TECHNOLOGY WHITE PAPER

SIP Technology in the Enterprise

Present enterprise communications systems and applications provide integration of voice and data services. Still, proprietary technology and architecture discrepancies make it difficult to implement and deploy new converged communications services.

The Session Initiation Protocol (SIP) brings user centric Internet Telephony. This model provides new converged communications services based on presence, mobility and user preferences. SIP technology promotes distributed architecture models, and seamless integration with IP technology. SIP is an open extensible Voice over IP (VoIP) protocol and is a framework for building voice and data communication applications. This article presents what SIP brings to enterprise communication applications and how customers can manage a smooth transition to SIP based services with the example of the OmniPCXIP communication server.

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SIP TECHNOLOGY IN THE ENTERPRISE

SIP technology will be the next step in IP telephony, bringing new Internet telephony services to enterprise users.

Introduction

What will be the impact of the emerging open Session Initiation Protocol (SIP) technology on enterprise communication systems and applications like Private Branch eXchanges (PBX), Internet Protocol (IP) PBXs, enterprise intranets, the Internet and portals?

In addition to using public domain carriers and the Internet, enterprises require Customer Premises Equipment (CPE), services hosted by service providers, or a mix of the two. Small and Medium sized Enterprises (SME), in particular, are willing to outsource some of their information technology and networking activities in a managed services mode. Nevertheless, whatever the model, enterprise users have the same needs in terms of desktop tools, applications and seamless integration with existing corporate resources.

SIP is an application-layer signaling protocol that can establish, modify and terminate interactive multimedia sessions over IP. The technology encompasses a suite of extensions to the protocol, several architectural models for communication applications, integration with other IP technologies, and a framework for building new communication services.

SIP promise significant improvements for business communications, first as a voice over IP (VoIP) protocol, and also because it integrates well with other business communication applications. Used in conjunction with other Internet technologies, SIP will bring new communication services to the user's office and mobile desktop, thereby increasing employee productivity while reducing operating costs.

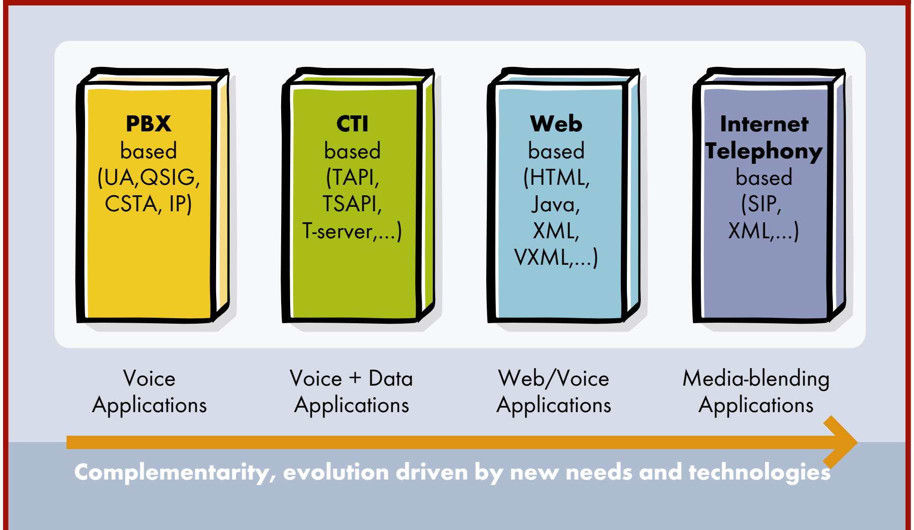
This article outlines SIP technology, explains "Internet Telephony enabled by SIP", looks at the factors that are slowing SIP deployment and considers the key elements for evolution to SIP. The Alcatel OmniPCX IP Communication Server offers a number of SIP features that will facilitate the smooth transition to SIP communications within an enterprise.

SIP Enables Internet Telephony

Figure 1 shows the evolution of communication applications in an enterprise environment.

Over the past couple of decades, traditional PBX voice applications (e.g. feature rich enterprise telephony, voice mail, call distribution) have been augmented by new Computer Telephony Integration (CTI) services, such as information pop-up on incoming call, click to dial and softphone (a PC-based telephone software application). Today, a new generation of unified communication applications based on web technology can be accessed from any web or voice terminal. The next step will be the introduction of media-blending[1] (converged) communication applications based on SIP technology.

Fig. 1 Evolution of communication applications in an enterprise environment



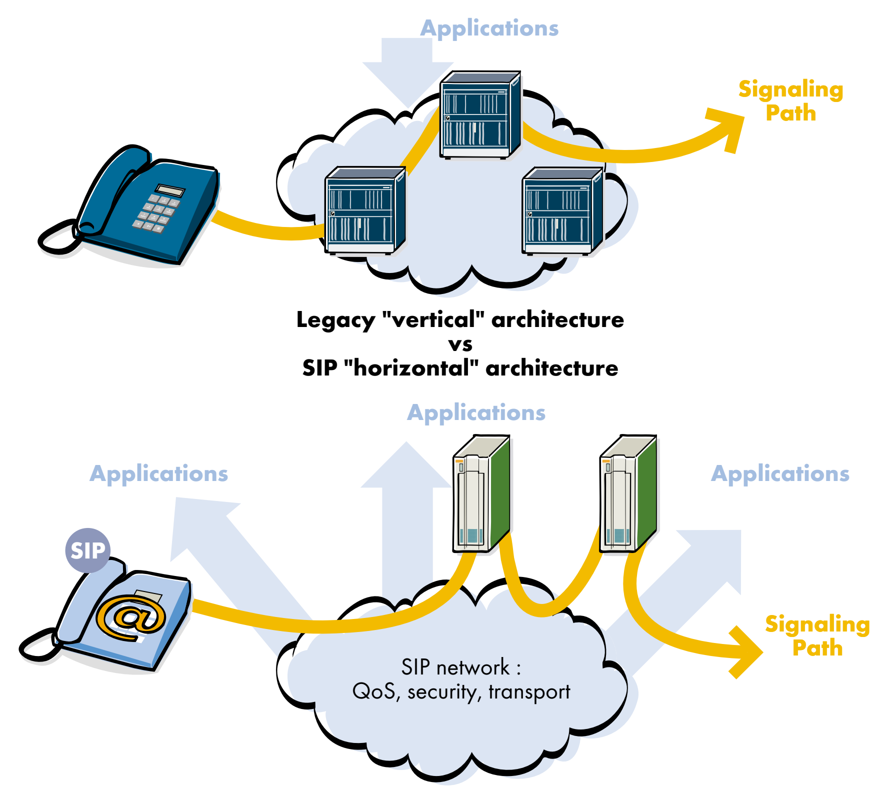
HTML: HyperText Markup Language **VXML:** Voice eXtensible Markup Language

1 Media blending refers to multimedia combined with the use of more than one type of terminal (mobile, PC, business phone, PDA, etc).

As shown in *Figure 1*, PBXs are based on protocols like User Access (UA) for proprietary phones, Q Interface Signaling (QSIG) for private networking, Computer Supported Telephony Application (CSTA) for CTI applications and IP for the latest IP Private Communication eXchange (PCX) generation. CTI servers are based on standard Application Programming Interfaces (API), such as the Telephony API (TAPI) and Telephony Services API (TSAPI), or on dedicated APIs (e.g. Genesys' T-server).

As the next evolutionary step after VoIP networks and IP telephony, legacy CTI applications and Internet services, SIP will bring Internet telephony to users' desktops. Internet telephony is the convergence of IP telephony, the Internet and various new technologies and standards, such as SIP and the eXtensible Markup Language (XML). It naturally integrates unified communications (unified messaging and telephony), user-centric media-blending communication and mobility services.

Fig. 2 **Comparison between the legacy "vertical" architecture (top) and the SIP "horizontal" architecture (below)**



QoS: Quality of Service

Internet telephony is neither device centric nor is it legacy telephony over the Internet (i.e. it is not simply VoIP) but a powerful evolution of telephony services using the full potential of Internet technology. By combining voice with IP services like presence, instant messaging, web and e-mail, it is possible to develop a wide variety of innovative applications. As an example, web conferencing is based on voice conferencing combined with web-based management, participant presence and data conferencing tools.

New media-blending communication applications are the real long-term driver for implementing SIP in the enterprise market.

Overview of SIP Technology

SIP is an application-layer signaling protocol that can establish, modify, and terminate interactive multimedia sessions over IP between intelligent terminals. It shares most of the features that made the HyperText Transfer Protocol (HTTP) a success: it is a clear text client/server protocol using Uniform Resource Locators (URL) for addressing. SIP has been developed by the Internet Engineering Task Force (IETF), the organization that standardized IP. It relies on Internet technology.

SIP goes beyond the scope of VoIP to provide building blocks for new enterprise communication applications:

- Powerful addressing schemes (URLs) for user-centric services.
- Features and media negotiation for improved plug-and-play, easily upgraded media-blending applications and terminals.
- Seamless integration with existing enterprise IP networks and applications: integration with network Domain Name Servers (DNS), and with the corporate directory using the Lightweight Directory Access Protocol (LDAP), etc.
- Built-in extensibility to other information technologies used in enterprises: e-mail, documents transported as Multipurpose Internet Mail Extension (MIME) attachments, etc.
- New features: the subscription / notification mechanism is suitable for transporting user presence and terminal state information. Multimedia instant messaging is also supported.

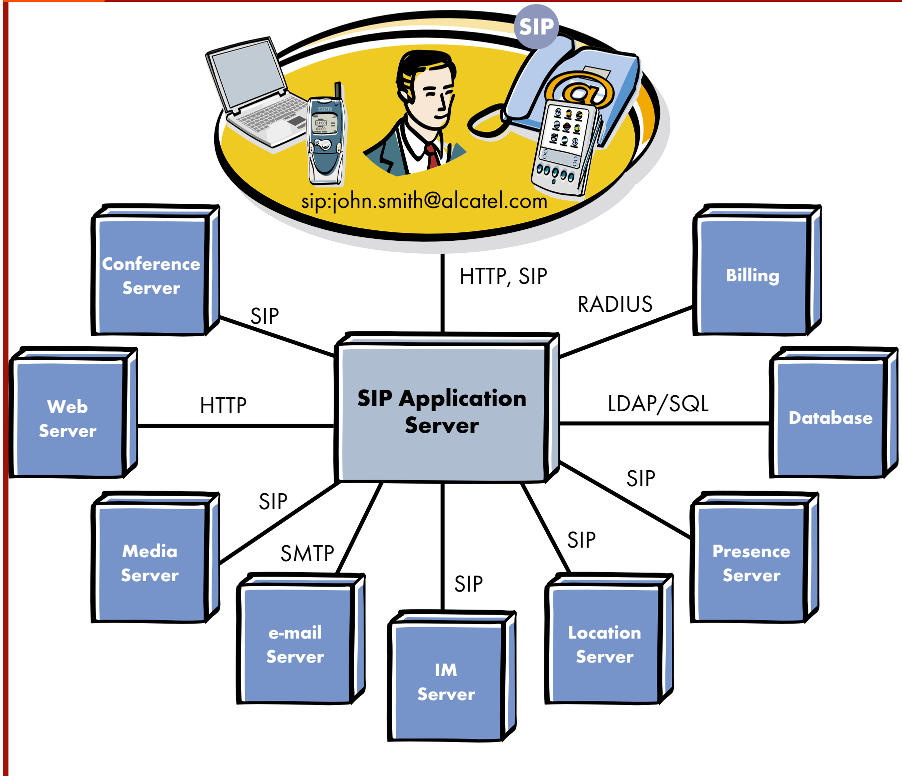
SIP technology promotes peer-to-peer architectures (see *Figure 2*) in which intelligence is provided at the edge of the network (terminals, application servers). The network provides stateless, scalable functions (transport, access control, routing, bandwidth control, etc), thereby enhancing scalability and robustness. Centralized, stateful real-time services (call monitoring, call queuing) are implemented at the network edge in “application servers”. They can also be provided by stateless entities that are monitored by applications, or that interpret intelligence (scripts, code) pushed by stateful entities.

SIP offers address indirection and content indirection. Address indirection is the routing process that enables SIP entities to receive SIP messages on behalf of a user in order to provide services. Like e-mail addresses, SIP URLs describe a user domain. Messages are first routed to the domain, then forwarded by the domain network entities to terminals, other network servers or applications. Bindings between “public” SIP URLs and the destination URLs are kept in a location database, which can be dynamically updated by SIP messages. These mechanisms lessen the constraints on where and how services are implemented, simplifying the roll-out of new services, and enabling carrier-hosted and corporate-hosted services to be mixed.

Content indirection means that SIP does not transport data content but the URLs that edge entities can use to retrieve the content. It fosters a unique converged access to application content. SIP terminals give access to enterprise applications. Today, portals and thin web clients (web browser only) let the customer decide on the presentation and graphical aspects of the content of enterprise information. Enterprise SIP terminals must comply with these requirements.

SIP provides end-to-end integrity protection and confidentiality. Hop-by-hop security in the network is addressed at the network transport level. VoIP requires specific SIP extensions and associated protocols when crossing firewalls and Network Address and Port Translators (NAPT).

Fig. 3 Functional distributed architecture of a SIP application



SQL: Structured Query Language

SIP for Building Enterprise Applications

Figure 3 shows the functional distributed architecture of a SIP application server, which handles not only the SIP protocol, but also various standard protocols to build user-centric media-blending applications. The media involved in each communication session (voice, video, web, e-mail, etc) are dynamically allocated according to the users’ resources and preferences.

Using such an architecture, the application logic is centralized; when needed it dynamically calls other highly reusable services components which are distributed throughout the network (conference server, web server, etc). Open protocols are used, such as HTTP for transport, Simple Object Access Protocol (SOAP) for service invocation, Simple Mail Transport Protocol (SMTP) and SIP. This model enables application developers to focus on the customer application as the services components are already available. As a result it should be possible to build new and better applications more easily, more quickly and at lower cost.

Openness, APIs and ease of service creation are key issues for a SIP-enabled application platform.

SIP Service Creation Environments (SCE) are already available:

- Call Processing Language (CPL), an XML-based scripting language for describing call routing services. Users can create their own SIP services using tools that produce CPL scripts.
- SIP Common Gateway Interface (CGI) (similar to HTTP CGI).
- SIP Servlets (similar to HTTP Servlets).

Typical services that can be created using the SCE are: call redirect to a web page or e-mail if the called party is busy; web interactive voice response (display of a web page to enhance the voice response); presence-based personal call routing; user follow-me enabled by the user's location (for called party mobility); and dynamic media allocation (voice, video, e-mail, instant messaging) based on presence, available terminals and user preferences.

The rapid deployment of Microsoft Windows XP (offering a SIP softphone) will increase the user demand for a SIP SCE. Similarities between SIP and Internet technology will help developers quickly to provide new SIP services. However, higher level APIs will be required for easier integration of advanced enterprise features. The difference from existing CTI technology for VoIP, such as H.323 terminal Microsoft Netmeeting controlled by the Microsoft telephony API, is that SIP addresses issues and services that go beyond telephony. Nevertheless, SIP may not be able to provide CTI services similar to those available today in legacy environments. Hybrid solutions therefore make sense: centralized real-time services enhanced by end-to-end distributed features.

SIP makes it easy to build routing applications, notification applications and basic call control (click-to-dial). The real challenge is to build advanced call control and call monitoring without breaking the SIP model. SIP communication between applications requires extensions to enforce multi-vendor interoperability at the feature level.

Where Do We Stand Today?

SIP has made significant progress since last year and will soon be part of the technology used by enterprises to conduct business. The SIP players (equipment vendors, service providers and software developers) and all telecom vendors are working on the subject.

Standardization within the IETF continues to progress. The SIP core protocol (RFC 3261) is available, so the focus is now on SIP extensions (presence, advanced telephony services, security, etc). However, the current economic problems are slowing down the deployment of SIP services. Today it is forecast that the first commercial enterprise products using SIP solutions could be available in 2003, with deployment in 2004/2005.

What is Slowing SIP Deployment?

A number of factors are slowing down the deployment of SIP:

- Cost of the evolution to VoIP networks that are fully adapted for real-time communication.
- High cost of SIP phones.
- Interfacing with legacy networks. Public Switched Telephone Network (PSTN) charging is not defined in SIP. Also, SIP does not support some popular advanced PBX features. Mapping between legacy and SIP addressing schemes is difficult.
- Addressing issues: no provider domain portability and mapping with PSTN numbers.
- No real response to the legal requirements on telecom systems: call intercept, emergency calling: 911, etc.
- Security issues: VoIP technology is subject to spam, attacks and privacy violations; it does not cope well with firewalls and NAPTs.

What is the Key for Evolution to SIP?

SIP is the basis for the convergence of technology and features in carrier and enterprise networks. The applications that this technology makes possible will put increasing pressure on both carriers and enterprise product vendors to build products and services based on open, standard implementations of SIP.

The probable future proliferation of wireless mobile devices supporting Internet telephony capabilities also strongly reinforces the value of SIP. It will make SIP-enabled Internet telephony services available to the growing population of teleworkers.

Enterprises clearly have an increasing need for business applications that enhance productivity and improve employee efficiency within and outside the enterprise.

Business and Key Players

The current difficult economic situation and the immaturity of the SIP market are making it difficult to have a clear picture of the business opportunities. In the short term, the main revenue-generating opportunities are presence and messaging services for mobile carriers.

Today the first key SIP adopters are:

- Microsoft: every PC running Windows XP, with 350 million users foreseen in 2005.
- AOL Time Warner for presence and instant messaging interoperability.
- 3rd Generation Partnership Project (3GPP) for next-generation mobile devices.
- Telecom carriers like Worldcom, Song Networks, Telia, Delta Three, Level3, AT&T, Radiant Telecom and BT.

Widespread deployment of SIP is essential if it is to be used successfully in the enterprise environment. First it means that users will be accustomed to the technology being present on their desktops (SIP softphone of Microsoft Messenger embedded in Windows XP). Second it is essential to ensure interoperability (access and service ubiquity) with mobile workers, customers and partners thanks to SIP carriers and service providers.

SIP Enterprise Services

The OmniPCX IP Communication Server is completely designed around native IP packet technologies. It offers feature-rich telephony and coupling to other enterprise communication applications. Further information can be found elsewhere in this issue [1].

The OmniPCX IP Communication Server is a SIP-enabled call server that integrates SIP proxy and SIP gateway functions. As described below and in *Figure 4*, the SIP gateway/proxy of the OmniPCX Enterprise takes advantage of existing investment, allowing smooth migration to Internet telephony and associated new services by offering interoperability with legacy enterprise applications and devices (analog, Reflexes™, IP Reflexes™, H.323) and trunks (PSTN, ISDN). Today, the OmniPCX IP Communication Server is unique in that it integrates SIP technology and offers extensive telephony, voice mail and addressing interworking services.

The strategy for product evolution will be to integrate new SIP-based applications and features with existing telephony components, thereby protecting past investment and not disrupting service and usage patterns.

Smooth Transition to SIP with the OmniPCX Communication Server

SIP is a key element in bringing Internet telephony to the OmniPCX IP Communication Server.

OmniPCX SIP Features

Management services for SIP clients

A user with a SIP terminal is part of the OmniPCX system and can therefore be assigned a directory number in the enterprise numbering plan, an entry in the enterprise phone book (for calling name presentation or for calling by name), a Class Of Service (CoS) for call barring and accounting when SIP entities communicate with legacy systems, and a voice mail account.

Routing legacy calls to external SIP entities (proxy, gateway, client) is part of the automatic route selection process provided by the OmniPCX.

SIP gateway services

This gateway is responsible for handling the services required for communication between SIP terminals and legacy terminals. These services encompass telephony interworking services, voice mail services and address mapping between SIP URLs and legacy directory numbers.

The SIP gateway provides the following telephony interworking services:

- Local calls between SIP and legacy users.
- Out/inbound calls: SIP user <—> PSTN, legacy user <—> Internet SIP domain, with user accounting and barring for PSTN access.
- Supplementary services (if supported by SIP terminals): SIP caller/called party identity, hold, consultation hold, attended transfer, call forwarding (unconditional, busy, no reply), 3-way conference, do not disturb, Dual Tone Multi-Frequency (DTMF) (in-band G.711 or RFC 2833 for OmniPCX voice mail).

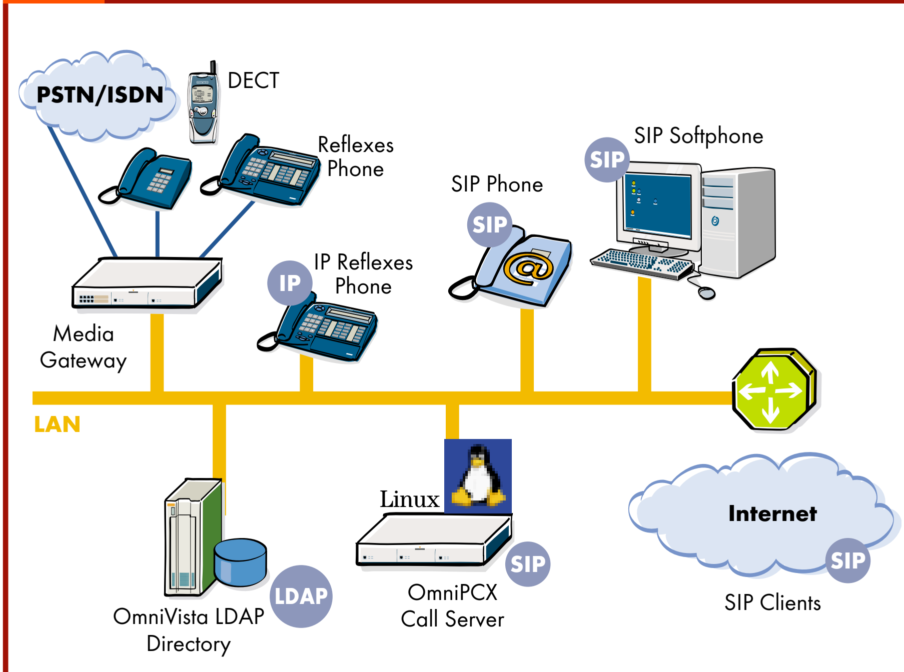
A user with a SIP terminal can forward incoming calls to an OmniPCX voice mail box which sends message waiting indications back to the terminal. The user can then use the SIP terminal to listen to and manage his or her voice mail by following the voice prompts.

SIP proxy services

The proxy provides dynamic location and routing services for SIP communications. The proxy implements parallel forking which enables a user to be called simultaneously on several SIP terminals. The location database is dynamically updated by the OmniPCX SIP registrar[2] when it receives notifications that users are online.

The proxy is open to other SIP proxies in the same domain or in different SIP domains thanks to

Fig. 4 Functional distributed architecture of a SIP application



GW: GateWay
ISDN: Integrated Services Digital Network
LAN: Local Area Network

2 A SIP logical function that collects user registrations.

calls to the enterprise DNS. It handles connection-oriented or connectionless network transport: Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).

The proxy authenticates (digest HTTP authentication) the sender of a SIP message to prevent malicious calls and identity spoofing.

The proxy function of the OmniPCX IP Communication Server is powerful compared with the products of other major vendors.

SIP media services

To improve the quality of VoIP, whenever possible media streams are established directly between terminals, even between SIP terminals and legacy VoIP terminals.

The whole range of OmniPCX VoIP voice compression algorithms (G.711, G.723, G.729) is available for communication between legacy and SIP terminals. DTMF is available either as in-band G.711 or as required by RFC 2833.

Conclusion

SIP offers fundamental new communication models based on presence, mobility and user preferences, as well as on the integration of all forms of communication, events and applications.

In the SIP open Internet model, there is space for distributed services and applications at all levels (customer premises, carrier network, service provider, Internet). For example, some major applications, like unification of a user's messages (e-mail, voice, instant messages) or a user's presence, need to be managed at both the enterprise and carrier levels.

Today, equipment vendors, service providers and application developers are all investing in SIP technology as it has become the *de facto* standard for real-time converged communications in the IP world, as well as for presence and instant messaging. SIP makes it possible to build media-blending enterprise applications based on a distributed architecture at lower cost.

Another key driver for SIP technology evolution is the increasing need of enterprise customers for converged applications which can be used by people on the move, and which are equivalent to those on their office desktops. The converged communications applications enabled by SIP are the real long-term drivers for the evolution of IP telephony to Internet telephony.

In conclusion, for enterprise business, SIP is simple, standard, scalable and allows new media-blending services to be created easily. It is built on proven, promising and evolving Internet technologies.

Alcatel has the right answer with its OmniPCX IP Communications Server which leverages existing investment in order to facilitate the smooth and cost-effective migration to Internet telephony.

Reference

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Extensions: http://www.ietf.org/html.charters/sipping-charter.html.
Presence and instant messaging extensions:
http://www.ietf.org/html.charters/simple-charter.html.
[2] SIP Forum: http://www.sipforum.org.
[3] SIP Center: http://www.sipcenter.com/.

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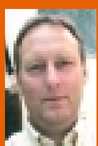

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Abbreviations

3GPP 3rd Generation Partnership Project
API Application Programming Interfaces
CGI Common Gateway Interface
CoS Class of Service
CPE Customer Premises Equipment
CPL Call Processing Language
CSTA Computer Supported Telephony Application
CTI Computer Telephony Integration
DNS Domain Name Servers
DTMF Dual Tone Multi-Frequency
HTTP HyperText Transfer Protocol
IETF Internet Engineering Task Force
IP Internet Protocol
ISDN Integrated Services Digital Network
LDAP Lightweight Directory Access Protocol
MIME Multipurpose Internet Mail Extension
NAPT Network Address and Port Translator
PBX Private Branch eXchange
PCX Private Communication eXchange
PSTN Public Switched Telephone Network
QoS Quality of Service
QSIG Q Interface Signaling
RFC Request For Comments
SCE Service Creation Environments
SIP Session Initiation Protocol
SME Small and Medium sized Enterprises
SMTP Simple Mail Transport Protocol
SOAP Simple Object Access Protocol
SQL Structured Query Language
TAPI Telephony API
TCP Transmission Control Protocol
TSAPI Telephony Services API
UA User Access
UDP User Datagram Protocol
URL Uniform Resource Locators
VoIP Voice over IP
XML eXtensible Markup Language

ARCHITECTS OF AN INTERNET WORLD