Quality of Service (QoS) in the New Public Network Architecture

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Abstract

This article provides an overview of Quality of Service in the new public network architecture that is committed to replace the traditional IP network to include multimedia services. Focus is on the different mechanisms and models available and the important aspect of end-to-end implementation of quality of service in the new public network domain. Quality of Service based on different service levels is considered in every side of the network - the user, the backbone network access, and the IP core network.

Sommaire

Cet article présente un survol de la qualité du service dans la nouvelle architecture de réseau qui est dédiée à remplacer le réseau IP traditionnel afin d'inclure les services multimédias. L'attention est portée sur les différents mécanismes et modèles disponibles et l'aspect important de l'implantation complète de la qualité du service dans le nouveau domaine des réseaux publics. La qualité du service basée sur des niveaux différents de service est considérée de tous les angles-l'utlisateur, l'accès à la structure du réseau et le réseau IP central.

1.0 Introduction

In recent years, the most active area in networking is - data, voice, and video integration. Business users are beginning to combine real-time applications such as voice and video, which have a limited tolerance for network latency, with non-real time data traffic. With Voice over IP (VoIP) technology - defined as the ability to make telephone calls (real-time voice) over IP-based data networks with a suitable QoS and a much superior cost benefit - systems can provide simultaneous voice and Internet access over the same connection, or integrate existing phone connections with the Internet through VoIP Gateways.

1.1 What is Quality of Service (QoS)?

QoS refers to the capability of a network to provide better service to selected network traffic over various technologies including Ethernet and 802.1 networks, Wireless networks, IP-routed networks, Asynchronous Transfer Mode (ATM), and Frame Relay (FR) that may use any or all of these underlying technologies. It can also be interpreted as method to provide preferential treatment to some arbitrary amount of network traffic, as opposed to all traffic being treated as “best effort”.

1.2 Factors affecting Quality of Service

Following factors can profoundly impact the quality of service:

a) Delay: Echo and talker overlap are the problems that result from high end-to-end delay in a voice network. Round trip delay should be less than 50 ms to avoid echo. Since VoIP has longer delays, such systems must address the need for echo control and implement some means of echo cancellation. The ITU recommendation G.168 defines the performance requirements that are currently required for echo cancellers. Talker overlap (problem of one caller stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 ms. Delay can be attributed to - accumulation delay, processing delay and network delay. The choice of a fast CODEC like the G.729 CS-ACELP takes care of the accumulation and processing. Network delay describes the average length of time a packet traverses in a network. The network delay is handled by a good network design that minimizes the number of hops encountered and by the advent of faster switching devices like Layer 3 switches, tag switching system like MPLS (Multi Protocol Label Switching) systems and ATM switches.

b) Jitter (Delay Variability): This is the variation in the inter-packet arrival time (leading to gaps, known as jitter, between packets) as introduced by the variable transmission delay over the network. Removing jitter requires collecting packets in buffers and holding them long enough to allow the slowest packets to arrive in time to be played in correct sequence. Jitter buffers cause additional delay, which is used to remove the packet delay variation as each packet transits the network.

c) Packet Loss and Out of Order Packets: IP networks do not guarantee delivery of packets, much less in order. Packets will be dropped under peak loads and during periods of congestion. Approaches used to compensate for packet loss include interpolation of speech by re-playing the last packet, and sending of redundant information. Out of order packets are treated as lost and replayed by their predecessors. When the late packet finally arrives, it is discarded.

d) Bandwidth available: Maximal data transfer rate that can be sustained between two end points affects service quality. Techniques used to minimize congestion loss in the network may reduce the available bandwidth for an application. With current advancements in transmission media technologies, plentiful capacity is a reasonable assumption for a controlled, localized environment, such as a corporate LAN, but it is currently unrealistic across a global network such as the Internet.

2.0 Different Service Levels

Service levels refer to the actual QoS capabilities, meaning the ability of a network to deliver service needed by a specific network application from end-to-end. This can also include edge-to-edge, as in the case of a network that connects other networks rather than hosts or end systems, (the typical service provider network, for example), with some level of control over bandwidth, jitter, delay, and loss, provided by the network.

Essentially, QoS can provide three levels of strictness from end-to-end or edge-to-edge: best effort, differentiated, and guaranteed.

2.1 Best-Effort Service

Also known as lack of QoS, best-effort service is basic connectivity with no priorities or guarantees. It provides basic queuing during congestion with first-in, first-out (FIFO) packet delivery on the link. Examples of this type of traffic include a wide range of networked applications such as low-priority e-mail and general file transfers.

2.2 Differentiated Service

Also called “qualitative QoS / Soft QoS”, differentiated services treats some traffic better than the rest (faster handling, more bandwidth on average, lower loss rate on average), however, there is no hard and fast guarantee. With proper engineering, differentiated service can provide expedited handling appropriate for a wide class of applications, including lower delay for mission-critical interactive applications, packet voice applications, and so on. Typically, differentiated service is associated with a course level of packet classification, which means that traffic gets grouped or aggregated into a small number of classes, with each class receiving a particular QoS in the network.

2.3 Guaranteed Service

Also called "quantitative QoS / Hard QoS", guaranteed service is an absolute reservation of network resources, typically bandwidth, which implies reservation of buffer space along with the appropriate queuing disciplines, and so on, to ensure that specific traffic gets a specific service level. This type of service is for delay-sensitive traffic, such as voice and video. The Guaranteed Service level is intended for applications requiring a fixed delay.

3.0 QoS mechanisms

Several mechanisms have been proposed to support real-time and multimedia traffic at different layers of networking.

3.1 Data Link layer

At this layer (Layer 2) media access control needs to be modified to provide service differentiation so that QoS guarantees can be supported. Asynchronous Transfer Mode (ATM) is associated with wide area network (WAN) and in the local area network (LAN), Frame Relay (FR) in the WAN and IEEE 802 style in the LAN media.

ATM: Constant Bit Rate (CBR) and Variable Bit Rate (VBR) are best suited for telephony and voice applications, and for multimedia applications such as video. Available Bit Rate (ABR) and Unspecified Bit Rate (UBR) are designed for best-effort delay-insensitive traffic such as file transfers and e-mail.

FR: attempts to provide a simple mechanism for arbitration of network over subscription. Committed Information Rate (CIR) confirms to the commitment on the part of the network to provide network delivery.

IEEE 802.1: 802.1p specification provides a method to allow preferential queuing and access to media resources by traffic class, on the basis of a "priority" value signaled in the frame. This value will provide across the sub-network a consistent method for Ethernet, token ring, or other MAC-layer media types. The priority field is defined as a 3-bit value, resulting in a range of values between 0 and 7, with 0 assigned as the lowest priority and 7 indicating the highest priority. Packets may then be queued based on their relative priority values.

3.2 Network layer

At this layer (Layer 3) too real-time services should be distinguished from non real-time services.

IP precedence utilizes the three precedence bits in the IPv4 header's Type of Service (ToS) field to specify class of service for each packet. These bits may be assigned by an application or a user, or by destination and source subnet, and so on. Typically this functionality is deployed as close to the edge of the network as possible, so that each subsequent network element can provide service based on the determined policy.

Packet marking: The ingress router must mark the packets as they enter the network with appropriate values so that interior routers can handle packets differentially. The marking of the IPv4 packets use the ToS octet.

Packet classification: Routers must check all received packets to determine if the packets should receive differential treatment. The traffic can be policed and shaped to the network in order to maximize the probability that the traffic will meet the service required and receive the desired quality of service.

Packet queuing: In interior routers, packets must be handled differently. The routers may employ multiple queues along with some scheduling disciplines such that delay-sensitive traffic will be serviced sooner.

- **FIFO queuing:** In a traditional IP router, first-come first-serve is the scheduling policy used. This is a fair algorithm and same delay is imposed on all queued packets. It is necessary to alter this fairness and introduce mechanisms such that preferential treatment may be given to differentiated classes of traffic.
- **Priority queuing:** There is a queue for each distinct priority levels and queues are serviced in order of priority. Highest priority traffic receives minimal delay but lower priority queues may be prevented from being serviced leading to their starvation. This simple priority mechanism must be used with some other mechanism to police traffic into the queues.
- **Weighted Round-Robin:** Queues are serviced round-robin in proportion to a weight assigned for each queue. The assigned weight is normalized by dividing it by the average packet size for each queue. Normally, at least one packet is transmitted from each non-empty queue in every round.
- **Deficit Weighted Round-Robin:** Each non-empty queue has a deficit counter that begins at zero. The scheduler reads the packet at the head of each non-empty queue and tries to serve one quantum of data. A packet is served if the counter is zero and if it is less than or equal to the quantum size. If the packet cannot be served, then the value of the quantum is added to the deficit counter for that queue.
- **Weighted Fair queuing:** It schedules interactive traffic to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth among high-bandwidth flows. It ensures that queues do not starve for bandwidth, and that traffic gets predictable service.

Figure 1 shows this classification and queuing of packets to provide differential treatment.

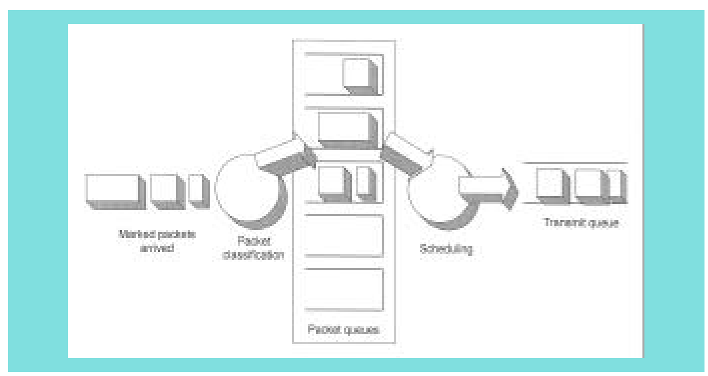

Figure 1: Packet classification and queuing

3.3 Transport and Application Layer

Packets may be marked and classified by transport layer and application layer. Routers could use port numbers, however, they will have to locate the transport-level header that might be behind the optional IP header. By adding application-specific information to packet payloads, the routers need to know the many application-level protocols.

The transport and application levels must however provide new functionality to support real-time applications. The real-time transport protocol (RTP) is the standard for real-time data transmission on an IP-based network. RTP provides no QoS capability but implements specific framing for real-time media, such as sequence numbers and time stamps, to the user datagram protocol (UDP).

4.0 DiffServ and IntServ Models

The main architectures and techniques defined for IP QoS are:

- The Integrated Services: IntServ
- The Differentiated Services: DiffServ

4.1 IntServ Model

The Integrated Services architecture for the Internet was proposed in RFC 1633 to support real-time traffic as well as "best-effort" traffic. It is founded on reservations-based traffic engineering, where resources for traffic are explicitly identified and reserved. Network nodes classify incoming packets and use reservations to provide differentiated services. It performs resource reservation using a dynamic signaling protocol and employs admission control, packet classification, and intelligent scheduling to achieve desired QoS. This model is relatively complex and has difficulties in scaling to large backbones. This architecture is based on the Resource Reservation Protocol (RSVP).

RSVP is an IETF Internet Standard (RFC 2205) protocol for allowing an application to dynamically reserve network bandwidth. It enables applications to request a specific QoS for a data flow. Hosts and routers use RSVP to deliver QoS requests to the routers along the path, and to maintain router and host state to provide the requested service, usually bandwidth and latency. Bandwidth reservation is based on mean data rate, the largest amount of data the router will keep in its queue, and the minimum QoS required.

The specific standards and definitions for services developed by the Integrated Services (IntServ) working group in the IETF fall under Guaranteed QoS. Technologies that can provide guaranteed service for portions of the end-to-end connection include:

- IP-WFQ combined with RSVP signaling or guaranteeing bandwidth on a single link,
- Ethernet- Subnet Bandwidth Manager (SBM) (when used with a compliant switch),
- ATM-Variable Bit Rate (VBR) and Constant Bit Rate (CBR), and
- FR- Committed Information Rate (CIR).

4.2 DiffServ Model

The DiffServ working group in the IETF is working on specific standards and definitions for services that fall under Differentiated QoS. It is looking at a more scalable model and more likely to be easier to implement than IntServ/RSVP model for identifying flows. It is based on traffic aggregation rather than individual per-application instance flows. The DiffServ model largely focuses on the use of the ToS field in IPv4 header or the IPv6 Traffic Class octet as a QoS mechanism. These bits are used to mark a packet to receive a particular forwarding treatment, or per-hop behavior, at each network node. Classification, marking and policing are done at the network edges and only packet handling requirements need to be provided in the core of the network.

Technologies that can provide differentiated service for portions of the end-to-end connection include:

- IP-WRED, WFQ, combined with IP Precedence signaling or prioritizing traffic on a single link
- ATM-Unspecified Bit Rate (UBR) and Available Bit Rate (ABR), especially if no Minimum Cell Rate (MCR) can be specified in the implementation
- Frame prioritization in campus switches in conjunction with 802.1p signaling.

5.0 End-to-End Implementation

In order to provide end-to-end QoS in the new public network architecture, it requires that every element in the network path - router, switch, firewall, host, client, etc. - deliver its part of QoS (Figure 2). The Service Providers should therefore ensure that QoS elements are available throughout in the Intranet/Internet, or some other mechanisms such as reserving the bandwidth are available to support QoS in the network.

On the host side, an access device may receive voice packets and/or data packets that are differentiated based on the ports on which they are received. For packets arriving at the voice port, its related service priority is provided in the Layer 2 and/or Layer 3 header by the access device. For packets arriving at the data port, best effort service is provided if not specified in its Layer2/Layer3 header. In case of same port used for both voice and data packets, differentiation is based on Layer 2 header (IEEE 802.1p), and/or Layer 3 header (IP ToS) fields. Prioritization field is added to the Ethernet packets by most of the recent terminal

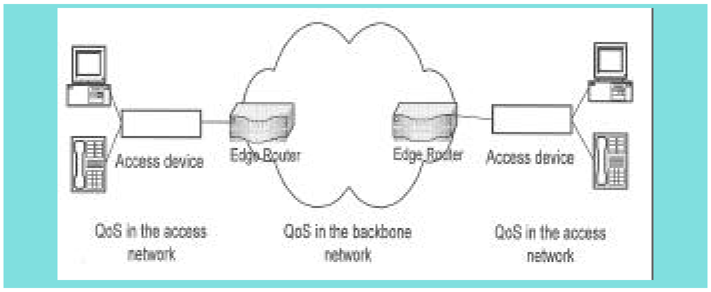

Figure 2: End-to-end QoS

equipments providing voice traffic, to differentiate voice from data on the same port.

Router switches that can forward packets and apply traffic conditioning at wire speeds are essential to provide QoS delivery in the IP backbone network. The presence of legacy routers will potentially limit service offerings and the QoS level will default to the capability of the lowest performing router. Routers should therefore be quality aware and be able to classify delay-sensitive traffic from non real-time traffic. They must be configured to handle packets based on their IP precedence level, or similar semantics expressed by the bit values defined in the IP packet header. Any priority scheme that was used at Layer 2 should be mapped to a particular IP precedence value.

It has been observed that higher-layer protocols, such as TCP/IP, provide the end-to-end transportation service in most cases. Although it is possible to create QoS services in the lower layer of other protocol stacks (for example, ATM), such services may cover only part of the end-to-end data path. It is therefore not sufficient to have a lossless ATM subnetwork from the end-to-end performance point of view. In addition to a large ATM cell header overhead, the disadvantage of using ATM networks would be to still use routers at the boundary of the network, and to maintain two sets of configurations: one for routers and the other for ATM switches. One of the possible solutions would then be to confine legacy routers to the best-effort traffic only and the QoS-sensitive traffic to send over the ATM network (Figure 3).

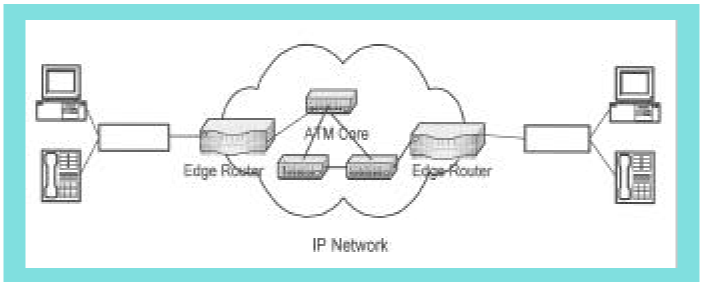

Figure 3: Delay-sensitive traffic channeled to ATM core

Another solution would be the use of Multi Protocol Label Switching (MPLS) with Differentiated Services (DS) whereby which router networks can also provide QoS and Traffic Engineering. The Service Provider must have a Service Level Agreement (SLA) with the customers specifying the service levels supported. The access network marks the DS fields of individual packets to indicate the desired service and the edge routers classify, police and possible shape the incoming packets based on the SLAs using the First-in First-out queuing, Weighted-Fair queuing, Priority queuing or other queuing mechanisms. To support interactive traffic, the router should also be able to support fragmentation of large datagrams and interleaving of delay-sensitive packets with the resulting smaller packets.

MPLS is a forwarding technique that offers simpler mechanisms for packet-oriented traffic engineering allowing Service Providers to deliver new services that are not readily supported by conventional IP routing techniques. It provides a solution that describes the integration of Layer 2 switching and Layer 3 routing with a decreased complexity of mapping between the two distinct architectures. It therefore allows networks to be built using an overlay model in which a Layer 3 IP runs over and is independent of an underlying Layer 2 switched topology. MPLS can be used together with Differentiated Services to provide QoS. In such a architecture, a Label Switched Path (LSP) is first configured that is followed by all packets between each ingress-egress pair. The core routers process the packet based on its labels and class of service field. With such schemes, MPLS effect is confined within the service providers that use MPLS. Its effect is transparent to other Service Providers. Figure 4 shows the position of MPLS in the new public network.

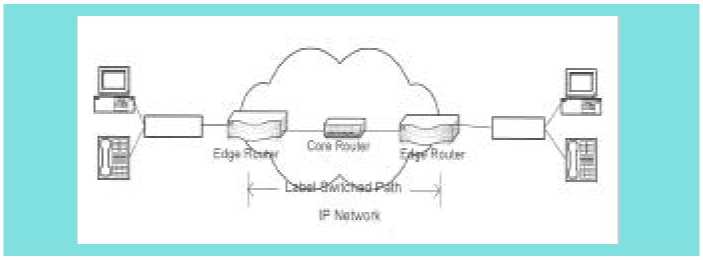

Figure 4: MPLS based Service Provider Network

6.0 Conclusions

With Internet usage doubling each year, more and more companies have started to develop products for Internet Telephony and other real-time applications. But the basic problem still remains of providing QoS to such applications in the global Internet. This paper presents the different QoS mechanisms available that can support customer's different service level agreements. This paper also proposes architecture to implement end-to-end QoS. Both access network and IP network should recognize and treat packets belonging to real-time traffic with priority. This involves marking such packets, classifying the packets based on the markings so that they are given differential treatment, and allowing the scheduling mechanisms to transmit the packets in a timely manner.

7.0 References

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About the author

Anjali Agarwal received her Ph.D. from Concordia University, Montreal in 1996. She is an Assistant Professor at Concordia University since June 1999.

Prior to that she worked as a Protocol Design Engineer and a Software Engineer in industry, where she was involved in providing specifications and design issues for TCP/IP and VoIP support, and in the software development life cycle of real time embedded software. Her current research interests are in various aspects of real-time and multimedia communication over Internet and over the access network. Most recently, she has been working on IP telephony for Broadband Wireless Access networks.

Letters to the Editor / *Lettres envoyées au rédacteur*

Summer issue of the Review

Nice issue as usual. Since Queen's Physics lives in Stirling Hall, opened by John B Stirling, I do have to “twit” you mildly about the spelling on page 13!

Dr. Howard J Wintle,

Thanks for Canadian Review

Congratulations and many thanks for the Summer 2000 issue of the Review. Enjoyed all articles cover to cover, and been informed too.

Gordon Chen LS

My complements to you.

The IEEE-USA Editorial Board was very impressed with our Canadian Review publication. Their primary comment was “Why cannot they produce a publication as good as ours”!!!!

Terry Malkinson, U of Calgary

IEEE Vancouver Section Millennium Awards - Addendum

Missing from the list of winners of the IEEE Millennium Awards for the Vancouver Section was **Roger K. Nelson** (see CR35, page 29).

Nick Keenan, IEEE Vancouver Section

Surfing the net

An architect, an artist and an engineer were discussing whether it was better to spend time with the wife or a mistress. The architect said he enjoyed time with his wife, building a solid foundation for an enduring relationship. The artist said he enjoyed time with his mistress, because of the passion and mystery he found there. The engineer said, “I like both.” “Both?” Engineer: “Yeah. If you have a wife and a mistress, they will each assume you are spending time with the other woman, and you can then surf the net.”

Bob McLoud, Markham, ON

Fully Digital Real-Time Simulation

With regards to the article “Fully Digital Real-Time Simulation” appearing in Issue No. 34 of the IEEE Canadian Review, we congratulate TransÉnergie Technologies and École de technolgie supérieure (ÉTS) on their accomplishment. We discovered in 1994, while implementing our first “big” system, that achieving real time for large scale power system simulations is no small task.

As it may not be known to the authors of the above mentioned article, we wanted to provide some information about our most recent large scale simulator which is about to be shipped to the Korean Electric Power Corporation (KEPCO). The RTDS® Simulator purchased by KEPCO has successfully represented a power system including 160 buses, 41 generators, 131 single and twin (counted as one line) circuit lines, 78 transformers, and more than 60 dynamic load models. Continuous, real time operation was achieved for the system with a simulation timestep of 50 microseconds. Since only 60% of the available processors were used in the simulation, it is expected that a power system with as many as 250 or more buses could be represented by KEPCO's simulator. The Simulator and its application by KEPCO were described in a paper titled “Overview of the Development and Installation of KEPCO Enhanced Power System Simulator” presented at ICDS '99, an IEEE sponsored conference on real time digital simulators.

For any information, you can contact us by phone at (204) 989-9700 or by email at rtds@rtds.com.

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