True Number Portability and Advanced Call Screening in a SIP-Based IP Telephony System

Ismail Dalgic, Michael Borella, Rick Dean, Jacek Grabiec, Jerry Mahler,

Guido Schuster, and Ikhlaq Sidhu

3Com Corporation

ABSTRACT Custom local area signaling service features offered in the PSTN have certain limitations due to the closed nature of PSTN network signaling. The adoption of telephony over IP (IP telephony) will enable a new paradigm of services and features that are not possible to implement in today's PSTN. This is especially the case for services that make use of personal, trusted information, which can be provided by a user's personal digital assistant. In this article we demonstrate how personal information can be coupled with an IP telephony service to provide user-customized call handling by the network. In particular, we describe a demonstration architecture that includes Ethernet-attached phones running SIP, with an interface to synchronize with PDAs that supply personal information. The proposed architecture is quite flexible; it can support enhanced versions of the current PSTN and private branch exchange services, in addition to many new features and services. We describe true number portability and advanced call screening as examples of new services in a hybrid PSTN/IP telephony environment.

For many years, the public switched telephone network (PSTN) has provided custom local area signaling service (CLASS) features to residential customers, such as call blocking, caller ID, and call forwarding. However, these services have certain limitations, primarily due to the closed PSTN signaling architecture which does not allow access to network signaling protocols by terminal devices. The adoption of IP telephony will enable the development and deployment of a new paradigm of services and features that are not possible to implement in today's PSTN. This is especially the case for services that make use of personal, trusted information. Personal digital assistants (PDAs) are very suited to providing such information, because they store personal data and are carried by their owner most of the time.

We have designed and built an experimental system to investigate how personal information can be coupled with an IP telephony network service to provide user-customized call handling by the network. In particular, we use a PDA which supplies personal information to an Ethernet-attached phone running the Internet Engineering Task Force (IETF) Session Initiation Protocol (SIP) [1] (from here on referred to as a *SIP-Etherphone*). By synchronizing the owner information on the PDA with the SIP-Etherphone, the owner of the PDA registers with the phone. This allows for *true number portability*, since the call is forwarded to the SIP-Etherphone where the owner of the PDA was registered most recently. Furthermore, the address book data and the appointment book data in the PDA can be synchronized with the SIP-Etherphone to program an *advanced call screening* service in the local SIP server. For example, the user may opt to accept calls from only those callers who are in the address book with a certain priority, and to forward calls from everyone else to a third party or to voice mail. More sophisticated services could check the appointment book. If the current meeting is of lower priority than the current call, the phone would ring; otherwise, the call would be forwarded. Moreover, the PDA can provide the user with a flexible user interface to customize and control the phone services.

CURRENT TELEPHONY SERVICES

In this section we discuss current telephony services that go beyond the setup, transport, and teardown of calls, with the objective of identifying the issues with regard to implementing similar features as discussed above for PSTN and private branch exchange (PBX)-based switching environments.

TRADITIONAL PSTN CLASS FEATURES

For many years, the PSTN has provided CLASS features to residential customers. Once a customer subscribes to a CLASS feature, they may activate and/or deactivate it through the use of "*" directives (e.g., *69 to automatically return a call to the most recent caller). CLASS features may also be implemented with the use of out-of-band data. CLASS feature data is transmitted between local Class 5 switches using the Signaling System No. 7 (SS7).

Local exchange carriers (LECs) or similar organizations maintain CLASS offices that contain a database entry for each customer. The database allows specification of which CLASS services a customer has subscribed to, as well as any information, such as lists of phone numbers, associated with those features. Customers may edit these lists online via a touchtone interface. A list of all phone numbers that have originated or terminated a call with each customer is often included in the CLASS office database. For each customer, only the most recent number on this list is stored by the local Class 5 switch.

Some of the more popular CLASS features are described below:

- Call blocking: The customer may specify one or more numbers from which he or she does not want to receive calls. A blocked caller will hear a rejection message, while the callee will not receive any indication of the call.
- Call return: Returns a call to the most recent caller. If the most recent caller is busy, the returned call may be queued until it can be completed.
- Call trace: Allows a customer to trigger a trace of the number of the most recent caller.
- Caller ID: The caller's number is automatically displayed during the silence period after the first ring. Requires that the customer's line be equipped with a device to read and display the out-of-band signal containing the number.

- Caller ID blocking: Allows a caller to block the display of their number in a callee's caller ID device.
- Priority ringing: Allows customers to specify a list of numbers for which, when called by, they will hear a distinctive ring.
- Call forwarding: A customer may cause calls to their number to be automatically forwarded to another number for a period of time.

PBX Features

The PBX is a stored program switch similar to a Class 5 switch. It is usually used within a medium- to-large-sized business for employee telephony service. Since a PBX is typically operated by a single private organization, there are a wide variety of PBX services and features. Custom configurations are common. All PBXs support their own versions of the CLASS features, as well as other features that go above and beyond the scope of CLASS. Most PBX features are designed to facilitate business and group communications.

A summary of typical PBX features is as follows:
- Call transfer: An established call may be transferred from one number to another on the same PBX.
- Call forwarding: In addition to CLASS call forwarding, a PBX number can be programmed to automatically transfer a call to another number when the first number does not answer or is busy.
- Camp-on queuing: Similar to PSTN call return, a call to a busy number can be queued until the callee can accept it. The caller can hang up their phone, and the PBX will ring them when the callee answers.
- Conference calling: Two or more parties can be connected to one another by dialing into a conference bridge number.
- Call parking: An established call at one number can be put on hold and then reestablished from another number. This is useful when call transfer is not warranted.
- Executive override: A privileged individual can break into an established call. After a warning tone to the two participants, the call becomes three-way.

In addition to these features, PBX systems can also be integrated into intercom and voice mail systems.

PDA Enhanced IP Telephony Features

While borrowing the concepts of caller ID, call blocking, call forwarding, and priority ringing, the calling features discussed in this article improve on CLASS features in a number of ways. From a customer perspective, the most obvious difference is that call forwarding is initiated from the physical location to which calls are to be forwarded, rather than the location from which calls are to be forwarded. This results in a call forwarding mechanism that is truly portable, since it follows the user wherever he/she goes. PDA integration also enables a more sophisticated priority mechanism. It allows calls to be either completed, blocked, or sent to voice mail, depending on the priority assigned to the caller in the address book of the PDA. The action taken can be varied based on the callee's schedule stored in the PDA. Finally, since the features are controlled using the PDA, the user interface is more intuitive and flexible, and allows better integration with other PDA and desktop computer applications.

In addition to the above mentioned features, many other new applications and services are enabled by the coupling of two new paradigms:
- Shifting intelligence from the network to the telephones
- Providing personal information and location information to the network by means of synchronizing the information on the user's PDA with the network

Some examples of those new applications and services are:
- The PDA may store and download to the phone the preferences of the user about phone operation, such as the ringer volume and tone.
- The PDA may act as a smart card, providing authentication information for making toll calls.
- The user may program the system through the PDA so that, depending on the time of day and the date book information in the PDA, the phone forwarding information is dynamically updated. For example, during business hours the default location to forward calls could be set to be the user's office, and during other hours his/her cellular phone or pager.
- If the PDA has voice playback capability, it can download voicemail and play it back offline. On a LAN, this would be implemented as a file transfer, which is much faster than playing audio back. This feature would be useful if the user could not spend much time on the phone to check his/her voice mail. For example, a traveler at an airport may download his/her 30 min worth of voicemail in a few minutes just before taking a flight, and may listen to those messages during the flight.

Session Initiation Protocol Overview

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions or calls [1]. There are two major architectural elements to SIP: the *user agent* (UA) and *network server*. The UA resides at SIP end stations, and contains two parts: a user agent client (UAC), which is responsible for issuing SIP requests, and a user agent server (UAS), which responds to such requests. There are three different network server types: a *redirect server*, a *proxy server*, and a *registrar*. A basic SIP call does not need servers, but some of the more powerful features depend on them.

The most generic SIP operation involves a SIP UAC issuing a request, a SIP proxy server acting as an end-user location discovery agent, and a SIP UAS accepting the call. A successful SIP invitation consists of two requests: INVITE followed by ACK. The INVITE message contains a session description that informs the called party which type of media the caller can accept and where it wishes the media data to be sent. SIP addresses are referred to as *SIP uniform resource locators* (SIP URLs), which are of the form sip:user@host.domain.

Redirect servers process an INVITE message by sending back the SIP URL where the callee is reachable. Proxy servers perform application-layer routing of the SIP requests and responses. A proxy server can either be stateful or stateless. A stateful proxy holds information about the call during the entire time the call is up, while a stateless proxy processes a message and then forgets everything about the call until the next message arrives. Furthermore, proxies can either be forking or nonforking. A forking proxy can, for example, ring several phones at once until somebody takes the call. Registrar servers are used to record the SIP address (SIP URL) and the associated IP address. The most common use of a registrar server is to register after startup so that when an INVITE request arrives for the SIP URL used in the REGISTER message, the proxy or redirect server forwards the request correctly. This registration-based forwarding feature forms the basis of the network support for the true number portability service described in this article. Note that usually a SIP network server implements a combination of different types of servers.

THE SYSTEM ARCHITECTURE

In this section we describe the components that make up the system and the interactions among them. The system architecture is shown in Fig. 1.

SIP-ETHERPHONE

An Etherphone is a telephone that can be plugged into an Ethernet port. In most cases, Etherphones support IP using an IP address that is either statically configured or obtained via Dynamic Host Configuration Protocol (DHCP). There are two major parts to an Etherphone: the signaling stack and the media engine. While currently two different standards (SIP and H.323) and several proprietary approaches exist for the signaling stack, the media is almost exclusively transported via the Real-Time Transfer Protocol (RTP [2]), which itself is carried inside User Datagram Protocol (UDP).

The purpose of the signaling stack is to set up, manage, and tear down a call. During the setup phase the location of the endpoint is discovered, and communication capabilities such as the supported voice codec types are exchanged. During the management phase the voice channel is established, and other parties are invited to the call if needed. During the teardown phase, the call is terminated gracefully. We chose SIP as the signaling protocol for our architecture because it is a lightweight protocol that is easy to implement and supports all the functions needed.

The purpose of a media engine is to sample the voice, to encode the samples, and to build the RTP packets on the sending side. On the receiver side, in addition to performing the reverse operations, it must also manage a receiver buffer to compensate for network jitter.

The SIP-Etherphones in our system have pre-programmed phone numbers, represented as SIP URLs of the form *sip:4085551212@3com.com*. After power-up, each SIP-Etherphone sends a SIP REGISTER message to the default registrar. When a call arrives at the registrar for any of the registered SIP URLs, the SIP server will forward the call to the appropriate destination. If a SIP-Etherphone is moved to a new location, all calls to the associated SIP URL will still be properly routed to that device. In other words, IP telephony provides device mobility in the sense that calls will "follow" the SIP-Etherphone according to its SIP URL. (This is especially useful if the SIP-Etherphone is running the DHCP protocol so that when the location is changed, the IP address is also automatically changed. Indeed, our prototype implements DHCP.) Hence, no interaction with an IT department is necessary, which is currently a major cost when relocating employees.

As far as we know, our prototype SIP-Etherphone is the first fully functional SIP phone.[1] In addition to the above-mentioned features, it also has a serial port through which a PDA can be attached. We have also implemented the necessary software on the Etherphone to synchronize with the PDA.

THE PDA

The PDA maintains personal address and date book information (among other personal information). It includes a graphical user interface and can dynamically synchronize with a host computer via a serial port, or a wireless infrared or radio link.

The PDA offers three fundamental features for IP telephony:

- Authentication of the owner
- A graphical user interface
- Personal data of the owner

While the PDA significantly enhances the system, as discussed in subsequent sections, it is not necessary for the PDA to communicate with the SIP-Etherphone for it to provide basic phone service.

In our prototype we used the 3Com Palm III™ organizer, which is a palm-sized computer with built-in organizer applications and the ability to load and run third-party applications. The Palm III organizer has a touch-sensitive LCD display, on which a stylus is used to input data and navigate the user interface.

[1] *All other LAN telephones we know of use proprietary signaling protocols.*

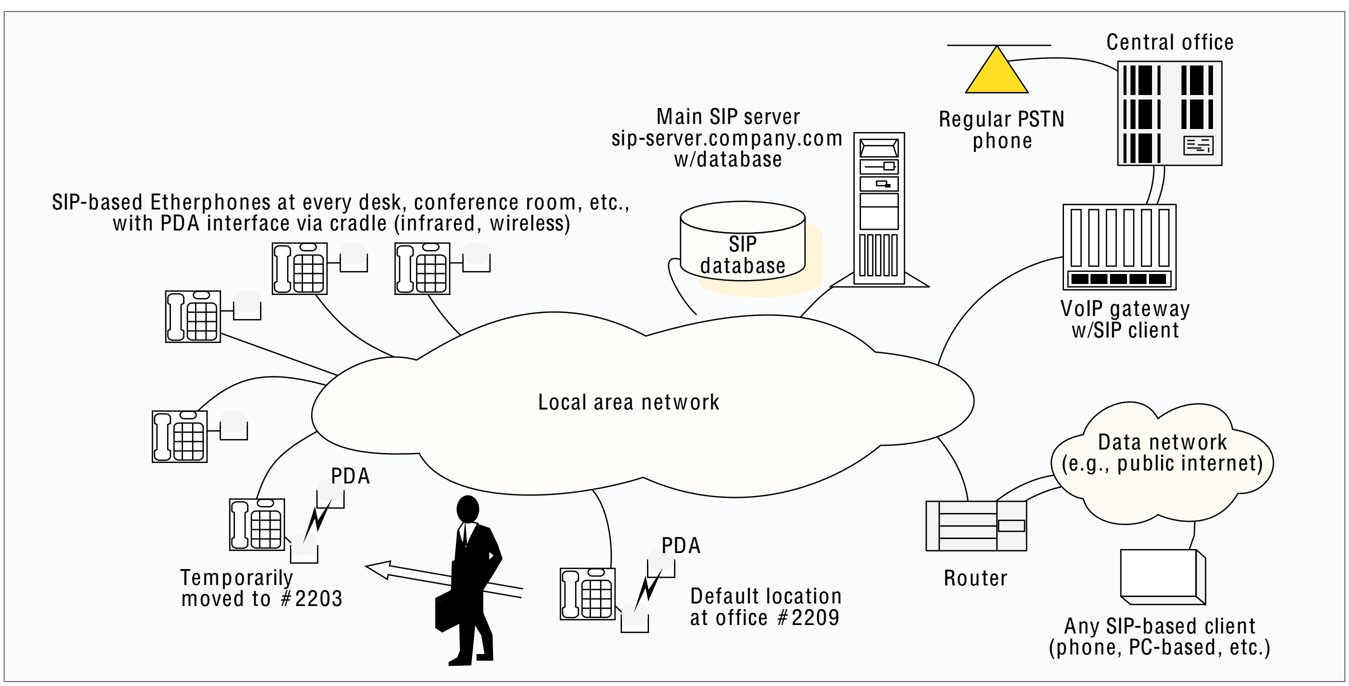


Figure 1. *System architecture.*

THE MAIN SIP SERVER

A central registrar/proxy server, which we refer to as the *main SIP server*, is the primary destination of all SIP messages that originate within the administrative domain trying to establish a connection. It is also the only destination advertised to the SIP clients outside the LAN on behalf of all the SIP-Etherphone clients residing on the LAN. The main SIP server relays all SIP INVITE messages to the appropriate final destination (or another SIP proxy), based on a database lookup. It allows all mobile clients to register with their current locations. The decision of whether or not to accept the call is left to the clients. This approach requires intelligence in the clients, which is in line with our IP telephony vision. Its main advantage is that it does not require the transfer of potentially large files (e.g., the address book and schedule information stored in the PDA) across the network from the electronic organizers via SIP-Etherphones to the server.

THE GATEWAY

The final component of the system is the PSTN/SIP gateway, which provides connectivity with the PSTN. On the LAN side, the gateway acts as a SIP UA on behalf of the PSTN phones. The gateway is connected to the PSTN via a number of T1 trunk interfaces. The PSTN phones have SIP-URLs of the form *sip:phone-number@gateway.domain*.

TRUE NUMBER PORTABILITY

In the previous section, we described how device mobility works in a SIP environment. It is important to note that in most cases, a caller does not want to reach a particular phone but rather a particular person. In this section, we show how a particular callee can be mobile, but still be located by the network. We call this feature true number portability.

As with device mobility, true number portability is based upon the REGISTER request of SIP. In order to use this feature, a person authenticates himself to the SIP-Etherphone by synchronizing their PDA with the SIP-Etherphone. After the synchronization, the SIP-Etherphone knows the identity of the PDA's owner. Reading the address book entry of the owner, the SIP-Etherphone also knows the owner's SIP URL (which could be based on their e-mail address, e.g., *sip:Dilbert@3com.com*). At this stage, the SIP-Etherphone registers the user with the main SIP server as being reachable at the SIP-Etherphone's SIP URL. Since every incoming SIP request goes through the main server, these requests are all sent to the correct SIP-Etherphone. If the owner moves and registers with another SIP- Etherphone, the old registration is erased and a new registration entry, pointing to the new location, is created at the main server. This is extremely useful not only in a corporation, but also in a hotel, at a meeting place, or at a convention center. Note that after the synchronization of the PDA with the SIP-Etherphone, no further communication between the two devices is required for the system to operate as described.

Our prototype supports multiple users to register with one SIP-Etherphone by simply synchronizing one after the other at the same SIP-Etherphone. This allows sharing the same SIP-Etherphone during a meeting. Of course, in order for this feature to be practical, the phone may need to support distinctive ringing, or even better, an automated voice announcement of the SIP URL of the callee.

When a user is finished with a meeting, they may sign off from the phone in the meeting room, in which case the main SIP server would forward all calls to voice mail until the user synchronizes with another phone.

In order for the true number portability to become a widely accepted service, it is necessary to provide security mechanisms to validate the user's identity and to prevent eavesdropping. While we did not include such mechanisms in our current prototype, it is straightforward to implement them using known techniques, such as shared secret authentication and public key encryption.

Note that PSTN service providers could offer a similar service by means of a web site that would enable users to forward calls to their current location. However, this is not as convenient to use as synchronizing a PDA wherever the user is located, since a computer is required to access the web site. Even if one exists on the site, it is more time consuming to connect to the web site and enter the pertinent forwarding information. The PDA synchronization makes the process automatic and easy.

ADVANCED CALL SCREENING

Advanced Call Screening features make use of the address book and date book of the PDA. We allow for the assignment of priorities for meetings and addresses. (If the PDA software does not support priorities, a priority indicator string such as "Priority=" can be inserted in an appropriate field to assign the priority level.) The SIP-Etherphone uses the priorities to screen incoming calls. For example, if the user is at a meeting that is assigned a higher priority than a particular caller, then the caller hears a busy signal and the SIP-Etherphone never rings. The user can optionally program the system to forward the low-priority calls directly to voice mail.

In addition, the graphical user interface of the PDA can be used to define the particular behavior for particular callers, e.g., a friend with priority 3 gets a different voice mail greeting than a co-worker with the same priority. Since most PDAs allow the assignment of more than one meeting for a given time, one could, for example, set the entire day as priority 3, so that only calls of priority 3 or higher would reach the owner of the PDA. If there is a meeting between 2 p.m. and 3 p.m. with the CEO, either that meeting can inherit the priority of the CEO in the address book or an alternate priority can be assigned.

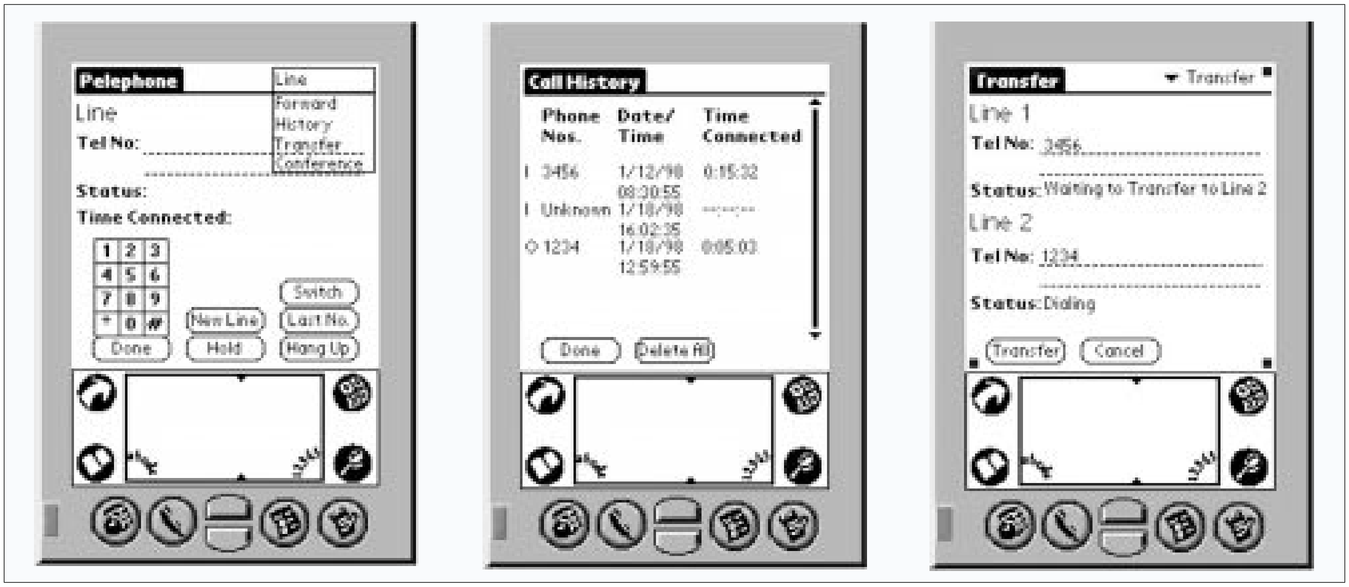
An additional use for personal call screening is to block unwanted calls. For example, one can program the SIP-Etherphone such that callers who are not in the address book get forwarded directly to voicemail.

A USER-FRIENDLY GRAPHICAL INTERFACE

The two features described above do not require any changes to the PDA software. We have further enhanced the system by providing a user interface on the PDA for the SIP-Etherphone system. This application provides the following features:

- An easy-to-use dial-out screen.
- Automatic dialing of numbers in the address book.
- SIP URL-to-name mapping from the address book information.
- Easy configuration of advanced features and services such as conferencing, call forwarding, transfers, etc.
- Voice mail integration.
- Internet access for the PDA through the LAN interface of the SIP-Etherphone.

For the design of this user interface, we have followed the PDA user interface design guidelines provided in [3]. These guidelines are derived from more general graphical user interface guidelines [4], and adapted specifically to the PDA envi-



■ **Figure 2.** *Examples of user interface screens for the PDA phone control application.*

ronment with its screen-size and data-input limitations. We have paid particular attention to:

- Consistency of the look and feel within the application, as well as with the other PDA applications
- The organization of the screens and data input objects in accordance with their usage frequency (e.g., making a basic call is much more common than making a conference call; therefore, the basic call can be made from the main screen)
- The avoidance of screens that are too busy
- Logical layout, alignment, and labeling of objects on each screen
- Fast navigation, selection, and data entry

The user interface application communicates with the SIP-Etherphone via a lightweight protocol. By using such a customized, streamlined protocol, we have achieved a 64-kbyte footprint for the protocol and the user interface. Figure 2 shows some example displays from this application program. Figure 2a corresponds to the main screen that the user interacts with; a call can be made or terminated, and other call features can be accessed from this screen. Figure 2b shows the call history screen, which stores the most recent incoming and outgoing calls. The character I indicates an incoming call, and the character O an outgoing call. The user also has the option to redial any phone number stored in the call history by tapping on the number. Figure 2c shows the call transfer screen, which allows a second logical phone line to be activated, and then the first logical line to be transferred to the second line.

CONCLUSIONS

We have successfully demonstrated true number portability and advanced call screening in a hybrid environment consisting of SIP-based Ethernet phone systems which obtain their current users' information from their PDAs, interconnected to PSTN phones via a SIP/PSTN gateway.

The proposed architecture is flexible and can support enhanced versions of the current PSTN and PBX services, as well as many other new features and services. Furthermore, the architecture is robust, as it does not have single points of failure due to the following design features:

- A large portion of the service intelligence is implemented in the phones; therefore, the main SIP server can be stateless with respect to calls, allowing a backup server to take over transparently in case of a failure without affecting any of the ongoing calls.
- The network can be designed to be highly available through redundant links, switches, and routers.
- The user information is stored in a distributed fashion in the PDAs.

The distributed storage of personal information also removes the burden of backups from the IT department (or the phone company) to individual users. PDAs are regularly synchronized with users' personal computers, which makes this an automated task for the user.

While we focused on SIP in this article, many of the underlying principles and techniques discussed here apply equally well to ITU-T H.323 [5] and other IP telephony signaling protocols [6]. We must also note that although we have used 3Com PDAs and remote access servers for this demonstration, the same architecture and concepts are applicable to similar equipment by other vendors, as long as they have the required features that were discussed throughout this article. Similarly, the same concepts could be applied in a straightforward manner to SIP phones that may be built with network interfaces other than Ethernet, such as ATM or Token-Ring.

The presented PDA-enhanced SIP-Etherphones and the SIP proxy/registrar server clearly show the potential of Internet telephony to greatly surpass circuit-switched telephony service in terms of enabling new applications and offering tremendous service customization capabilities to the end user.

Flexible new telephony services are only part of the many new location-dependent applications that will be enabled by the integration of PDAs with the Internet, especially with the advent of the next-generation PDAs with wireless network connectivity. Some examples of such applications are:

- Travelers at an airport or a train station can receive constant updates about schedules.
- Shoppers can make purchases at a store simply by tapping a few buttons on their PDAs, and the money would be transferred from the PDA to the merchant.
- Shoppers can also instantaneously receive detailed information on location about the products at the store.
- Presenters can remotely control computer presentations using their PDAs.

It is clear that such new applications will bring great value by allowing mobile users to save time and money, and to be more connected at any location.

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BIOGRAPHIES

ISMAIL DALGIC (Ismail_Dalgic@3Com.com) has been a member of the 3Com Corporate Technology Development Center, Santa Clara, California, since 1996. His areas of interest include QoS and signaling aspects of packet voice and video communications, LAN performance evaluation, and information appliances. He has authored various publications and patents in these areas, and represents 3Com in various standards bodies such as ITU-T Study Group 16, ETSI/TIPHON, and IETF. Prior to joining 3Com, he received his Ph.D. degree in electrical engineering in 1996, his M.S. degree in computer science in 1989, both from Stanford University, and his B.S. degree in electrical engineering in 1987 at Middle East Technical University, Ankara, Turkey.

MICHAEL BORELLA (Mike_Borella@3com.com) received a Ph.D. degree from the University of California at Davis in 1995. He is currently a distinguished member of technical staff at the 3Com Advanced Technologies Research Center, Mount Prospect, Illinois. Prior to joining 3Com he was a radio station manager, music journalist, technical writer, software engineer, and university professor. His research focuses on next-generation IP telephony systems, architectures, and protocols. He has authored numerous papers and patents in various areas of networking, including optical networks, performance analysis, multicasting, network packet delay and loss characterization, IP telephony, and MAC layer protocols. He has served on the program committees of the IEEE INFOCOM and International Symposium on Internet Telephony conferences.

RICK DEAN (Rick_Dean@3com.com) completed his B.S. and M.S. in electrical engineering at Northwestern University. He has been employed by Motorola, Andersen Consulting, and Caterpillar. He currently researches Internet telephony for 3Com.

JACEK GRABIEC (Jacek_Grabiec@3com.com) received a B.S. degree from the Department of Electrical Engineering, Automation and Electronics of the University of Mining and Metallurgy, Krakow, Poland. He is currently a member of technical staff at the 3Com Advanced Technologies Research Center, Mount Prospect, Illinois. His research interests include modeling and simulation of communication systems, networks performance evaluation, and data traffic analysis.

JERRY MAHLER (Jerry_Mahler@3com.com) received B.S. and M.S. degrees in electrical engineering from Northwestern University, Evanston, Illinois, in 1995 and 1997, respectively. He is currently a member of technical staff at the 3Com Advanced Technologies Research Center, Mount Prospect, Illinois. His research interests include networking, performance evaluation, multimedia, and wireless communications.

GUIDO SCHUSTER (Guido_Schuster@3com.com) received an Ing HTL degree in elektronik, mess- und regeltechnik in 1990 from the Neu Technikum Buchs (NTB), St. Gallen, Switzerland. At NTB he was awarded the gold medal for academic excellence, and was also the winner of the first annual Landis & Gyr fellowship competition. He received M.S. and Ph.D. degrees, both in electrical engineering, from Northwestern University, Evanston, Illinois, in 1992 and 1996, respectively. In 1996 he joined the Network Systems Division of U.S. Robotics, Mount Prospect, Illinois (now the Carrier Systems Business Unit of 3Com). He co-founded the 3Com Advanced Technologies Research Center and is currently associate director of the Center. He is also an adjunct professor in the Electrical and Computer Engineering Department at Northwestern University, Evanston, Illinois.

Ikhlaq Sidhu (Ikhlaq_Sidhu@3com.com) received his B.S.E.E. degree from the University of Illinois, Urbana-Champaign, and M.S.E.E. and Ph.D. degrees from Northwestern University, Evanston, Illinois. He currently directs the Advanced Technologies Department, which is the applied research and advanced development program for the Network Systems Business Unit of 3Com Corporation. His research interests include architectures, technologies, and new services for next-generation communication infrastructure and network access.