Delivering Enhanced Voice Services over the Internet

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Abstract: - This paper focuses on deployment of voice over the Internet for the customers of a global services provider which needed to extend cost-effectively the portfolio of enhanced voice services including unified messaging, post paid and prepaid calling card services along with related validation, billing and payment systems.

Key-words: VoIP, Internet, SS7, Compression, QoS

1 Introduction

Data traffic has traditionally been forced to fit onto the voice network (using modems, for example). The Internet has created an opportunity to reverse this integration strategy — voice and facsimile can now be carried over IP networks, with the integration of video and other multimedia applications close behind [1, 2, 3, 4, 5, 6]. VoIP's appeal is based on its capability to facilitate voice and data convergence at an application layer. Increasingly, VoIP is being seen as the ideal last-mile solution for cable, DSL, and wireless networks because it allows service providers to bundle their offerings. VoIP also offers service providers the ability to provision standalone local loop bypass and long distance arbitrage services. Additional bandwidth will be needed to support distributed, multimedia voice over IP applications building to multigigabit network backbones based on Gigabit Ethernet or Asynchronous Transfer Mode.

Customer demand is driving service providers to round out their service portfolios with a mix of voice, data, fax, and video services, and this trend is fuelling the convergence of voice and data network architectures. Merging voice traffic onto an existing data network represents significant cost savings and revenue opportunity for service providers. The popularity of the Internet is also driving the emergence of multimedia applications, many of which are ripe for the integration of voice and data traffic over a single network. These applications include Web-enabled call centres, unified messaging, real-time multimedia video/audio conferencing, distance learning, and the embedding of voice links into electronic documents. In fact, the full business potential of such applications is only beginning to be discovered [7]. But one thing is clear: these integrated voice-and-data applications will require a converged IP network.

Managing company's communications media requires speedy, reliable, and cost-effective messaging solutions. With the emergence of Internet-based applications, enhanced messaging systems are fast becoming tools that can dramatically increase employee productivity, simplify an IT management and reduce operating costs enterprise-wide. A unified messaging solution provides a single location for voice, fax, and email messages; users can have access to all of their messages from their desktop, cellular or wireless phone, or over the Internet and respond, forward, save, or delete messages. Messaging systems with conference calling capabilities let us automatically call the message sender, or click on a screen icon from a desktop and link several parties to the same discussion.

With unified messaging a single application can be used to store and retrieve an entire suite of message types. Voice-mail messages stored as WAV files can be downloaded as e-mail attachments while travelling, a response recorded and returned to the sender, all recipients, or an expanded list. e-mail can be retrieved via a telephony user interface, converted from text to speech, and reviewed from an airport lobby phone or cell phone. Infrastructure is decreased as now a single application can provide voice, e-mail, and fax. Productivity is increased because what were once disparate message types can be retrieved via the most convenient, or the user's preferred, interface.

The Internet and the corporate Intranet must soon be voice-enabled if they are to make the vision of "one-stop networking" a reality.

2 VoIP Overview

VoIP is an emerging technology that allows the systems and wires that connect computer networks to act as an alternative to phone lines — delivering real-time voice to both standard telephones and PCs. VoIP converts standard telephone voice signals into compressed data packets that can be sent locally over Ethernet or globally via an ISP's data networks rather than traditional phone lines [8]. VoIP is vital, according to some analysts, to the continued existence and profitability of Internet Service Providers (ISPs), Application Service Providers ASPs), Local Exchange Carriers (LECs) and Inter Exchange Carriers (IXC). Enhanced services such as unified messaging, “follow-me”, Web-based services, calling cards, and billing are revenue-generating services carriers can offer with very low overhead and a high payback [9]. Figure 1 illustrates the IP network protocols that are currently being used to implement VoIP.

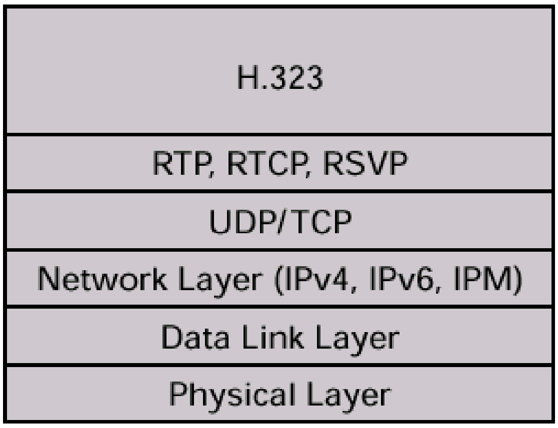

Figure 1. VoIP protocol structure

H.323 is an ITU (International Telecommunication Union) standard that enables audio, video, and data communications across IP-based networks. H.323 standard has been defined to describe terminals (i.e. client end points), equipment and services for multimedia communication over networks (such as LANs or the Internet) that do not provide a guaranteed QoS [10]. H.323 is a family of software-based standards that define various options for compression and call control. RTP (Real-time Transport Protocol) is an IETF real-time end-to-end protocol utilising existing transport layers for data that has real-time properties. RTCP (RTP Control Protocol) is an IETF protocol to monitor the QoS and to convey information about the participants in an ongoing session; provides feedback on total performance and quality so that modifications can be made. RSVP (Resource Reservation Protocol) is an IETF general purpose signalling protocol allowing network resources to be reserved for a connectionless data stream, based on receiver-controlled requests.

The most important consideration at the network level is to minimise unnecessary data transfer delays. Providing sufficient node and link capacity and using congestion avoidance mechanisms (such as prioritisation, congestion control, and access controls) can help to reduce overall delay.

3 Objective

An advanced services provider planned to deliver telephone portal services with phone and Web access to customer’s e-mail, fax, address book, calendar, file storage service and Web News — anywhere, anytime, from any phone. A converged network infrastructure should be deployed at a regional corporate site and integrated with a network infrastructure at the corporate headquarter. A global enabler of enhanced Internet services would provide cost-efficiently its European subscribers with telephone portal services (VOGO) [11] including telephone access to e-mail, unified messaging and other personal communications services.

In addition to this general capability, specific features supporting voice transmission must also be implemented in the aimed network platform. These features include [12]:

- Compression — Low bit-rate voice compression significantly reduces the amount of bandwidth used by a voice conversation while maintaining its high quality.
- Silence suppression — The ability to recover bandwidth during periods of silence in a conversation makes that bandwidth available for data transmission and other network activities.
- Quality of Service (QoS) functionality — Assuring priority for voice transmission is essential. On the Internet and other IP networks, QoS functionality is provided by the Resource Reservation Protocol (RSVP), which reserves resources across the network for a voice call.
- Signalling for voice traffic — Data network equipment can provide more sophisticated services (such as least-cost routing and virtual private networks) than simple voice transmission, by recognising and responding to voice signalling.
- Voice switching — Data network equipment can not only perform sophisticated voice transmission between company locations, but can provide private branch exchange (PBX) functionality by performing call processing and voice switching capabilities either within a campus or over the WAN.

The challenge is to design such an information network that could be quickly and easily extended according to the projected business growth. This means that no large, up front investments for the future years should be required. The whole information system has to be manageable via user friendly and easy-to-use graphical user interface, which would reduce training and hand-over period at minimum. It is important to deploy management software that would include all standard options relating to control, configuration and voice over IP traffic analysis possibilities. This management software has to be modular and easy to upgrade in case of future increasing of the value-added services.

4 Requirements

Deployment of VoIP application for public use involves much more than simply adding compression functions to an IP network. Anyone must be able to call anyone else, regardless of location and form of network attachment (telephone, wireless phone or PC). Everyone must believe the service is as good as the traditional telephone network. Long-term costs (as opposed to simply avoiding regulatory costs) must make the investments in the infrastructure worthwhile. Any new approach to telephony will naturally be compared to the incumbent and must be seen as being no worse (i.e. the telephone still has to work if the power goes off), implying that all necessary management, security, and reliability functions are included.

Some of the functions that are required for practical VoIP solution include:

- Fault Management: One of the most critical tasks of any telecommunications management system is to assist with the identification and resolution of problems and failures. Full SNMP management capabilities using MIBs should be provided for enterprise-level equipment.
- Accounting/Billing: VoIP gateway must keep track of successful and unsuccessful calls. Call detail records that include such information as call start/stop times, dialled number, source/destination IP address, packets sent and received, etc. should be produced. This information would preferably be processed by the external accounting packages that are also used for the PSTN calls. The end user should not need to receive multiple bills.
- Configuration: An easy-to-use management interface is needed to configure the equipment (even while the service is running). A variety of parameters and options are involved. Examples include: telephony protocols, compression algorithm selection, dialling plans, access controls, PSTN fallback features, port arrangements and Internet timers.
- Authentication/Encryption: VoIP offers the potential for secure telephony by making use of the security services available in TCP/IP environments. Access controls can be implemented using authentication and calls can be made private using encryption of the links.

5 The VoIP Solution

With reference to Figure 2, the following networking hardware had been deployed to deliver Voice over IP to subscribers of a rapidly evolving services provider:

- Nokia's DX 220 IP Access integrates the PSTN and IP worlds to offer a high capacity solution for dial-up traffic, enabling a PSTN subscriber to access IP networks: company Intranets or ISP networks [13]. It offers telecom reliability and security for IP traffic, all in a compact package. DX 220 IP Access also provides the Always on Net service, which gives customers a constant connection to the Internet, and opens the window to next generation web-based businesses and services.
- Nuera's Open Reliable Communications Architecture (ORCA) GX-21 trunk gateway is the cost-effective solution for Digital Circuit Multiplication Equipment (DCME) applications. ORCA uses digital compression techniques to increase the capacity of cable links carrying voice, fax and voice-frequency modem traffic [14]. The ORCA GX-21 provides excellent line capacity gain and high quality transmission at all times, even under heavy traffic conditions in the facsimile demodulation mode. DCME applications transport many more voice channels over a transmission circuit than can normally be accommodated. ORCA-based DCME applications provide user-configured compression ratios from 4:1 to 20:1 with up to 17 compressed T1/E1 DCME trunks per chassis. Therefore, an E1 circuit that accommodates 32 voice channels can provide 640 voice channels when a DCME is deployed at each end. As a result, utilisation of the long-distance transmission circuits is dramatically increased. This large difference can turn a financially-threatened business case into a profitable venture. DCMEs provide more than just voice service. They also provide fax and modem services. The method used to multiply the throughput of the transmission circuit varies depending on what type of service is being provided. The multiplication ratio varies depending on the type of traffic:
    - Voice traffic multiplication is achieved by using voice compression technology to lower the bit rate. Over the years, lower and lower bit-rate vocoders have been developed. Traditional vocoder rates are 64Kb/s while currently 8Kb/s vocoders, with nearly the same voice quality, are common. Nuera offers the highest performance low-bit-rate vocoders in the industry, as proven in numerous independent lab tests. Generally, two people don't speak at the same time while having a conversation. In addition, there are intervals of time when neither party is speaking. Therefore, when using newer generation packet technology, these silence intervals (which add up to more than 50% of the two-way session) can be used to carry more voice traffic from other conversations.
    - Fax traffic multiplication is achieved by demodulating the fax modem (being carried in the 64Kb/s duplex PCM channel) at the near-end DCME to recover the data being transmitted. The base band data (which is typically 9.6Kb/s) is then transmitted to the destination where it is remodulated and delivered on a PCM (Pulse Code Modulation) channel. In the reverse direction, little information is transmitted. Therefore, the 64Kb/s of bandwidth in the reverse direction is available for carrying traffic from other channels.
    - Modem traffic multiplication is achieved by transporting the modem waveforms in a lower-bit-rate vocoder when compared to 64Kb/s PCM. Typically, the DCME network bandwidth used in each direction is three times the modem rate and supports modems up to 19.2Kb/s.

Therefore, a 9.6Kb/s modem uses approximately 30Kb/s in each direction as compared to 64Kb/s in each direction on a traditional circuit switched network.

- The Digitalk service node is an open standards-based switch, utilising industry-standard computer telephony hardware with support for tandem switching and prepaid interactive voice response [15]. Employed Digitalk service node includes a signalling Gateway Server (consisting of two Compaq ProLiant 1600R machines), an SQL Data Server (Compaq Proliant 3000R) dedicated as a Data Warehouse, the DataKinetics Septel ISA E1/T1 PC Line Card for SS7 (providing 2E1 trunks with SS7 channel) [16] and an Internal ISDN modem for Remote Access Service (RAS). Each Compaq Proliant 1600R handles 16 E1's (4 quadspan E1 boards/4slots). The system features of deployed Digitalk service node include:

- Full Windows NT hosted administration and system configuration tools;
- SS7-based Call Control providing a seamless interoperability to the PSTN and 99.999% reliability;
- Fully featured least-cost routing based upon dialled digits or time of day/day of week;
- Alternate routing on primary route congestion or port failure, automatic re-routing programmable on call failure codes;
- Dynamic call editing allowing number modification for routing via alternate carriers;
- Sophisticated SQL based call record logging and system load profiling tools;
- Built-in protocol analyser tools and diagnostic test utilities;
- Fully integrated remote management tools.

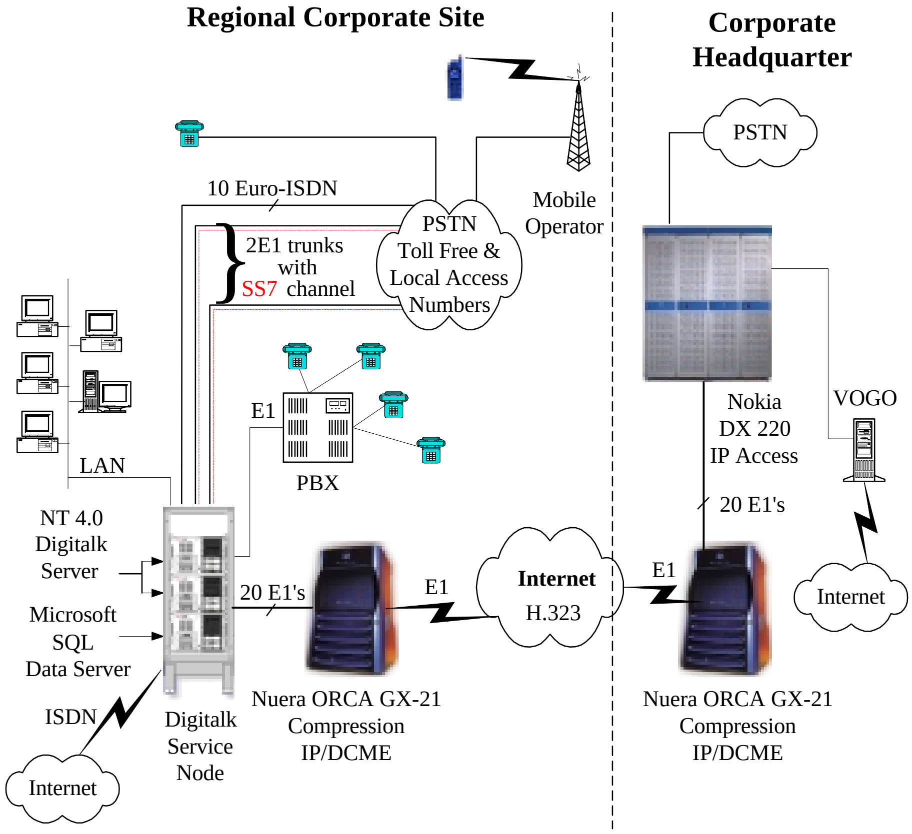

Figure 2. Deployed solution for delivering VoIP

Digitalk platform can be easily extended to a Tandem (redundant) switching architecture (see Figure 3) cost effectively by saving the original information platform. In this scenario, new technologies such as ATM and SQL server are integrated to allow massively scaleable solutions to be deployed.

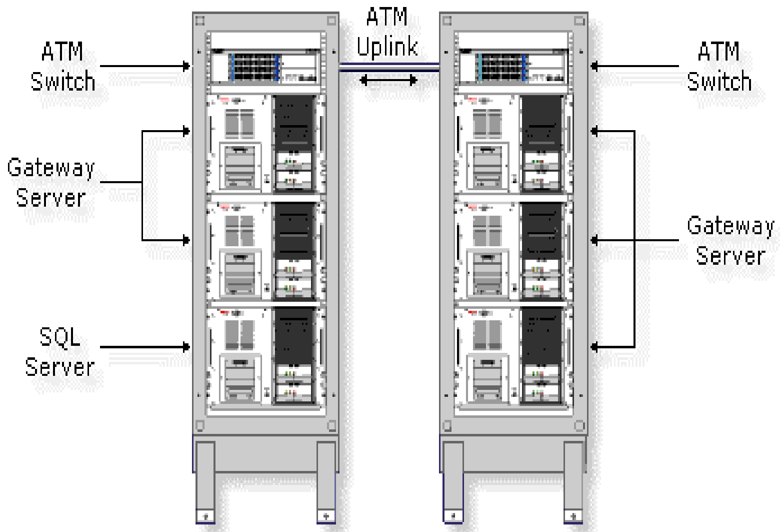

Figure 3. Tandem switching architecture

Digitalk service node can be linked to other nodes via 155Mb/s ATM fibre links connected into a 4.2Gb/s backbone (up to 24 switch nodes). Incoming calls on any node in the system can be routed transparently to any other node via the ATM switching backbone.

It is possible to host a combination of applications on the same switch that would allow new trial services to be launched on existing hardware without the need for a fixed allocation of hardware and software resources.

6 Conclusion

Rather than building separate networks to offer multiple services, a leading supplier of enhanced services integrated various traffic flows onto efficient, packet switching infrastructures based on statistical multiplexing, which maximises the use of network capacity. Using the same infrastructure for all traffic also holds the promise of significant savings because the equipment handles multiple functions and the operational costs are reduced.

Deployed solution for delivering Voice over IP provides the following features:

- Mediation or gateway services between the PSTN and the IP network for transparent internetworking.
- A high degree of scalability — the ability to affordably add processing and switching power as network expands.
- The deployment of new and enhanced services.
- Multiple services over a common infrastructure.
- Automated and integrated management functions.
- The ability to mix and match best-of-breed equipment and software from different vendors.

While enabling new value-added services for European customers, it was in foresight for the near future to deploy similar information systems in other countries worldwide, providing a global delivery of Voice over IP to the international subscribers.

Convergence is on its way. Companies that delay for too long the adoption of first-wave convergence technologies, most notably VoIP, will suffer a substantial competitive disadvantage as these technologies enter the mainstream. There are, however, clear risks that need to be avoided at this early stage of converging technologies. To avoid these risks while still practising due diligence in advancing the corporate technology portfolio, decision-makers should carefully consider the use of intelligent multi-path gateway switching.

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