White Paper

Testing Convergence for Next-Generation Communications

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First in a series of whitepapers discussing the viability of converged communications.

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Introduction

The Internet and mobile communications have had an immense impact on business communications methods and costs. Industry-wide changes have come at an almost frantic pace and, according to Gartner DataQuest, the amount of traffic on packet networks surpassed that of time-division multiplexing (TDM) networks in the year 2000.

According to Jupiter Media Metrix (2001), use of instant-messaging applications in the workplace increased 48 percent in 2001—from 2.3 billion to 4.9 billion minutes—making it arguably the most rapidly growing technology in history. Cell phones, pagers and personal digital assistants are now a mainstream communication tool for co-workers, friends and family members. Many people rely on mobile and wireless devices to access essential information quickly and easily, reporting that it helps them be more effective in their work and personal lives.

To meet the demands of the ever-changing business environment, enterprises and service providers are working to deploy converged voice-and-data services based on the Session Initiation Protocol (SIP). With a foundation in Internet protocols, SIP allows the integration of traditional voice services with Web-based data services, including self-provisioning, instant messaging, presence, and other mobility services.

To better understand the challenges presented by implementation and utilization of new converged solutions, the Intel Embedded and Communications Group (ECG) has internally developed a proof-of-concept (PoC) Remote Office/Branch Office (ROBO) solution at Intel's Chandler, Arizona campus. Based on Intel® architecture processors, this ROBO pilot supports the complete convergence paradigm where all data, voice, video, and internet access is available on a single unified Ethernet-based network, and all voice, video and data are simultaneously delivered via one IP network. The network is 100% isolated from the internal IT-supported network, thus simulating the behavioral characteristics of a small "single site" company.

The pilot utilizes the CAP24-120 PoC platform at the edge of the network. There are currently 100 users within the Intel environment utilizing the ROBO system as the main connectivity method for voice (via VoIP phones), for all PC data traffic, and for "VPNing" into the Intel IT network. The system is a pure SIP voice play utilizing the relatively new "SIP Trunking" paradigm, with service provided by Broadband.com. A wireless local loop (point-to-point WiFi) is provided by LastMile Communications.

This paper discusses the suitability of convergence as the foundation for next-generation communications networks. It describes SIP history and trends, SIP in the small-to-medium business (SMB) environment, and ROBO—its implementation and some early findings within the Intel pilot.

Origin of the Session Initiation Protocol

In early 1999, the Internet Engineering Task Force* (IETF) defined SIP in rfc2543. The definition was the culmination of years of work in the IETF's MMUSIC working group to provide a mechanism to integrate voice, video and data over the same network. SIP supports seamless integration with existing TDM networks, allowing integration with e-mail, the World Wide Web, and next-generation technologies such as instant messaging and 3rd Generation Partnership Project (3GPP) mobile networks (www.ietf.org).

Since its inception, rfc2543 has undergone changes and improvements, and work continues to further define and advance the SIPs. At this time, the IETF has expanded to three working groups:

- SIP-WG—Defines the underlying SIP and all extensions to this protocol
- SIPPING-WG—Defines common application usage of SIP
- SIP for Instant Messaging and Presence Logical Extensions (SIMPLE)-WG—Defines how SIP can be used to build instant messaging and presence. (Both Microsoft/MSN and AOL have publicly announced that they will embrace SIMPLE as the standard for their next-generation of instant messaging products.)

The most notable SIP characteristics are its usage of an Internet-like distributed architecture and its reuse of existing Internet technologies:

- Domain Name System (DNS) and URLs for naming
- Session Definition Protocol (SDP) for capabilities exchange
- Multipurpose Internet Mail Extensions (MIME) for application integration
- HTTP/text for message formatting

SIP Components

A SIP-based network is made up of several logically discrete components:

- SIP user agent—Any network endpoint that can originate or terminate a SIP session. This might include a SIP-enabled telephone, a SIP PC client (known as a "softphone"), or a SIP-enabled gateway.
- SIP proxy server—A call-control device that provides services such as routing of SIP messages between SIP user agents.
- SIP redirect server—A call-control device that provides routing information to user agents when requested, giving the user agent an alternate uniform resource identifier or destination user-agent server.
- SIP registrar server—A system that stores the logical location of user agents within a domain or sub domain. A SIP registrar server stores the location of user agents and dynamically updates its data via REGISTER messages.
- Back-to-back user agent—A call-control device that provides routing similar to a proxy server, but allows centralized control of the network call flows. This device empowers SIP networks to replicate some of the traditional telephony services that require centralized knowledge of device state, such as call park and pickup.
- SIP-aware network devices—Devices that have knowledge of the SIP and allow the network to function more efficiently. This type of device might be a firewall or network address translation unit that can allow SIP traffic to traverse network borders. It could also be a load-balancing switch that allows requests to SIP servers to be handled more efficiently.

Role of Convergence in a Small-to-Medium Business Network

SMBs are always looking for ways to reduce costs and improve productivity within limited resources. Converged networks are believed to be a rational way to collapse multiple networks (TDM, ATM and IP, for example) into a single, unified IP-based system. Benefits include lower costs to support and administer the network, as well as lower connection costs for voice transport through VoIP service providers. Instead of multiple networks separating voice and data, with resulting support costs for each network, it is now possible to drive voice and data through an all-IP-based network, essentially retiring the analog-based TDM voice infrastructure and its associated PBX or key system. An IP-PBX based on Intel architecture becomes the signaling and voice mail system for the entire enterprise (i.e., a single-site company, as simulated within the context of the Intel ROBO trial). With the performance available in a mid-range processor and chipset, literally hundreds of concurrent SIP endpoints can be supported on a single unified system. In an environment requiring 99.999% (five nines) availability, a simple failover mechanism can be incorporated utilizing a second, warm standby system.

Once the SMB has implemented an all-IP communications network, unified messaging, firewalls, gateway and email services become much less costly to provision, manage and support. This reduces procurement costs and total cost-of-ownership (TCO)—essentially resulting in lower capital and operating expenditures—which directly improves the bottom line of a business.

Networks and Services SMB Solutions

With the maturation of IP Telephony, many services are now being deployed using SIP-enabled products, and major carriers have deployed or are deploying this technology. Examples of networks and services currently being deployed include:

- Interexchange Carriers—WorldCom IP Communications combines voice and data applications on one network, allowing customers to transition efficiently from a traditional voice platform to IP-based communications.
- Tier 1 Internet Service Providers—One of the world's largest packet voice networks has been deployed by Genuity, Inc., with a worldwide fiber backbone spanning more than 15,000 miles of routes in the United States alone, and carrying more than 100 million minutes of use per month. The Genuity Network presently covers more than 80 percent of the U.S. metropolitan market, serving other service providers and enterprise customers, and can support 80,000 simultaneous voice calls.
- Application Service Providers (ASPs) and Communication Services Providers—Taking advantage of the strengths of SIP as an Internet-based protocol, ASPs are deploying a variety of enhanced services. Services that fall into this category include conferencing, hosted call centers, voice portals, and debit card applications.
- Alternate Carriers—Alternate carriers are actively deploying both managed business and residential services based on Cisco SIP-enabled products.

ROBO@Intel

The goal for the ROBO pilot is to establish a living laboratory in a commercial setting–to explore technology and human factor issues and the limitations of a converged IP PBX using SIP session management, all running on Intel's CAP 24-120 PoC platform.

True convergence has often been viewed by customers as a difficult task requiring many disciplines and adding management headaches. There is a lot of conflicting and sometimes unsubstantiated information regarding the pros and cons of transitioning to a converged system, and many customers have found it difficult, if not impossible, to get the data they need to clearly understand the issues in their drive to adopt SIP. At the same time there are those alleging that SIP and IP phones are less secure than the standard TDM system. By implementing a complete ROBO system, Intel engineers and platform architects have been able to address these and other issues, using data to provide clear and concise results.

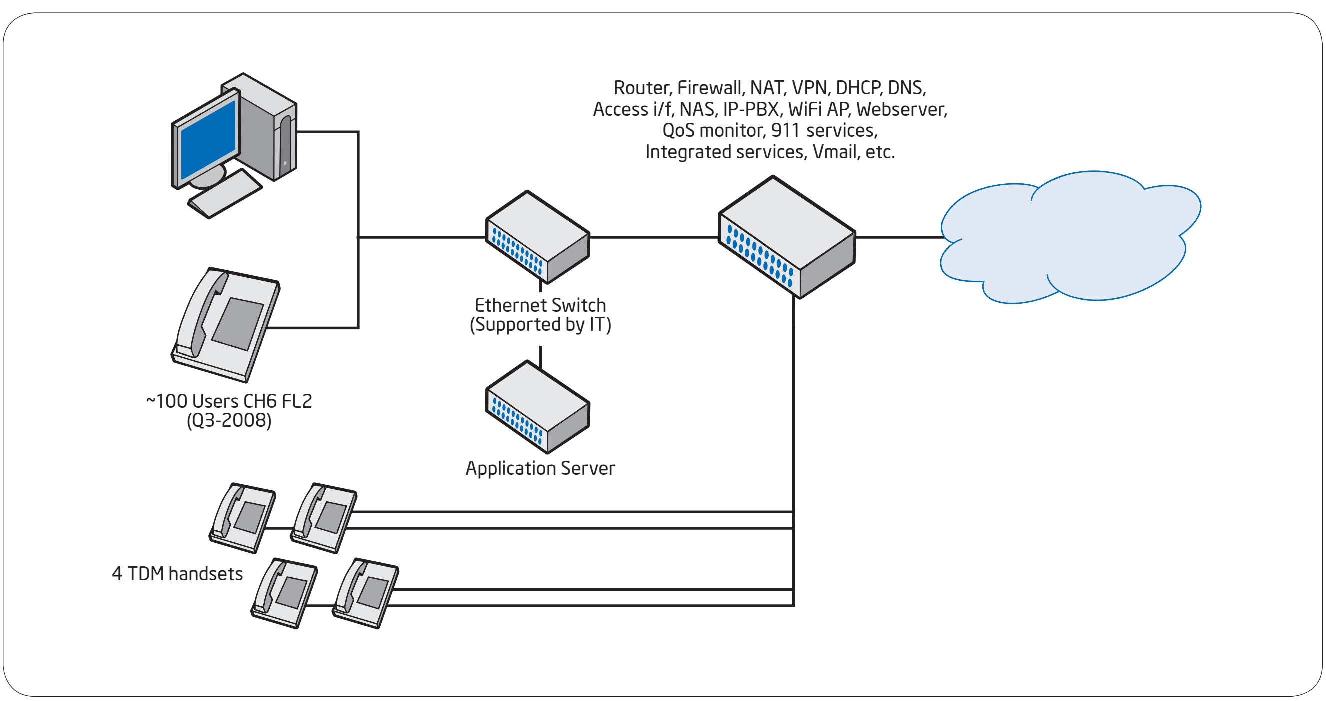

Figure 1. ROBO Systems Element Overview

As some of the mystery surrounding ROBO systems has been solved, many customer concerns have turned out to be unwarranted. One of the first issues slowing adoption is the alleged ability to eavesdrop on SIP-based conversations. To eavesdrop on a TDM-based call, one would need to connect a bug to a terminal block or punchdown block in a wiring closet. The bug would then allow the straightforward analog recording of conversations at a separate location. This capability is generally thwarted by securing the wiring closet or switch room from unauthorized access. To tap a phone conversation in an all-IP environment, one would need exactly the same access to a co-resident switch in order to sniff packets on the call for storage and later review. If the communications link requires SRTP (voice encryption), then all voice packets are encrypted at the endpoints so even if a person with nefarious intent were to gain access to the data stream, in most cases this would result in the reception of unintelligible voice data.

There have been anecdotal reports that VoIP and IP telephony are unreliable and deliver poor quality. By implementing the ROBO system we wanted to determine the factors causing this perception so that we could work to eliminate the problem and drive faster adoption of IP voice and data services.

ROBO Software

The software stack runs on CENTOs, a Red Hat Linux* derivative that has been stripped down to attempt to provide a more robust, reliable operating system on the Intel® Pentium® M processor. On top of the OS, we added Digium's Asterisk* software (includes a series of Intel architecture-based host media processing services) running Fonality's trixbox* GUI. This has resulted in a robust implementation with point-and-click management and monitoring. An added benefit of using Linux and open-source software has been the tremendous support with updates and patches to assure long-term system uptime and reliability in a cost-effective manner. This simple stack provides advanced IVR, monitoring and voice services operating within 512 MB RAM. We then updated to 1 GB DDR memory to improve performance as we continue to add users throughout 2008.

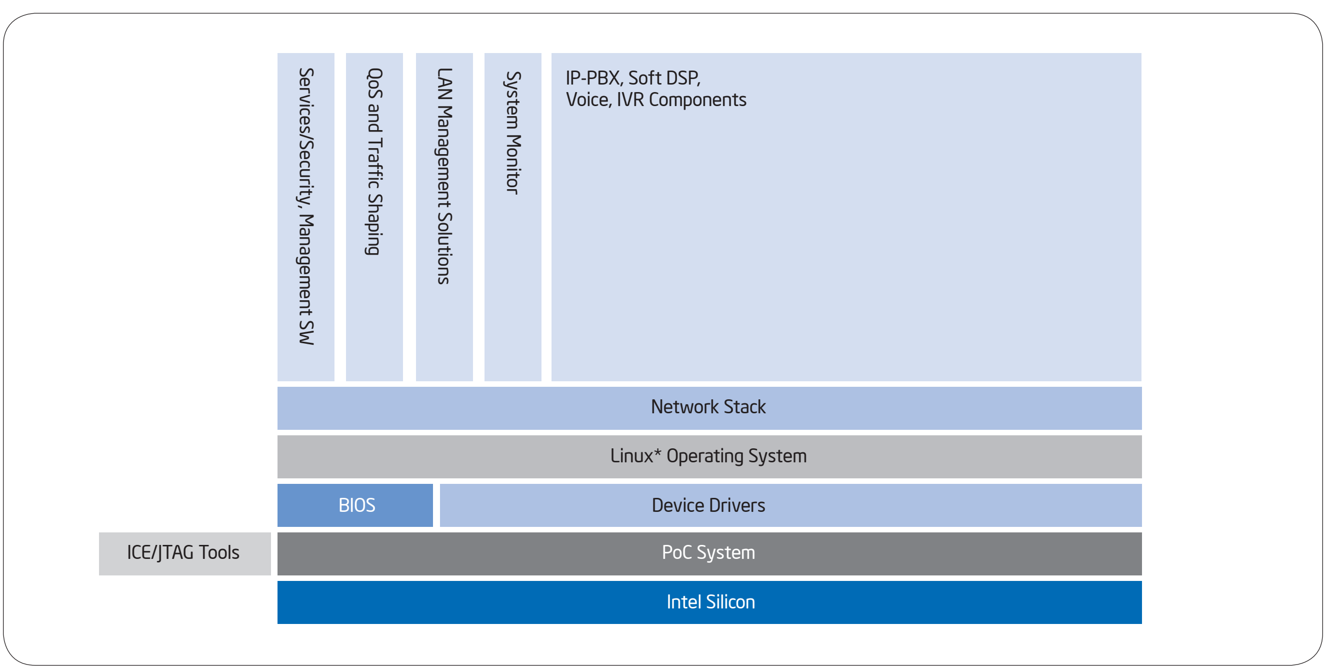

Figure 2. Software Stack Representation

Early ROBO Results

Utilizing a purpose-built Converged Applications Platform (CAP), Intel determined that the Intel Pentium processor on the CAP24-120 was idle more than 80% of the time. Running at 1.8 GHz, the processor can support a large application set of added capabilities without impacting voice and data services. We are now embarking on doubling the number of users to better load the system to understand and share our findings in this use case.

System reliability has surpassed expectations. In 30 weeks of continual use, the system achieved greater than four nines availability and we expect to be able to ultimately achieve five nines availability.

Voice quality has been universally applauded by Intel employees utilizing the ROBO services. The issues of bandwidth-constrained voice quality degradation that occur on home VoIP systems have been largely eradicated with careful network design, attention to details that affect QoS, and by utilizing a network processing stack from Intoto, Inc.

Security has been implemented with strong passwords on the IP-PBX CAP as well as a robust firewall between the internal LAN system and the external or WAN system. We experienced one broadcast storm from a rogue switch that resulted in a 20-minute outage. The rogue system was taken offline by the service provider.

Operational expenses have also dropped. From an initial eight hours needed to set up the system, it has been found that active system administration has been reduced to less than 15 minutes per week. This has taken the form of password resets and the occasional question on how to use the IP phones.

Performance data is continually being recorded by an open source utility called MUNIN, available from SourceForge.net. This utility will form the basis for performance findings on the next whitepaper in this series.

Conclusion

Taking advantage of its foundation in Internet protocols, SIP enhances the ability to create services that integrate telephony with Web-based services such as instant messaging and presence. Though SIP has its roots in Internet technology, it facilitates seamless interoperability with traditional telephony networks and helps packet-based networks achieve the scaling, reliability, and voice quality of the global switched telephone network. With the addition of new service providers and rollout of more infrastructure, CAP drives a new paradigm for collapsing multiple networks into one seamless, unified SMB system. This reduces costs and improves interoperability between historically disparate networks. The ROBO system has been shown to be a reliable IP-PBX with an astounding number of cycles available for value-added applications running on the hardware. With 99.99% uptime and about fifteen minutes per week spent in management, Intel's pilot ROBO system has demonstrated that a carefully designed, purpose-built platform can be as reliable as any TDM-based PBX, with many value-added services not cost-effectively available to the analog world.

The next step for the Intel ROBO pilot is to add additional users in remote sites that will be part of the PBX's internal numbering plan, accessing remote workers as part of the single-site SMB model. In addition, the user population will be roughly doubled, generating more converged traffic through the solution. Intel expects to publish these additional findings in Q3'08.

For more information, visit www.intel.com/go/viop

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