Integrated Internet Appliances: More than Just a Phone

Henning Schulzrinne
Internet Real-Time (IRT) Laboratory
Dept. of Computer Science
Columbia University
New York, New York
hgs@cs.columbia.edu

February 15, 1999

1 Introduction

For the first time in a generation, basic communication services like sending letters, making phone calls, listening to the radio or watching TV have begun to change, with a migration to email, Internet telephony, Internet radio and TV in various stages.

In this white paper, we discuss some of the new services that we envision in a communication environment dominated by the Internet. We first describe the components that we have developed and then provide details how these components can be combined to implement these services.

We have developed a set of four related components:

e*phone Ethernet "phone": The Ethernet phone (e*phone) is a low-cost Internet audio device that connects to any twisted-pair 10 Mb/s Ethernet. It features a high-performance digital signal processor that can handle both low bit-rate voice encodings such as G.723.1 or G.729, as well as high-quality music encodings such as MPEG I layer 3, commonly known as MP3. The hardware has a smaller liquid crystal display for the local address book, dialing, status, configuration and the selection of audio content. Audio can be directed to the speaker, the handset or line output, with audio input from the handset, a microphone or line input.

The e*phone is the first complete Internet phone that can communicate directly with any other Internet-standards-compliant software application. It is anticipated that the phone would cost roughly the same as a better home speaker phone, around $50 in parts, but it supports, beyond standard PBX phone functionality, additional functionality, such as high-quality audio (see Section 8) and interfaces to external sensors (see Section 9). The phone will implement a complete IP telephony protocol stack, including IP, UDP, RTP for audio transport, DHCP for automatic IP address assignment, DNS for name lookup, SIP for session setup and teardown ("signaling"), RTSP for voice mail access and SDP for session description.

The e*phone is initially targeted for business environments, since it requires an Ethernet connection, but can also be used with cable modems and ADSL, since both have an Ethernet interface to the home. We plan to add other interfaces, including wireless Ethernet, in the future.

sipd **SIP call handling server:** The SIP server provides network support for SIP-capable end systems. It allows redirecting and proxying (forwarding) calls and can interface to a number of user location services. The behavior for calls can be programmed (see Section 5). User location mechanisms currently include interfaces to LDAP databases (in progress), recursive lookup and login-based mechanism. In addition, a variety of names can be mapped to the canonical user identifier, for example, `j.smith`, `john.smith` and `j-smith` will all reach the same user, if unique, or return a list of possible matches, if not. Calls can be proxied to several locations, so that a single "phone number" can ring on any number of dispersed end systems, with the first to pick up receiving the call. Sequential search is also possible, where the server first attempts to ring a primary address, and then switches to a secondary location if there is no answer or the phone is busy. Simply plugging in a second phone registered to the same user will cause both phones to answer, with more flexible search orders setable by each subscriber.

The sipd server currently runs on Windows NT, Sun Solaris, Linux and FreeBSD.

rtspd **RTSP multimedia server:** The RTSP multimedia server can stream general stored media using the RTSP protocol. Unlike other servers, it can also record media and is thus suited as a voice mail server. Stored media and voice mail can be retrieved from any RealNetworks G2 client or other clients implementing RTSP. We are currently adding the capability to stream MP3 audio from that server.

The rtspd server currently runs on Windows NT, Sun Solaris, Linux and FreeBSD.

sipc **SIP client:** The SIP client complements the e*phone hardware. It runs on standard workstation and PC platforms and can originate and receive SIP-based phone calls, interface to local address directories and manage sessions, i.e., adding and deleting participants and transferring participants to another destination. It does not implement audio and video coding, leaving that to applications of the user's choice.

2 Common, Open Protocols vs. Proprietary Solutions

Traditionally, there are two types of phone "customer premise equipment": traditional analog phones that work, more or less, worldwide, subject to regulatory approval and variations in plugs, and PBX-based phones that are proprietary to a particular PBX vendor. There are also digital ISDN phones, but they have not found much use outside a few European countries that have heavily promoted ISDN.

For all three types of phones, the available functions and how to invoke them depends strongly on whether one makes calls within the same company, within the same LATA or elsewhere. For example, caller ID may not work across providers and call transfer and parking is usually only allowed within the same organization.

Internet phones, if designed right, avoid both problems, in that they work the same everywhere and with any "Internet PBX". However, this is true only if the Internet phone is based on an open standard. Two standards are required for minimum interoperability, beyond the physical interface and the network layer: voice data transport and signaling. For voice data transport, the Real-Time Transport Protocol (RTP) [1, 2] has become the accepted standard. For signaling, i.e., the setup and teardown of phone calls, two standards are currently being developed, H.323 [3] by the ITU and SIP [4] by the IETF.

We believe that SIP is more closely aligned with the existing Internet architecture, being more closely related to a web- rather than traditional-telephone model. SIP offers integration into web and email by its use of addressing structure, the ability to accommodate a diversity of media types beyond audio and video and the ability to return web pages and email addresses. SIP already has a fully developed and interoperable "wide-area" model, while H.323 is still defining how inter-gatekeeper communications should work. Currently, SIP call setup is significantly faster than H.323, since it can use UDP and thus avoids the use of multiple TCP connections as in H.323. A basic SIP implementation can be implemented in an embedded device with minimal hardware, while existing H.323 implementations have only been demonstrated in full-fledged operating systems such as Windows or Unix. A more detailed comparison between the two protocols can be found in [5].

Just like for the two email protocols, SMTP/POP and X.400, a few years ago, we anticipate that both SIP and H.323 will be used for signaling.

SIP is now an Internet Proposed Standard, the first level of Internet standardization. A number of vendors have started to develop prototypes and products based on this standard[1]. Products include both large and small gateways into the telephone systems, as well as servers.

SIP has also been used, by the PINT working group within the IETF, for controlling regular phone calls and Internet call waiting.

3 Internet Phone Functionality

A network-connected phone has a number of conveniences that go beyond what is commonly available in standard residential or PBX phones, such as:

- high-quality audio, sounding like FM radio or audio CDs rather than "telephone quality";
- simplified call transfer to any address, without "hairpinning";
- indication of priority and subject of the call, as well as name, address and organization of the caller;
- indication of return calls for call filtering;
- flexible multi-party calls, using either a mesh of unicast connections, a bridge or multicast;
- dialing by name, with a directory lookup in the network;

[1] see http://www.cs.columbia.edu/~hgs/sip for a partial listing

- “web based IVR”: instead of a long voice menu, the caller receives a web page with further directions and simply clicks on the person or department she wants to reach;
- automatic downloading of speed-dial lists that can be edited and stored on a server;
- automatic clock setting;
- “do not disturb” button; if set, it forwards incoming calls to a designated location such as a secretary or voice mail;

4 Intelligent vs. Dumb End Systems

Recently, two architectures have emerged for providing Internet telephony service. They can be roughly described as “smart”or peer-to-peer vs. “dumb” or master-slave end systems. Smart end systems have a complete IP stack and can make calls on their own, without assistance of another network device. They may, however, use network devices such as proxy servers for services, particularly user location and call screening. SIP-speaking devices fall into the “smart” category.

On the other hand, “dumb” end systems cannot initiate calls on their own, but rather rely on a controller to tell them what to do, e.g., to ring, send packets to a particular destination, or to stop sending packets. The devices are often called media gateways (MGs), since they may well allow the attachment of one or more “classical” telephones. The MGs are then controlled, master-slave fashion, by a media gateway controller (MGC). There currently is no standard MG-to-MGC protocol, but MGCP [6] is the one most completely specified at this point. (In all but very small installations, MGCs need to communicate with each other in a peer-to-peer manner. SIP has been proposed by a number of companies, including Bellcore, Cisco, L3 and MCI-Worldcom as the inter-MGC protocol.)

The primary motivation, or at least the one people will admit to, is that peer-to-peer protocols are too complicated to implement for cheap end systems. This motivation to be primarily guided by the complexity of H.323 end systems that require a full workstation or PC, with a full-function operating system. Our e*phone shows that this need not be the case if the peer-to-peer protocol is chosen correctly. From our analysis, implementing a master-slave protocol like MGCP would not be significantly less complicated than the current SIP implementation.

The choice between the two models has other consequences. With a peer-to-peer model, it is up to the end system to decide which services to perform locally and which to delegate to one or more servers, possibly furnished by different service providers. Dumb end systems do not have this ability – all services must be provided by a server in the network.

If an end user wants to change service providers, they can do this in both models by reconfiguring their end system. However, in the master-slave model, they can then receive calls only through their new provider. In SIP, it is easy to register with several providers at the same time and change their relative preference.

From the perspective of the network provider, the master-slave model allows to charge for calls and services in a manner similar to today’s charging model, with services deployed as the provider sees fit. With the peer-model, control rests with the end user, making traditional per-call and per-service charges less likely.

Because of the different incentive structures, we anticipate that both models will exist for some time, just like multi-standard or multi-homed cellular phones. We are considering adding a master-slave capability to the e*phone to demonstrate that both can co-exist within the same end system.

5 Programmable Telephone Services

Currently, telephone services are prescribed by the telephone company, with the exception of very large corporations that can create a limited set of customized services. This is quite different for web and email. For web services, a large number of people have added programmed features to their web pages, from Javascript and simple visitor counters or email forms to full-fledged e-commerce applications. These features are often added using scripting languages such as Perl, Tcl, Python or Visual Basic, which allow rapid development of small programs, offering a higher abstraction than language like C++ or Java. (Much of server backend of yahoo, for example, is written in Perl.) Similarly, an increasing number of email users have installed email filters, either as part of their email application, as scripts like procmail or in the server.

There is no reason that phone calls could not be equally programmable, by both local system administrators and end users. We have developed two approaches, sip-cgi and a call-processing language, that make it easy to write call processing logic.

sip-cgi [7] is derived from the web CGI (common gateway interface) [8], which is the most common interface between web servers such as Apache or IIS and the scripts that implement services. sip-cgi provides a familiar interface to web programmers, adding the ability to proxy and redirect calls. sip-cgi scripts are invoked by every request and response. With sip-cgi, interfaces to proprietary databases or other call-routing applications can be added without modifying the server itself. SIP is particularly suited for scripting, as it uses a textual format for conveying request and response parameters.

We anticipate that most sip-cgi scripts will be written by administrators, service providers and "power users". For users who do not want to program, we are designing a call processing language (CPL) [9] that is geared to be generated automatically by graphical user interfaces. The language is based upon XML, and is designed to be safe, predictable in its runtime resource requirements, and verifiable when written.

sip-cgi and CPL scripts can execute in either the server or the client. In our server implementation, they are invoked both for outgoing and incoming calls. Administrative server scripts can be used to enforce corporate policies, look up "phone numbers" and bill for services, while individual user scripts can be used to restrict and direct incoming calls. More detailed examples for their use can be found in [9].

6 True Multimedia Sessions: Beyond Audio and Video

Current multimedia conferencing systems focus on traditional business conferences, with audio, video and possibly slides. However, the concept of sessions in the Internet can be far reacher. Sessions can include any Internet application, including games, chat or shared virtual reality. Inviting somebody to a game of Doom, where participants can see and hear each other, is not fundamentally different from a regular phone call. SIP makes it easy to include any "media" and network application in the session setup process, as well as add other media later during a session.

7 Internet Paging

Internet paging has two meanings and both apply to the e*phone: audio paging and text paging (short message service). Audio paging can be implemented as a special multicast group that automatically overrides the current audio selection.

8 Internet Radio and Remote CD Player

Our e*phone can operate not just as a telephone, but also as a "radio" or "remote CD player". In radio or *multicast mode*, it "tunes" to one of a set of multicast groups, equivalent to radio stations. Each such multicast group broadcasts audio in one of several audio formats, including CD-quality MP3 (MPEG layer 3). A single Ethernet can carry about 70 radio stations. The radio stations are displayed by title on the device and can be selected by pressing a number key.

In *on-demand mode*, the e*phone device acts like a CD player, except that the music is stored on a PC-based server rather than on traditional CDs. We envision that the display would show selections by genre and title, with standard CD player remote-control functionality like pause and next track. A single PC server should be able to supply about 50 concurrent music streams [2].

This combination of radio and telephone has obvious cost advantages, but it is particularly useful in specialized environments such as hotels, hospitals or cruise ships. For example, even luxury hotels often have $9.95 radio clocks, which can barely receive two stations and may or may not display the correct time. With the integrated e*phone device, wakeup by phone or radio can be programmed on the same device, with decent-quality audio and station listings. While an in-room CD player is rarely found, the hotel can now offer a large selection of music to the guest.

Having incoming phone calls automatically interrupt the current radio broadcast or pause the current audio-on-demand session is a small side benefit.

9 Health and Environmental Monitoring

The e*phone can also serve as a monitor for non-audio signals. Adding sensors to the device is straightforward, as it already has both digital and analog inputs. Digital sensors might include burglar alarm switches, motion detectors or fire alarms, while analog sensors can include sensors for temperature, humidity, light intensity, or wind speed. Thus, the e*phone can simultaneously serve as the room thermostat and intrusion alarm.

Another application is medical and health monitoring. Here, we envision that a wireless version of our e*phone could transmit on-going monitoring data (e.g., for ambulatory care after hospitalization or chronic conditions such as diabetes). Using the same system for both audio (telephone) and measurement decreases cost and reduces the number of devices that the patient has to wear.

[2]This is a rough estimate.

The subscribe/notify mechanisms (Section 11) is a natural fit for event-based monitoring. For example, if a particular medical parameter reaches a threshold, a voice connection can automatically be established to advise the patient.

There is almost no additional cost incurred by adding sensing capabilities, as they directly connect to the existing logic and A/D converter.

10 Internet PBX and Centrex

An Internet PBX uses the normal local area network and external Internet for voice transport. Thus, no telephone switches in the traditional sense are required. The simplest Internet PBX does not require any computer: simply plug a number of e*phones into an Ethernet hub. For larger installations, a SIP proxy server adds directory and other services to the PBX. An Internet voice mail server (Section 12) enhances the Internet PBX.

For the next few years, most phone calls originating on an Internet phone will still terminate on a regular phone. Thus, gateways to the regular phone network are a necessity. These gateways can either be located on the premises, or be provided by the ISP or phone company. Having the gateway outside the company, as a form of “Internet Centrex”, has the advantage that the business then only needs a single high-speed data connection to the ISP. Also, cost per line decreases if a larger trunk is purchased; with larger trunks, higher average utilization is possible while maintaining the same blocking probability. The Centrex model is particularly attractive if the business is connected via ADSL and cable modems, as their per-circuit cost is significantly below that for regular voice lines. A single connection to the Internet makes it possible to more efficiently share bandwidth between voice and data applications, particularly if their usage peaks are offset in time.

An Internet Centrex solution is independent of the artificial restrictions imposed upon PSTN Centrex. For example, “extensions” can be anywhere in the world, not just in the same LATA.

We anticipate that many ISPs will want to offer such Internet Centrex services, just like they are offering web hosting or email storage.

11 Instant Messaging and Presence

Instant messaging (IM) is one of the most popular applications for consumer ISPs. For example, AOL moves 42 million e-mails, but 309 million instant messages among its 16 million members per day. Instant messaging actually encompasses two aspects: tracking the presence and availability of other subscribers (“buddies” in AOL terminology) and sending text messages. In the long run, we anticipate that the presence mechanism will be used not just to know when one can send a text message, but also to initiate phone calls, multi-player games and multimedia conferences. (In the game case, a player might want to be notified when a sufficient number of similarly-skilled players have “arrived”.) This is a generalization of the call-back and call-queueing mechanism currently found in telephone systems, except that they avoid putting callers on hold for long periods of time. It also makes it possible to find out if the caller is currently receptive to phone calls, avoiding interrupting a meeting.

SIP can readily be extended to handle presence information [10] by adding a subscribe/notify mechanism. Many of the underlying features of SIP are rather useful in this environment, for example, the ability to branch or redirect requests.

The subscribe/notify approach can be extended beyond indicating the presence of people. For example, one can subscribe to certain events and then be notified in real time. Such events might include measurements taken by the e*phone (Section 9), computer system conditions or events in “real life”. One could imagine a “surf notification” feature that automatically sends an event notification when the weather conditions are appropriate.

We are currently experimenting with integrating the subscribe/notify extensions into our SIP application.

12 Voice Mail and Unified Messaging

Unified messaging is the ability to present all forms of store-and-forward communications, such as voice mail, fax and email, in a single interface. While there are a number of emerging proprietary solutions, we propose to use Internet standards, implemented in our servers, to offer Internet voice mail. In our setup, the e*phone, the sipc client or the sipd server forward the call to the voice mail server. The voice mail server, currently under construction, is itself a SIP user agent and “picks up” the call. It then generates RTSP requests for the outgoing message, intercepts DTMF (touch tone) and triggers recording. At the end of the message, a URL pointer is sent via email to the recipient. The recipient can then pick up the message from any Internet-connected host using any standard Internet multimedia tool, e.g., RealNetworks player. Since the message is a regular email message, it can be forwarded to anybody, without relying on proprietary systems. Note that the audio samples remain on the server, allowing higher quality and avoiding the delivery of huge email messages.

13 Internet Telephony “Portals”

Unlike their “classical” predecessors, Internet telephony and radio allows to separate the transport of bits, the provision of content and the delivery of indexing and filtering services. For example, for Internet telephony, bits can be transported by any ISP; content such as voice announcements or voice mail recording can be provided by a web hosting company and services can be provided by the subscriber’s business, professional association, alma mater or any other group or organization. This allows competition and new business models.

Unlike mail forwarding, telephone management services do not require large amounts of storage or bandwidth, as only the setup and possibly teardown request reaches that service. The subscriber to such a service simply configures the service through a web page. Examples of services geared towards consumers and individual business users might include:

Calendar-based filtering and forwarding: Particularly if the server has access to the user’s calendar, it can intelligently refuse or forward calls. For example, a user may choose to reject calls from callers she does not know between 5 and 7 pm. If the calendar contains an entry for the time of the call, the call can be automatically forwarded to voice mail, with a better time to call provided as well. “Special” callers may still be able to get through, or be provided with more detail, so that family and friends may receive “on vacation until Monday” while random callers will just be rejected without further explanation.

Rule-based filtering: Subscribers can readily formulate rules such as “during the day, try home and business number in parallel; at night, just ring at home”. SIP-based Internet phone calls can be identified as returning an earlier call, by a subject, the name of the caller and callee, the caller’s organization and the call’s urgency.

Rules can also be cooperative, so that one may automatically subscribe to a service that filters calls from businesses that have not been approved by (say) the Better Business Bureau. Cooperative telemarketer filtering might allow a group of people to share dislikes, so that the telemarketer only reaches the first person from that group.

Multiple identities: Unlike scarce telephone numbers, it is quite common for users to have multiple email addresses. Similarly, for SIP telephone service, subscribers may well maintain different addresses based on their employment, college they went to, affiliation with a sports club or professional organization or a particular pastime. All of these identities are then routed, on demand, to one or more end systems, possibly with different filtering. This makes it possible to have a much more fine-grained privacy mechanism compared to just a listed or unlisted phone number. Indeed, we are developing algorithms that allows callees to identify callers, but only if they know the caller or submit the caller’s identity to law enforcement authorities (even after the nuisance call).

Local number portability: Efforts are under way to provide local number portability, i.e., the ability to keep one’s number even if changing telephone companies. Servers and mechanism for this are specialized and expensive. Using Internet telephony mechanisms, even if the caller and callee are using traditional circuit-switched phone services, would greatly simplify the lookup process. Once Internet telephony has taken hold, there does not need to be a tie between one’s “bit transport provider” and user identity.

For businesses, services previously only available to large corporations now can be offered to even the smallest enterprise. For example:

Automatic call distribution: The proxy server can track the number of calls accepted and pending for a particular agent. Unlike with traditional phones where caller-id information is sometimes coarse, it is easy to route subsequent calls from the same person, regardless of where they are calling from, to the same agent. Calls can be distributed regardless of the physical location of the caller or agent; the voice call itself is not routed to the ACD device. This makes it possible to maintain one call center with worldwide reach. In addition, SIP calls can indicate the language preferences of the caller, making call routing and voice menus more efficient.

Virtual private networks: As discussed in Section 10, Centrex services can span any geographic area, so that a distributed office with small branches in a number of cities can appear like a single company to the outside world.

Voice mail and unified messaging: See Section 12.

We have seen the success of the separation of services from transport in the case of email, where hotmail.com, for example, now has 28 million e-mail accounts, with 150,000 subscribers being added daily [3]. Among other reasons, it allows people to maintain separate work and private identi-

[3] `http://microsoft-online-sales.com/hotmail.ASP`

ties, provides spam filtering and avoids being tied to an ISP's domain name in one's email address. Similarly, colleges and universities have begun to offer permanent email forwarding accounts to their alumni [4].

Indeed, given that Internet telephony may limit the potential for differentiating oneself based on price or voice quality, ISPs and other organizations may well come to view the provision of services as a mechanism to bind users, just as the portal sites have all begun to offer free email and calendaring services. (Indeed, it is conceivable that portal sites may offer this service once Internet telephony becomes sufficiently popular.)

14 IP Telephony as a Means for Achieving Local Competition

IP telephony may well provide a means for ISPs, interexchange carriers and cable television providers to enter the local exchange market, as long as the access to the customer is available on an equal basis. Instead of provisioning a whole service, only the physical copper plant to the first switching element (switch or SLIC) needs to be made available to the competitor, simplifying the issue of operations support systems (OSS). If the DSLAM is owned by the monopoly local provider, changing IP providers implies changing the DHCP server that the ADSL modem listens to.

As a side note, the notion of choosing a different long distance "telephone company" does not directly apply. In an IP world, the choice of IP access provider determines who carries the packets end-to-end. There are two natural places where hand-off between the local access monopolies such as the coax or copper providers can take place, at the LATA or at each switch. There are 165 "real" LATAs, but 18,803 switches (1982). A more realistic number is the number of local dialing areas. Nationwide ISPs seem to have beetween 300 and 500 local access numbers to ensure no-toll calls.

15 Status

The four components are currently in various stages of development:

e*phone Ethernet "phone": A prototype version of the e*phone is operational, having been demonstrated successfully to MCI and 3Com. We are currently adding DHCP and DNS support, as well as echo cancellation and additional SIP features such as address books, "do not disturb" and call forwarding. A second hardware version, with somewhat more memory, is being designed.

sipd **SIP call handling server:** The server is currently operational in redirect mode, with the final touches for proxy mode being worked on. sip-cgi is supported in redirect mode, with support for LDAP in progress. We expect to have version 1.0, with proxy support, ready for outside availability late February. A number of prototype name resolution "plug ins" are ready and tested. See http://www.cs.columbia.edu/~hgs/sipd for details.

[4] *Online, Universities Link to Alumni and Their Wallets*, New York Times, March 26, 1998

rtspd **RTSP multimedia server:** The RTSP server is currently operational in playback mode and with limited file type support (audio files); record mode is being added. We anticipate a version usable for voice mail by May 1999.

sipc **SIP client:** A basic client has been operational for a number of months. Enhancements for the integration of encryption, call control functionality (transfer, multi-party calls) will be tackled after the server "ships".

In addition, we plan to derive a user agent SIP library from our server code. This makes it possible to rapidly develop SIP clients in C.

16 Summary

The architecture and vision outlined above requires new end system and new software. The IRT research group is developing the necessary technology, hardware designs and software to provide major pieces:

- the e*phone as the common hardware platform for telephone, pager, radio, remote CD player and environmental/health monitoring;
- the `sipd` SIP server for network intelligence and programmable services, described in more detail at http://www.cs.columbia.edu/~hgs/sipd;
- the `sipc` SIP workstation/PC client for advanced services.
- the `rtspd` RTSP server for voice mail.

Together, this white paper has shown, how they provide an evolvable platform for improving traditional communication services as well as new services and new applications.

References

[1] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: a transport protocol for real-time applications," Request for Comments (Proposed Standard) 1889, Internet Engineering Task Force, Jan. 1996.

[2] H. Schulzrinne, "RTP profile for audio and video conferences with minimal control," Request for Comments (Proposed Standard) 1890, Internet Engineering Task Force, Jan. 1996.

[3] International Telecommunication Union, "Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service," Recommendation H.323, Telecommunication Standardization Sector of ITU, Geneva, Switzerland, May 1996.

[4] M. Handley, H. Schulzrinne, E. Schooler, and J. Rosenberg, "SIP: session initiation protocol," Internet Draft, Internet Engineering Task Force, Jan. 1999. Work in progress.

[5] H. Schulzrinne and J. Rosenberg, "A comparison of SIP and H.323 for internet telephony," in *Proc. International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV)*, (Cambridge, England), July 1998.

[6] M. Arango, A. Dugan, I. Elliott, C. Huitema, and S. Pickett, “Media gateway control protocol (MGCP),” Internet Draft, Internet Engineering Task Force, Feb. 1999. Work in progress.

[7] J. Lennox, J. Rosenberg, and H. Schulzrinne, “Common gateway interface for SIP,” Internet Draft, Internet Engineering Task Force, Nov. 1998. Work in progress.

[8] K. Coar and D. Robinson, “The WWW common gateway interface version 1.1,” Internet Draft, Internet Engineering Task Force, Dec. 1998. Work in progress.

[9] H. Schulzrinne and J. Lennox, “Call processing language requirements,” Internet Draft, Internet Engineering Task Force, Aug. 1998. Work in progress.

[10] J. Rosenberg and H. Schulzrinne, “SIP for presence,” Internet Draft, Internet Engineering Task Force, Nov. 1998. Work in progress.