QoS-Enabled Voice Support in the Next-Generation Internet: Issues, Existing Approaches and Challenges

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This work was supported in part by a grant from Research Grants Council (RGC) under contract CRC98/01.EG03.

ABSTRACT

The Internet is under rapid growth and continuous evolution in order to accommodate an increasingly large number of applications with diverse service requirements. In particular, Internet telephony, or voice over IP is one of the most promising services currently being deployed. Besides the potentially significant cost reduction, Internet telephony can offer many new features and easier integration with widely adopted Web-based services. Despite these advantages, there still exist a number of barriers to the widespread deployment of Internet telephony such as the lack of control architectures and associated protocols for managing calls, a security mechanism for user authentication, and proper charging schemes. The most prominent one, however, is how to ensure the QoS needed for voice conversation. The purpose of this article is to survey the state-of-the-art technologies in enabling the QoS support for voice communications in the next-generation Internet. In this article, we first review the existing technologies in supporting voice over IP networks, including the basic mechanisms in the IETF Internet telephony architecture and ITU-T H.323-related Recommendations. We then discuss the IETF QoS framework, specifically the Intserv and Diffserv framework. Finally, we present two leading companies' (Cisco and Lucent) solutions to offering IP telephony services as examples to illustrate how real systems are implemented.

INTRODUCTION

While the Internet has served as a research and education vehicle for more than two decades, the last few years have witnessed its tremendous growth and its great potential for providing a wide variety of services. In particular, using the Internet to carry phone conversations, known as Internet telephony or voice over IP (VoIP), is taking the telecommunications industry by storm. Not only does it represent the best opportunity so far for companies and telcos to facilitate voice and data convergence, but it also promises to deliver a new era in cheap telephone calls. Five years ago, Internet telephony was regarded by many to be far too unreliable for mass market deployment. But over the past few years, reliability and quality have quickly improved, and Internet telephony is now one of the fastest-growing industries.

The reason behind Internet telephony's success is that it can potentially bring enormous benefits to end users, telcos, and carriers. There are several compelling reasons that carriers are interested in IP telephony, including:

- It is cheaper for end users to make an Internet telephony call than a circuit-switched call, mainly because operators can avoid paying interconnect charges.
- Internet telephony gives new operators an easy and cost-efficient way to compete with incumbent operators by undercutting their pricing regimes, while avoiding many of the regulatory barriers to standard voice provision.
- Engineering economics favors Internet telephony. While a circuit-switched telephony call takes up to 64 kb/s, an Internet telephony call only takes up to 6–8 kb/s and possibly even less bandwidth.
- In the longer term, it offers exciting new value-added opportunities such as high-fidelity stereo conferencing bridges, Internet multicast conferencing, and telephony distance learning applications, phone directories and screen popping via IP, or even "voice Web browsing," where the caller can interact with a Web page by speaking commands.
- Internet telephony gives carriers the ability to manage a single network handling both voice and data. Internet telephony will also create end user opportunities and demand for new services. VoIP aims to ultimately bring end user benefits in terms of communication management — effectively meaning that people will be able to control different media and different types of terminals: Global System for Mobile Communications (GSM), fixed phone, PC, and so on from their Web browsers. Users will be

able to set up conference calls from their homes, route home calls to a GSM phone or centrex voice mail, look at the state of their accounts — even bar their children from accessing certain audiotex services. These are all services people will start to demand from their telephony service providers as the market matures.

As a result, the switch to IP as the main delivery mechanism for telecom services in the future is looking increasingly promising. Unfortunately, Internet telephony technology is also relatively immature, with quality and latency still being major issues. They are, however, both being addressed. Voice quality has improved greatly from early versions of the technology, which was characterized by distortions and disruptions in speech. Improved technologies for voice coding and lost packet reconstruction have also yielded products where speech is easy to understand. Latency, a factor that affects the pace of a conversation, is also being addressed. Humans can tolerate about 250 ms of latency before it has a noticeable effect, and voice services over the public Internet today typically exceed this figure. Latency will, however, continue to improve, driven by three factors: improved gateways (developers are just beginning to squeeze latency out of the first generation of products); deployment over private networks — by deploying gateways on private circuits, organizations and service providers can control the bandwidth utilization and hence latency; Internet development (today's Internet was not designed with real-time communications in mind). The Internet Engineering Task Force (IETF), together with Internet backbone equipment providers, is addressing this with technologies like Resource Reservation Protocol (RSVP), which will let bandwidth be reserved. While it will take some time for the world's routers to be upgraded and operational aspects (e.g., how to bill for high quality of service, QoS) to be resolved, the Internet world is moving fast and in the right direction.

The objective of this article is to review the recent developments and key enabling technologies in providing QoS supporting for voice communications in the next-generation Internet. The rest of the article is organized as follows. We first review the existing technologies in supporting VoIP networks, especially the basic mechanisms in the IETF Internet telephony architecture. We describe International Telecommunication Union — Telecommunication Standardization Sector (ITU-T) H.323-related Recommendations for enabling multimedia communications in packet-based networks. We then discuss the IETF QoS framework, specifically the integrated services model (Intserv) and differentiated services (Diffserv) architecture. We present two leading companies' (Cisco and Lucent) solutions in offering IP telephony services as examples to illustrate how the real systems are implemented. We then conclude the article.

Internet Telephony Standards

To support Internet telephony and other related applications, standards are being recommended and developed to insure interoperability. In particular, the ITU H.323 specification for Internet telephony is gaining widespread acceptance among software vendors. In addition, the IETF is developing protocols such as Session Initiation Protocol (SIP) for multimedia session initiation, and RTSP for controlling multimedia servers on the Internet that can work together with H.323.

Interwoven with all of the above protocols is the Real-Time Transport Protocol (RTP). It is used by H.323 terminals as the transport protocol for multimedia; both SIP and RTSP were designed to control multimedia sessions delivered over RTP. Its main function is to carry real-time services, such as voice and video, over an IP network. It provides payload type identification so that the receiver can determine the media type contained in the packet. Sequence numbers and timestamps are also provided so that packets can be reordered, losses detected, and data played out at the right speeds. RTP was designed to easily be used in multicast conferences. To this end, it guarantees that each participant in a session has a unique identifier, providing applications a way to demultiplex packets from different users.

RTP also contains a control component, called the Real-Time Control Protocol (RTCP). It is multicast to the same multicast group as RTP, but on a different port number. Both data senders and receivers periodically multicast RTCP messages. RTCP packets provide many services. First, they are used to identify the users in a session. One RTCP packet type, the Source Descriptor (SDES), contains the name, e-mail address, telephone number, fax, and location of the participant. Another, the receiver report, contains reception quality reporting. This information can be used by senders to adapt their transmission rates or encodings dynamically during a session. It can also be used by network administrators to monitor network quality. It could potentially be used by receivers to decide which multicast groups to join in a layered multimedia session.

One of the key components supporting VoIP is a signaling protocol, which has to provide the following functions: user location, session establishment, session negotiation, call participant management, and feature invocation [1]. Within the IETF, two protocols are defined to implement these tasks: SIP [2] and Session Description Protocol (SDP) [3].

SIP is used to initiate a session between users. It provides user location services, call establishment, call participant management, and limited feature invocation. SIP is a client-server protocol. This means that requests are generated by one entity (client), and sent to a receiving entity (the server), which process them. Since a call participant may either generate or receive requests, SIP-enabled end systems include both client and server. There are three types of servers. SIP requests can traverse many proxy servers, each of which receives a request and forwards to the next-hop server, which may be another proxy server or the final user agency server. A server may also act as a redirect server, informing the client of the next-hop server so that the client

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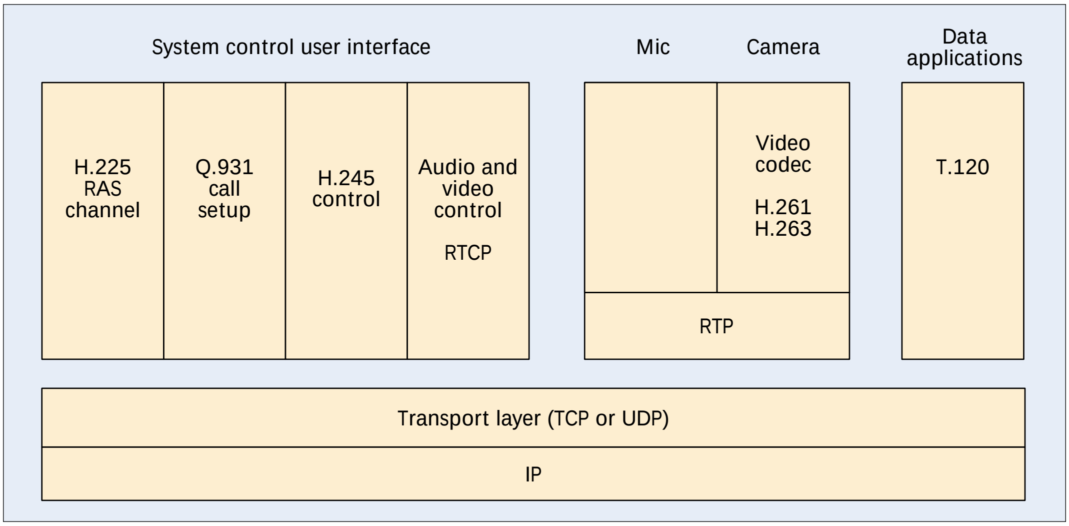

Figure 1. *The H.323 protocol stack.*

can contact it directly. SIP defines several methods, the client requests invoke method on severs. A client sets up a call by issuing an INVITE request. This request contains header fields used to convey call information. Following the header fields, there exists the body of the message that contains a description of the session to be established.

SDP is used to describe multimedia sessions for both telephony and distributed applications. The protocol includes several kinds of information, as follows. Media streams convey the type for each media stream. For each media stream, the destination address (unicast or multicast) is indicated by Address; Ports define the UDP port numbers for each sending or/and receiving stream. Payload type conveys the media formats that can be used during the session. For a broadcast-style session such as a television program, start and stop times convey the start, stop, and repeat times of the session, and Originator names the originator of the session and how that person can be contacted.

Basic Mechanisms in H.323

H.323 are a series of Recommendations of the ITU-T to enable multimedia communications in packet-switched networks [4]. H.323 is designed to extend the traditionally circuit-based services including audiovisual and multimedia conferencing services into packet-based networks. The Internet Telephony can be based on a subset functions of the H.323 for voice only support. Therefore, one of the primary objectives of H.323 is the interoperability with the existing circuit-switching systems (PSTN and ISDN).

The basic elements defined in H.323 architecture are: *terminals*, *gateways*, *gatekeepers*, and *multipoint control units* (*MCUs*), in which the terminals, gateways, and MCUs are collectively referred as *endpoints*.

A terminal is an end user device, which can be a simple telephone or PC/workstation. Its main responsibility is to participate in H.323-defined communications, including both point-to-point calls and multipoint conferences.

A gateway, as the name suggests, is an intermediate device to provide interoperation between H.323 compliant devices and non-H.323 devices, in particular PSTN and ISDN devices. The main functionalities contain the translation of signaling, media encoding, and packetization. There exist a number of different types of gateways; for example, gateways for PSTN devices and gateways for ISDN (H.320) videoconferencing devices.

A gatekeeper manages a set of registered endpoints, collectively referred as a *zone*. Its main functions include call admission (or call authorization), address resolution, and other management-related functions (e.g., bandwidth allocation). Each endpoint before initiating a call or conference has to register with the designated gatekeeper within the zone. The gatekeeper provides the address resolution to a specific transport address of the target recipient. It also determines whether to accept or reject the call connection request based on the available bandwidth or other system parameters.

An MCU provides the necessary control needed for multiparty video conferences. It contains two logical components: a *multipoint controller* (MC) for call control coordination and a *multipoint processor* (MP) to handle audio or video mixing.

The H.323 protocol stack is outlined in Fig. 1. The key protocols used in the call setup are the *Registration Admission Status* (RAS) *protocol*, a *Q.931-based signaling protocol*, and an *H.245 media and conference control protocol*.

RAS protocol is responsible for registration of endpoints (terminals, gateways, and MCUs) to the correspondent gatekeeper. RAS messages carried in User Datagram Protocol (UDP) packets contain a number of request/reply messages exchanged between the endpoints and gatekeeper. Besides the registration, RAS protocol also provide a means for the gatekeeper to monitor the endpoints within the zone and manage the associated resources.

The Q.931-based signaling protocol is derived from the integrated services digital network (ISDN) Q.931 signaling protocol tailored for use in the H.323 environment. The signaling messages are carried in reliable TCP packets. It provides the logical connection between the calling and called parties.

The H.245 media and conference control protocol is used for the two connected parties (after Q.931 establishment) to exchange various information related to their communications; for instance, type of messages (audio, video, or data) and format. In addition, it provides a set of control functions for multiparty videoconferences.

RTP and RTCP, described earlier, are used for actual message transmission.

A summary of the H.323 protocol phases is given in Fig. 2. The RAS protocol is used in phases 0, 1, and 6 for registration and shutdown process. Signaling protocol is involved in phases 2, 5, and 6. The H.245 media and conference control protocol is active during phases 3 and 5, and media exchanged based on RTP/RTCP is carried out in phase 4 [4].

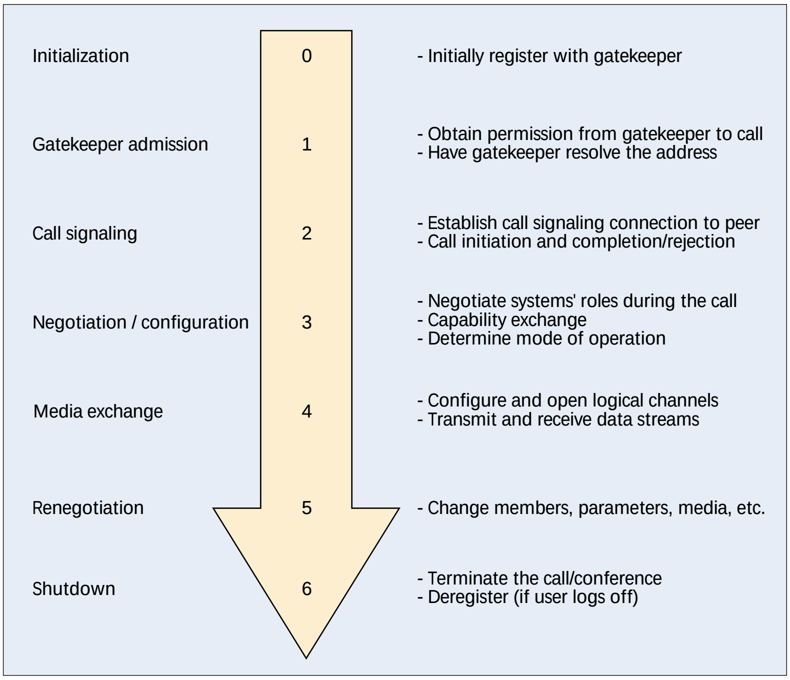

Figure 2. *The H.323 protocol phases.*

QoS Issues in the Internet

The existing Internet service (i.e., the *best-effort service* of IP) cannot satisfy the QoS requirements of emerging multimedia applications, primarily caused by the variable queuing delays and packet loss during network congestion. There has been a significant amount of work in the past decade to extend the Internet architecture and protocols to provide QoS support for multimedia applications. This has led to the development of a number of service models and mechanisms. In this section we discuss two key models: Intserv and Diffserv.

The Integrated Service Model

The Intserv model was proposed as an extension to support real-time applications. The key is to provide some control over the end-to-end packet delays in order to meet the real-time QoS. Specifically, the Intserv model proposes two service classes in addition to best-effort service. They are:

- Guaranteed service for applications requiring a fixed delay bound
- Controlled-load service for application requiring reliable and enhanced best-effort service

The fundamental assumption of the Intserv model is that resources (e.g., bandwidth and buffer) must be explicitly managed for each real-time application. This requires a router to reserve resources in order to provide specific QoS for packet streams, or *flows*, which in turn requires flow-specific state in the router. The challenge is to ensure that this new service model can work seamlessly with the existing best-effort service in one common IP infrastructure.

Intserv is implemented by four components: flow specification, the signaling protocol (e.g., RSVP), admission control routine, and packet classifier and scheduler. Applications requiring guaranteed or controlled-load service must set up path and reserve resources before transmitting their data. *Flowspec*, describing the source traffic characteristics, has to be provided to the network. Under the Intserv framework, two separate parts of the Flowspec are defined: one describes the flow's traffic characteristics (the *Tspec*), and the other specifies the service requested from the network (the *Rspec*). Admission control routines determine whether a request for resources can be granted. When a router receives a packet, the packet classifier will perform a classification and put the packet in the appropriate queue based on the classification result. The packet scheduler will then schedule the packet accordingly to meet its QoS requirement.

The IETF Differentiated Services Framework

The Diffserv architecture as specified by IETF offers a framework within which service providers can offer each user a range of network services which are differentiated on the basis of performance [5]. The Diffserv architecture is based on a simple model where traffic entering a network is classified and possibly conditioned at the boundaries of the network, and assigned to different behavior aggregates (BAs), with each BA being identified by a single Diffserv codepoint (DSCP). Users request a specific performance level on a packet-by-packet basis, by marking the Diffserv field of each packet with a specific value. This value specifies the per-hop behavior (PHB) to be allotted to the packet within the provider's network. Within the core of the network, packets are forwarded according to the PHB associated with the DSCP.

Sophisticated classification, marking, policing, and shaping operations need only be implemented at network boundaries or hosts (Fig. 3). Network resources are allocated to traffic streams by service provisioning policies which govern how traffic is marked and conditioned upon entry to a Diffserv-capable network, and how this traffic is forwarded within that network. A wide variety of services can be implemented on top of these building blocks.

A salient feature of the Diffserv-iframework is its scalability, which allows it to be deployed in very large networks. This scalability is achieved by forcing much complexity out of the core of the network into boundary devices which process smaller volumes of traffic and less numbers of flows.

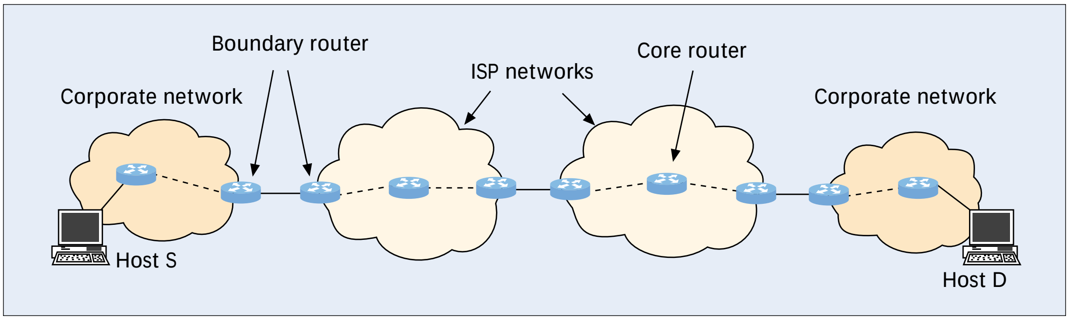

Figure 3. *End-to-end transport from host S to host D under the Diffserv architecture.*

A salient feature of the Diffserv framework is its *scalability*, which allows it to be deployed in very large networks. This scalability is achieved by forcing much complexity out of the core of the network into boundary devices which process smaller volumes of traffic and fewer flows, and by offering services for aggregated traffic rather than on a per-microflow basis. That is, complex traffic classification and conditioning functions are only implemented at network boundary nodes; inside the core network, PHBs are applied to aggregates of traffic which have been appropriately marked using the Diffserv field in the IPv4 or IPv6 headers. PHBs are defined to permit a reasonably granular means of allocating buffer and bandwidth resources at each node among competing traffic streams. Per-application flow or per-user forwarding state need not be maintained within the core of the network.

A Diffserv architecture can be specified by defining or implementing the following four components:

- The services provided to a traffic aggregate
- The traffic conditioning functions and PHBs used to realize the services
- The Diffserv field value (DSCP) used to mark packets to select a PHB
- The particular node mechanism to realize a PHB

Services — A service defines some significant characteristics of packet transmission in one direction across a set of one or more paths within a network.

There are two approaches to provide Diffserv:

- The first approach specifies the QoS in deterministically or statistically quantitative terms of throughput, delay, jitter, and/or loss. Such approach is called *quantitative Diffserv*.
- The second approach specifies the services in terms of some relative priority of access to network resources and is called *priority-based Diffserv*.

Conditioning Functions and PHB — In order for a user to receive Diffserv from its Internet service provider (ISP), it must have a service-level agreement (SLA) with its ISP. A SLA basically specifies the service classes supported and the amount of traffic allowed in each class, respectively.

Users can mark Diffserv (DS) fields of individual packets to indicate the desired service at hosts or have them marked by the access or boundary router (Fig. 3). At the ingress of the ISP networks, packets are classified, policed, and possibly shaped. The classification, policing, and shaping rules used at the ingress routers are derived from the SLAs. When a packet enters one domain from another, its DS field may be remarked, as determined by the SLA between the two domains. Such traffic control functions at hosts, or access or boundary routers are generically called *traffic conditioning* [5].

PHB refers to the externally observable forwarding behavior applied to a Diffserv behavior aggregate at a Diffserv-compliant node. PHBs are defined to permit a reasonably granular means of allocating buffer and bandwidth resources at each node among competing traffic streams.

DS Codepoint — An IPv4 header contains a type of service (ToS) field, while an IPv6 header contains a traffic class byte. The IETF Differentiated Services Working Group has defined the layout of this byte (the DS field). By marking the DS field of packets differently and handling packets based on their DS fields, various Diffserv classes can be created. Six bits of the DS field are used as a codepoint (DSCP) to select the PHB a packet experiences at each node, while the other two are currently unused (CU).

A Node Mechanism for Achieving PHB — PHBs are implemented in nodes by means of some buffer management and packet scheduling mechanisms. PHBs are defined in terms of behavior characteristics relevant to service provisioning policies, not in terms of particular implementation mechanisms. In general, a variety of implementation mechanisms may be suitable for implementing a particular PHB group.

EXISTING SOLUTIONS

In this section we present two leading companies' solutions to offering IP telephony services as examples to illustrate how real systems are implemented. The Cisco IP telephony system described is targeted at enterprise networks, while the Lucent solution is for carrier networks.

THE CISCO SOLUTION: ENTERPRISE IP TELEPHONY

The Cisco solution for IP telephony in enterprise networks includes hardware, such as switches, routers, IP/PSTN gateways, desktop IP phones, and software, such as the call manager. An IP telephony system can be built by utilizing these products in

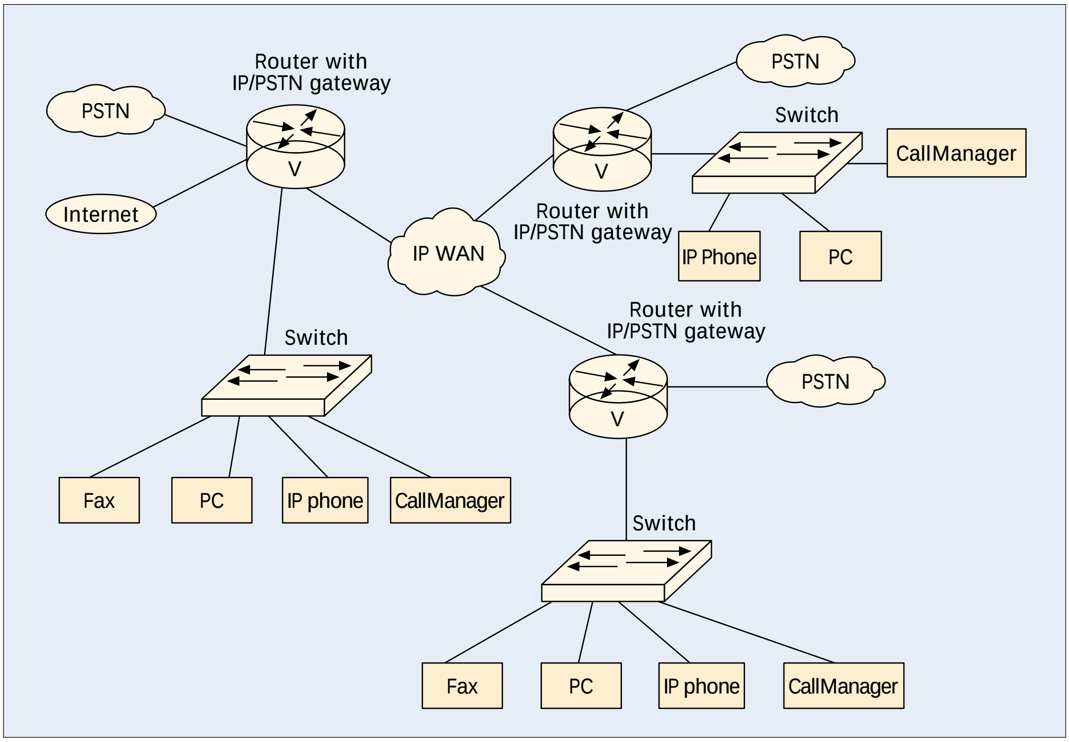

Figure 4. *The Cisco data and IP telephony network configuration.*

The QoS guarantees are primarily provided by two mechanisms: the call manager, equipped with a resource reservation protocol (such as RSVP), and a priority queue mechanism.

the current IP infrastructure. Figure 5 illustrates a typical scenario of a Cisco IP telephony system.

In this IP telephony system, voice and data can be integrated in the wide area network (WAN) by permitting long distance calls to traverse the existing data infrastructure between remote locations. By using routers and gateways to connect the PBX, voice traffic can be carried over data IP networks. Call management software and IP telephones are deployed in the existing IP networks at each remote site. This will reduce the cost of WAN consolidation while at the same time eliminating the cost of installing a second network at each remote location. Using the analog access gateway (at the remote site), local calls can be enabled for remote users. Long distance calls can be routed over the WAN link and consolidated from the central site. With this approach, the transport for IP telephony becomes transparent to users, who will be unable to distinguish whether a call is placed over a packet network, a circuit-switched network, or a combination of both. The networks can support multiple classes of services (CoSs) and provide guaranteed QoS to real-time communications. QoS functions and mechanisms are distributed between cooperating edge/aggregation devices and core/backbone switches. Packet classification and user policies are applied at the edge of the network. Packet classification identifies and categorizes network traffic into multiple classes. The Cisco IP phone can set the IPv4 ToS at the ingress to the network.

The QoS guarantees are primarily provided by two mechanisms: the call manager equipped with a resource reservation protocol (e.g., RSVP) and a priority queue mechanism. The priority queue mechanism is maintained in the core routers, and is responsible for high-speed switching and transport as well as congestion avoidance. Congestion avoidance uses packet discard mechanisms such as weighted random early detection (WRED) to randomly drop packets on a congested link. WRED ensures that the voice packets will get higher-priority services while no one user monopolizes network resources.

Lucent Gateway Solution for Service Provider Networks

The Lucent Gateway approach is target for service provider networks [6]. In this architecture an H.323- or SIP-compliant terminal (e.g., an IP phone) is connected to the IP switch or router. The edge switches or routers serve as access points and concentrators for the core IP network, which comprises higher-capacity IP routers or switches. A directory server is connected to the core network and serves multiple edge nodes.

The core network can be implemented using several different technologies: IP routers, IP switches, IP-over-ATM (asynchronous transfer mode) switches, IP over a synchronous optical network (SONET), and IP over dense wavelength-division multiplexing (DWDM). To the end terminal, the network is an IP network regardless of the underlying technologies.

Two gateways are added to the IP network architecture as interfaces to the public switched telephone network (PSTN). The first added is a *connection gateway* (*CG*), which performs signaling interworking between the IP protocol (e.g., H.323 or SIP) and PSTN protocols. The second is a *voice gateway* (*VG*), which converts time-division multiplexed (TDM) signals into IP packet and vice versa.

The gateways allow a local area network

The core network can be implemented using several different technologies: IP routers, IP switches, IP over ATM switches, IP over a synchronous optical network (SONET), and IP over dense wavelength division multiplexing (dense WDM).

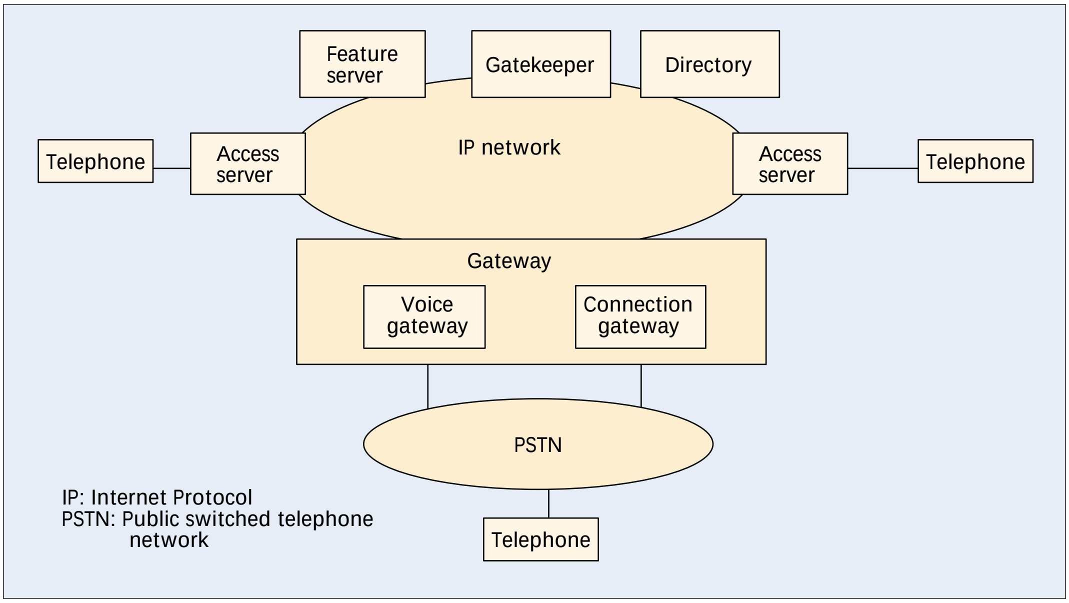

Figure 5. *Lucent IP and PSTN architecture.*

(LAN) telephone to call another LAN phone on the network and eliminates the need for a voice gateway. In addition, a LAN telephone can also call a regular plain old telephone service (POTS) phone through the gateway. Note, however, that within an IP network there is no distinction between local and long distance calls.

The Lucent router implements a straightforward scheme for QoS. It simply extracts ToS information from incoming IP packets and sets up a series of prioritized queues. These queues can control packet flow based on the CoS value, which allows the router to prioritize voice data and move fax data to a lower priority, thereby minimizing delay on real-time information at the expense of less time-critical information.

The difference between these two approaches lies in the fact that the Cisco system is targeted for the enterprise network, in which per-flow end-to-end QoS guarantee is possible. However, the requirement for setting up a path might not be feasible for the Internet, due to its poor scalability. The Lucent approach is used for carrier networks, which is more scalable but relies on the underlying IP network to provide the needed QoS.

Conclusions

There has been significant work done to establish the foundation to support VoIP. However, much remains to be done in order to ensure the QoS for VoIP and for multimedia traffic in general in the next-generation Internet. This article surveys the existing technologies to support VoIP, in particular the basic mechanisms in the IETF Internet telephony architecture and ITU-T H.323-related recommendations. It then reviews the IETF QoS framework and major components in providing such QoS guarantees, including the Intserv and Diffserv models. In addition, this article also presents two leading companies' (Cisco and Lucent) solutions to offering IP telephony services as examples illustrating how real systems are implemented.

One other major issue currently under active development is internetworking with legacy networks (i.e., PSTN). There are a number of proposals within the IETF, in particular Media Gateway Control Protocol (MGCP) [7]. It is anticipated that such services will be available soon.

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Biographies

Bo Li [SM] (BLI@cs.ust.hk or boli@acm.org) received B.S. (summa cum laude) and M.S. degrees in computer science from Tsinghua University, Beijing, China, in 1987 and 1989, respectively, and a Ph.D. degree in computer engineering from the University of Massachusetts at Amherst in 1993. Between 1994 and 1996 he worked on high-performance routers and ATM switches in IBM's Networking System Division, Research Triangle Park, North Carolina. He joined the faculty of the Computer Science Department of Hong Kong University of Science and Technology in January 1996. He has been on the editorial board of *ACM Mobile Computing and Communications Review* (MC2R), *ACM/Baltzer Journal of Wireless Networks* (WINET), *Journal*

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There has been significant work done to establish the foundation for supporting VoIP. However, much remains to be done in order to endure the QoS for VoIP and for multimedia traffic in general in the next-generation Internet.