include a "friendly name" (e.g. John Smith).
To	A SIP URL that identifies the recipient of the request. May include a "friendly name".
Via	Indicates the path taken by the request so far. The Via parameter is used to prevent looping of requests, assures that replies take the same route as requests and assists in unusual routing situations.

Table 4: INVITE parameters

Response messages

SIP Response messages indicated either call progress information or final status information. Response messages contain a Status-Code and a Reason-Phrase. The Status-Code is a three digit integer that indicates the outcome of the request. The Reason-Phrase provides a textual description intended for humans. Table 4 presents a summary of SIP response message categories and their use.

Status-Code	Category	Example information
1xx	Informational	trying, ringing, call is being forwarded, queued
2xx	Success	OK
3xx	Redirection	Moved permanently, moved temporarily, etc
4xx	Client error	Bad request, unauthorized, not found, busy, etc
5xx	Server error	Server error, not implemented, bad gateway, etc.
6xx	Global failure	Busy everywhere, does not exist anywhere, etc.

Table 5: Response codes

References

[1] Information Sciences Institute University of Southern California, “Internet Protocol”, Request For Comment 791, September 1981.

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Additional information on SIP may be found at:

http://www.cs.columbia.edu/~hgs/sip

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About The Author

Thomas Doumas has been a member of the technical staff at the Hewlett-Packard Network Systems Test Division (NSTD) since 1984, now Agilent Technologies. Thomas has been involved in the management and software development of network test equipment for WAN, ATM, and LAN networks. Thomas holds a BSEE and MSEE from the University of Wisconsin-Madison.

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