IP Broadcasting over Unidirectional Satellite Link

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Abstract

We consider sharing a unidirectional satellite link between bandwidth hungry applications typically found in broadcasting environment, namely, time sensitive real-time streaming and bulk reliable multicasting. With unidirectional satellite environment, there is a long latency delay but high bandwidth on the downstream link while return paths could be with low bandwidth. Though reliable multicast is not time sensitive but delays on the transmission can cause congestions or ACK implosion on the return path. To ensure data reliability and congestion control, our reliable multicast protocol on unidirectional satellite link (RMUS) monitors the current condition of the network and adjusts the transmission rate accordingly. With a scheme to provide dynamic QoS configuration or D-QoS, high quality real-time broadcasting can occupy the entire transmission channel to ensure minimum delay, loss or jitter causing packet loss on other flows. The co-existence of these two types of applications on a satellite link is typical due to the broadcasting characteristic of the satellite. In this paper, we propose a scheme where RMUS can co-exist with D-QoS without disrupting the operation of the link nor effecting end results for respective applications running on the same link.

Keywords : Reliable Multicast, Unidirectional Link, Satellite Internet, Dynamic QoS

1. Introduction

IP multicast over the satellite link provides the broadcasting capability on the Internet for mass or bulky information delivery. In general, IP multicast is a network-layer Internet group communication with point-to-multipoint and multipoint-to-multipoint communication which provides an unreliable datagram multicast service with no guarantee whether a given packet can reach all the intended recipients. This characteristic does not pose any problem for real-time applications, such as teleconferencing and video streaming, which are more concerned with time than reliability; it does, however, create problems for applications such as software or massive text file distribution. Mixing time sensitive streaming with massive content distribution applications on the satellite link poses a challenging bandwidth allocation problem for IP broadcast environment.

In this paper, we propose a time-sharing scheme to mix these two groups of flows so that both classes of broadcasting packets can be delivered efficiently while satisfying both the time and reliability constraints.

2. Unidirectional Satellite Link

The unidirectional satellite link architecture comprises 2 types of stations: the feed station which can only send datagrams through the Unidirectional Link (UDL), and the receiver which receives datagrams through the UDL.

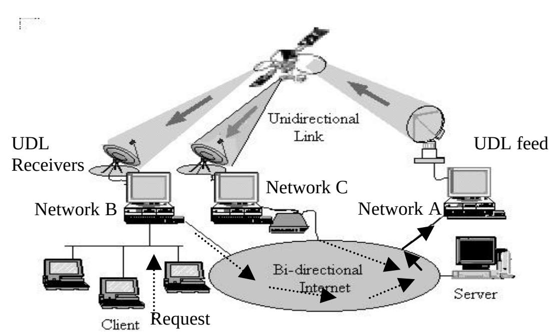

Figure 1. The architecture of unidirectional satellite link.

Figure 1 illustrates a UDL environment where there is a server on the upstream network, network A, sends packets through the feed. These packets are broadcasted to all receivers which are then forwarded to the clients, who are members of the communication group, on the receivers' downstream networks, network B and network C. The request or acknowledgment packets from the network B and C are sent back to the receiver station which forward them to the feed through terrestrial Internet connection. The feed then forwards the acknowledgement packets to the server on its network.

The UDL architecture is clearly asymmetric. The unidirectional downstream satellite link is normally of high bandwidth but with a long delay due to satellite latency while the return paths may vary; it might be regular Internet, dial-up modem connection or another satellite uplink. In general, the bandwidth of the return path is relatively small as compared to the bandwidth of the downstream link. Both upstream and downstream links are asymmetric in many respects, bandwidth, delay, and error rate, affecting the reliability of the multicast transmission.

3. D-Qos for time sensitive streaming

Time related attributes can be considered as crucial problems to applications with stronger timing requirements such as internet telephony, telemedicine and audio/video conferencing. In [3], a scheme called dynamic QoS (D-QoS) was introduced to ensure dedicated bandwidth allocation by reconfiguring the network to handle such high quality of service flow. D-QoS model allows the QoS requirements be reconfigured dynamically. Users can request a network interruption to guarantee its own smooth traffic flow at the expense of possibly interruptions or blockages of other lower priority traffic flows. Based on the concept of active IP network, interruption mechanisms can be triggered by sending an active packet requesting for an interruption with specified interruption level to the network. Under normal circumstances, D-QoS model adopts Differentiated Services (DiffServ) based on Class-Based Queueing (CBQ) with Random Early Detection with In and Out (RIO) and, in case of a prioritized flow interruption, Priority Queue (PQ) is employed. The two queueing mechanisms operate alternately in response to the programs or data sent via active packets. With D-QoS, upon receiving a super user flow request, an interruption is generated on all the receiving nodes and the normal DiffServ service is then suspended where the network output queueing mechanism on nodes is automatically reconfigured. The highest priority flow could thus be the sole occupier of the link depending on its bandwidth requirement and the link capacity. Normal DiffServ will resume after the super user flow has completed. This allows high priority telemedicine traffic to flow through with the highest possible bandwidth at the expenses of other lower priority traffic. Figure 2 shows an example of network interruption of a 10 Mbps telemedicine flow. During the interruption period, the telemedicine flow is given the highest priority and the highest bandwidth allocation, while other two flow types share the rest of the link bandwidth.

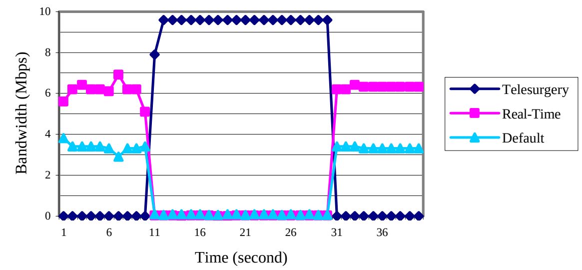

Figure 2. Bandwidth Sharing in D-QoS with the Interruption of Telesurgery Flow.

Since the highest priority flow occupy the link for sometime, other flows may be suspended due to a large number of packet loss while paving the way for the highest priority traffic. To deploy D-QoS in a fragile UDLR environment, this would quickly create a large number of NACK causing congestions on the narrow return paths due to the ACK/NACK implosion and consequently network failure.

4. Reliable Multicast on Unidirectional Satellite Link (RMUS)

In [2], a reliable multicast protocol for unidirectional satellite (RMUS) was proposed. Since the satellite UDL has large RTT, adjustments of the transmission rate within each RTT help make effective use of the available downstream bandwidth and also avoid congestion on the downstream link. If the transmission rate is too high, it could cause congestion and packet loss on the downstream link. If the transmission rate within one RTT is too low, the protocol is not utilizing the available bandwidth efficiently and taking longer time to deliver information through the link. Therefore, the multicast transport protocol should be able to dynamically adjust the transmission rates appropriately so as to keep up with the current network conditions.

RMUS congestion control manages the data transmission rates so that the available bandwidth is utilized efficiently with the presence of other traffic without causing network congestion. It monitors the status of the network by dividing the data transmission into a number of sessions, called monitoring region as shown in figure 3. The interval of each RMUS monitoring region, T_{MON}, is fixed at (1).

$$T_{MON} = T_{SEND} + RTT_{UDL} + T_{BACKOFF} \quad (1)$$

T_{MON} : Time interval of the monitoring region

T_{SEND} : Time sender takes to feed all intended packets
RTT_{UDL} : Round trip time of the unidirectional link
$T_{BACKOFF}$: Backoff timeout to prevent ACK implosion

At the beginning of each region, the sender sends a number of data packets to the multicast group and waits for the reports of reception results. RMUS adjusts the transmission rate by changing the amount of data sent within each monitoring region, instead of changing network monitoring period. When congestion occurs, the transport protocol needs to adjust the transmission rate. The transmission rate is gradually decremented until it finds the rate at which it can share the link with other traffics without causing congestion.

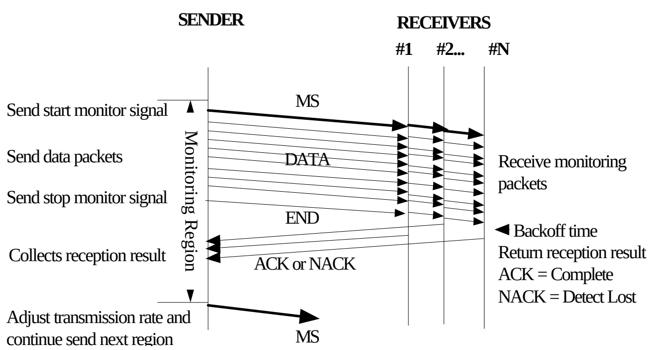

Figure 3. Activities of RMUS in each $T_{MON.}$

5. D-QoS and RMUS

With D-QoS, we allow top priority flow to occupy the downstream link of the UDL while reliable multicast traffic of RMUS would be put to a complete halt. We thus examine if we could relax the interrupt condition of D-QoS sufficiently enough for RMUS traffic to survive the bandwidth starvation. Since the transmission of packets of RMUS occurs at the beginning of a T_{MON}, we consider interleaving the RMUS traffic with D-QoS such that D-QoS would release the link for a fraction of the T_{MON} sufficiently for RMUS to transmit its data packets, as shown in Figure 4.

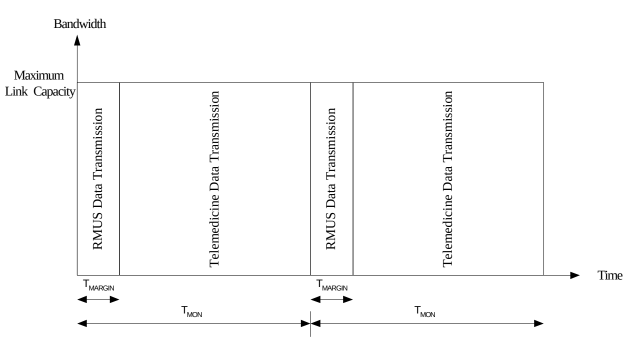

Figure 4. Interleaving of RMUS and D-QoS.

Unlike DiffServ, our scheme relies on time-sharing instead of bandwidth sharing. A T_{MON} is shared by RMUS traffic and D-QoS such that RMUS traffic will not be completely pushed out.

Let ΔT be the time taken for a packet to travel from the source to the destination on the satellite link and ΔT_{MAX} is the maximum time difference between the sending and the receiving ends of a streaming application which is acceptable for the users. For example, for a telesurgery application, the acceptable ΔT_{MAX} is 330 msec [1].

$$T_{MARGIN} = \Delta T_{MAX} - \Delta T \qquad (2)$$

T_{MARGIN} is the time within which RMUS can steal from D-QoS within a single T_{MON}. Thus RMUS, can adjust its transmission rates such that it sends $B * T_{MARGIN}$ data per T_{MON} where B is the bandwidth.

Figure 5 displays the data size of RMUS that can be transmitted within a T_{MON} under the interleaving scheme, when the interruption for full link capacity is made for each of the three time-critical applications. The ΔT_{MAX} for different types of applications[1,5,6] are shown in Table 1. The ΔT used here is an approximation of satellite propagation delay, 250 msec[4].

Application	Telesurgery	VoIP	Video Conferencing
ΔT_{MAX} (msec)	330	400	500

Table 1. ΔT_{MAX} for each application

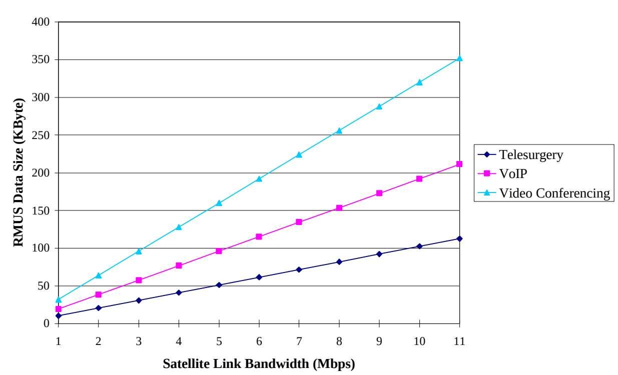

Figure 5. RMUS data size in each $T_{MON.}$

Figure 6 illustrates the transmission time required when sharing a satellite link with two broadcasting flows using DiffServ and our proposed time-sharing or interleaving scheme with T_{MON} value equals to 850 msec. One flow is a telesurgery of 7 Mbps as used in [1] and another is an RMUS flow with a 10 MB file transfer. With DiffServ, the first part of the graph (1-9.33 Mbps), there is only the file transmission flow on RMUS since the bandwidth is inadequate for the telesurgery flow.

The multicast file transfer on RMUS can be allocated the entire link capacity as it is the only flow on the link. Similarly, on our proposed scheme, RMUS traffic can occupy the entire link when it is not feasible to fit telesurgery traffic in.

For DiffServ, telesurgery flow can be transmitted at the point where the link capacity is 9.33 Mbps, the file transfer acquires 25% of the capacity. Lower link capacity is considered inadequate to support the transmission of the 7 Mbps flow without significant packet loss. With D-QoS and RMUS, a 7 Mbps telesurgery flow can be accommodated as from a link bandwidth as small as 7.73 Mbps with no data loss. The file transfer over RMUS with telesurgery would take about eight times longer to complete than without telesurgery when the link bandwidth is tight (7.73-9.33 Mbps). As for DiffServ, file transfer over RMUS with telesurgery can start operating when the link capacity is at least 9.33 Mbps. As from that point on, the file transform performance from either one of the schemes is the same. Thus the interleaving scheme allows the telesurgery and the file transfer flow to share the satellite link smoothly even when the link bandwidth is insufficient for DiffServ to operate.

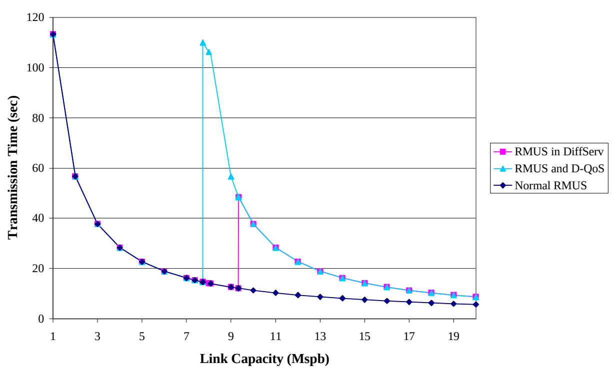

Figure 6. Transmission time for a 10 MB file

6. Conclusion

We have demonstrated a scheme for deploying D-QoS in a UDLR environment without causing bandwidth starvation for RMUS. We have shown that the proposed scheme can survive the network and can effectively deliver the services promptly to both traffic flows at the smaller minimum link capacity requirement.

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