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Experiences with PC-Based Real-Time Multimedia Collaboration over IP

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Experiences with PC-Based Real-Time Multimedia Collaboration over IP

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ABSTRACT

Intel recently completed a trial of standards-based real-time multimedia collaboration tools running on laptop computers over an Internet Protocol (IP) network. Key goals for the trial included validation of usage models and user benefits while utilizing the multimedia collaboration tools in a production work environment. Trial participants were equipped with a multimedia "softphone" application, a headset, and a Webcam that enabled them to establish high-quality small-group (multiparty) voice and video calls. Call setup was performed with the standard Session Initiation Protocol (SIP) and open-source products. These components provided a cost-effective, easy-to-setup and use collaboration environment, where all communications and collaboration were integrated into one device, the PC. This capability provided increased productivity because all information that people needed to do their jobs was literally at their finger tips.

The trial included an evaluation of high-quality voice calls using wideband codecs. These codecs encode twice the frequency range as that provided in the public telephone system, thus conveying most of the frequencies made by the human voice which enables them to deliver more lifelike speech and increased intelligibility. Because wideband algorithms are computationally intense, they most often appear in PC voice products. Thus the use of the PC for voice calls can deliver better quality conversations than we normally hear with telephones and cell phones.

The trial also tested PC client-based voice and video conferencing provided by the softphone application. Client bridging provides an easy way to conduct meetings on demand without requiring scheduling of conferencing bridges. The trial showed that laptops based on Intel® Centrino® mobile technology deliver the performance to conduct small group meetings of up to the limits of the "softphone" application (5 video or 7 audio-only participants).

In the trial, users found value in the integration of their communications environment into the PC, and in fact were looking for integration beyond the experience we provided. However, we also learned that both the tools and the infrastructure will need to be further optimized before users would be willing to fully replace their time-tested desk phone with a PC-based collaboration solution.

In this paper we describe the architecture, deployment, and key learnings for the trial. We also portray the business value of wideband audio and peer-to-peer conferencing.

INTRODUCTION

Collaboration capabilities are a key element of Intel's corporate growth strategy, as many employees participate in teams which span the globe. The ability to allow individuals to effectively collaborate remotely from their home sites as virtual teams, rather than physically attending meetings, allows employees to be more efficient with their use of time, and allows the corporation to reduce expenses by avoiding unnecessary travel. Many Intel employees are mobile–moving between conference rooms, cafeterias, labs, and their cubicles within their home campuses, working from home, traveling to other Intel campuses as well as visiting customer or supplier sites. Several years ago, Intel Information Technology (IT) launched a program to enhance employee productivity by equipping most Intel employees with notebook computers. Doing so allowed employees to carry with them the data and applications necessary to do their jobs. Over the past three years, IT has enhanced access to corporate data by equipping all Intel sites with wireless local area networks to enable access from anywhere within a site. IT continues to enhance the notebook computing environment by improving connectivity and collaboration tools to increase employee effectiveness.

Last year Intel launched a Digital Office initiative, which focuses on enabling real-time business by applying technology in four key focus areas for employee and IT manager benefit. The four key areas or "pillars" are 1) Seamless Collaboration, 2) Pervasive Connectivity, 3) Embedded IT, and 4) Information Assistance. Seamless Collaboration allows an employee to collaborate anywhere with anyone to accomplish his/her task. Since the Seamless Collaboration pillar highly correlates with the way Intel employees work, the Digital Office and IT organizations jointly defined a proof of concept trial to validate several key capabilities of Seamless Collaboration, including the following:

- Integrated collaboration consisting of voice telephony and client-to-client video, all running on the trial participant's notebook computer.
- Client-based (notebook computer) small group conferencing allows small group audio and video meetings conducted just on the laptop (with one laptop mixing the multiple audio and video streams), with no requirement to have other conferencing equipment nor to schedule the meeting ahead of time.
- "Click to call" capability uses a directory on the PC, which gives the user the ability to click on one or more names to instantly call people and establish a conference.
- High-fidelity audio conferencing, enabled with the use of wideband codecs on the PC.
- Unified messaging, the storage of voice mail messages in e-mail with the ability to listen to a message on the PC.

IT saw the trial as an opportunity to validate its strategic direction to transition multimedia calling to a Session Initiation Protocol (SIP)-based environment. Consequently, the trial was architected to include a SIP-compliant notebook softphone application (CounterPath Solution, Inc.'s eyeBeam* softphone [1]) along with the open source SIP proxy (SIP Express Router (SER) [2]) and PBX-gateway/unified message server (Asterisk* [3]) applications.

In order to focus the trial on the user experience and eliminate the potential of performance problems due to network congestion, the 100 trial participants were selected from four large U.S. Intel sites: Santa Clara and Folsom in California, Hillsboro in Oregon, and Chandler in Arizona. These sites are connected by high-bandwidth links with no congestion. Participants were asked to test the collaboration services while using wired connections (100 Mbps Ethernet) from their laptops to Intel's network. They also had the freedom to use the trial services while connected via 802.11b wireless LANs and via remote access over the Internet using VPN.

To prepare for the trial, a SER server was installed in Folsom and Asterisk servers were installed in the four U.S. sites to provide connections to the existing site TDM PBXs. Each participant was equipped with the eyeBeam client application, a headset, and a Webcam. A SIP dialing plan was created to give each participant the same number as their existing Intel phone number. This enabled trial participants to establish client-to-client IP calls in a way which is similar to normal Intel dialing, and to use the Asterisk gateways to make calls to people outside the trial. Furthermore, by forwarding their Intel phone to their local Asterisk server, every participant could receive calls from non-trial people on their eyeBeam client. Any incoming call that was not answered was handled by the Asterisk server, which took a voice mail message and routed it to the user's e-mail box.

The eyeBeam client includes several audio and video codecs. To enable the testing of wideband audio, the client was configured to use the open source Speex* codec [4] for calls between trial participants and the standard Public Switched Telephone Network (PSTN) G.711 codec for calls to people outside the trial.

Trial participants were selected as part of teams that frequently collaborate and were encouraged but not forced to use the capabilities exclusively. While it is understood that productivity benefits would increase with more consistent usage, the trial was specifically designed to evaluate how quickly and under what circumstances users would "take to" this new way of collaborating, rather than asking them to abandon their desk phone altogether.

Figure 1: The eyeBeam interface

They also were free to use the application in any way and for any purpose they chose (as long as the capability was enabled), while having access to a user guide with a detailed description of the features and capabilities. Feedback was collected via a Web page as well as a survey and selected interviews to correlate specific usage models with the user experience.

DETAILS OF TRIAL COMPONENTS AND DESIGN

Client

The eyeBeam PC softphone application (see Figure 1) provides a familiar telephone-like interface that is intuitive and can be used effectively with little training required. It provides many new telephony features such as "drag and drop" to place a call, auto answer, and auto conference. It includes basic telephony features, including hold and transfer, caller ID, speed dial, and microphone and speaker level controls. For the trial, a list of contacts (trial participants) was distributed to all participants in a format that could be imported into the client to facilitate speed dialing. The eyeBeam client also simplifies establishing small audio/video conferences (up to 7 users) with an easy-to-use interface. Users can be conferenced together at any time by clicking on the conference button.

VOICE AND VIDEO OVER IP

Voice over IP (VoIP) allows voice traffic to be integrated with an existing IP data network. This provides the advantage that a user can take their telephone with them anywhere as an application on their laptop. This also provides opportunities to integrate voice with other networked PC applications, such as presence, instant messaging (IM), and video.

VoIP uses codec technology to send speech across a network. When a person speaks, the sounds are picked up by the PC's microphone and digitized and then encoded using a voice codec (coder/decoder), which is a piece of software integrated into a softphone or other VoIP device. At the receiving end another codec decodes the voice and converts it back to sound waves via a headphone or speaker.

VoIP audio and video calls can take place directly between distributed PCs (peer-to-peer) rather than through a central PBX. The voice codec can be negotiated on a per-call basis rather than being centrally determined; any codec that is supported on both ends can be used. This provides an opportunity to use a wideband codec, which encodes a much wider audio spectrum than a normal (narrowband) codec traditionally used in telephone systems (such as the PSTN). As shown in Figure 2, a wideband codec can capture and reproduce a frequency

Figure 2: Frequency comparison

range of about 50 Hz to 7 kHz (out of the human vocal range of 80 Hz to 10 kHz) whereas a narrowband codec is limited to transmitting 200 Hz to 3.4 Hz, eliminating significant frequencies at both the bottom and top of the spectrum. Wideband audio brings telephone audio fidelity much closer to actual conversation–it sounds more like talking to someone face to face. The higher frequency range supported by wideband codecs can provide better understanding of intonations and accents and consonant discrimination (e.g., "s" versus "f"). This is becoming increasingly important as Intel's workforce and business becomes more global.

The eyeBeam client supports use of multiple codecs. High-quality wideband codecs can be used between eyeBeam clients while narrowband codecs are used to communicate with traditional telephones. In this trial, between eyeBeam clients, the Speex wideband codec is used; this provides very high-quality audio at a moderate 40 kbps bit rate (including IP overhead). Use of Speex also actually saves bandwidth compared to the standard G.711 uncompressed narrowband codec which is used in the public telephone network and which consumes at least 80 kbps (with IP overhead).

Video is provided using the standard H.263 codec at Common Intermediate Format (CIF) (352x288 pixels) and Quarter CIF (QCIF) (176x144 pixels) resolutions, by default implemented at up to 128 kbps. This generally provides a good picture in a window that is typically about 2 x 1.5 inches (depending on the monitor). The video codec is independent from the audio codec, so wideband audio can be used during audio-video calls.

One of the most significant factors in the quality of voice and video are the devices used to acquire the signal–a headset and camera. High-quality, moderate-cost devices were recommended for all trial participants to provide a baseline for signal acquisition.

Ensuring call quality with VoIP can be challenging, particularly on WAN links, and the technology is still being developed. For these reasons, the trial was focused on four US core sites, which are interconnected with unconstrained WAN bandwidth links.

Session Initiation Protocol

SIP is an evolving standard signaling protocol from the Internet Engineering Task Force (IETF) for Internet conferencing, telephony, presence, events notification, and IM. SIP is independent of the packet layer and only requires an unreliable datagram service, as it provides its own reliability mechanism. SIP is used for all signaling in this trial.

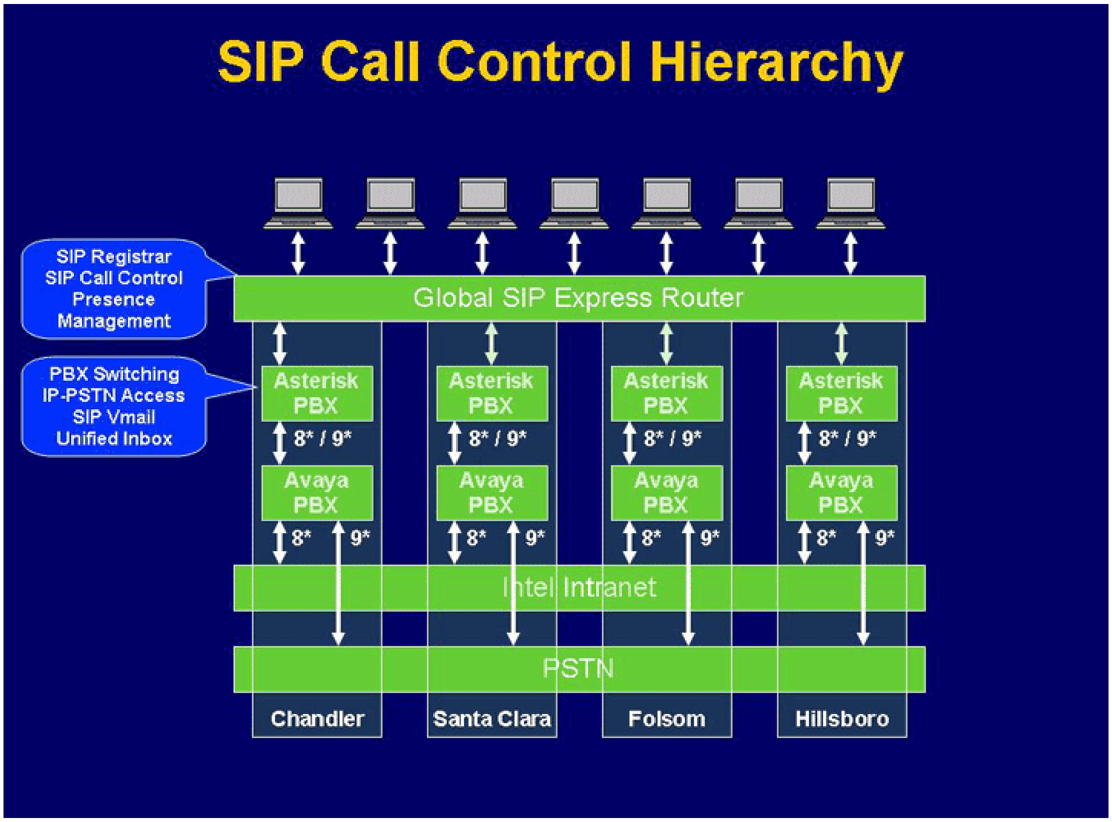

Figure 3: Trial architecture

The basic functionality for call setup is provided by a SIP registrar and proxy. The registrar/proxy for this trial is the open-source SER, which runs on Linux*. All users register with a single SER server, automatically providing each user a SIP address similar to an e-mail address. End users need only use the proxy as the SIP host; the proxy will use the PCs' IP addresses (provided through registration) to send the calls to the end PCs (eyeBeam clients). This allows a consistent SIP address regardless of a user's location; calls can be received on a single phone number anywhere in the world, regardless of one's physical location.

Since the only functions on the SIP proxy server are to register clients and proxy calls to end nodes, a single SIP proxy has minimal system requirements (less than 1 MB) and can scale to thousands of calls per minute. Multiple proxies could be used to establish SIP domains and redundancy, but for this simple trial only one proxy was required.

* Other names and brands may be claimed as the property of others.

Asterisk PBX

The trial also incorporates the open-source Asterisk PBX, running on Linux. Asterisk supports a long list of call features (see http://www.asterisk.org/features*), but in this trial is primarily used for voicemail, voicemail to e-mail integration, and as a SIP gateway to the TDM PBX infrastructure.

Unlike SER, which directs calls between IP end points (PCs), Asterisk can also terminate calls from the PCs (for voicemail and gateway functionality). The codecs that can be used are limited to those supported by Asterisk. These are typically narrowband codecs, which have lower CPU overhead. In the trial, G.711 is used between the client PC and Asterisk.

Overall Trial Design

The flexibility of SER and Asterisk provided numerous design possibilities. The trial design (see Figure 3) incorporated centralized call control and distributed voicemail and gateway functionality. This allowed all users to register on a single server (with a consistent client configuration), and still use a dialing plan similar to their existing TDM one.

Each of the four trial locations (Chandler, Folsom, Oregon, and Santa Clara) has an Asterisk server. In Folsom, the Asterisk server is running on the same system as SER. Each Asterisk server connects into a local PBX using a T1 ISDN line, each capable of 23 simultaneous calls. Two of the larger campuses involved have over 60 users. Traffic studies indicate that the Asterisk gateways to the PSTN were utilized 25% on average with peak traffic hitting 50-70% of trunk utilization. During some peak gateway testing a noticeable decrease in call quality was observed. The cause appeared to be isolated to the gateway card technology and the amount of processes being utilized on the server to support the gateway card. One possibility to resolve this call-quality degradation is to ensure that each card function runs on a single process.

Dialing to regular phones is provided by the local Asterisk server at the trial locations. To keep the dial plan consistent for end users, the trial used the same dial plan as Intel's existing phone system.

ENGINEERING KEY LEARNINGS

One of the main goals of this trial was to increase the knowledge of the IT network engineers on the topics of SIP, open source, and VoIP support issues. The trial was very successful from this perspective and produced key learnings in the areas of open-source software, SIP interoperability, VoIP monitoring and measurement, and client configuration issues.

Use of open-source software has both advantages and challenges. The upfront cost and source code access is seen by many as an advantage. The lack of a formal support model is seen by some as a disadvantage, and in this trial resulted in some implementation delays. Rather than direct support, most problems must be solved by sifting through discussion archives on the Internet. Some of the developer solutions that were utilized in the design did not function as the documentation indicated, and further research and experimentation were needed to resolve these feature/functionality issues.

This experience produced some guidance for future projects. Extra time should be allocated for lab testing before the proof of concept and design changes should be anticipated. Engineers can learn a considerable amount through trial-and-error lab testing, but if an organization wishes to initiate a proof of concept quickly or it does not have network and telecom engineers with the technical experience to resolve issues independently, it should consider employing an outside consultant with open-source experience.

There appears to be a significant opportunity for companies to pre-package/bundle open-source VoIP solutions in much the same way Linux distributions are now bundling solution stacks. Open source VoIP technology is a disruptive yet driving force in the VoIP PBX industry. Open source leverages the collective efforts of many developer groups and technical experts to create new technologies and applications. With the high demand for telephony customization and application integration, open source will continue to drive vendors to either embrace it or stay ahead of its development.

The trial showed that basic SIP functionality was interoperable among a variety of products. However, the basic IM and presence functionality proved more problematic. Support for the IM and presence capability of the softphone client was part of the initial plan, but presence management was not reliable with the version of SER used in the deployment, so the trial was rolled out without this capability.

Monitoring and measurement, particularly call detail recording, is a challenge with open source IP PBX solutions. The data files were collected during the trial, but the information provided was very limited. Further tool integration or software is required to manipulate the data records into a meaningful reporting format. Though there are numerous open source tools available for SER and Asterisk, none were integrated into this trial environment.

Client stability is important when calls are made through the PC, and this provided some challenges due to the interactions of the real-time communications capabilities with various IT-mandated client applications (such as virus detection, firewall, and remote management). This trial was designed to evaluate the usage of a real-time application in an unconditioned environment, so the clients and network were not tuned to support this usage model. One of the key learnings was that the introduction of telephony software as a real-time collaboration tool on the PC requires characterization of the entire PC software environment to ensure coexistence of all applications before global deployment of a fully supported solution can be attempted.

USER EXPERIENCE

This trial was designed primarily to establish the value of "seamless collaboration," as opposed to a product-centric proof of concept where the focus is on the tools themselves. To this end, the users were requested to use the eyeBeam client as their primary communication device for all their audio collaboration needs, but not given any specific scenarios to test. Wherever possible, intact teams that normally collaborated with each other were selected for the trial, so they would get the most out of the solution, and so there would be a high likelihood of peer-to-peer video and telephony calls utilizing wide band audio.

However, from a user perspective, it becomes difficult to separate the value proposition from the day-to-day issues encountered with the way the tools work. The user feedback falls mainly into the following categories:

1. Understanding the capabilities and functionality of the solution.
2. Desired functionality and usability issues with the interface.
3. Feedback on technical limitations and audio quality.

The users were provided with a detailed Intel-specific user guide explaining the setup and features of the eyeBeam client at the start of the trial, along with a link to a FAQ page, a support email account, and a Web page where they could post their feedback. The FAQ page was updated regularly throughout the trial based on the feedback received.

Most of the issues encountered during the first couple of weeks were related to the setup of the eyeBeam client, audio tuning, and volume control (such as use of the headset, mixing of PC and phone audio, and coping with different levels of volume for different participants on the call). Most users would like to see a client with auto configuration so that manual set up is not required to start using the technology. It will be very important to support an automated installation and configuration process for large global deployments. Further development work with vendors' developer tool kits can assist to minimize user setup time, configuration errors, and trouble calls upon initial setup.

Redirecting all incoming calls to the eyeBeam client rather than the desk phone proved a bit challenging as well. Users were given the opportunity to utilize their existing desktop phone or the proof of concept technology on their laptop. This architecture allowed for minimal productivity loss if issues arose with one or the other technologies.

Users commented a lot on the dialing functionality, ranging from requests for e-mail integration ("click-to-dial" a name) to drag-and-drop, cut-and-paste and number editing functions, to voice recognition similar to cell phone functionality. Some would have liked a quick access toolbar in other office applications, rather than having to switch to the eyeBeam application to initiate the call. Several users would have liked either IM functionality within eyeBeam or at least strong integration with Intel's existing IM application.

Answering an incoming call via eyeBeam also took some getting used to. While the application always pops to the front when a call is received, in some cases users experienced "stolen keystrokes" which produced unpredictable results within eyeBeam or promptly sent it to the background again while typing in other applications. This lead to some missed calls. Since all audio goes to the headset, users also felt they were at risk of missing calls if they did not wear the headset all the time at work.

Many users compared the eyeBeam client to their experience with other softphones. In some cases, the comparison was in favor of eyeBeam, in others less so. Some found the user interface too complex and were not able to navigate the different buttons, others got hooked on the AutoAnswer and AutoConference features and actually changed the way they scheduled meetings as a result.

At the same time, users commented that the ability to make a call by a simple drag and drop of a name, the ability to click-to-dial, and the autoconferencing features are a great improvement over desk phones. These softphone features provided them increased productivity.

We received a wide range of feedback on audio quality. Several users heard significant pitch changes at times during calls when there seemed to be resource contention, either because they were using other applications at the same time, or because they were making calls over a VPN connection, from a wireless hotspot or their home office. For some users calls over their VPN broadband connections were of good quality while the experience for other users over VPN broadband rendered the client useless. During a call, when the PC was performing a resource-intensive task, such as when a large e-mail attachment came in over VPN or when opening a large presentation, a slowdown of the desktop and changes in voice quality were evident. These were considered annoying but not detrimental to productivity.

At the same time, all users that experienced true wideband (eyeBeam to eyeBeam) calls reported amazement at how much clearer and easier to understand the audio was compared to their regular desk phones. For some, it even sounded better than other Internet telephony tools. Given that many users still dialed into a standard audio bridge for their conference calls, this is one of the benefits that can only be fully realized if a large number of users has access to and is utilizing the technology.

Conversely, the Voice Activity Detection (VAD) feature, which sends "digital silence" when there is no voice detected, was unexpected and confused users. To many trial participants it seemed to be a loss of audio or a dropped call because it was too quiet. Once users turned off VAD there tended to be more "static" on the line but the experience was more familiar. Eventually, softphone developers will incorporate side tone, which will ameliorate this issue.

Advanced technical users suggested that eyeBeam should provide diagnostics on packet error rate, jitter, and latency so the user can "monitor" the health of the conversation and adjust as needed. Being told that the other party could not hear them certainly was perceived as a negative, even though in most instances this was related to a weak signal from a Bluetooth* headset rather than the inherent audio quality in eyeBeam itself. Those users also would have liked to have a call automatically reconnect itself once the signal from the Bluetooth headset was within acceptable range again.

Headsets in general drew some interesting comments. Most users liked the quality they got via the recommended headsets but found them too bulky to travel with. Headsets utilizing the audio jack on the PC did not produce the same high fidelity. The inbuilt speaker and microphone on the PC were not attractive due to the lack of sufficient echo-cancellation software and array microphones. Not being able to accept a call directly on a Bluetooth headset also was a usability issue.

Little feedback was received about the video. Call detail records on the server showed that very few video calls were actually established, indicating that many users were either not comfortable using video, could not be bothered to try it, or did not have the time or patience to figure out how it works. Some users did feel the video aided comprehension in remote one-on-one meetings.

Unexpected was the fact that sometimes calls seemed to go into "dead air"; this was traced to a user initiating PC hibernation without logging off of eyeBeam. Closing the eyeBeam client and therefore unregistering the phone prior to going into hibernation mode was an easy enough work-around but needed to be explained first.

Regrettably, many users also experienced hard lock-ups on their PCs while using the softphone client, which discouraged usage to a certain extent. The hard lock-ups were traced to an interaction with an IT-mandated application load, which similarly affects other real-time-softphone and collaboration clients. We recommend characterization of the current PC operating system (OS) and any additional software components before deploying real-time applications into production.

In general, traveling users seemed to get most value and benefit out of the tool since their desk phone essentially followed them. Numerous user anecdotes were received regarding the value of the softphone in environments where there was no corporate phone system phone (such as a lab, hotel room, or phoneless cubicle). Also, it made settling down in a remote campus very easy, especially given the lack of office space and traveler workspaces equipped with phones at some Intel sites. Being able to receive their voice mail directly in e-mail contributed to this positive experience and was perceived as a significant value-added feature.

Generally, the comments were encouraging and confirmed that employees can be more productive with real-time collaboration tools on the PC since there are incremental time savings in many repetitive tasks, as well as increased support for mobility and improved comprehension in conversations.

CONCLUSION

The user trial was conducted for a period of about two months. It revealed that users found value in the integration of their real-time communications with their PC collaboration environment, but wanted even more integration, such as with their e-mail client and IM. Users particularly enjoyed the wideband audio experience, the "click to call" capability, unified messaging, and the ability to be mobile and still communicate from a variety of locations. However, they also were bothered by the inconsistency of the experience, as the trial included no mechanisms to ensure that the real-time communication was given priority on the PC and the network. Video was not used very much, even though some value was noted for one-on-one meetings. Finally, the trial showed that some SIP products interoperate, but it takes work (especially with open source) and possibly multiple products to obtain a desired feature set.

Overall, the user feedback was positive on the added mobility and productivity enhancements this PoC provided. IT's plans are to further develop the integrated communications and collaboration environment with the PC, taking advantage of the advanced collaboration features that will be delivered with the next-generation laptops based on Intel Centrino mobile technology.

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* Bluetooth is a trademark owned by its proprietor and used by Intel Corporation under license.

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