SIP PROTOCOL: IMPACT ON TELEMATICS INDUSTRY AND EDUCATIONAL ASPECTS

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Abstract: *Session Initiation Protocol (SIP) is an application-layer signaling protocol that can establish, modify and terminate interactive multimedia sessions over Internet Protocol (IP) networks. Actually, SIP has the potential to support the development and rapid deployment of new telematic applications on wireline and wireless networks. SIP uses the Internet model and maps it onto the telecommunications world, thus, offering convergent voice and data services.*
In this paper, the impact of SIP protocol on the telematics industry will be presented, while educational aspects of the protocol will be examined by laboratory experimentation on a SIP network architecture (set up at the Technological Educational Institute of Crete/Department of Electronics).Various implementation issues of SIP components (user agent, proxy server, registrar server) and communication scenarios will be discussed and evaluated.

Keywords: *Session Initiation Protocol (SIP), Internet Engineering Task Force (IETF), convergent communications, telematics, hands-on experimentation*

1. INTRODUCTION

Session Initiation Protocol (SIP) is an application-layer protocol for controlling (i.e. to initiate, modify and terminate) real-time multimedia unicast or multicast network sessions. From its initial use in Internet telephony, SIP or extensions of the protocol is spreading into many new telematic application areas and services, including advanced telephony applications, interactive multimedia conferencing, instant messaging (IM), presence and event notification applications or management of other session types (such as distributed games, voice-enriched e-commerce, web page click-to-dial, etc.). In setting up sessions, SIP is offering services similar to traditional circuit-switched telephony signaling protocols such as Q.931, Q.SIG or ISUP, but in an Internet context. Actually, SIP has been designed to be a general purpose open protocol that is independent of the packet layer. Furthermore, SIP is an access-independent protocol offering seamless service capabilities between fixed and mobile networks and supporting personal mobility (i.e. users can maintain the same identifier even as they change attachment points to the network or use different devices). Thus, SIP can be regarded as a key element in making the promise of telecommunication networks and telematic applications convergence a reality.

The main forum of SIP standardization is in the Internet Engineering Task Force (IETF), which is the primary standards body for Internet protocols. Actually, SIP is part of the overall IETF multimedia architecture that has emerged over the past few years. This architecture includes the Real Time Transport (RTP) for transporting audio, video and other time-sensitive data, the Real Time Streaming Protocol (RTSP) for setting up and controlling on-demand media streams, the Session Description Protocol (SDP), the Media Gateway Control Protocol (MGCO) and Megaco for controlling media gateways, as well as a suite of resource management and multicast address allocation protocols, [6]. All the aforementioned protocols serve as the signaling, media transport and Quality of Service (QoS) protocols that enable Voice over Internet Protocol (VoIP) and other multimedia communication, as depicted in Figure 1, [7]. Therefore, from an educational point of view, treatment of SIP protocol includes topics from traditional communications area as well as from computer networking. As it concerns teaching and experimentation of networking concepts and protocols, a theoretical and mathematical orientation accompanied by simulations is often not adequate to provide a through understanding. On the other hand, hands-on experimentation is ideal to supplement the theory and demonstrate network protocol issues in real life situations.

The purpose of this paper is twofold: At first to report on SIP protocol influence on telematics industry (Section 3), and, second, to provide (Section 4) hints for teaching SIP-associated issues through experimentation in the framework of a course developed at the Technological Educational Institute of Crete (TEIoC). In Sections 2 and 5, a technical overview of the SIP standard and conclusions are presented, respectively.

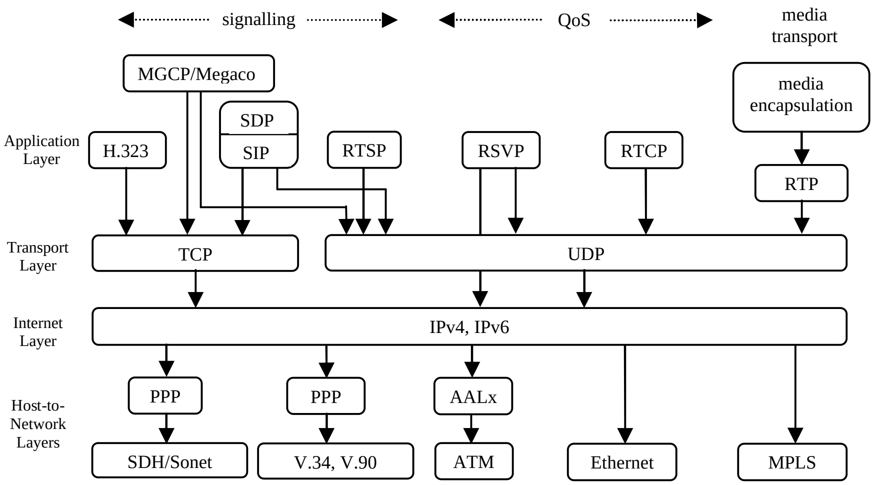

Figure 1: Protocol stack for IP communications

2. OVERVIEW OF SIP STANDARD

2.1 SIP ARCHITECTURE AND OPERATION

The services provided by the SIP protocol include, [5]:
(a) User location: determination of the end system or device to be used for communication
(b) Call setup: ringing and establishing call parameters at both called party and calling party
(c) User availability: determination of the willingness of the called party to engage in communications
(d) User capabilities: determination of the media and media parameters to be used
(e) Call handling: the transfer and termination of calls

Four logical types of entities participate in a SIP system architecture: user agents, registrars, and proxy and redirect servers. User agents (UA) are endpoint devices that terminate the SIP signaling; they can be clients (UAC) that initiate request, servers (UAS) that respond to requests, or more normally a combination of the two. Registrars are specialized UASs that keep track of users within their assigned network domain (e.g., all users with identifiers x@chania.teicrete.gr register with the registrar in the chania.teicrete.gr domain). Proxy servers are devices in the signaling path between UAs that route requests on towards their destination. In addition, they may add parameters to the requests and may reject requests, but they may not initiate requests or respond positively to any request that they receive. A redirect server also receives

requests, and determining a next-hop server. However, instead of forwarding the request there, it returns the address of the next-hop server to the client. It should be mentioned that it is quite common to find registrar, proxy, and redirect servers implemented within the same physical SIP entity.
The SIP protocol is a client-server protocol that shares many of the HyperText Transfer Protocol (HTTP) features. In particular, it is using Uniform Resource Identifier (URI) or Uniform Resource Locators (URLs) for addressing and, as in HTTP, the UAC requests invoke methods on the SIP servers. Requests and responses are textual messages, and contain header fields which convey call properties and service information. The aforementioned messages include INVITE (for inviting a user to a call), BYE (for terminating a connection between the endpoints), ACK (for reliable exchange of invitation messages), OPTIONS (for getting information about the capabilities of the call), REGISTER (gives information about the location of a user to the SIP registrar server) and CANCEL (for terminating the search for a user) messages.
The action that a communications device takes on receipt of a SIP request in not determined purely by the SIP protocol; it is also determined by the application. An application may decide to forward the request on to another server throughout the network for further processing (e.g., for authentication, billing or web browsing). The generic term for such a device is an application server (AS). From a SIP protocol point of view, an AS may behave as a UA, a proxy or a combination of the two, depending on the situation. Thus, an AS handles not only the SIP protocol, but also various standard protocols (SMTP, HTTP, RADIUS, etc.) in order to build media-blending applications and telematic sessions (Figure 2). Thus the adoption of the model of Figure 2 enables application developers to focus on the users' application as the service components are already available, [4].

2.2 STANDARDIZATION OF SIP

The IETF has set up three working groups to work on SIP protocol and its application, i.e. the SIP working group (responsible for enhancements to the core protocol), the SIPPING working group (for applications of SIP), and the SIMPLE working group (which covers IM and presence applications of SIP). Other IETF working groups whose areas touch on SIP include IPTEL (for Internet routing of telephone calls), MMUSIC (responsible for SDP), MIDCOM (for firewall and Network Address Translation (NAT) traversal issues of SIP), SPIRITS (PSTN-Internet telephony interconnection), and ENUM (Internet use of traditional Public Switched Telephone Network (PSTN) phone numbers).
However, several industry groups are also discussing how to standardize the use of SIP in their environment. These include PacketCable (for use of SIP for telephony over cable), 3G Partnership Project-3GPP (for use of SIP in 3G cellular networks), ETSI TIPHON (to ensure that SIP is suitable for deployable telephony applications), and Multi-service Switching Forum (MSF). All the aforementioned standardization

efforts potentially lead to conflicts (between the requirements of the traditional telecommunication carriers, who need to provide an end-to-end billable solution that meets their regulatory requirements and those involved in the less controlled environment of the Internet) as well as extensions of the base SIP standard, resulting in concern over interoperability of the SIP protocol. For example, MSF has defined SIP-T conformance levels and is now working to ensure that SIP can be deployed in large scale PSTN networks, while the multi-vendor SIP-B effort is to outline a set of advanced features for business telephony networks.

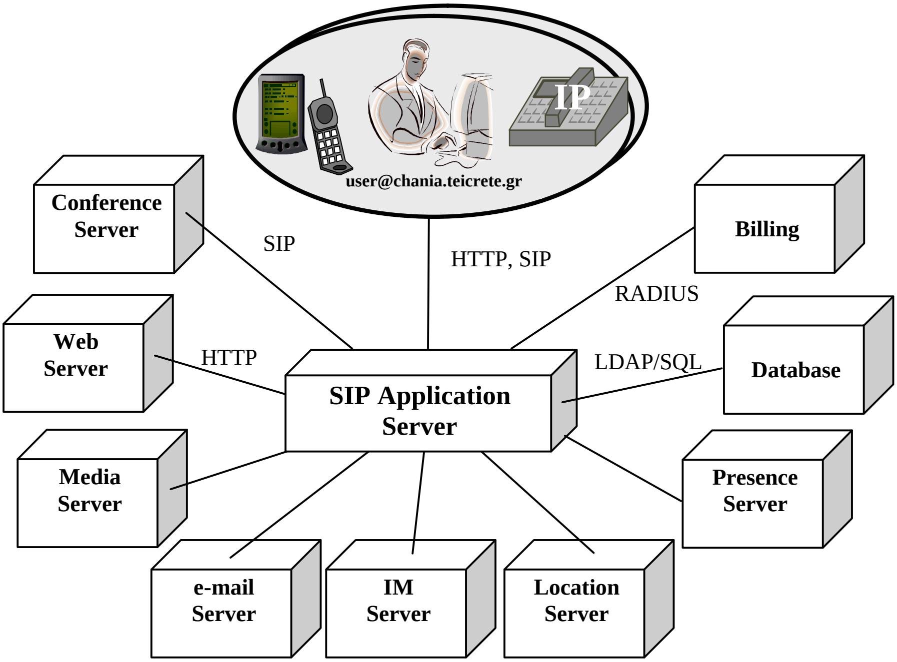

Figure 2: Functional architecture of a SIP application

3. SIP AND TELEMATICS INDUSTRY

Telephony is the most developed SIP application, due to the fact that the initial driver for SIP adoption was cost. However, as the monopoly of the telephony service providers has been reduced, prices have been dropped in many markets to a level where cost is no longer a significant factor. Actually, it is expected that future SIP adoption will not be driven primarily by cost, but by the new services that it can provide and the convenience offered by the underlying converged (fixed and wireless) telecommunications infrastructure. As it concerns SIP integration with the legacy PSTN and the Private Branch Exchange (PBX) market, some of the technical, interoperability, economical and regulatory issues that arise is as follows:

(a) A high level of interoperability with the PSTN Signaling System No. 7 (SS7) is required, as SIP cannot readily handle all PSTN features. Furthermore, a fully functional replacement of a traditional phone by a SIP phone in a circuit-switched TDM PBX communications environment, should be accomplished by interoperability of SIP protocol with the proprietary PBX protocols.
(b) An efficient numbering scheme is necessary for provision of a standard mapping between traditional phone numbers (E.164 standard) and SIP addressing. In such a way the creation of an integrated PSTN and IP telephony system.
(c) SIP protocol architecture and features have an impact on regulatory issues of IP telephony. These issues include (see [1] for more details) wire tapping (there is no longer a simple central point at which to monitor the calls) and emergency calls (due to the location-independent address identifying the domain of a SIP user).
(d) Although voice communication requires a fairly low bandwidth to be available on demand with consistent latency to provide good sound quality, the SIP standard contains no mechanisms for controlling network bandwidth and latency availability. Therefore, there exists the need of an end-to-end QoS comparable to that of PSTN and TDM voice networks to be provided to SIP phones connected to an IP-PBX, for example.
(e) While in traditional telephony an availability of 99.999 percent (the well known “fine-nines” term) has been achieved since decades by using expensive fault tolerant hardware, IP networks and Internet, in particular, are far away this level of availability. To make matters worse, SIP messages are, generally, forwarded by the UDP protocol, since the later avoids the TCP connection setup and teardown overhead. Actually, SIP attempts to provide availability using Domain Name Service (DNS) to reroute the SIP messages around failures. However, as other proprietary mechanisms and use of redundant hardware are also examined to allow reliable communication, the issue of availability for a SIP communications environment remains open.
The aforementioned discussion makes SIP introduction and its wide deployment a challenge for Internet Service Providers (ISPs), also. In order to get a return of investments, they must offer not only plain bit transport. Furthermore, we should take into account that the mature H.323 series of signaling protocols has been available for several years and ISPs have built out many large H.323-based networks. As a consequence, addition of VoIP gateways supporting the SIP protocol and interoperation with H.323 network deployments, should be addressed by ISPs. In such a way, new converged services will help ISPs differentionate themselves from the competition and expand their customer base beyond Internet access subscribers.
Convergence of IP networks and mobile networks cellular presents another challenge for SIP protocol exploitation to support call signaling and mobility management. As 3G networks have introduced support of IP mobility through the Mobile IP (MIP) protocol and the 3GPP has defined and standardized a network infrastructure (the IP multimedia subsystem (IMS)) based on SIP for supporting a multitude of services to 3G users, new technical and nontechnical challenges arise, [2]. It is expected,

however, that extensions of SIP protocol will cope with issues such as the integration of the application-layer (SIP) and IP layer (MIP) solutions within the scope of the IMS and with respect to the IPv4 to IPv6 transition. Furthermore, due to bandwidth limitations of the radio-based environment, compession of SIP messages, [3], for the provision of VoIP, IM and streaming services in 3G networks may be introduced.
Finally, the information technology is facing new challenges as it concerns software for SIP-based services as well as SIP phones and softphones (A softphone is a software-based telephone, usually operating on a PC). Traditional telephony services have been either hard-coded into switches, made available by application programming interfaces (APIs), or made configurable in limited ways by proprietary service creation environments. For SIP-based services, however, other approaches are available including language-based APIs (Jain, Parlay), dedicated languages such as the Call Processing Language (CLP) (an XML-based language that can be used to describe and control Internet telephony services), SIP common gateway interface (CGI) scripts (HTTP CGI compatible extensions to providing SIP services on a SIP server) or SIP servlets (a Java code that receives calls from the SIP server and instructs the SIP server how to handle requests).

4. EXPERIMENTATION ON SIP PROTOCOL

As IP-based communication starts assuming a dominant role in worldwide communications, higher level educational institutions need to keep up with technological developments and prepare their students and future engineers to use this technology and be able to adapt in subsequent advancements. Actually, as exemplified by the development and the early deployments of the SIP protocol, the large proliferation of communication, networking and information technology standards provides a daunting amount of theoretical material to be presented to students. Furthermore, the skills required to evaluate, integrate, and administer networking equipment as well as to develop telematic services, is considered important for students.
As it concerns teaching and experimentation of networking concepts and protocols, a theoretical and mathematical orientation accompanied by simulations is often not adequate to provide a thorough understanding. On the other hand, hands-on experimentation is ideal to supplement the theory and demonstrate network protocol issues in real life situations. Such a hands-on experimentation on SIP protocol is carried out during a laboratory activity of the course “Convergent Communication Systems and Services”, developed at the Department of Electronics/Division of Telecommunications of TEIoC. The testbed for laboratory experimentation consists of PCs, where the appropriate software has been installed (Figure 3). In particular, free software (www.sipcenter.com) publicly available (registration is needed on a regular basis, however) for research and development purposes was used to emulate a real SIP communications environment. Our SIP communications system is composed of three or more UAs, two SIP proxies and a registrar server. Various communication

scenarios are examined during students' experimentation, while the exchange of SIP messages is captured by a free software protocol analyzer (www.ethereal.com). In such a way students have a detailed view of the procedures taking place when establishing-terminating a SIP connection as seen by the proxy server (Figure 4), or when communication of two UAs through the two SIP proxy servers is taking place (Figure 5). Students have, thus, the chance to implement feature and applications supported by the SIP protocol (call forwarding, call hold, call forking, call reject, find me/follow me, etc.).

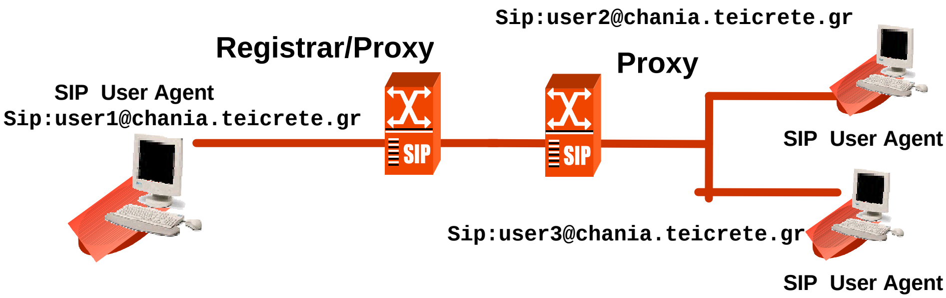

Figure 3: Experimental setup with SIP Signaling

Source	Destination	Protocol	Info
194.177.198.84	194.177.198.126	SIP/SDP	Request: INVITE sip:user3@chania.teicrete.gr
194.177.198.126	194.177.198.84	SIP	Status: 100 Trying
194.177.198.126	194.177.198.84	SIP	Status: 180 Ringing
194.177.198.126	194.177.198.84	SIP/SDP	Status: 200 OK, with session description
194.177.198.84	194.177.198.126	SIP	Request: ACK sip:194.177.198.126:12837
194.177.198.126	194.177.198.84	SIP	Request: BYE sip:user2@194.177.198.84
194.177.198.84	194.177.198.126	SIP	Status: 200 OK

Figure 4: Establishment-Termination of a connection as seen by the SIP proxy server

Source	Destination	Protocol	Info
194.177.198.86	194.177.198.126	SIP/SDP	Request: INVITE sip:user2@cometta.chania.teicrete.gr
194.177.198.126	194.177.198.82	SIP/SDP	Request: INVITE sip:user2@cometta.chania.teicrete.gr
194.177.198.82	194.177.198.84	SIP/SDP	Request: INVITE sip:user2@194.177.198.84:5062
194.177.198.84	194.177.198.82	SIP	Status: 180 ringing
194.177.198.82	194.177.198.126	SIP	Status: 180 ringing
194.177.198.126	194.177.198.86	SIP	Status: 180 ringing
194.177.198.84	194.177.198.82	SIP/SDP	Status: 200 OK, with session description
194.177.198.82	194.177.198.126	SIP/SDP	Status: 200 OK, with session description
194.177.198.126	194.177.198.86	SIP/SDP	Status: 200 OK, with session description
194.177.198.86	194.177.198.126	SIP	Request: ACK sip:194.177.198.126:5060;lr
194.177.198.126	194.177.198.84	SIP	Request: ACK sip:user2@194.177.198.84:5062
194.177.198.84	194.177.198.126	SIP	Request: BYE sip:papoutsis@194.177.198.86
194.177.198.126	194.177.198.86	SIP	Request: BYE sip:papoutsis@194.177.198.86
194.177.198.86	194.177.198.126	SIP	Status: 200 OK
194.177.198.126	194.177.198.84	SIP	Status: 200 OK

Figure 5: Communication of two UAs through two SIP proxy servers

5. CONCLUSIONS

Telematics industry is investing in SIP technology, as it offers an open environment for adding value for telecommunication carriers, ISPs, mobile operators, application developers and equipment vendors. SIP standard promises to offer converged communications for PC users, enterprises, and people on the move, through a variety of end-users devices (SIP phones, softphones, cellphones, PDAs, etc.). The simple, scalable, evolving and IP-based nature of the SIP protocol allows the implementation of advanced applications. For example, Erlanger Health Systems is developing software that will soon let nurses in a hospital to receive SIP-based instant messages from devices such as IV pumps, heart monitors or other medical devices.
From an educational point of view, TEIoC/Department of Electronics tries to keep its graduates fully aware of current and projected developments in communications technology. Thus, in order to connect theoretical material taught in an elective senior lecture course, experimentation and performance evaluation of the SIP protocol are carried out.

6. ACKNOWLEDGMENTS

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