Scheduling TCP-Friendly Flows over a Satellite Network

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Abstract—**We discuss in this paper basic requirements of an RCST gateway supporting VoIP traffic without strict QoS provisioning. Besides easing the system design, a QoS management not based on reservation is suitable to accomplish changes in traffic scenarios and to support new applications. The main elements of this architecture are illustrated throughout the paper following a layered structure. A preliminary set of simulations is presented to point out benefits and to identify major issues. Implications of buffer management, different scheduling algorithms, and bandwidth allocation protocols are analysed.**

Keywords—**TFRC, Scheduling DCCP flows, BoD**

I. Introduction

The telecommunications industry is embracing itself for an all-IP next generation network architecture requiring several access networks to seamlessly integrate into the Internet. An all-IP network architecture provides a plethora of new applications creating new revenue opportunities to service providers. Voice over IP (VoIP) is considered to be one such killer application and a considerable increase in the volume of VoIP traffic is forecasted in the future. This raises the issue of stability in the Internet. Many commercial VoIP applications do not implement congestion control and, in general, are not interoperable with existing TCP flows. Currently, the amount of traffic generated by these applications is not such to constitute a real threat of congestion, but, if the current trend continues, analysts presage a collapse.

In this paper we discuss an architecture for satellite access gateways suitable for congestion controlled voice flows, which departs from the classical class based QoS paradigm [2]. We argue that a satisfactory perceived audio quality can be achieved, up to a certain level, without enforcing strict QoS mechanisms. Though our proposed architecture is not meant to be the ultimate solution, we illustrate necessary components and their interactions. We show algorithms required at the transport layer, the network layer and the MAC layer pointing out their influence on VoIP performance.

In our framework, congestion control and network stability are guaranteed coupling a transport protocol suitable for multimedia flows, such as TCP friendly rate control (TFRC) [7][11], with active queue management algorithms (AQM). More specifically, the datagram congestion control protocol (DCCP) [6] provides a choice of congestion control mechanisms specified by congestion control identifiers (CCIDs). CCID-3 based on the specification was especially designed for multimedia flows over the Internet [8]. Random early drop (RED) and its variants have been proved effective [12] to improve bandwidth fair share and network stability. Early dropping or marking (ECN) of packets is here suggested for VoIP flows as a mean to reduce queuing delay and packet drops. Packets scheduling at IP layer is also a critical part of this QoS architecture. FIFO buffers are inadequate for supporting VoIP flows in that DCCP regulated flows may suffer an unfair treatment when competing with large data transfers. Fair queuing schedulers can alleviate this problem, but at the cost of increasing system complexity. Frame-based schedulers [3], such as deficit round robin (DRR), are particularly appealing in the design of high-speed systems for their low computational complexity and fairly well performance.

Another important component of our QoS architecture is the bandwidth allocation protocol. Traditional schemes that allocate on demand a fixed bandwidth are nowadays progressively abandoned in favour of more efficient adaptive schemes. Though these new schemes allow us significant bandwidth savings, they introduce a control loop at MAC layer that may interfere with TFRC. Hence, the lack of cross-layer design may cause drastic performance degradations. Rate based algorithms have been recommended for slowly variable flows and are considered in simulation section. The rest of paper is organized as follows. The principal components of our architecture are detailed in Section II following a top-down approach. In Section III, simulation results point out important aspects of our architecture. Conclusion and future work conclude the paper.

II. Network Architecture

A. *Architecture Overview*

Figure 1 shows the main elements of our reference architecture for an access gateway to a satellite mesh network. This architecture aims at being compliant with digital video broadcasting with return channel via satellite (DVB-RCS) standard. This technology is particularly popular in the DVB-S (DVB for satellite) format [1]. In next paragraphs, we describe, for the top layer to the bottom layer, the major components of our architecture.

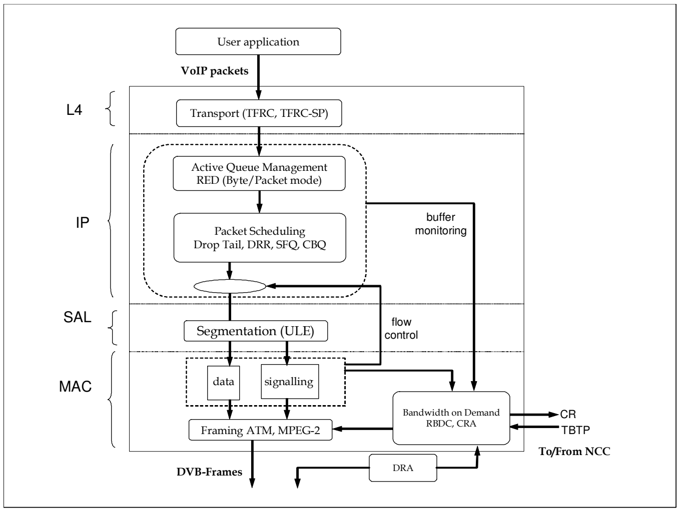

Figure 1 RCST access gateway networking diagram

B. *Datagram Congestion Control Protocol*

The transmission of voice packets generated by user application is regulated by datagram congestion control protocol (DCCP) [6]. DCCP provides a choice of congestion control mechanisms specified by congestion control identifiers (CCID). In particular, CCID-3, based on TFRC specifications, is suited for multimedia flows over the Internet. It is a rate-paced equation-based congestion control mechanism that requires both the sender and receiver to participate in determining the allowed sending rate.

A transport-layer control loop is formed where the receiver periodically sends a feedback report informing the recent loss event rate that has been witnessed by a multimedia flow. The sender uses an equation that models the equivalent throughput that would have been obtained by a TCP flow to calculate the allowed sending rate for the next path RTT period. This includes sum of all link RTTs along the path between the sender and the receiver. The equation calculates the throughput that a TCP connection would receive under steady-state conditions given the loss event rate and the RTT of the connection.

TFRC was designed to be reasonably fair when competing for bandwidth with TCP connections using the same packet size, and does not perform well when a low-bandwidth TFRC flow using small packets shares a bottleneck with high bandwidth TCP flows using large packets, because the TFRC flow is unfairly slowed down. To solve this problem, TFRC-SP, a Small-Packet (SP) variant of TFRC was designed for applications such as VoIP that send small packets [7]. TFRC-SP seeks to achieve the same bandwidth as a TCP flow using packets of up to 1500 bytes. TFRC-SP also enforces a minimum interval of 10 ms between packets, to prevent a single flow from sending small packets arbitrarily frequently.

C. *AQM & Packet Scheduling*

Packets forwarded by DCCP enter a system of queues at IP layer. Two major functions are considered at this level: active queue management and packet scheduling. The simplest IP queue, consisting of a single FIFO drop tail buffer, is not adequate for VoIP flows, which require short queuing delay and possibly no packet losses. Thus, we consider the use of Explicit Congestion Notification (ECN) that allows DCCP sources to detect, and hence react, quicker to congestion. Congestion feedbacks are signaled using an apposite field in the IP packet before raising buffer overflow.

Buffer management is coupled in our system with a packet scheduling that enables isolation between data and voice flows, and improves fairness in bandwidth sharing among flows. Packets are serialised towards different virtual queues based on their source/destination addresses and the next packet scheduled for transmission is chosen based on a fair queuing scheduling algorithm. In this paper, we consider two practical implementations of the ideal fluid scheduler (GPS), namely fair queuing (FQ) and Deficit Round Robin (DRR).

The fair queuing (FQ) model is a good approximation of GPS. The FQ server picks the first packet that would complete the service in a GPS simulation if no additional packets arrive after the time of packet scheduling. This scheme requires the evaluation of GPS finishing time at each packet arrival and the insertion of the packet timestamp into a linked list. Since this operation is time-consuming, we consider a simplified version of FQ called stochastic fair queuing (SFQ).

Deficit Round Robin (DRR) [3] the strategy to visit, in a round robin fashion, all the non-empty queues assigned to served flows, which ensures to each flow a service opportunity proportional to its bandwidth share. To avoid penalizing flows, the DRR scheduler maintains a per-flow state variable, the deficit count, which stores the difference between the amount of data sent so far and the amount that should have been sent. This deficit count is summed to the quantum of data scheduled for transmission in the subsequent round. Thus, if a flow has received a too small service during a round, in the next round it will have the opportunity to send more data, compensating the previous unfair treatment.

D. *MAC layer & BoD*

At MAC layer, IP packets are first encapsulated by multi protocol encapsulation (MPE) or using unidirectional lightweight encapsulation (ULE). This encapsulation also adds the MAC address of the destination satellite earth terminal. The encapsulated data is then delivered using a transport stream of MPEG-2 or ATM cells.

In DVB-RCS time slots are negotiated by means of explicit capacity requests (CRs) sent by RCST to the network control centre (NCC). The NCC periodically broadcasts a traffic burst time plan (TBTP) detailing the assignments for a superframe. DVB-RCS describes several methods for dynamically deciding how to allocate time slots to RCSTs, which are known as bandwidth on demand (BoD) methods. The one considered in this paper is the rate base dynamic capacity (RBDC): CRs are calculated as the incoming traffic rate seen by the terminal from the terrestrial network. Every explicit request overrides the previous one and new requests are submitted only when needed.

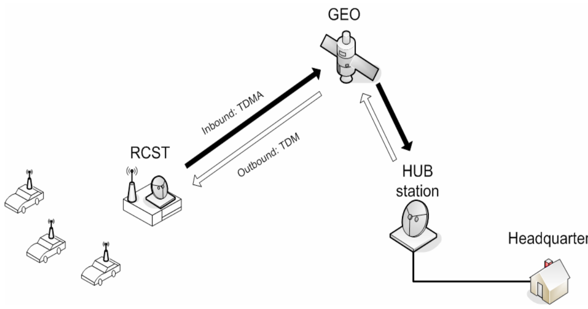

Figure 2 RCST and Hub Network Topology

III. SIMULATION RESULTS

This section describes three simulation tests that point out important aspects of proposed system. Table I summarises the parameters used in simulations. Results are designed to be illustrative, rather than providing detailed analysis of representative systems and traffic. Thus, as a first approach, we used an algorithm that intentionally over-allocates capacity to ensure acceptable performance. A simple single carrier per channel (SCPC) does not take into account cross-layer trade-offs possible in assigning free capacity.

The following results were obtained by simulating a single VoIP flow using DCCP competing for bandwidth with several FTP flows. Figure 2 depicts the topology of our simulation: sources compete for the return link bandwidth through a RCST gateway. Although voice calls typically transmit audio in both directions, we consider here only the inbound direction. The outbound part of the connection is indeed sent on a TDM carrier and does not impact the performance assessment.

The model for the voice encoder is consistent with the widely used G.711 codec with a media rate of 8 kb/s and 20 ms RTP frames. Voice activity detection (VAD) is assumed, resulting in talk-spurts and silence periods. Each VoIP session simulates a randomly timed sequence of talk-spurts and silence periods, with a decaying exponential distribution of 0.352 s and 0.65 s respectively [4]. Since voice is active only 35% of time, the VAD enables a saving in capacity and compensates for the overhead of VoIP encapsulation. Considering the overhead introduced by RTP/UDP (40B) encapsulation and link encapsulation headers (14B, assuming ULE), each call requires on average 6.6 kbps of capacity per session. This represents a saving in capacity with respect to a constant rate encoder.

A. *Fairness criteria for VoIP flows*

In this part, we highlight the necessity of fair queuing to ensure fairness between data and voice flows. We adopt a fairness criterion called max-min fairness [5]. Roughly speaking, an instance of allocations is said max-min fair when, in order to increase the rate assigned to any flow, we have to decrease the rate assigned to a smaller rate flow. For instance, the allocation between a large data flow and a voice flow would not be considered fair, if bandwidth is assigned to the large data flow reducing the allocation of the small voice flow. For a dumbbell topology with M greedy TCP flows and N DCCP flows, whose rate is limited to r by the application, the max-min fair share (normalized to the available bandwidth) for a DCCP flow is $FS_{DCCP} = \min\{ r/B \ , \ 1/(M+N) \}$, where B is the available bandwidth.

Figure 3 compares the fair share achieved by DCCP with a drop tail and DRR queuing systems. We note that the system is not able to guarantee fair share among VoIP and FTP flows when a drop tail FIFO discipline is used. The lack of fairness is mainly due to the difficulties in correctly estimating TCP equivalent throughput at DCCP sender when flows have different packet sizes. Although this problem may be solved at the transport level introducing more sophisticated estimation algorithms, such as the already mentioned SP option, a simple mechanism to offset this problem is using fair queuing. As we can note, DRR, a simples form of fair queuing, is able to restore fairness.

Figure 4 shows the delay d (in ms), which accounts for the average end-to-end packet delivery latency and jitter, and the drop rate e, which accounts for the amount of packets lost along the path and missing the play-out deadline. These parameters can be used to calculate the R-score, a synthetic audio quality metric between zero (poor) and hundred (high). A derivation of R-score for the environment under study is

$$\begin{cases} Rscore = 94.2 - I_e - I_d \\ I_e = 0.134 \cdot d - 19.5 \\ I_d = 30 \cdot \ln(1 + 15 \cdot e) \end{cases} \quad (1)$$

where I_e and I_d measure the impact of loss and delay, respectively. The maximum R-score for a satellite environment with a delay of 280 ms and no losses corresponds to 76.2, which is considered Medium quality according to ITU-T recommendations. R-score can be used to approximate the widely used mean opinion score.

TABLE I. TEST CONFIGURATION

SD & SI parameters	*value*
Satellite propagation delay	280 ms
LAN delay local to RCST	10 ms
LAN delay local to hub	50 ms
Outbound capacity	10-32 Mbps (continuous)
Forward link	DVB-S, MPEG-2
Forward encapsulation	MPE/ULE
Inbound (MF-TDMA max)	0.512 Mbps
Return PHY	MF-TDMA
Return Link	MPEG*
Return Encapsulation	MPE/ULE*
Synchronisation	1 per second
Frame/Superframe	1
Superframe Period	40 ms
CRA	1 slot/superframe
CR	on SYNC
BoD	RBDC
Max-RBDC	Carrier Capacity
Allocation Delay	800 ms

*Future simulations will explore ATM, AAL5

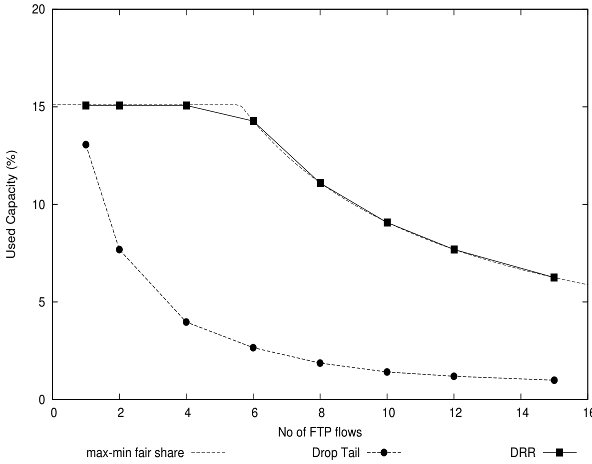

Figure 3 Fairness for VoIP flows

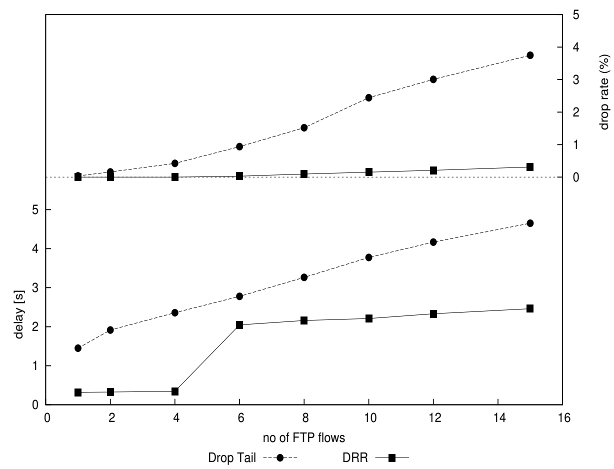

Figure 4 VoIP drop rate and delay for Drop Tail and DRR.

B. Scheduling & AQM

Results in this section aimed at showing that the specific algorithms used for buffer management and packet scheduling have a non negligible impact on overall system performance. Figure 5 reports R-scores for a scenario using TFRC-SP as transport protocol, and a MAC layer configured as in Table I. In addition to DRR and SFQ, we consider also a class-based queuing (CBQ) discipline with two priorities. We expect performance significantly enhanced by implementing some QoS handling at the satellite terminal that differentiates the different types of traffic (e.g. VoIP be prioritised over other data). Furthermore, we consider the case that buffer overflows happen when queue length measured in packets or bytes is reached, referred in graphs as Packet mode and Byte mode respectively.

When VoIP traffic is prioritised, performance is clearly independent of the pattern of other traffic and quite high scores are attained. However, DRR and SFQ achieve performances not far apart from CBQ (an R-score of 70 or higher is considered good quality) up to a large number of competing FTP flows. SFQ performance lowers when the number of flows approaches the number of virtual queues. When the gateway operates in byte mode (Figure 6), the VoIP have better performance with both RED and drop tail queuing. This result is inline with what observed in [7] regarding dropping rates for small packets: the goal of TFRC-SP is to achieve fairness in terms of sending rate when competing with larger TCP flows packets, if each packet has roughly the same drop probability.

In a scenario where large packets are more likely to be dropped than small packets, or where flows with a bursty sending rate are more likely to have packets dropped than flows with a smooth sending rate, TFRC-SP flows could receive more bandwidth than its fair share. This can be observed in Figure 6, where RED in byte mode allows the VoIP flow to get a performance almost similar to CBQ. Although this inconvenient can be compensated enforcing a limitation in TFRC-SP sending rate (100 pps), we believe that this behaviour is reasonable for the Internet where the majority of the routers operates in Byte mode.

C. Effect of BoD over DCCP flows

There are two components needed to counter the effect of the different types of traffic: a more advanced radio resource management that helps alleviate the effects of increased load, some QoS handling at the satellite terminal that differentiates the different types of traffic; e.g., VoIP (expedited forwording data) may be prioritised over other data.

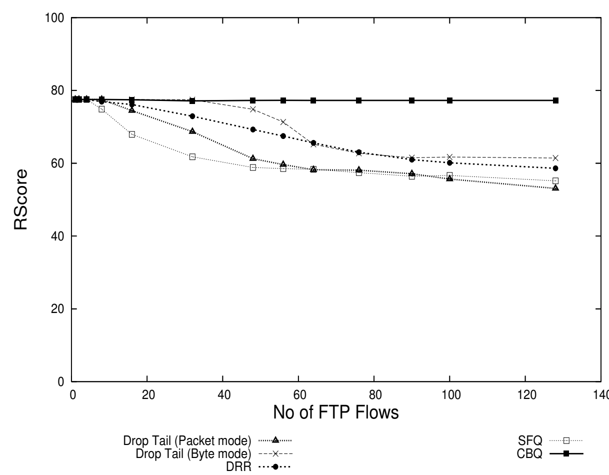

Figure 5 R-score of VoIP flows for different scheduling mechanisms.

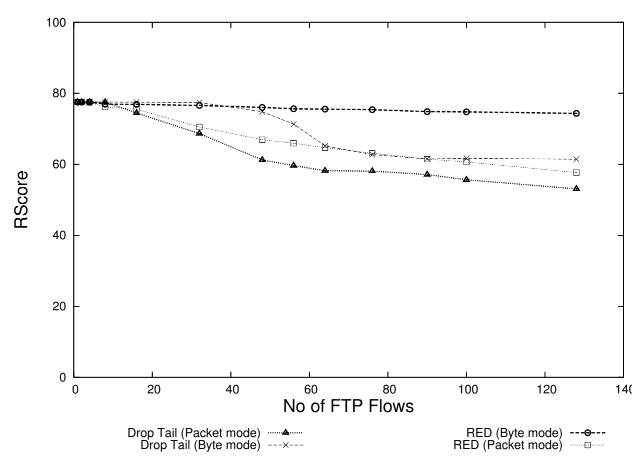

Figure 6 R-score of VoIP flows for different buffer management.

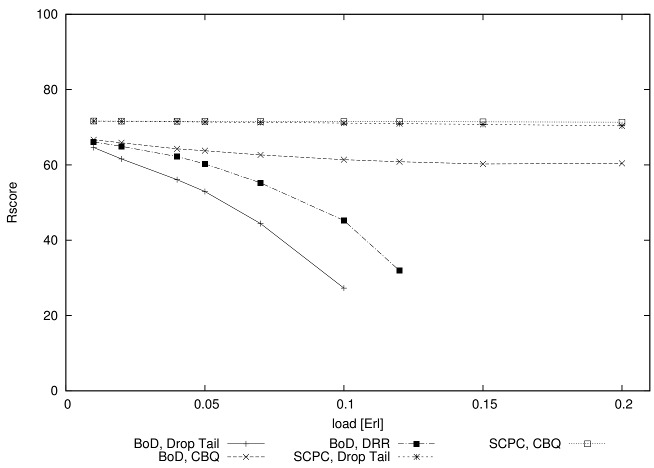

Figure 7 R-score of VoIP flows comparison for BoD and SCPC.

This is considered in the last test that illustrates the effects of BoD when multiplexing of several VoIP calls (eight parallel flows in current simulation) and their impact on performances. RBDC is used to assign capacity on demand, allowing the capacity allocation to be varied with time to reflect the needs of the traffic to be sent by the RCST. In order to ensure continuity in bandwidth allocation, one slot per superframe in addition to BoD is allocated to the RCST. This over-allocation is beneficial since it assures an acceptable one-way delay for a VoIP session. It is expected that the traffic characteristics for voice will result near to constant assignment.

We estimated an average R-score of 67.3 versus an R-score of 71.66 when SCPC is used. This small performance deterioration is swapped for a considerable bandwidth saving. Since VAD is used, RDBC theoretically allows a saving of 65%. However, CRA and the high traffic variability requires on average to allocate slightly more than the expected mean bit rate.

Figure 7 shows the average R-score of 8 VoIP calls using TFRC-SP. VoIP calls share the forward link with web traffic generated starting a new HTTP session of L bytes every T seconds, which corresponds to a load of $8L/(C{\cdot}T)$ Erlangs being C the forward link capacity. The SCPC system is compared with BoD system using drop tail or CBQ. Since the capacity in SCPC is by far larger than the peak rate of the traffic mix, the type of scheduling mechanism used weakly affects performance. On the other hand, a BoD system is more influenced by the IP level mechanism employed. For instance, when the network layer implements a simple drop tail queue, perceived audio quality reduces drastically even for very small traffic loads (2-3%). Introducing some form of scheduling (e.g. CBQ in this experiment) is thereby necessary to avoid making the system unavailable.

IV. CONCLUSIONS & FUTURE WORK

In this paper, we presented an analysis of an architecture supporting multimedia traffic. This architecture does not consider sophisticated QoS mechanisms to handle multiplexing of VoIP and data flows. Though increasing the priority of VoIP messages would offsets the performance degradation caused by web workload, this method leads to considerable design issues. Indeed, a practical method would require a sophisticated scheduler that supports more than two levels of priority (e.g., other traffic classifications will also require special treatment) and will need to integrate QoS with congestion control to optimize the overall performance (especially when traffic is classified at multiple levels and capacity varies as a function of ACM and fading). In addition, when traffic is originated by RCST, the performance degradation due to the delay induced by TDMA can be observed.

Simulations were performed using ns2 to provide illustrative results for the VoIP architecture. Schemes that insist on VoIP performance have been considered. At transport layer TFRC/TFRC-SP is used to avoid congestion, which cooperates with AQM at network layer. Fair queuing is considered to equalise traffic flows and to improve fairness. Bandwidth on demand is implemented at the MAC layer to enhance the link capacity utilization.

Future work will need to assess how to optimise performance both of the end-to-end service for different traffic classes, and to ensure efficient utilisation of the satellite capacity. Moreover, we require the development of a more detailed simulation environment and design decisions reflecting size/complexity of the system.

ACKNOWLEDGMENT

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