White Paper

Session Initiation Protocol (SIP) and MCI Advantage®

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Overview

Global IP communications over the Internet is probably the most significant development since the emergence of telephone networks over 120 years ago. In this paper, we explore the new communication services enabled by the Internet, including examples of new and potential services.

One such new service, MCI Advantage®, leverages SIP (Session Initiation Protocol) based IP communications to provide a network-based convergence solution. SIP is an open, non-proprietary standard modeled after HTTP that seamlessly integrates with the Internet. Because of its easy integration and ability to inter-operate with other protocols and vendors, SIP is the foundation for flexible and scalable VoIP (Voice over IP) solutions.

Present telecommunication networks and services provide adequate and universal telephone service at affordable rates. Why would enterprises want to transition from their traditional circuit-switched networks to the MCI Advantage or another VoIP alternative? An enterprise has two major incentives to complement existing services or migrate to IP communications:

1. Potential short-term savings by moving to end-to-end voice over IP and mixed IP-PSTN (Public Switched Telephone Network) and also by moving PBX (Private Branch Exchange) voice to VoIP usage.
2. Increasing overall productivity by the integration of voice, data, and productivity applications.

It is the second of these incentives, increasing overall productivity, that is the focus of this paper. We begin in Section 1 by providing a generic description of SIP-based Internet communications. Section 2 highlights new functionality enabled by SIP, while Section 3 details how SIP-based applications improve upon many existing PSTN limitations. Requirements for IT managers are presented in Section 4 with summary guidelines for the implementation of IP communications. Finally, in Section 5 we conclude this paper by outlining how the MCI Advantage leverages SIP technology to provide an exceptional convergence solution. We draw the attention of the reader to the fact that the MCI Advantage service is an emerging, radically new service environment, and not all capabilities envisaged for SIP are implemented at the time of this writing.

White Paper

Contents

White Paper

1. SIP-Based Internet Communications

In this section we present the generic capabilities of SIP-based Internet communications' that may be considered a roadmap for commercial products, though it may be some time before all services are commercially available.

VoIP Protocols

Before we discuss SIP in greater detail, below is a brief overview of existing VoIP protocols, including H.323, MGCP/MEGACO and others. A large number of voice over IP protocols are in use at present, reflecting their origin or the voice strategy of particular vendors.

H.323 was introduced by the ITU-T in the late 1990s. It complements H.320, H.321, H.322, and H.324, which are for audio/video conferencing using analog, ISDN, BISDN (ATM) and the public telephony networks respectively. H.323 was thus mainly modeled by ISDN and modified to work on LANs. It was used later by vendors for IP LANs and WANs.

The Internet community started their IP communications approach somewhat later around 1997, based on Internet multimedia conferencing. The key protocol is Session Initiation Protocol (SIP), which was modeled after HTTP used on the World Wide Web.

The circuit switch vendors applied the central control model as used in PBXs for servers to control desktop phones over the IP LANs and the result were several master-slave protocols, such as MGCP and MEGACO/H.248. The ITU-T H.248 protocol was actually designed with voice over ATM in mind, at a time when circuit switch vendors were still promoting ATM as the next generation public network. The central controllers used by such master-slave protocols are also referred to as "softswitches," though the definition of a "softswitch" is more commercial than technical. Sometimes, the media gateway between the PSTN/PBX and IP is also called a softswitch.

Leaving the fine protocol details aside, we remark that SIP is essentially a peer-to-peer protocol, while the softswitch protocols and even H.323 presume some central control. SIP can also use servers for network-based services, but the user can choose whether to use them, and which servers they prefer, similar to pointing a browser to any Web server of choice for the home page.

As for the many vendor proprietary protocols, such as used by IPBX vendors internally, we believe they carry all the disadvantages of vendor lock-in of the customers into proprietary servers and IP phones, for which in most instances there is no second source and therefore no benefits from competition.

IP Communications

IP communications do not necessarily have to emulate the present telephone system, but should be broadly based on the Architectural Principles of the Internet[2] and those of the World Wide Web.[3] We will avoid here a tutorial on the Internet and Web architecture, but only state the generic requirements for the end-to-end control principle of the Internet (the "dumb network"[4]), its scalability and resilience, and also the principles of simplicity, decentralization and the seamless interworking of *independently developed* applications across the Web. The notion of the "dumb network" does, however, not exclude communications between endpoints and servers, such as between IP phones and telephony servers. Such servers need not be placed on user premises or be maintained by end-users, but rather can be hosted by a service provider.*

*What is the difference between "network based" services and "hosted" services? Telecommunication services are based "in the network," actually in service nodes belonging to the telecom service provider and in the embedded infrastructure, such as switches and transmission systems. Customers cannot ask for services that are not available from the telecom service provider. Moving to another network will not result in moving the services as well (though telephone numbers are portable by Government regulation), but buying whatever services are available by the new telecom network provider. Enterprise networks may suffer considerable disruptions when moving from one telecom network to another network, and this is actually not an oversight in the telecom industry!

By contrast, services hosted on the Internet by Internet service providers (ISPs), such as file servers, Web, e-mail and now also IP communications can, if so desired, have services developed and controlled entirely by the enterprise customer themselves. Customers have a free choice to move between hosting service providers or to distribute their particular services among several ISPs.

Using the development of the Internet communication protocols RTP/RTCP[6] (Real-time Transport Protocol/RTP Control Protocol), SDP[7] (Session Description Protocol) and SIP,[8] IP communications enable radically new capabilities and applications that we will describe here.

Integration of Communications and the Web

Probably the most far-reaching DISRUPTIVE engineering decision of IP communications is to integrate voice and all other media with the World Wide Web with regards to addressing, protocols, and data formats.

The Internet development community prefers text-based messages for easy code development and debugging. SIP uses text-based messages for simplicity and easy troubleshooting. In addition to working with standard telephone numbers and extensions, SIP can also use email-like addresses. These addresses can be imbedded in documents and Web pages for “click-to-dial” applications. For example, SIP addresses can look like:

- sip:henry@example.com
- sip:3145551212@example.com

Multimedia Communications

IP communications should work consistently for all media, such as text, voice, video, games, etc. This is best exemplified in PC/laptop/PDA soft clients, such as the Microsoft® Windows® Messenger®, which is based on SIP and shown in Figure 1. Figure 1 shows also several compatible SIP phones from such companies as Pingtel, Cisco, snom AG, Siemens and Mitel, as well as Windows CE 2002 based PDA’s and three soft clients for PDA's such as Microsoft Portrait, and softphones from XTEN and SJLabs. The SIP soft clients from HotSIP and Pingtel are also shown respectively next to the Windows® Messenger. The soft clients work on any Windows XP powered laptop or tablet as shown.

Clients such as Microsoft’s Windows® Messenger can display the familiar phone pad for dialing phone numbers, as well as dialing from a “buddy list” displayed as icons with various meanings. We will review this later in the paper when discussing presence (make “polite” calls for example, when there is an encouraging buddy icon).

Figure 1. User devices for Internet communications based on SIP.

White Paper

Better Than PSTN Voice Quality

Present PSTN voice quality is based on the 1920s analog long distance transmission channel bandwidth of 3.1 kHz and echo control required by 4-wire/2-wire hybrids, limitations that do not exist for voice on the Internet. Actually, a phone call between two IP devices is the ultimate 4-wire connection. If correctly implemented, there is fundamentally no signal attenuation, no signal frequency/amplitude distortion, no nonlinear distortion, no noise and no echo. The default G.711 codecs for VoIP have close to 4 kHz bandwidth and therefore provide equal or slightly better than PSTN quality of voice, while wideband codecs such as G.722 or GIPS (Global IP Sound Corp.) provide near 8 kHz bandwidth.

Speech quality in the IP environment is therefore as a rule, equal or better than PSTN and using the PSTN-derived quality indicators as reference, is misplaced in our opinion. High-end IP communication systems are much closer to studio quality voice than to the frugal 3.1 kHz PSTN telephony.

Compressed codecs, such as G.729 and G.723.1, use as little as 5-8 kb/s data rates yet still provide business quality voice, even over narrowband facilities, such as frame relay or dial-up access respectively.

Most Internet backbone networks have packet loss well below the loss concealment capabilities of G.711, G.723.1, G.729 and GIPS decoders and can carry voice in a flawless manner. Well operated IP backbone networks have rare instances of packet loss bursts due to route flapping when routers are taken out of operation for maintenance or other reasons. Even voice calls over the Internet that traverse 3-5 public networks can be flawless based on our experience.

The Access Bandwidth Problem

Packet loss, mostly due to congestion on overloaded access lines, may cause severe speech clipping, unless so-called “QoS Appliances” are deployed to limit TCP traffic, such as e-mail and Web, and to prevent them taking up all the access bandwidth available with narrowband access. Also some form of session counting is required to limit the number of simultaneous voice calls, commensurate to the speed of the access link. In summary, Internet voice, correctly implemented, can be and is in many cases at least equal or better in quality than PSTN voice.

While QoS is a concern in access networks (“the last mile”), the core backbone of the Internet does not have any QoS issues, but is rather experiencing a fiber bandwidth glut as reported in the trade press.

The solid increase of broadband in many countries across the world, including the USA, will enable small offices and consumers to enjoy high-quality, IP-based voice communications already available in large office buildings.

Higher Than PSTN Resilience

The architectural principles of the Internet—avoiding single points of failure, avoiding single paths, avoiding state in the network, and end-to-end control—have proven to support more reliable communications than circuit-switched networks, in spite of their “carrier strength” network elements. On 9/11/2001 in New York, telecom services were blacked out for days and weeks, while communications on the Internet continued to function without any reported outages to our knowledge, including an ad-hoc SIP telephony service at Columbia University. There are abundant references about Internet resilience and there is a glut of commercial IP network bandwidth that assures high resilience for the Internet. Internet service has degraded somewhat when experiencing powerful virus attacks, but has never failed and has been proven so far to meet the design expectations.

SIP-enabled networks are also distributed by design and have most state and control pushed to the edge of the network, in compliance with Internet architecture. Internet resilience is based on IP endpoints—autonomous hosts that communicate over any available path and do not depend on the network for state. This is a fundamental difference from PSTN systems where a central authority activates and shuts down all boxes having a stake in the communication and where all systems in a path have to keep state for every call going through.

2. Innovative IP Communications Based on SIP

We will now detail the very rich portfolio of services that are available through SIP-based communications. This functionality will drive improved productivity by making it easier and more efficient for end-users to communicate. At the same time, SIP-based communications enhance flexibility by providing enterprises a variety of platforms, endpoints, and servers to utilize.

Presence-Based Communications and Instant Messaging

Presence is a new and core communication capability that is not available in circuit-switched telecom networks. A simple example of "presence" is an instant message client's "buddy list," which lists the user's "buddies" and their current state—online or offline. Additional state information, such as whether they are currently active or idle and whether they are currently typing a message response or not, is also provided using presence. This type of primitive presence coupled with an advanced signaling and presence protocol such as SIP will give rise to a new set of advanced services currently unknown to telephony.

Presence can be used for such services as:

- Make "polite" calls, only when you see a smiley icon
- Avoid phone tag during busy hours
- Automatic call-back on presence
- Ad-hoc conference calls based on presence
- Avoid waiting for call center agents—replace ACD (Automatic Call Distributor) with agent presence
- On-the-air presence for mobile phones
- Presence coupled with location

The popularity of multifaceted icons for buddies in Instant Message (IM) clients illustrate the potential not only of presence, but also the possibility of replacing the familiar phone pad with an IM-like caller's "dial-pad." New interfaces in communication devices may have much more in common with the IM interface than with the telephony dial pad. Presence is also the key for tying endpoints across multiple network boundaries and thus will be a critical tool for fixed-mobile convergence.

Several features distinguish enterprise IM services from consumer presence and IM:

- Security that can adequately support authentication and privacy for sensitive business communications
- Ability to expand presence functionality beyond short text communications; it also can be used to initiate voice calls, start conferences, and generate automatic call-back
- Standards-based approach to signaling, message transfer and XML-based data formats
- Using the same SIP-enabled infrastructure, applications and data for *all communication* services in the enterprise: Text, voice, video, data and application sharing
- Choice of clients, servers and applications from different vendors
- Avoiding duplicate databases for customer and service related data

Unified Messaging: Voice Mail, E-mail, IM, SMS

IP communications supports the seamless integration of voice and video mail with e-mail, instant messaging and wireless short message service (via mobile network to SIP/IP gateways). An application example is shown in Figure 2 for voice mail.

In Figure 2, a mobile user or a user on the PSTN is trying to call a user that has an IP communication service and can be reached at a SIP phone and/or IP devices. The call (1) is first routed by the PSTN gateway to the SIP server for the called party (2) and from there to the SIP phone (3). If the called party does not answer (4), the SIP servers will proxy the call to the unified messaging (UM) server. The caller can now deposit the voice message (5).

The notification for 'Message Waiting' will appear (a) as a flashing light on the SIP phone—as is usual on PBX and mobile phones, (b) as an email notification and also (c) on a UM (Unified

Messaging) Web page. Message waiting notification by e-mail and the Web has the advantage of playback control, using any standard compliant Web media player. An SMS (Short Messaging System) message can also be sent to a mobile phone. Voice mail can also be retrieved the old fashioned way by calling in from the PSTN or from a mobile phone as shown in Figure 2.

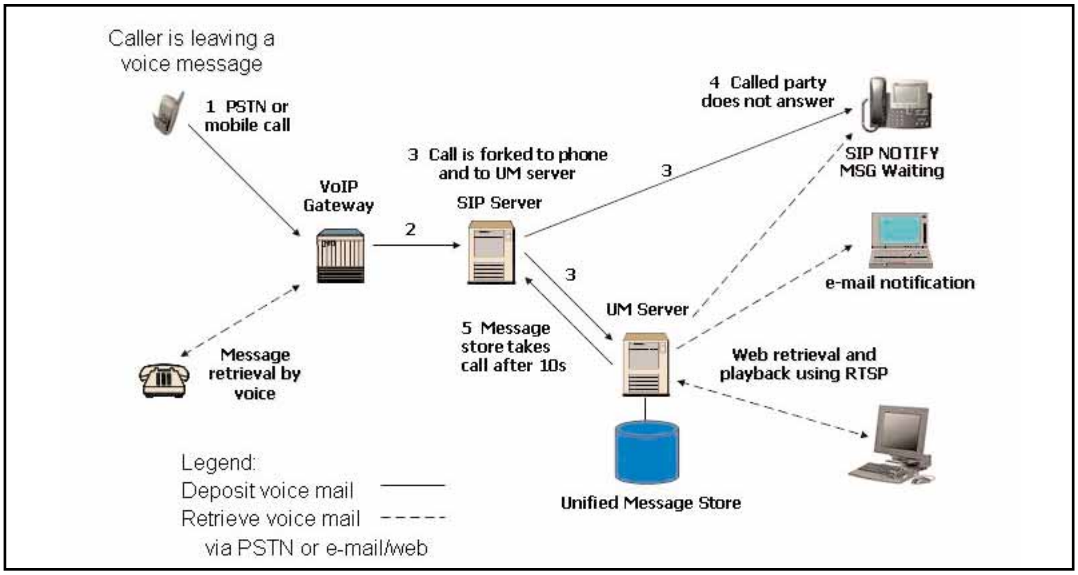


Figure 2. Voice mail example as part of unified messaging.

Figure 3 shows a screen for UM as developed by Interactive Intelligence (inin.com) using Microsoft® Outlook®. The list of messages shows both e-mail and voice mail and can also display fax messages. Upon clicking on a voice mail entry, the media player will play out the voice message.

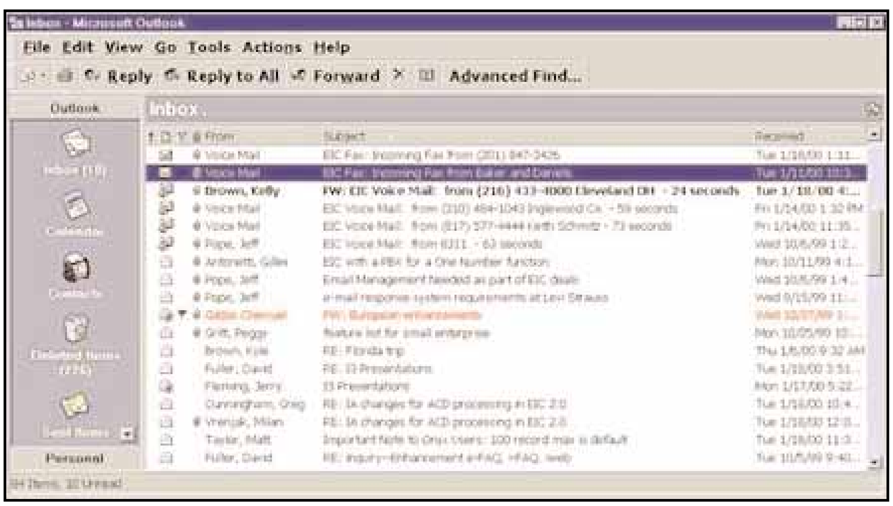


Figure 3. Example of the integration of voice mail notification with e-mail.

Multiple Conferencing Modes and Media

SIP-based conferencing can be implemented in many models,[8] and covers scheduled PSTN-like conferences, small ad-hoc conferences, the transition from a two-way call to a multi-party conference and multicast group conferences. SIP conferences can be used for text only as in IM, for voice,

voice and video (see Figure 2), with data and for games. As mentioned, presence can be used for the setup of ad-hoc conferences, and it enables a new spectrum of conference services, from spontaneous to rigorously scheduled controlled conferences.[9]

Call Routing Options and Route Recording

In the PSTN, the commercially most valuable services of the Intelligent Network are based on call routing, depending on various criteria, such as origination, destination, time of day, etc. The SIP protocol takes routing to its full potential and also adds routing control by the end-user as an additional feature in caller preferences.

ENUM and Directory Services

E.164 telephone NUMber [ENUM] mapping to IP[10] allows finding all communication contact addresses, such as mail, PSTN, mobile phones, fax and SIP when given a standard E.164 telephone number. ENUM technology has many service routing applications, among them directory services that can return all of an individual's listed phone and fax numbers, e-mail addresses, IM, SMS and paging address, etc.

Contact Addresses

The SIP protocol allows you to resolve an *address of record* to multiple *contact addresses* with preferences selected by the user, so that calls can be routed based on who the caller is, based on time/date and on the location of the called party (work, travel, home). The call will then be routed to the preferred communication device.

Secure Communications

The IP communication architecture deploys multiple Internet security mechanisms for secure and confidential signaling and media. The high security possible for IP communications has to be harmonized with the legal intercepts required by law enforcement agencies and is a topic still under discussion.

SIP provides the ability to authenticate callers based on a shared secret, such as username and password. It will also provide, in the future, the ability to send all signaling messages over a secure transport, such as SSL (Secure Sockets Layer), so that the nature of SIP signaling would not be discernable from captured IP packets without breaking the encryption scheme.

Global Mobility

Application level mobility for IP communications has three facets:

Personal Mobility

The user is reachable under the same identifier on several networks, at several terminals, and possibly at the same time.

Service Mobility

Ability to obtain the same services while roaming: Services, personalized GUI (Graphical User Interface), directory, "dial plan" and other "sticky" service features.

Terminal Mobility

The user is reachable by the same SIP address, using dynamically acquired IP addresses. This can support pre-call mobility, mid-call mobility and recovery from disconnect.

A Web page with references on SIP mobility is http://www.cs.columbia.edu/~hgs/sip/drafts_mobility.html

User Preferences

Callers can specify via the SIP User Agent the routing and disposition of their calls, while the called party can specify a rich portfolio of preferences for receiving calls.[11] For example, a caller can specify if they wish to avoid speaking to machines and a called party may block all work-related calls during non-work hours. Many services can be built using these capabilities. The important point to keep in mind is the standard nature of communicating these preferences when routing calls between networks from different service providers—a capability that is impossible between proprietary IP PBXs or proprietary softswitches.

Gateway Services

While the majority of the global public telephone traffic still uses the PSTN, and most of the installed PBXs are also circuit-switched of the TDM (Time Division Multiplexed) type, most phone calls to and from IP devices require gateways, such as:

- Network gateways in carrier networks that support SS7 (Signaling System 7) for high call volumes
- Enterprise gateways to connect legacy PBXs to a local area or wide area networks
- Local gateways for 800 calls, 911 emergency calls, and 411 directory calls

3. How SIP Improves upon PSTN Functionality

Leaving aside for now the significant voice traffic migration to mobile telephone networks, the advent of the Internet and its associated technologies are forcing telecommunication companies and their vendors to face the facts that (1) telecommunication networks have a competitor in the Internet and (2) the deployed technology is obsolete.

What Is a Closed System?

Closed systems, such as the traditional PBX or telephone company switch have the design and change control entirely in the hands of the respective vendor. To enhance this control, switch vendors will also protect the design with as many patents as possible, so that intellectual property claims may keep the competition at bay.

As a result of the Intelligent Network concept, telephone companies have embraced the need for control of the services to be outside the switch, in server farms that may be procured from other sources. End-users still have no control over the service, and innovation by third parties is practically impossible.

The Internet and the World Wide Web by contrast are built entirely on public standards and without intellectual property claims or the notion of end-to-end control. Services are in the hands of users, which provides a wide-open field for innovation.

In the context of voice over IP, a closed system cannot be expanded with components from competing vendors using public standards. Examples of closed systems are “softswitches” and proprietary IP PBXs. In such systems, it is not possible to choose the following from different sources:

- IP phones
- IP-PSTN gateways
- Service controllers
- SIP servers
- IVRs, VoiceXML and speech recognition systems
- Media and announcement servers
- Conference bridges
- Voice mail and unified messaging

  ...as well as other components

Closed systems may support standard protocols to the outside, such as having ISDN trunks or SIP ports on the IP side, but this does not make them open systems. Besides open protocols between all the mentioned components, standard data formats are also required. An example for standard data formats are those for presence and instant messaging (IM) defined by the IETF SIMPLE working group[12] and the protocols and data formats for the auto-configuration of SIP phones.[13, 14] Such open protocols and data formats between components are not found in closed systems.

One of the penalties of closed systems is the need to maintain multiple instances of similar user and service data in multiple systems. This is not a trivial task and may lead to increased cost of ownership, besides the occasional errors when duplicating data.

An interesting point is the claim that "open" Application Programming Interfaces (API) qualify softswitches and some IP PBXs to be open systems. APIs have usually two intellectual property owners: (1) The owner of the operating system and (2) the owner of the softswitch system. Java and Jinni occupy an interesting middle ground, since no accredited standards body has intellectual property and change control. The Internet in most cases does not standardize APIs, but rather protocols, for architectural reasons. It does not matter how an IP device is implemented (this is best left to the designer and to the competition in the marketplace), but only how it speaks and behaves over the Net.

Examples of How SIP Improves Upon Traditional Telephony

The fast ascent of the Web and e-mail was due in part to the fact that telecom services simply had nothing comparable to offer. But even for such telecom services as telephony, the existing telephone networks are technically obsolete regarding the services potential. Following are several examples of traditional telephony functionality that are enhanced in a SIP environment.

Call Transfer in Voice Mode Only

Many or most telecom engineers are challenged when trying to transfer a voice call due to the difficult and primitive user interfaces (compared to icons and mouse clicks) found in various telephony systems. But even attendants who understand how to transfer a voice call, may prefer to do the "attended" part of an attended call transfer using instant messaging (IM) with the transferred-to party, before making the transfer. Thus when a call from a customer comes in, the attendant may quickly send an IM to ask what to do, while still talking to the caller. Upon receiving an IM reply, the call may be transferred to the requested called party or somewhere else, such as voice mail.

Automatic Callback

Presence can be used for automatic callback. Thus if the called party becomes available, for example, has the cell phone on the air, or the called IP phone is no longer busy, an application can initiate a callback, thus saving unsuccessful call attempts and the resulting phone tag.

Internet communication engineers believe presence to be a key ingredient of the radically new capabilities of IP communications, since there are many other applications enabled by presence, for example ad-hoc conference calls and the replacement of automatic call distributors (ACD) in call centers.

Ad-Hoc Conference Call

Scheduling multiparty conference calls to accommodate busy calendars is often hard work. Many e-mails are exchanged to find a suitable date/time for a multiparty conference call. Presence enables applications that automatically invoke a conference call when the intended participants are online and available to users in a convenient fashion.

Call Distributor Queue

Customers calling an 800 number for assistance are often frustrated by long waiting times to get to an agent that can answer specific questions, only to wait again when transferred from one agent to another. Presence enables agents with the desired skills, selected by the calling customer to call back, thereby avoiding waiting by the customer online.

Web-Sharing and Data Collaboration

An agent talking to a customer or employees in a conference cannot exchange documents during the call, except sending files by e-mail or by looking up Web pages so as to share information. The telephone network has no means of pushing Web pages during a voice call using the same signaling mechanisms as used to set up and maintain the conference call. There are no shared whiteboard capabilities, file transfer or data capabilities of any type; no possibility to facilitate understanding between speakers familiar with different languages, so the Internet has to be used anyway.

Mobility

The most sophisticated PBX or PSTN/Centrex services are lost to the user outside the premises of the enterprise. Even such common services as 800 numbers are not valid outside the country or calling area. This is in contrast to the Internet and the Web where public IP addresses and URLs have global significance, and services like e-mail and Web pages can be accessed in the same way wherever the user may be and, typically, whatever device, operating system and application is employed for access to Internet services. Advanced enterprise IP services can also be securely accessed by roaming users from anywhere.

These are only some examples to illustrate the technical obsolescence of local Class 5 and PBX services for consumers and for enterprises alike. There are many applications enabled by integrating voice communications with Internet and Web applications. Such applications are given in various Internet drafts related to the SIP protocol, though we believe some of the most successful may not yet be invented.

4. Making a VoIP Investment

Now that we have highlighted SIP functionality and how it can improve your productivity and communications network, we will provide a brief overview of investment considerations for implementing a VoIP application. Responsible IT managers and ISP (Internet Service Providers) network planners will remember the golden rules for making new investments:

New Services

Invest as a rule in new services only. IT managers will look for new services that increase the core capabilities of the company and the productivity of the workforce and management. Service providers have a vested interest to support their customers with state of the art new services. New services are a main source for revenue increase. As we have shown, IP communications support an extraordinary portfolio of new services that may actually take quite some time to roll out and to be absorbed by the market.

Cost Reduction

Cost reduction for existing services may be possible by replacing existing services, such as PSTN/PBX telephony with IP voice communications. Rich end-to-end controlled IP communications, such as those supported by the Windows® Messenger soft client illustrated in Figure 1, can be introduced and supported at a cost that is truly negligible compared to the huge investments and operational cost required by traditional telecom PSTN networks.

User Satisfaction and Retention

Giving the user a wealth of new IP communication services at reasonable cost is bound to increase user satisfaction. The large amount of enterprise customization and support for end user preferences make IP communications a very “sticky” Internet application.

Beyond these basic considerations, generic technical requirements are:

Benefits From Full Internet Standards Compliance
Complete and faithful adherence to the relevant Internet standards of the IETF is required. This ensures good design and can be tested for interoperability between several vendors. This is also the basis for the following benefits:

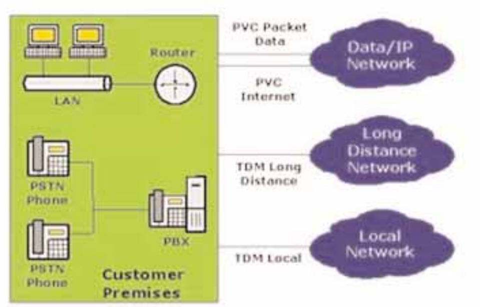
Unbundled and Modular Design and Multi-Vendor Support
Various user and network components, such as IP phones, PC clients, SIP servers, application servers, gateways and media servers should be available from more than a single vendor. A key requirement is for all the components to interwork 100%, without any knowledge by the designers of one component about the design of another component. This loose coupling between components[15,16] allows, for example, an Interactive Voice Responder (IVR) to interwork with a service controller and with voice mail, media servers and other components from various vendors or even with specialized outside service providers.

Even in turnkey systems from one single vendor that simplify management for the user, we believe that a clean decomposable internal SIP system design will make its operation and inevitable future enhancements much less expensive and faster to implement.

5. SIP-Based Communication Service: MCI Advantage

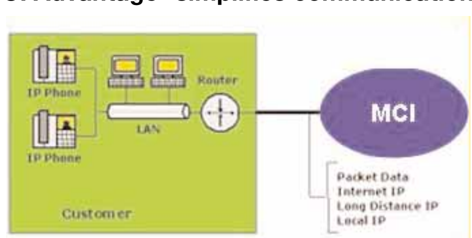
Having discussed in detail the framework and benefits of SIP, let's now review an actual VoIP application using SIP. There are a number of emergent networks and services providing Internet communications based on SIP—one example shown in Figure 4 is MCI Advantage. This offering is a network-based IP Communications solution that can converge all of a customer's Local and Long distance voice and data traffic onto the customer's IP network.

Too many networks are inefficient and expensive.



Old World
- **Three networks**
- **Duration/geography-based pricing**
- **Expensive moves, adds, and changes**
- **Standalone applications**
- **Closed PBX architecture**

MCI Advantage™ simplifies communications management.



New World
- **One source for all services**
- **Network access to PSTN from the WAN**
- **Centralized, network-based services**
- **MCI provides the technology**
- **Easy mobility**
- **Efficient communication with Web and e-mail**
- **Choice of vendors for CPE**

Figure 4. MCI Advantage consolidates networks and services.

The rich portfolio of voice services and the technical approach is described at the MCI Web site http://www.mci.com.

The variety of customer premises equipment cannot easily by condensed in Figure 4, so we prefer to show here in Figure 5 a simple home office/small office (SOHO) setup that besides SIP phones and the Windows® Messenger client is also using a Universal Plug and Play (UPnP) small wireless gateway, such as the Microsoft MN-500 exemplified here. This office configuration example is actually similar to the home office of one of the authors.

The Windows® Messenger shown in Figure 5 has the rich multimedia communications and data collaboration features shown in Figure 1.

The internal Web site contained within the SIP phones allows dialing from the PC desktop using the address book, etc. We are mentioning these communication examples to illustrate the power of open, global IP communications, versus the voice-only emulation or localized capabilities of the PBX, be it circuit-switched or IP-based.

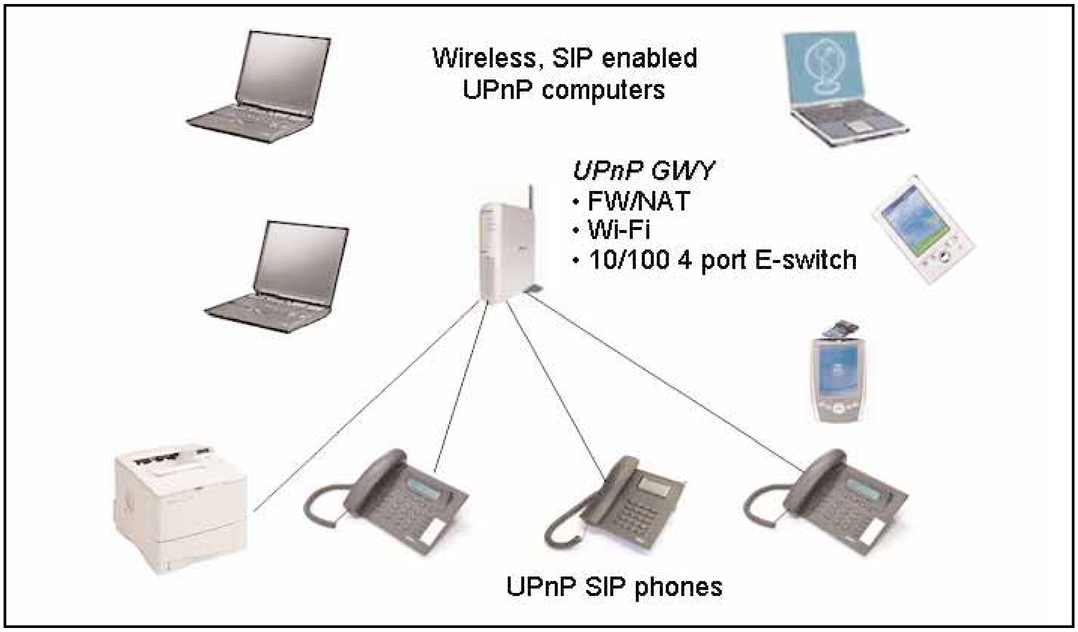


Figure 5. Small office example with UPnP gateway and UPnP SIP phones.

Today, MCI Advantage is primarily about connecting SIP IP telephones and gateways; in the future, (the timing depends on market developments) it could be used with many customer's IP PBXs that support SIP, either in a native mode or at least as an external interface. However, without native SIP, the set of features and services available is limited by the IP PBX, and many of the advantages of the carrier services cannot be used.

The industry trend is to move away from proprietary, closed systems such as these to a pure SIP environment that can take full advantage of MCI Advantage. Thus, the service is valuable for these customers even during this transition period. Customers currently investing in Internet communications equipment should carefully consider any purchase that is not fully SIP-enabled, in native mode, as this will likely require further investments in the future to fully capitalize on Internet communications.

A remarkable feature of the MCI Advantage offering is its complete reliance on existing Internet standards and on avoiding proprietary systems. The main telephony features are described in detail in Internet Drafts that are on the IETF (Internet Engineering Task Force) standards track.[17, 18, 19]

6. Conclusion

This paper has discussed Internet Communications in general and MCI Advantage as an example of the advanced communications and telephony solutions possible using SIP. As voice is now viable over IP, the advantages of using the global IP network for converged voice and data services are just beginning to reach the marketplace. This paper has outlined these new services, particularly the rich services enabled by Session Initiation Protocol (SIP). The leading example of a SIP-based solution is MCI Advantage, which provides both innovation and potential cost savings to enterprises.

7. More About the Authors

Dr. Alan Johnston, Distinguished Technical Member, has co-authored the SIP standard, RFC 3261 and other work on the standards track such as “SIP Call Flows” and “SIP Service Examples.” MCI Advantage is based on these standards and facilitates customers and partners to “connect” to MCI entirely based on public standard protocols, call flows and message formats, without any need to exchange proprietary information or license fees.

Dr. Henry Sinnreich, Distinguished Member of Engineering, has also contributed to Internet standards work in the areas of SIP-based IP communications. Dr. Sinnreich received the Voice ON the NET Pioneer Award in 2000 in Stockholm and is a frequent speaker at technical conferences. Dr. Sinnreich is a founding Board of Directors member of the international SIP Forum.

Alan Johnston and Henry Sinnreich have been actively involved in the initiation, design and development of MCI Advantage service since the inception of the project and are presently contributing to its further development.

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