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Video Streaming with Multi-TFRC and Uplink Queue Management

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***Abstract*—This paper considers multi-connection video streaming across a tandem network with a WiMAX wireless link. The paper shows that Active Queue Management has a role and that a small number of connections results in reasonable video quality with increased throughput. The results will be of interest to those dimensioning a mobile network for an interactive multimedia service.**

I. INTRODUCTION

In Brazil, mobile WiMAX is the basis of a digital TV service but there is also interest in exploitation of uplink interactive multimedia services [1], which requires effective video streaming. Therefore, a way of streaming video over a WiMAX uplink is desirable. Two problems arise. The first is that the IEEE802.16 standard [2] (known as WiMAX) in all its versions (parts a–e) does not specify queue management algorithms at the Subscriber Station (SS) (or Base Station (BS)), allowing these to be designed by vendors as a means of commercial differentiation. Choice of queuing discipline is sensitive to application type and service class, whether it is implemented on the BS or the SS. The second problem is that a means of congestion control is needed if the video stream's path subsequently takes it over the wired Internet.

Chen and Zakhor proposed MULTTFRC [3] as a multi-connection version for tandem networks of the standardized TCP-Friendly Rate Control (TFRC) [4] congestion controller for the wired Internet. In a tandem network with wireless link and Internet path combined, whatever path the multiple connections take over the Internet, they are assumed to cross a single wireless uplink (UL) before reaching their destination. In Fig. 1, once the WiMAX BS has allocated bandwidth between the SS, each SS must manage its queue according to the data arrival rate from user applications. However, an SS streaming video must also respond to congestion across the best-effort Internet that lies beyond the wireless uplink.

TFRC itself uses IP/UDP to transport real-time services. To avoid a risk of congestion collapse, TFRC models a TCP Internet connection at the application layer by an equation parameterized by packet-size, round-trip-time, and packet

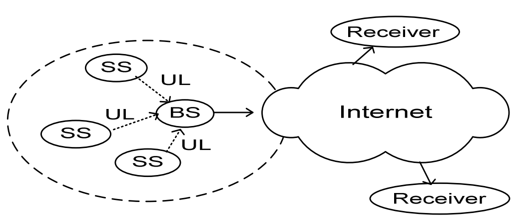

Fig. 1. IEEE 802.16 uplink service architecture

loss. The intention is to avoid TCP's aggressive rate control which can lead to excessive packet loss and 'saw-tooth' like rate oscillations. These respectively lead to damaging error propagation across a compressed video stream and unsettling changes in video quality for the viewer. However, high packet loss rates at a wireless link can cause TFRC to unnecessarily staunch its rate in anticipation of congestion. Therefore, MULTTFRC represents a lightweight way to retain TFRC for the Internet path but avoid complex means of suppressing channel loss feedback to TFRC over the wireless link. For example, in the SNOOP approach [5], TCP acknowledgments are suppressed if they arise from channel loss and *not* through congestion-induced buffer overflow at the wireless link. However, setting up a SNOOP-like system at each WiMAX uplink would be burdensome for a mobile SS.

Unfortunately, to date [3], though the principle of multiple TFRC streams has been established and though it is certainly true that multi-TFRC connections are an effective replacement for SNOOP-like systems, certain crucial and practical questions that affect its viability remain unanswered. These are: how exactly is a single video stream multiplexed over multiple connections; how many connections are feasible for broadband wireless; what is the effect of different queue management disciplines, including Active Queue Management (AQM) [6]; and what is the likely video quality. In fact, it turns out that for WiMAX there is a trade-off between video quality, throughput and associated packet-wise latency and jitter that can be achieved with a small number of connections, depending on co-existing traffic.

II. SIMULATION

To answer such questions, we used the well-known ns2 simulator with the WiMAX module from Chang Gung Univ., Taiwan. The settings in Table I are typical except the 1:3 sub-frame ratio in favor of the uplink (UL) for evaluation purposes only and the 1 km range to avoid propagation effects in delay timings. The simulated traffic sources from the three SSs in Fig. 1 are tabulated in Table II. We were interested in the behavior of a Variable Bitrate (VBR) video source as it entered the real-time Polling Service (rtPS) service class buffer. In tests, 1000 frames of the reference 'Paris' clip H.264/AVC VBR-encoded at 15 Hz (frame/s) at Common Intermediate Format (CIF) (352×288 pixel/frame) with initial quantization parameter set to 30.

We compared several queuing disciples. Drop-tail (FIFO) queue management has the advantage that it scales well and shares delay between different connections. However, drop-tail may lock-out some connections to the advantage of others occupying the buffer. Random Early Detection (RED) is

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TABLE I

SIMULATED WiMAX SETTINGS.

Parameter	Value	Parameter	Value
PHY	OFDMA	FFT size	1024
Duplex mode	TDD	DL/UL ratio	1:3
Frame length	5 ms	Fragmentation	Yes
Bandwidth	6 MHz	Range	1 km

TABLE II

SIMULATED TRAFFIC CHARACTERISTICS

SS-UL	Service type	Traffic type	Protocol	Packet size (B)
1	rtPS	VBR (video)	MULTTFRC	Variable
		CBR	UDP	1000
	nrtPS	FTP	TCP	
2	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	
3	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	
SS-DL				
1,2, 3	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	

known to under-perform when there are few coexisting flows in a queue, because buffer occupancy fluctuates rapidly, which suggested that some improvement might come with extra connections. Random Exponential Marking (REM)'s packet marking probability grows considerably more aggressively than RED's. Its congestion measure is based on any mismatch between input rate and link capacity and between and target length, though in practice rate change is measured by the rate of change of the queue length. Rather than queue occupancy, BLUE employs a history of the current buffer packet overflow rate and link utilization to form the packet dropping (marking) probability. The queue buffer sizes were each set to 50 packets. Results were averaged (arithmetic mean) over ten simulations. For RED, the dropping thresholds were set to 20% and 80% of the buffer size.

For ease of interpretation, each connection was statically serviced on a Group of Picture (GOP)-by-GOP basis, with GOP size set to 30 frames. A dynamic scheme is to select according to next available GOP and with smaller GOP sizes (12 or 15). A reorder buffer is needed at the receiver. The Baseline profile of H.264 was used to reduce computation at the mobile SS.

III. FINDINGS

Table III shows the mean Peak Signal-to-Noise Ratio (PSNR) and standard deviation (σ) of the video quality. Highlighted are the results for three connections because 30 dB is an approximate threshold below which video quality generally ceases to be good. As is apparent, with growing number of connections the quality variation grows considerably, with BLUE [6] giving a 1 dB advantage (recall this is a logarithmic scale). Table IV summarize the throughput for the best-performing BLUE. The period (time over which the connections complete streaming) reduces according to connection. For BLUE, the mean (maximum) delays were:

TABLE III

VIDEO QUALITY ACCORDING TO CONNECTIONS AND QUEUE TYPE

	Mean PSNR (σ) (dB)			
	Drop Tail	RED	REM	BLUE
1 Conn.	36.0 (0.2)	36.0 (0.2)	36.0 (0.2)	36.0 (0.2)
2 Conn.	33.2 (5.5)	31.4 (7.2)	33.2 (5.5)	33.2 (5.5)
3 Conn.	30.9 (7.6)	31.3 (7.6)	29.7 (8.0)	31.4 (7.2)
3 Conn.	28.1 (8.4)	29.4 (8.4)	20.3 (7.6)	29.4 (8.4)

TABLE IV

THROUGHPUT ACCORDING TO NUMBER OF CONNECTIONS

BLUE	Period (s)	Max.(kbps)	Mean (kbps)
1 conn.	70.5	258.6	120.8
2 conn.	35.5	505.7	237.3
3 conn.	23.9	726.4	347.2
4 conn.	17.9	819.4	456.5

1 connection 0.006 (0.072) s; 2 connection 0.013 (0.131) s; 3 connections 0.020 (0.184) s; and 4 connections 0.029 (0.196) s. Thus, the main penalty of multiple TFRC connections is a rise in maximum latency for the occasional packet, which could affect buffer dimensioning and cause interruptions in interactive services if not countered.

IV. CONCLUSION

Simulation suggests that a single video stream sent over multiple congestion-controlled connections can maintain good quality video. Results are best-case, as they do not include channel error or congestion drops in a best-effort Internet stage. Small numbers of connections (three in the simulations) with AQM are feasible but there is a risk of the occasional long packet delay, which could be compensated for by error concealment or resilience if delayed packets are dropped at the decoder. AQM does bring the possibility of content-dependent packet dropping which will further improve the delivered video quality.

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