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New IP QoS Architecture for Voice and Data Convergence over DSL Lines

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Abstract: Digital subscriber lines (DSL) offer the possibility to deliver broadband services over the existing telephone network. Before deploying DSL, the subscriber loops must be tested to see whether they can support high-speed data services, and at what rate. With regards to Voice over IP, VoIP converts analog voice signals into IP packets and distributes them across a WAN. VoIP is a delay-sensitive application. For POTS-quality voice, the network must be fine tuned end-to-end before implementing VoIP. This fine-tuning should incorporate a series of optimization techniques in order to improve the overall quality of service (QoS). Furthermore traffic shaping must be used to ensure VoIP reliability. There are two essential requirements to ensure VoIP quality: minimization of end-to-end delay and jitter. In this paper, the performance of QoS architecture was analyzed.. The basic metrics that were evaluated are the end-to-end delay of voice packets across the access network and the bandwidth consumed by a voice call.

Key words: VoDSL • SLIP • ACIS. DSL • QoS • PSTN • VoATM • VoIP • PPP • DSLAM • NSP• CPE

INTRODUCTION

Digital Subscriber Line (DSL) is a family of related technologies that bring high-speed network access to homes and small businesses over ordinary telephone lines [1, 2]. The basics of DSL networking refers to a collection of technologies used for the transmission of high-speed data over copper twisted-pair lines. It is used to connect the Network Service Providers (NSP) and the customers, which are usually residences or small-to-medium sized businesses. At the customer's home or office, a device called the Customer Premises Equipment (CPE) provides access to the NSP's network. The CPE connects to a DSL Access Multiplexer (DSLAM) located in the Central Office (CO) of the NSP. The DSLAM aggregates traffic from different customers and sends it over a high-speed uplink towards the core of the network.

There are two essential requirements to ensure VoIP quality: minimization of end-to-end delay and jitter. End-to-End Delay represents one of the most crucial factors in implementing VoIP is minimizing one-way, end-to-end delay. VoIP traffic is real-time traffic; if delay is too long, speech becomes unrecognizable. Less than 150ms is considered an acceptable delay for VoIP. There are two types of delays inherent in today's telephony networks:

Propagation delay;- is caused by the speed of light traveling from point A to point B. The longer the distance, the longer the delay.

Serialization delay: - is caused by the devices that handle voice information. Serialization delays significantly degrade voice quality in a packet network.

Jitter: - is the variation delay between the time a voice packet is expected to be received and when it actually is received. Jitter causes discontinuity in the real-time voice stream. Excessive jitter can result from congestion on LANs, Access Links, and low bandwidth WAN links or from the transmission of large data packets on the same link.

DSL service is delivered over conventional copper loops from DSL Access Multiplexers (DSLAMs) in the Central Office (as shown in Fig.1). For those customers who receive only data services over DSL, these loops are terminated at the customer premises with a DSL modem or router. For combined voice and data services, the DSL loop is terminated typically by a device that provides integrated voice and data access. Such devices typically offer an Ethernet port for data and multiple analog POTS ports for voice.

Digital Subscriber Line (DSL) technology has helped quench our thirst for bandwidth in recent years, extending the life of existing copper twisted-pair networks that now serve over 100 million subscribers around the globe with broadband internet connectivity. Whilst DSL technology has been hugely successful, incumbent telephone operators are increasingly faced with stiff competition from the decreasing cost of optical fiber-fed leased lines, and aggressive cable television companies serving subscribers from much higher bandwidth Hybrid Fiber-Coax (HFC) cable

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networks. The DSL connection to the customer makes use of a packet protocol such as ATM or frame relay to support voice and data. The DSLAM serves as a packet concentrator, delivering traffic from multiple customers over a high-speed uplink to a metropolitan or regional packet network. The principle data service that is offered to DSL customers is Internet access, so the packet network is connected to the Internet, typically through a device known as a Subscriber Management System. Connections to enterprise data networks may also be present, to support telecommuters and home-based workers.

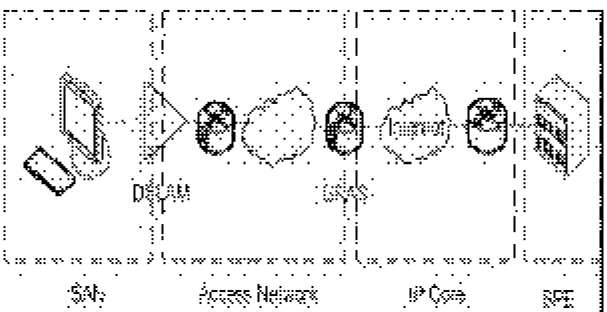

Fig.1 DSL reference architecture

Voice over DSL (VoDSL) uses the existing DSL access network to provide voice services in addition to data services. Voice is packetized at the customer premises and the packet-switched DSL access network is used to deliver the voice packets to a voice gateway. The voice gateway converts packetized voice into circuit-switched voice traffic and sends it to the PSTN. Thus, a single copper pair can be used to provide data services and also one or more voice lines to the customer. This eliminates the cost of provisioning separate copper pairs for voice and data and additional copper pairs for multiple voice lines. Another advantage of having packet-based voice is that voice calls consume bandwidth only when they are active.

DSL has several variants that use different transmission technologies, offer different data rates and support different types of services. The main interest is concerned in Asymmetric Digital Subscriber Line (ADSL), since it is the most widely deployed DSL technology. ADSL provides higher Bandwidth from the NSP to the customer (downstream) than from the customer to the NSP (upstream). The generic network topology can be implemented in a variety of forms, to suit the specific requirements of service delivery. The two main architectural variants are: the "centralized" and the "distributed" architectures, meet the needs of different kinds of service providers by locating the voice gateway optimally in relation to the other network elements. The current trend is to use Asynchronous Transfer Mode (ATM) as the transport technology for VoDSL. This is also called the Voice over ATM (VoATM) approach. VoATM takes advantage of ATM's built-in Quality of Service (QoS) mechanisms to guarantee low End-to-end (ETE) delays for voice packets. ATM also uses a lightweight protocol for carrying voice, there by ensuring high bandwidth efficiency.

For ATM transport of voice traffic, there are two main areas: N x 64 kb/s structured data transfer, also known as composite cell transport. A network operator may aggregate individual voice calls into N x 64 kb/s blocks for transport to an intermediate point in the network. At that point the individual calls need to be remapped into new M x 64 kb/s blocks for the next stage of their transport. Architectural alternatives and advantages of remapping are investigated; Interworking of signalling. Interworking of signaling is of two types: network interworking, in which the end users are connected to non-ATM networks and an ATM network is used for backbone transport; and service interworking, in which an ATM user is connected to a non-ATM user. Functionality and architectural alternatives for network and service interworking are evaluated.

An alternative approach of the above is to use the Internet Protocol (IP) as the transport technology for packet voice. This is also called the Voice over IP (VoIP) approach. Compared to VoATM, VoIP suffers from several potential problems. The performance of voice traffic degrades in the presence of competing data traffic. ETE IP QoS mechanisms that prioritize voice traffic are not yet standardized. Bandwidth efficiency is another issue in VoIP. The VoIP protocol stack adds a number of headers to the voice packets, which impose a considerable overhead on the voice packets. These difficulties have hampered the deployment of VoIP for VoDSL and made ATM the preferred protocol.

However, there is considerable interest in using IP-based data networks to replace PSTNs as carriers of voice in core networks, since it is cheaper to packetize voice and carry it over a data network. IP is certainly the preferred protocol in the core, owing to its universality. IP can run over any kind of core network, such as those based on ATM, SONET, Gigabit Ethernet or Frame Relay. Also, data traffic is growing much faster than voice traffic and hence it makes sense to use IP in the core, as it the predominant protocol for carrying data traffic over the Internet.

If the performance of VoIP in the DSL access network can be improved using IP QoS mechanisms, then the deployment of VoIP for VoDSL would yield several other benefits. First, the voice packet format used in the DSL access network would be compatible with that used in the core. This eliminates the need for a voice gateway and paves the way for ETE IP telephony. Also, VoIP can run over any kind of DSL network-ATM based, frame-based or Point-to-Point Protocol (PPP) based. Since IP is already being used in DSL to carry data traffic, it would be easy for voice traffic to run over IP too. Thus, a solution that achieves VoIP performance comparable to that provided by VoATM would be of enormous benefit in the DSL access network. In this paper, we assess VoIP (shown in Fig.2) as a solution for VoDSL. For VoIP deployment to be feasible, the used IP QoS mechanisms must be able to guarantee a service similar to that guaranteed by ATM's built-in QoS mechanisms. This involves prioritization of voice packets and protection of voice traffic from competing data traffic and employing Admission control called Admission Control by Implicit Signalling (ACIS). In addition to this, techniques to alleviate the bandwidth overhead imposed by the VoIP protocol stack must be employed.

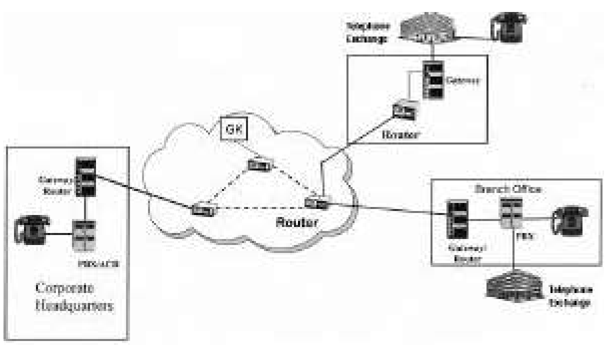

Fig.2 VOIP in Service

VoIP architecture could replace the existing architecture to deliver comparable voice quality. In addition, it also offers the benefit of voice packet compatibility with the core network, which is already employing VoIP to carry voice inexpensively over data networks. Thus, the idea of end-to-end IP telephony is conceivable in the future. Unlike most data applications, voice is very sensitive to delay. Good voice quality provides a faithful recreation of the conversation, with the same tone, inflection, pauses and intonation used by the speakers. Long and variable delays between packets result in unnatural speech and interfere with the conversation. Dropped packets result in clipped speech and poor voice quality. Fax transmissions are even more sensitive to the quality of the transmission and are less tolerant of dropped packets than voice.

One way to deal with the problem of delay and congestion is to add bandwidth to the network at critical junctures. Although this is feasible in the backbone, it is a costly and ineffective solution in the access arena, defeating the "bandwidth sharing" benefits of packet networks. The best solution is to implement mechanisms at the customer premises, access node and backbone which manage congestion and delay - without increasing bandwidth - such as setting priorities for different types of traffic. Therefore, smart access equipment was developed, that could implement procedures to reduce network congestion and the delay of voice packets without adding bandwidth.

LITERATURE REVIEW

Voice over Internet Protocol (VoIP) is a method used in data networks and broadband internet to establish voice calls. This is implemented by converting the analog voice calls into digital format that can be transmitted through the internet or through the intranet as shown in Fig. 2. As a result, voice signals will be transferred through a packet-switched network instead of being transmitted through a dedicated circuit switched voice lines. This method gives an advantage of reducing and sometimes completely bypassing the expensive fees imposed by telephone companies. There are, for instance, many software packages that enable this type of calls through personal computers. Another method that performs this task is to use a telephone adapter which establishes this type of calls. It is worth noting that VoIP can be implemented in any IP-based network such as local area networks (LANs). With the introduction of WLAN, many manufacturers started testing the ability of using it for VoIP service . Voice communication can thus benefit from the mobility offered by the network. After experimenting with this possibility, it was found that it is difficult to apply VoIP in a wireless environment. In such environments, the signal power at a certain point is difficult to determine, which leads to uncertainty in data rate. It is this pattern which creates difficulty in applying the Voice over IP in voice communication scheme.

The performance of the Voice over IP (VoIP) protocol is of interest when planning for public access. The technologies, which work well for limited use often, fail to scale-up to the user requirements. High-quality VoIP services are required as for the Internet communications to be an alternative towards Public Switched Telephone Network (PSTN). The deployment of VoIP in the Internet network does not promise a good Quality of Service (QoS), since Internet is a kind of best-effort networks . The privacy consideration is also of importance when provisioning voice services on the Internet; particularly from the business use perspective.

One of the key requirements for the widespread deployment of VoIP is the ability to offer a toll quality service equivalent to the existing PSTN. Indeed some carriers are even looking for Next-Generation Networks as a means for delivering much higher voice quality as a service. Perceived Voice quality is very sensitive to three key performance criteria in a packet network, in particular: · Delay, ·Jitter, · Packet loss IP, by its nature, provides a best-effort service and does not provide guarantees about the key criteria. Therefore it is necessary to implement a suitable QoS solution in the majority of cases where simple over provisioning cannot guarantee success. There are a large number of technologies that can be chosen to provide QoS support such as Diffserv, RSVP, MPLS and even ATM. However the objective of such a solution is always to guarantee prioritization of voice media streams over best-effort data, and to ensure that the voice service is not compromised by unforeseen traffic patterns.

IP QoS has been an area of active research in the recent years. The research has led to the emergence of several mechanisms for providing different levels of service to different kinds of applications. Several publications have studied buffer management and scheduling algorithms under the assumption that the network is over provisioned. Over provisioning implies that sufficient bandwidth has already been provisioned for classes of traffic that require a high level of service. In this scenario, the offered load for these classes would never exceed the bandwidth provisioned. [4] Studies the effectiveness of priority queuing in providing QoS to voice traffic, while [5] evaluates Class-Based Queuing for the same purpose. [6, 7] propose and evaluate their own buffer management and scheduling algorithms. All these schemes yield good results in situations with over provisioning. But very often, the resources provisioned for traffic classes requiring strict service guarantees are limited. Hence, admission control must be applied to limit the number flows (of these classes) entering the network. Otherwise, all flows belonging to a class will be equally degraded when the offered load exceeds the limit that can be supported by the network for that class. Keeping this in mind, we designed our QoS architecture to include not only buffer management and scheduling, but also admission control.

In the past, researchers have come up with several ways to enforce admission control. Endpoint admission control is an admission control technique in which the end-hosts themselves make the admission control decision, without any support from the routers. The basic idea is to have the end-host send probe packets into the network at the rate at which it intends to transmit the actual data packets, if admitted. The probe packets are sent to the receiver with which the end-host wishes to communicate. The receiver makes measurements on the arriving probe packets in order to estimate the QoS received by them. The results are reported to the sender, which makes the admission control decision based on the report. [8, 9] study the performance of endpoint admission control algorithms in general. [10-12] propose and evaluate their own endpoint admission control schemes. However, these algorithms rely on measurements made by the end-hosts and hence they might be inaccurate. On the other hand the end-hosts support ACIS operation by marking application layer signaling protocols appropriately, but the admission decision is taken by the router (CPE). The QoS architecture has been developed for the DSL access network and the end-to-end operation of endpoint admission control is not suitable for this. Besides, ACIS can provide better QoS guarantees since the admission decision is made by the network and not by the end-host.

Measurement-Based Admission Control (MBAC) is a concept in admission control where the decision to admit a flow into the network is made based on traffic measurements. Endpoint admission control algorithms are actually a special case of MBAC and are also called Edge-to-Edge MBAC (EMBAC) algorithms. Unlike endpoint admission control algorithms, the MBAC mechanisms do require router support. The involvement of routers offers MBAC a distinct advantage, since each router can make a more accurate estimate of local congestion. Consequently, the admission decisions in MBAC are based on accurate measurements made at each hop, rather than measurements made solely at the end-hosts. [13, 14] propose their own MBAC schemes.

Some QoS architectures like those proposed in [15,16] use a resource managing entity called a Bandwidth Broker (BB) to manage bandwidth resources within an administrative domain. The BB has information about the resources allocated within the domain and can thus make admission control decisions for the entire domain. The BB maintains the per-flow state for all routers across the network. ACIS can be operated in conjunction with BBs also. Rather than have the CPE maintain a fixed bandwidth reservation for premium traffic, a bandwidth broker (located perhaps at the DSLAM) can be used to keep track of the bandwidth available for premium traffic in the network service provider's entire domain. The CPE would contact the BB whenever requests for new connections arrive. A signaling protocol would necessary for communication between the CPE and the BB.

NETWORK TOPOLOGIES FOR VOICE OVER DSL

DSL service is delivered over conventional copper loops from DSL Access Multiplexers (DSLAMs) in the Central Office. For those customers who receive only data services over DSL, these loops are terminated at the customer premises with a DSL modem or router. For combined voice and data services, the DSL loop is terminated typically by a device that provides integrated voice and data access. Such devices typically offer an Ethernet port for data and multiple analog POTS ports for voice. The DSL connection to the customer makes use of a packet protocol such as ATM or frame relay to support voice and data. The DSLAM serves as a packet concentrator, delivering traffic from multiple customers over a high-speed uplink to a metropolitan or regional packet network. The principle data service that is offered to DSL customers is Internet access, so the packet network is connected to the Internet, typically through a device known as a Subscriber Management System. Connections to enterprise data networks may also be present, to support telecommuters and home-based workers.

Voice services are delivered to DSL customers by means of a voice gateway which connects the public switched telephone network (PSTN) to the packet network. Digital voice streams are converted into packet format for transport over the packet network between the voice gateway and the integrated access device on the customer premises. The voice gateway connects to the PSTN via a Class 5 switch. Since the voice gateway represents a digital access network from the point of view of the Class 5 switch, the connection between the gateway and the Class 5 switch typically makes use of a standard interface for digital loop carrier systems. The generic network topology just described can be implemented in a variety of forms, to suit the specific requirements of service delivery. The two main architectural variants, the "centralized" and the "distributed" architectures, meet the needs of different kinds of service providers by locating the voice gateway optimally in relation to the other network elements.

CENTRALIZED ARCHITECTURE

The centralized architecture meets the needs of service providers who wish to deliver voice services from a centrally located Class 5 switch via DSLAMs in multiple different Central Offices. This is normally the case with competitive local exchange carriers (CLECs) who provide DSL service from DSLAMs which are installed in collocation cages in an incumbent's Central Offices. In this case, the packet network which aggregates traffic from multiple DSLAMs is used to backhaul both voice and data traffic to one or more locations where connections are made to the Internet and to the PSTN. The benefit of this approach is that a single Class 5 switch can serve the voice needs of customers who are spread over a large metropolitan area. The packet network concentrates voice traffic from multiple different DSLAMs into a single gateway connected to the PSTN, which enables the service provider to offer voice services over an extensive geographic area with a modest initial investment, even in the early phases of deployment when market penetration is low.

DISTRIBUTED ARCHITECTURE

The distributed architecture meets the needs of service providers who own both DSLAMs and Class 5 switches in the same Central Offices, and who wish to drop off voice traffic from the DSL network locally in each Central Office. This architecture is appropriate for incumbent local exchange carriers (ILECs) who wish to serve their VoDSL customers from the same Class 5 switch as they would use to provide POTS services to these customers. In the distributed architecture, there is a requirement to separate voice traffic from data traffic between the DSLAM and the regional packet network. Data traffic will be passed over the packet network to a central location where connections into the Internet are made, while voice will be handed off locally to the Class 5 switch.

The separation of voice and data packets requires a packet switching function; however the installation of a separate packet switching device in each CO to accomplish this is generally regarded as highly undesirable. There are two different solutions to this architectural problem. The first solution is to use an enhanced voice gateway that provides "data pass through" functionality. This type of gateway connects to both the DSLAM uplink and the regional packet network.

Packets coming from the DSLAM are examined to see if they contain voice or data. Voice packets are dropped off inside the gateway and converted to circuit traffic for connection to the Class 5 switch, while data packets are passed on to the regional data network. In the reverse direction, data packets arriving from the regional packet network are merged with voice packets generated from PSTN circuits, and the combined packet stream is sent to the DSLAM. The second solution is to use a "switching DSLAM" that offers two or more high-speed uplinks. The packet connections in the DSLAM are configured to direct data packets out of one uplink to the regional packet network, while voice packets are directed out of the other uplink into a voice gateway for handoff to the Class 5 switch.

PACKET PROTOCLS FOR DSL ACCESS NETWORK

Packet networks are preferred without defining what kind of packet we are talking about. In the context of DSL access networks, the term "packet" could refer to an ATM cell, a frame relay frame, or an Internet Protocol (IP) packet. IP packets are typically carried as payload within ATM cells or frame relay frames. To further confuse the picture, packet switching within the regional packet network may be carried out at the ATM or frame relay layer by means of cell switches or frame switches, or it may be carried out at the IP layer by means of routers. And to make matters even more complicated, frame relay networks may be interconnected with ATM networks via frame-to-cell interworking functions. The delivery of combined voice and data services over DSL is implicit in the concept of VoDSL. The data services that are delivered over DSL are almost exclusively IP-based, so IP is universally supported by DSL access networks. But for VoDSL, the question that needs to be answered is: what kind of packet should be used for voice transport?. Before answering this, we need to look in more detail at how existing DSL access networks are constructed. And following the principles of one of the key market requirements identified above, which states that VoDSL solutions should overlay and not displace existing access network architectures, we need to identify a solution for packet voice that is compatible with current practice.

DSL ACCESS NETWORK REQUIREMENTS

The delivery of voice over DSL places certain requirements on the DSL access network infrastructure over and above those which apply if a data-only service is to be delivered. These requirements are generally not difficult to meet, but attention clearly needs to be paid to them.

ATM-BASED DSLAMs

ATM-based DSLAMs must provide Quality of Service on a per-virtual circuit connection basis. In practice this means the voice and data must be handled via different queues both at the DSL ports in the direction towards the user, and at the high-speed uplink port in the direction towards the network. The queues must be managed such that, provided the voice traffic does not exceed the provisioned cell rate, voice will always be given priority over data.

FRAME-BASED DSLAMs

Frame relay does not support the same set of Quality of Service capabilities as ATM. Nevertheless, frame-based DSLAMs can support mixed voice and data traffic successfully provided that prioritization is applied on a per-virtual circuit connection basis. As is the case for ATM, this means that voice and data VCCs need to be handled through different queues on all DSLAM ports, with voice being given priority. There is an additional requirement on frame-based DSLAMs which arises from the variable packet size property of frame relay. Where there is a mix of large data packets and small voice packets on a frame-based connection, voice packets will suffer from large variations in queuing delay even if the voice queue is given priority. This issue was identified above in relation to voice over IP, and it applies also to the case of voice over frame. The delay variation occurs because large packets take appreciable amounts of time to be transmitted over DSL connections, and once a data packet transmission has commenced, it must proceed until the entire packet has been sent. This is in contrast to ATM, where large data packets are segmented into small, fixed-length cells which can be interleaved with voice packets.

In a typical VoDSL situation, where the DSL bandwidth is 384 kbps and the data service supports the transport of 1500-byte IP packets, the voice packets will experience a variable queuing delay or jitter of about 30 ms. This arises because a voice packet that is queued for transmission when the line is not busy will be sent immediately, whereas a voice packet that arrives in the queue just after a 1500-byte data packet has started transmission will have to wait until the entire packet has been sent. The transmission time for a data packet is calculated as (number of bits in the packet) / (line rate), or in this case 1500 * 8 / 384 = 31.25 ms. If voice packets suffer variability of queuing delay in excess of 30 ms, then the jitter buffer at the receiver has to accommodate at least this amount of delay, which is additive to the packetization delay and other queuing delays in the network. The result is likely to be a total one-way transmission delay in the access network of 50 ms or more, which many voice service providers will regard as unacceptable. The solution to this problem is frame fragmentation, a technique that involves breaking large data packets down into a number of smaller fragments, permitting a much finer level of granularity in the interleaving of voice and data packets on the DSL connection.

QUALITY OF SERVICE REQUIREMENTS FOR VOICE OVER IP

The subject of Quality of Service for IP networks is one that has received a huge amount of attention in recent years. This has led to competing IP QoS solutions being developed. It is not the intention of the MSF to evaluate IP QoS solutions in the context of all possible types of IP traffic. The MSF is currently only concerned with an evaluation of the best mechanisms to solve the problem of supporting a voice service over an IP network. To this end it is important to consider the characteristics of a toll quality voice service, and what this means in terms of any IP QoS mechanism. In general any toll quality voice service requires the following performance.

1. Once a call has been accepted by call control and resources allocated to it the call should be carried to completion with the required voice quality.
2. Established calls must be protected from network disturbances as far as physically possible. One implication of this requirement, when applied to a connectionless IP network, is that stable calls must not be adversely affected by sudden loads caused by the re-routing of traffic from other parts of the network.
3. The network must be capable of supporting very high levels of call setup attempts. Existing narrowband exchanges may support millions of busy hour call attempts and a VoIP network must be able to support comparable volumes.
4. In the event of focused overload, calls that cannot be carried must be rejected without degrading the call carrying capacity of the network. The PSTN and thus any replacement IP network will occasionally be subjected to very high volumes of calls far beyond that which can be carried (TV and radio phone-in competitions or ticket sales for major events are prime drivers for this sort of overload), any resource reservation mechanisms must be able to deal effectively with this type of event.
5. Mechanisms must be available to ensure that emergency calls and high priority calls receive preferential treatment.
6. Call setup latency must be comparable to the existing network. The resource reservation mechanisms chosen must not introduce delays that mean the user notices a worse setup time on a packet network than they would on a traditional TDM network.
7. The network must be secure from denial of service attacks and spoofing. For example, only the call that has been allocated the resource must be able to use it and when the call is released the resource must again be available to the network.
8. Some networks may require the support for call pre-emption. In cases it may be required for a network to de-allocate resources that have been reserved for an existing call and re-allocate them to a new call.

The legacy PSTN network supports all of these requirements today using TDM narrowband switches. Additionally some network operators have migrated their TDM voice platforms onto ATM which is a relatively straight forward evolution because ATM is both connections oriented and rich in quality of service features. Where VoIP is considered, however, the underlying network is very different and it poses a number of challenges to operators wishing to support a toll quality voice service.

QOS SOLUTION FOR VOIP

There are various mechanisms that can be used to provide quality of service for IP networks and it is not possible to consider every solution here. Therefore it is proposed to examine the most likely candidates for solving the VoIP QoS problem specifically the following solutions need to be considered: Integrated Services (Intserv), Differentiated Services (Diffserv) and MPLS Traffic Engineering (MPLS-TE).

INTEGRATED SERVICES (INTSERV)

The integrated services, or Intserv, method of providing quality of service is to use a protocol for explicitly reserving bandwidth on a per flow basis. This protocol is the internet reservation protocol, or RSVP. It is important to distinguish between RSVP itself and Intserv. RSVP is a signalling mechanism that is used to realise the intserv architecture. It is possible to use RSVP for other reasons, one example is RSVP-TE where it is used to facilitate traffic engineering for MPLS networks, and another example is aggregate RSVP that is proposed for realizing dynamic Diffserv service agreements. When used as part of Intserv RSVP provides a method for a user to request a particular quality of service for a session, in effect this reserves the bandwidth throughout the network for the duration of the session. In the case of a voice session the sender of the voice flow (a SIP client) would send an RSVP path message through the network to the user (the intended receiver). Each node along the path identifies that the Path message signifies a new RSVP session and checks its resources before sending on (a possibly modified) path message. Each Intserv capable node along the path is required to store a soft state for the session and RSVP path refreshes must be sent periodically through the network to hold a particular reservation. Once the Path message reaches the user, the traffic parameters contained within the path message are checked and if the user can support such a session, or wishes to modify the session, an RSVP reservation message is sent back through the network to the sender. Since RSVP reservations are uni-directional this process would have to be carried out in two directions for a bidirectional voice circuit to be established.

Although IP networks are connectionless networks, RSVP provides a mechanism to ensure that the reservation message returns by the same route as the path messages, although this route through the network may change over the duration of a session. Each router along the RSVP "route" checks the RSVP reservation message against its available resources and determines whether it can support the reservation request. If it is able to meet the request then the reservation message is sent onwards towards the sender of the data, otherwise an explicit path tear message can be sent clearing the reservation. Once established an Intserv session must be maintained by each router along the path of the session. RSVP Path and Reservation messages must be sent periodically along the path of the session (refresh messages) in order to prevent the soft state timing out in the routers. A given session persists until either it is explicitly torn down or until no refresh messages have been received within a given time period in which case the soft state in the routers times out.

DIFFERENTIATED SERVICES (DIFFSERV)

The Diffserv approach to providing QoS support differs fundamentally from Intserv in that it does not refer to a specific protocol for providing quality of service but rather an architectural framework designed to facilitate QoS. Diffserv proposes that QoS should be provided by the setting and enforcing of policy within a network to provide a set of Service Level Specifications (SLS) between networks (or customers and networks), effectively service level agreements (SLA). The key features of the Diffserv architecture are as follows:

- The network is divided into one or more Diffserv domains.
- Sources and sinks of traffic outside of the Diffserv domain are considered customers and would typically have an appropriate Service Level Specification that defined how much traffic and of what type they could pass into, and receive from the Diffserv domain. It is important to note that these sources may not be individual users but could be an entire network.
- The edge of the diffserv domain is made up of Diffserv boundary routers. A Diffserv boundary router performs traffic classification and traffic conditioning and policing. It must provide functions for admission control, policy enforcement. In general it is the purpose of the Diffserv boundary router to maintain the integrity of the Diffserv network, to enforce service level specifications and to shape and mark traffic for transport across the remainder of the Diffserv domain.
- Unlike Intserv, Diffserv QoS functions are not applied to a single flow from a customer. Diffserv classifies traffic into a series of classes (otherwise known as per hop behaviours) and applies the same treatment to all traffic within a class.
- The core of a diffserv domain is made up of Diffserv core routers. Diffserv core routers are intended to concentrate solely on traffic handling, processing each packet based on how the packet was marked at the Diffserv Boundary. In order to facilitate QoS Diffserv core routers are likely to have a number of traffic queues available corresponding to Diffserv classes.

Diffserv defines a mechanism whereby competing services and levels of traffic priority within a particular service are handled by core routers so as to guarantee the Service Level Specifications associated with each service can be met. Because Diffserv is architecture rather than a complete solution, supplementary elements must be added to the solution in order for it to be suitable for supporting a voice service. A key aspect of this is admission control and one way of providing it is to deploy bandwidth managers within the network.

MPLS TRAFFIC ENGINEERING (MPLS-TE)

MPLS traffic engineering extends the capabilities of MPLS to incorporate quality of service and as such provides a potentially useful tool to a network operator looking to support voice services. MPLS can be used inside a network to setup label switched paths between ingress and egress points in the network; in effect this creates tunnels down which appropriately tagged traffic flows. By assigning a bandwidth to the label switched path on establishment it is possible to ensure that traffic being carried over a label switched path is guaranteed to be delivered to the egress point provided that the total traffic admitted to the label switched path does not exceed the bandwidth allocated to it. This is a useful tool for IP networks carrying voice as it allows what effectively is an aggregate reservation between two points down which many individual flows can be carried without requiring the explicit reservation of resources for each individual flow. Furthermore this aggregate reservation can be varied with time to allow for fluctuating traffic flows in a network and when combined with MPLS fast re-routing it allows for a resilient network to be created where even significant network failures have very limited impact on the traffic being carried by a particular label switched path.

Given the difficulties with respect to scalability and security that any VoIP QoS solution faces and given the currently available tool set for solving such a problem it is possible to draw a number of conclusions.

1. The IETF Intserv architecture is not suitable for the support of large scale VoIP QoS. The high volumes of call attempts that will be required to be supported by any voice network means that the use of Intserv would place an unacceptable burden on the edge and core routers. The MSF recognizes that it may be necessary to interface with access networks that support Intserv and in these cases RSVP should be passed transparently through the MSF network.
2. The Diffserv architecture should be used to provide QoS by deploying Diffserv boundary functions at the edge of the network and providing suitable mechanisms to control the admission of individual flows into the network.

3. Within the core network QoS mechanisms should be provided that guarantee service but that do not require knowledge of individual user flows. There are a number of suitable technologies, of which MPLS-TE is the most promising; however alternative solutions such as aggregated RSVP and ATM may also be applicable in some networks.
4. To enhance scalability and to allow call control functions to be abstracted from the underlying network bandwidth managers should be deployed. Bandwidth managers act as an interface between the call control functions and the network specific bearer functions.
5. In order to support full PSTN equivalence a two stage resource reservation model should be applied with resources being reserved on initial call setup and committed at the point where a bearer is established.

SIMULATION ENVIRONMENT OF VOIP

The reliable methods of performance analysis and prediction are necessary to efficiently manage the current networks and systematically make plans for the future network expansion [1, 2]. That is, the network administrators must be able to design a network by exact modeling and analyze its performance through a verified tool than experiential knowledge. The simulation tool for this has to be easy to use and reflect the patterns and properties of the recent applications. The key characteristics of this particular system are its easy and intuitive usage, the real behaviors implementation of network devices and protocols, the actual generation and transmission of call signals and simulation traffic, the support of VoIP simulations and so forth. One of the recent studies with regard to the implementation of a simulation tool is the NCTUns [4] that features the use of the real-life FreeBSD or Linux's TCP/IP protocol stack, the supports of various protocols and wireless environment, and so forth. But, it cannot directly support VoIP simulations and the interconnection of the Internet and PSTN (Public Switched Telephone Network), which are our contributions and the primary objectives of this study. The OPNET [5] which is the most famous of commercial simulation tools enables the users to set up the detailed configurations with profiles and has the advantage of diverse experiments. The NS-2 [6, 7] which is the most representative of pubic simulators is a tool mainly to verify the reformed protocols and analyze their operations. With the NS-2, the users must modify the developed codes and make out scripts. These tools are complicated to apply and difficult to use because those require the knowledge and experiences on protocol development and programming. As far as IP telephony is concerned, the OPNET and the NS-2 can define the patterns of voice traffic partially and transfer the traffic. But, these existing tools analyze only the quality of voice transmission such as delay and include no details about VoIP-related equipment. On the contrary, the system is equipped with the various devices such as gateways and gatekeepers for the simulations of VoIP. The voice traffic and call signals for voice calls are added to the system. Through these methods, it is possible to analyze not only the quality of voice transmission but also the performance of VoIP devices such as call success rate with the interconnection of the Internet and PSTN.

ARCHITECTURE OF THE SIMULATOR

The system of simulation used in this paper is a tool for the network design and its performance analysis, which additionally supports VoIP simulations. A user builds up the whole network topology by using nodes and links and determines the performance of each device. After the user specifies traffic patterns, the designed network undergoes the auto-configuration procedure. Then, it is simulated and its performance is finally analyzed. This system consists of four managers and several related components. We added VoIP-related functions and components to our existing simulator called NetDAS (Network Design and Analysis System) which had already been implemented in reference [8, 9]. For more detailed information, therefore, refer to reference [8, 9].

CORE MODULES AND IMPLEMENTATION METHODS

This section describes the basic design concepts and the implementation methods about the functions and the core modules composing our system. But, we just describe only the items not presented in reference [8,9].

TIME MANAGEMENT AND SCHEDULING

Time management and scheduling problems on simulation are very important to virtually execute the simulations similarly to the real environment on single system. The concept of virtual time and used ns (nanosecond) as all of the time units was introduced. As for the scheduling problem, the OPNET and the NS-2 have the structure that a whole scheduler processes the jobs one by one based on events. On the contrary, the scheduling of our system conforms to the round-robin mechanism essentially. That is, our system assigns a quantum for the processing time to all objects such as node and link. Let TQ be the amount of time that is assigned to each round and TP be the processing or latency time in a certain object. Then, the maximum of TQ satisfying the condition of $TQ < \min\{TPi\}$ ($i = 1, 2, ..., n$) is designated as the time quantum. The reason is as follows. If TQ is greater than TP, there is no problem for the processing within a same object. But, when the order of processing is turned over to the next object according to the round-robin, the situation that a time limit is expired happens.

ADDRESS ALLOCATION

The OPNET and the NS-2 assign only the logical sequence number. On the other hand, the system analyzes the designed topology and distinguishes the logical networks in advance. And then, the system assigns the explicit 32-bit class B IP addresses and 48-bit physical addresses to all objects. The structures of these addresses are represented in reference [8, 9].

PACKET STRUCTURE AND TRAFFIC GENERATION

The generation patterns of traffic play an important role in simulation because the network performance may vary in accordance with the traffic patterns [10]. Therefore, a simulation system must be able to generate the various types and patterns of traffic with the different packet size, time interval, destination, and so on. This system supports the traffic generation by the various methods as shown in Table 1. Traffic parameters are specified by mathematical probability distribution functions [11], which is similar to the OPNET. While the OPNET and the NS-2 generate the traffic only in the frame layer, this system supports the traffic generation in the application layer as well as in the frame layer with the mixed mode. The NS-2 does not offer the methods for the generation of voice packets. On the other hand, the OPNET can generate the traffic corresponding to the voice information at the end nodes and it is possible to conduct coding. On the contrary, our system also supports the generation of the signal packets to establish and release call connections. And in our system, the establishment of a call connection is accepted or rejected according to the performance of each device such as gateways and gatekeepers. Moreover, differently from other packets, the voice information packets undergo several processes such as media conversion in these devices.

NODES AND PACKET PROCESSING

The end nodes such as host and host group can generate both data packets and voice packets. The general telephone node and the IP phone node to our system are added, whereas the OPNET and the NS-2 do not include these nodes. And we added the gateway and gatekeeper nodes for the VoIP processing, which are not the objects of the OPNET or the NS-2. All of the nodes have the functions of transmitting and receiving the packets on the links. Various protocols were embarked in the processing nodes such as routers and switches. Router nodes exchange the routing information dynamically and update the related tables. The routers also classify incoming packets into data packets, call signal packets, voice information packets, routing information packets, etc. and handle those classified packets individually according to the types.

This system keeps the records of the histories about all of the packet flows. The method and log information are similar to the OPNET and the NS-2 [6-11]. For example, reception time, processing time, transmission time, and input/output interfaces per packet are recorded. Furthermore, throughput, memory utilization, packet loss information and so forth are all saved as logs for the performance of nodes. Some processing nodes such as routers and switches must be able to provide the processing power in addition to the memory utilization and latency time as the analysis results. For this performance, a unit of PPS (packet per second) is used, which is calculated as follows:-

PPS = 1s / (Inter Frame Gap + Preamble Time + Frame Time). **(1)**

The original capacity is influenced by the factors such as the bus and backplane of equipment. In our system, the performance analysis is achieved at the end of the simulation after recording the processing states. The end-to-end delay time is calculated as follows by the sum of the elapsed times on all of the nodes and links which the packets go through.

E2E Delay = Serialization Delay + Propagation Delay + Switching Delay. **(2)**

The delay times except for the switching delay are the fixed values or directly (or inversely) proportional to the specific factors. So, it is important to reduce the switching delay time. This time is influenced by the network topology, the performance of the used devices, and so forth. IP telephony is one of the Internet applications and a technology for delivering the voice traffic by using the Internet and IP. For providing this service, specific devices and related technologies are required. In this paper, we implemented a simulator that can analyze the performance of VoIP. This section describes the detailed methods for supporting the VoIP technology in our simulation system.

VOIP SIMULATION RESULTS

The NS-2 is used to develop and test new protocols or mechanisms. By reason of this point, as for the purpose and the application areas, the NS-2 differs from our system. Especially, if the user does not modify the source codes as he or she wants, it can not distinguish the types of nodes and its user can not define the voice traffic patterns. Added to that, it can not simulate the performance of the VoIP-related devices in the same way as the behaviors of the real-life equipment. The OPNET can measure the performance of the general-purpose network devices differently from the NS-2. The end nodes of the OPNET can define the voice traffic and transmit those packets to the network. The voice traffic, however, is treated in the same way as the general data traffic in the processing nodes such as routers and switches. Traffic can be distinguished by the QoS profiles based on TOS (Type of Service) values, protocol types, or port numbers. The processing nodes just process the packets according to these values and queue scheduling types. Also, this tool does not support the interconnection of the recent Internet and the legacy PSTN. And the OPNET supports only SIP (Session Initiation Protocol) [12, 13] as a signal protocol for the connection establishment. That is, the OPNET does not include the recent VoIP gateway and gatekeeper product models, so this can not take measurements of their performance [14]. The OPNET can make simulations only for the voice traffic within IP networks. As results of the analysis about the voice traffic transmission, the amount of the transmitted/received packets, end-to-end delay, delay variation, and so forth are provided. Besides, as results of the analysis about SIP, the number of accepted/ rejected calls, the number of blocked calls, call duration time, and so forth are presented. As mentioned above, the existing tools can not be supporting VoIP completely. Therefore, through the interconnection with PSTN, we intend to model the voice calls variously and analyze the performance of VoIP-related devices. The details for our aim are as below. The nodes of the system for the hosts to generate both general data packets and voice traffic are designed, and for the IP phones and general telephones to generate only voice traffic. In this section, we deal with only the voice calls with the various types of traffic.

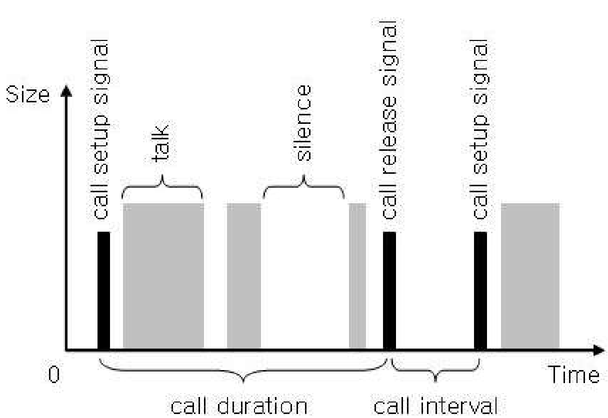

Fig. 3: Generation of call and voice

For the voice calls, nodes must be able to generate the signal for the call establishment, the actual voice traffic, and the signal for the call release individually. Fig. 3 shows briefly the relation between call signals and voice traffic. Calls are established or released by signaling protocols. A call session lasts between a call setup signal and a call release signal. And a call session consists of talk time and silence time. Another call can be tried after the random time elapses. So, through this system, it is possible to specify the parameters related to the call signals, the voice traffic, and their time intervals by using probability distribution functions [15-17].

The measurements are based on monitoring of RTP packets transmitted through different network scenarios. The figures below show the results. The X-axis of these graphs is the time in seconds. The left Y-axis shows the delay (gray) and the right Yaxis indicates jitter (black), both in milliseconds [18-20]. As we have used the normal internet, obtained results are affected by other network traffic at the time of observation. The results displayed here were selected as typical of each network scenario. In the LAN (Fig. 4), delay ranges from 3 to 40 ms. however some spikes observed during the session.

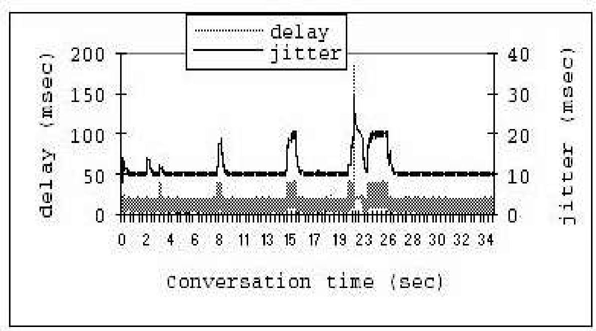

Fig. 4: Observed in LAN

CONCLUDED REMARKS AND FUTURE WORKS

Providing reliable, high-quality voice communications over a network designed for data communications is a complex engineering challenge. Factors involved in designing a highquality VoIP system include the choice of codec and call signalling protocol. There are also engineering tradeoffs between delay and efficiency of bandwidth utilization [21-24]. Voice over Internet Protocol (VoIP) is a rapidly developing technology. Most cable, broadband and phone service providers are planning to start adding Internet telephony service to their standard packages. Despite hardware constraints, the operators have to tackle the problem of providing voice quality over the current Internet. VoIP is a time sensitive application and requires real-time support for its quality of service (QoS) requirements. The traditional Internet, which uses a best-effort mechanism, fails to support the QoS requirement of most multimedia application like VoIP. Differentiated Service (Diffserv) is a scheme designed to support multimedia QoS requirements in a scalable manner [25-27]. Two per-hop-behaviours (PHB): Expedited forwarding (EF) and Assured Forwarding (AF) have been defined for Diffserv. They are designed to provide low loss, low latency end-to-end service and assured bandwidth service respectively. In addition, AF is capable of being configured as a low latency service. In this thesis, simulations of VoIP using RED and Diffserv were carried out which gave some indication of the potential importance of DiffServ.

There is a wide range of possibilities for future research in this area. It would be interesting to look at the performance of data traffic in our VOIP architecture. This would involve a study of TCP's ability to utilize the residual bandwidth (bandwidth unused by voice traffic) in the link. Research could be undertaken into an Adaptive variable codec which could respond to the variation of the traffic flow. This will improve the QoS as well as allowing user to experience better quality when network conditions do allow. Future research could be done on the automated DiffServ design and parameter setup so that DiffServ can be implemented from anywhere in the internet. DiffServ needs adaptive parameter configuration in order to achieve its potential.

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