

# **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, Ikhlaq Sidhu  
3Com Corporation**

Contact: Ismail Dalgic  
3Com Corporation  
Technology Development Center  
5400 Bayfront Plaza, Mail Stop 3219  
Santa Clara, CA 95052  
U.S.A.

Tel: (408) 326-1494  
Fax: (408) 326-8188  
Email: [Ismail\\_Dalgic@3Com.com](mailto:Ismail_Dalgic@3Com.com)

## **Abstract**

Custom Local Area Signaling Service (CLASS) features offered in Public Switched Telephone Network (PSTN) have certain limitations due to the closed nature of the PSTN network signaling. The adoption of Telephony over Internet Protocol (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 (PDA). In this paper 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 (PBX) 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.

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## 1 Introduction

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, call forwarding, etc. However, these services have certain limitations, primarily due to the closed PSTN signaling architecture which does not allow access to the network signaling protocols by the 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 (PDA) are very suitable to provide such information, because they store personal data, and they 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 IETF Session Initiation Protocol (SIP) [Hand99] (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.

## 2 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 PBX-based switching environments.

### 2.1 Traditional PSTN CLASS Features

For many years, the PSTN has provided Custom Local Area Signaling Service (CLASS) features to residential customers. Once a customer subscribes to a CLASS feature, they may activate and/or de-activate 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 #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 on-line via a touch-tone 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.

## **2.2 PBX Features**

The PBX, or Private Branch Exchange, 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 is a wide variety of PBX services and features. Custom configurations are common. All PBX's 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 below:

- *Call transfer*: An established call may be transferred from one number to another number 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 queueing*: 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 re-established 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.

### **3 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 paper improve upon 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 that calls are to be forwarded to, rather than from the location that calls are to be forwarded from. This results in a call forwarding mechanism that is truly portable, as it follows the user wherever he/she goes. The PDA integration also enables a more sophisticated priority mechanism. It allows calls to either be completed, blocked, or sent to voice mail, depending upon the priority assigned to the caller in the address book of the PDA. This 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 which are described in this paper, many other new applications and services are enabled by the coupling of two new paradigms: (i)

shifting intelligence from the network to the telephones; (ii) 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 the 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 on 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, their cellular phone or their pager.
- If the PDA has voice playback capability, it can download voice mail 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 cannot spend too much time on the phone to check their voice mail. For example, a traveler at an airport may download their 30-minutes worth of voice mail in a few minutes just before taking their flight, and may listen to those messages during the flight.

## 4 SIP overview

The Session Initiation Protocol is an application layer control protocol that can establish, modify and terminate multimedia sessions or calls [Hand99]. There are two major architectural elements to SIP: the *user agent* (UA), and the *network server*. The UA resides at the 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*

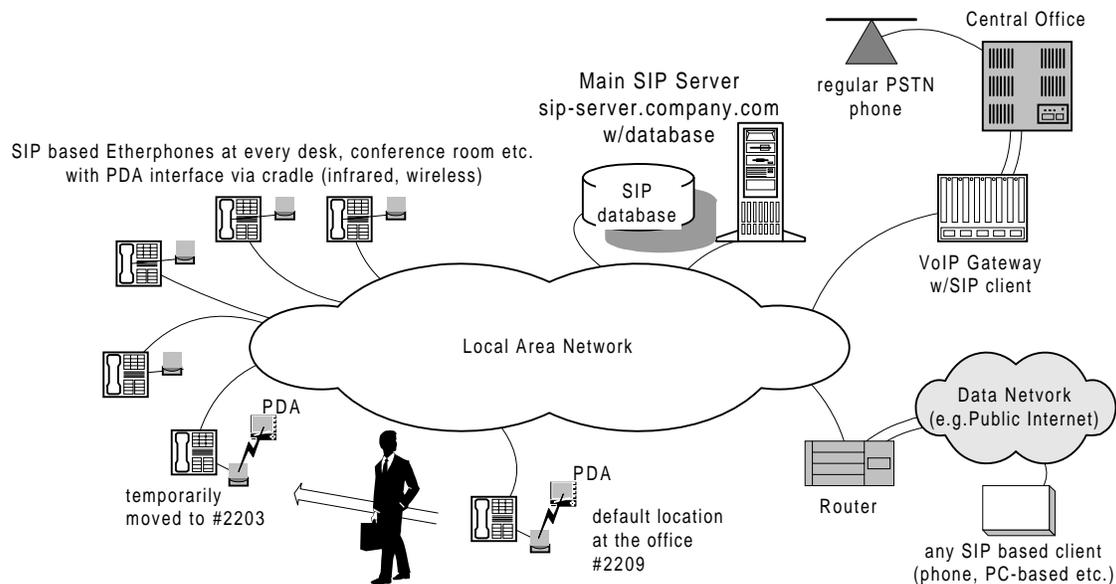
*server*, and a *registrar*. A basic SIP call does not need servers, but some of the more powerful features depend upon them.

The most generic SIP operation involves a SIP UAC issuing a request, a SIP proxy server acting as 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 session description that informs the called party what 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 non-forking. 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 (called a SIP URL) and the associated IP address. The most common use of a registrar server is to register after start-up, 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 paper. Note that usually a SIP network server implements a combination of different types of servers.

## **5 The System Architecture**

In this section, we describe the components that comprise the system, and the interactions among them. The system architecture is shown in Figure 1.



**Figure 1: System Architecture**

## 5.1 SIP-Etherphone

An Etherphone is a telephone that can be plugged into an Ethernet port. In most cases, Etherphones support Internet Protocol (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 protocol (RTP [Schu96]), which itself is carried inside of User Datagram Protocol (UDP).

The purpose of the signaling stack is to set-up, manage, and tear-down a call. During the set-up phase, the location of the endpoint is discovered, 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 tear-down phase, the call is terminated gracefully. We chose SIP as the signaling

protocol for our architecture because it is a lightweight protocol which is easy to implement, and it supports all the functions that were 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 also has to manage a receiver buffer to compensate for network jitter.

The SIP-Etherphones in our system have pre-programmed phone numbers, represented as SIP-URL's that are 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<sup>1</sup>. 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.

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<sup>1</sup> All other LAN telephones that we know of use proprietary signaling protocols.

## **5.2 PDA**

The PDA maintains personal address-book 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: 1) authentication of the owner, 2) a graphical user interface and 3) personal data of the owner. While the PDA significantly enhances the system as discussed in the subsequent sections, it is not necessary for the PDA to be communicating 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 to navigate the user interface.

## **5.3 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 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 (like the address book and schedule information stored in the PDA) across the network from the electronic organizers via SIP-Etherphones to the server.

## 5.4 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 that are of the form *sip:phone-number@gateway.domain*.

## 6 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 let users forward calls to their current location. However, this is not as convenient to use as synchronizing a PDA wherever the user is at, 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.

## **7 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 say 2 and 3 p.m. with the CEO, that meeting then can either 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 voice-mail.

## **8 User-Friendly Graphical Interface**

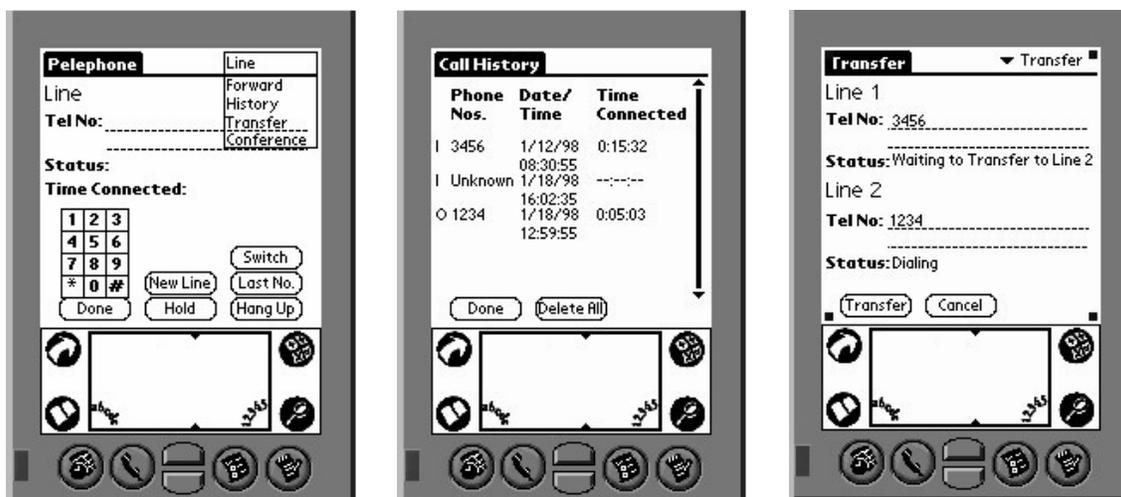
The two features described above do not require any changes to the PDA software. We have further enhanced the system by providing a user interface on the PDA for the SIP-Etherphone system. This application provides the following features:

- 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 [Cook98]. These guidelines are derived from more general graphical user interface guidelines [Ambl98], and adapted specifically to the PDA environment 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 the data input objects in accordance with their usage frequency (for example, making a basic call is a lot 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. Part (a) 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. Part (b) 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' indicates an outgoing call. The user also has the option to redial any phone number stored in the call history by tapping on the number. Part (c) 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.



(a) Main Screen

(b) Call History Screen

(c) Call Transfer Screen

**Figure 2: Example user interface screens for the PDA phone control application**

## 9 Conclusions

We have successfully demonstrated true number portability and advanced call screening in a hybrid environment consisting of a 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. This is because (i) a large portion of the service intelligence is implemented in the phones, and 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; (ii) the network can be designed to be highly available through redundant links, switches and routers; and (iii) 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 the individual users. PDAs are regularly

synchronized with the users' personal computers, which makes this an automated task for the user.

While we focused on SIP in this paper, many of the underlying principles and techniques discussed here apply equally well to ITU-T H.323 [H323-98] and other IP Telephony signaling protocols [Schu98]. 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 paper. 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: (i) travelers at an airport or a train station can receive constant updates about flight or train schedules; (ii) 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; (iii) shoppers can also instantaneously receive detailed information on location about the products at the store; (iv) 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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