INTERNET TELEPHONY

Mohsen Guizani *Ammar Rayes* *Mohammed Atiquzzaman*

The growth in Internet Protocol (IP)-based services in the last few years has been explosive. It is projected that this market will continue to grow at an even higher rate for several years to come. IP-based services allow carriers to provide differentiated services tuned to specific segments and customers. This includes virtual private network (VPN) and LAN service, electronic commerce (e-commerce), and of course Internet telephony. Table 1 shows an example of the forecast data for the total global IP telephony service market.

While there are more than 100 years of experience in designing, operating, and managing traditional circuit-switched networks, limited data is available about IP-based networks. The success of Internet telephony as well as other IP services depends primarily on a clear understanding of the overall technology and service requirements. As Internet technology evolves, companies have many questions to answer in order to carry voice over the Internet —or voice over IP (VoIP) — successfully. These questions include the economic advantages to both service providers and end users, the set of new services, and the quality of service (QoS) requirements.

Some see voice over the Internet as being nearly free and would like to take advantage of this apparent bonanza without having to worry about owning their own networks. Another advantage would be the ability to support "off-net" voice applications. Yet another is to allow a company to reach voice users outside of its own administrative boundaries. However, the ultimate benefits may be due more to voice compression at the source than from the transport vehicle itself (the Internet).

We received many interesting articles for this special issue. We have selected only eight articles that provide answers to a few of the questions in the field of Internet telephony. These articles cover a spectrum of topics that range from IP management and QoS to the standards and design of voice over IP components. The first article in this issue, by Rayes and Sage, discusses an integrated management architecture for IP-based networks, illustrating the functions needed to support the challenges of managing VoIP services. Unlike data-transfer-based applications, voice transport across an IP-based infrastructure demands low latency and jitter. Not only will the performance threshold triggers be different, the servicing of VoIP packets would be done based on a priority scheduling scheme. The particular scheme used and the type of traffic would determine the corrective measures needed to fix a problem. Examples of priority scheduling systems include expedited forwarding as defined in the differentiated services (DiffServ) framework, and Priority Queuing with Class-Based Weighted Fair Queuing. Integrated management architecture helps mediate these differences, limiting the exposure of the differences to management operators and end customers. The article describes the five FCAPS functional areas: fault management, configuration management, accounting management, performance management, and security management. It also discusses service restoration, traffic engineering, data collection, service activation, and network planning of IP networks.

Year	Average unit (millions/year)	Unit growth rate (%)	Yearly revenue (millions)	Yearly revenues growth rate (%)
2000	3987.2	256	388.75	209
2001	8544.0	114	640.80	65
2002	22,386.2	162	1511.07	136
2003	78,351.6	250	4701.09	211
2004	167,896.2	114	8814.55	88
2005	335,792.5	100	15110.66	71
2006	587,636.9	75	22036.38	46

Compound annual growth rate (1999–2006):

Note: All figures are rounded.

Table 1. *Total global IP telephony service market: unit and revenue forecasts (source: Frost & Sullivan).*

The second article, by Li, Hamdi, and Jiang, describes the IETF QoS framework, specifically the integrated services model (IntServ) and Resource Reservation Protocol (RSVP), differential service (DS) architecture, multiprotocol label switching (MPLS), and the buffer management and packet discard mechanism. Then, the authors discuss two different solutions offered by two vendors that offer IP telephony services. These examples are used to illustrate how real systems are implemented. Afterward, they present a new DS architecture with a novel buffer management and packet discard mechanism and show that it substantially improves voice performance over the current best-effort Internet.

Rao, Lin, and Chou propose in the third article a VoIP value-added service for mobile network called iGSM. iGSM service provides user mobility to subscribers, which allows them to use either

(Continued on page 46)

(Continued from page 44)

GSM handsets or H.323 terminals (IP phones or PCs) to access telecommunication services. They describe how the iGSM registration, deregistration, and call delivery procedures can be implemented without modifying the GSM network. They study how the tromboning effect in standard GSM system can be avoided when accessing the IP network. Then they investigate the misrouting problem caused by user mobility.

Liao and Liu discuss VoIP mobility in IP/cellular network interworking. They investigate how H.323 gateways provide interconnection between IP networks and switched circuit networks (SCNs). They also describe how H.323 is to provide interoperability with other SCN terminals while major efforts are focused on IP/wired SCN interworking. Then they discuss the potential problems associated with interworking between IP networks and cellular networks through H.323 gateways. After that, they propose a lightweight approach using the existing call transfer supplementary service to provide VoIP mobility in the H.323 IP telephony networks. The proposed approach uses existing components in the H.323 standard, thereby allowing VoIP mobility service in hybrid IP/cellular networks to be a value-added feature in existing H.323-compliant Internet telephony systems.

Iida, Kawahara, Takine, and Oie discuss the performance evaluation of architecture for end-to-end QoS provisioning. They mainly divide the Internet into three types of subnetworks: domain networks, access networks, and stub networks. They focus on issues arising in the former two networks for end-to-end QoS provisioning. First, the access networks are of rather low-speed links, so delay is still of major concern. They examine the statistical delay bound through numerical results derived from their analysis. The performance of some ways to reduce delay is also investigated. Then, they examine low-bit-rate voice flows so that effective flow management can be controlled. They pay more attention to a flow aggregation scheme, and evaluate its performance by analyzing its blocking probability and the amount of bandwidth required, keeping it less than some acceptable value.

The article by Pazos, Kotelba, and Malis describes how H.323 calls are established using H.323 gatekeepers and the H.225.0 call setup procedure to take advantage of transport over asynchronous transfer mode (ATM) networks. This approach imposes control termination on the H.323 and H.323 gateways, simplifying the switched virtual connections (SVCs) for the transport of H.323 Real-Time Transport Protocol (RTP) media streams. The article also discusses the opening of a logical channel and how H.245 messages are used to provide the necessary information to establish SVCs. This includes the mechanism to redirect traffic to the H.323–H.323 gateway.

The article by Bergmark and Keshav describes the ITX project at Cornell, which is an open source, portable, programmable platform for developing multimodal applications (i.e., applications that span the telephone network and the Internet). As an example, a phone recorder allows one to use a telephone to create a recorded message that will be stored in a server on the Internet. Performance measurements (e.g., latency) of the ITX have been reported. Commercial companies are currently testing the ITX.

Finally, the TCP/IP protocol was developed for non-real-time data transmission and hence does not provide QoS guarantees to real-time applications such as IP telephony. However, IP telephony allows integration of many applications and services such as integration of data, voice, and fax, video telephony, multiple telephone lines, support of multiple voice qualities, and enhanced teleconferencing which cannot easily be supported by the circuit-switched telephone network. Despite the advantage of IP telephony and the fact that it is gaining momentum and popularity, a lot of technical issues, such as packet loss, network delay, and packetization delay, still need to be solved to make it useful in practice. In the meantime, a number of commercial products are already available in the market. The article by Hassan, Nayandoro, and Atiquzzaman presents an overall, very broad survey of the applications and services supported by IP telephony, the technical challenges IP telephony faces, and some IP telephony products (along with their technical basis and the services they provide) available on the market. The article is meant to provide an overall picture of the state of the art and future of IP telephony.

BIOGRAPHIES

MOHSEN GUIZANI [SM] (mguizani@cs.uwf.edu) is currently chair of the Computer Science Department at the University of West Florida, Pensacola. From 1996 to 1999 he was an associate professor of electrical and computer engineering at the University of Missouri-Columbia. His research interests include computer networks, design and analysis of computer systems, parallel and distributed computing, fault-tolerant systems, modeling and performance evaluation of communication systems, wireless communications and computing, and optical interconnection networks. He has served as a guest editor for *IEEE Communications Magazine*, *Informatica*, *International Journal of Computer Systems and Networks*, *International Journal of Communication Systems*, and *International Journal of Computing Research*. He is founder and Editor-In-Chief of a new journal entitled *Wireless Systems and Computing* (Wiley). He is the author of two books: *Designing ATM Switching Networks* (McGraw-Hill, 1999) and Optical Networking and Computing for Multimedia Systems (Marcel Dekker, to appear December 2000). He is program chair of PDCS 2000 and ANSS 2001. He has more than 100 publications in refereed journals and conferences. He received his B.S. and M.S. degrees in electrical engineering; and M.S. and Ph.D. degrees in computer engineering in 1984, 1986, 1987, and 1990, respectively, from Syracuse University, New York. He is a member of the IEEE Computer Society, ASEE, ACM, OSA, SCS, and Tau Beta Pi.

AMMAR RAYES (rayes@cisco.com) is a solutions manager and principal solutions architect at Cisco Systems. He is currently working on network management/operation support systems for IP wireless, fixed broadband wireless, IP, ATM, VoIP, and WDM networks. Prior to joining Cisco Systems, he was a director in the Traffic Capacity Management and Planning Department of Telcordia Technologies (formerly Bellcore). He has authored/co-authored over 30 books, patents, and papers on advances in numerous communications-related technologies including a recent book on ATM switching and network design published by McGraw Hill. He received his B.S. and M.S. degrees in electrical engineering from the University of Illinois at Urbana in 1986 and 1988, respectively. He received his D.Sc. degree in electrical engineering from Washington University, St. Louis, Missouri, in 1994, where he received the Outstanding Graduate Student Award in Telecommunications. He has served as a board member of IEEE Computer Society press and is currently a member of the International Advisory Committee of the IEEE IP-Oriented Operations and Management Workshop (IPOM). He is also Industry Liaison Chairman of the International Conference on Parallel and Distributed Computing and Systems.

MOHAMMED ATIQUZZAMAN (atiq@ieee.org) received M.Sc. and Ph.D. degrees in electrical engineering and electronics from the University of Manchester Institute of Science and Technology, United Kingdom. Currently he is a faculty member in the Department of Electrical and Computer Engineering at the University of Dayton, Ohio. He is a senior editor of *IEEE Communications Magazine* and *European Transactions on Telecommunications* and International Journal and serves on the editorial boards of *Computer Communications Journal* and *Telecommunication Systems Journal*. He has guest edited many special issues of various journals, including feature topics about switching and traffic management for multimedia; architecture, protocol, and QoS; ATM switching and ATM networks; and parallel computing on clusters of workstations in *IEEE Communications Magazine*, *European Transactions on Telecommunications*, *International Journal of Computer Systems Science & Engineering*, and *Parallel Computing*. He has also served on the technical program committees of many national and international conferences including IEEE INFOCOM, IEEE GLOBECOM, AND IEEE Annual Conference on Local Computer Networks. His current research interests are in next-generation Internet, broadband networks, multimedia over high-speed networks, TCP/IP over ATM, multiprocessors systems, and image processing. His book, *TCP/IP over ATM Networks* (Artech House), will be published in the second quarter of 2000. He has taught many short courses to industry in the area of computer and telecommunications networking. He has over 100 refereed publications.