

Multi-Connection TFRC Video Streaming in a WiMAX Environment

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Abstract— Dedicated WiMAX multimedia services are proposed for areas that lack networked infrastructure. This paper proposes multi-connection uplink video streaming for mobile WiMAX. Establishing multiple TFRC connections for a single video stream has emerged as a promising lightweight way of coping with wireless channel losses in a congestion-controlled tandem network. This study shows the impact (in terms of video quality and latency) on multi-connection streaming performance in the presence of burst errors on the wireless link. It also establishes how many connections are feasible.

Keywords—multi-connection; TFRC; video streaming; WiMAX

I. INTRODUCTION

IEEE 802.16e (known as mobile WiMAX) [1] allows rapid deployment of video services in areas in the world unlikely to benefit from extensions to 3G such as High Speed Downlink Packet Access (HSDPA). WiMAX's uplink capacity is likely to exceed that of HSDPA's 384 kbps. In Brazil, mobile WiMAX is the basis of a networked digital TV service but there is also interest in exploitation of uplink interactive services [2], which could involve video streaming from mobile devices across the Internet. For these services, rate control of video streaming over the wireless uplink and onwards over the wired Internet will be important to reduce packet loss and latency.

The contribution of this paper is a detailed analysis of a multiple-connection rate control scheme over a mobile WiMAX uplink prior to the Internet. As a basis, we employ a form of MULTTFRC [3], which uses multiple TCP-Friendly Rate Control (TFRC) [4] connections. We ask what would occur if multiple TFRC connections were opened in the uplink (UL) from a WiMAX subscriber station (SS) to base station (BS) in the presence of cross traffic from other mobile SSs. In this situation, congestion will occur at the real-time polling service (rtPS) queue and packet loss will occur over the wireless channel. It appears from our results that, in a realistic setting, some gain in throughput and video quality arises from using a limited number of connections. However, another attraction is that using TFRC allows congestion control when the video stream (over multiple connections to the receiver) continues its path across the Internet, constituting a tandem or concatenated network.

In video streaming across the Internet, UDP transport

usually serves to reduce delay at the expense of some packet loss, while application-layer TCP emulation [5] such as TFRC acts as a form of cooperative congestion control. Notice carefully that TCP emulation by the application is *not* the same as TCP. TCP itself is unsuitable for delay-intolerant video streaming, because it introduces unbounded delay in support of a reliable service. Instead, TCP emulation mimics the average behavior of TCP, but is not 'reliable' and does not result in the 'saw-tooth'-like rate fluctuations that arise from TCP's aggressive congestion control algorithms.

In tandem networks with wireless link and Internet combined, the multiple connections are initially assumed to cross a single wireless link before entering the wired Internet. However, while TCP emulators are known [5] to work well over the Internet, they may not respond appropriately over a wireless link. Packet loss through channel error may be mistaken for congestion, leading to underutilization of the wireless link. In fact, burst errors, through slow fading, are common in mobile networks, exacerbating the problem.

The research in [3] proposed MULTTFRC originally as a form of downlink control. Any single TFRC connection responds to packet loss by reducing its output rate by increasing the inter-packet gap. By multiplexing a video stream across multiple connections it is hoped that the impact of packet loss on one or more of these connections will be mitigated by the rate across the remaining connections. 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 [6], wireless link packet loss feedback to the congestion controller is suppressed by a SNOOP unit, which requires intervention at the data-link layer and cross-layer interaction.

Prior work on MULTTFRC and its variants such as [7] appears largely confined to analysis of a generic link without other traffic. An exception is [8] in which a practical demonstration for IEEE 802.11 was included. However, even in the most definitive account of MULTTFRC so far, the work in [8], no video data ever appears to have been streamed, as there is no account of how a single video stream is multiplexed onto the multiple connections. Only generic packet loss and delay statistics are reported even though the type of error pattern is known to affect video quality (PSNR) by several dBs. Therefore, after the initial development of the concept, it is now necessary to investigate how the scheme works for a

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particular wireless technology and what the QoS concerns are. This is the intention of our paper.

In [9], multiple connections over an IEEE 802.11e link were used to send different layers of a scalable video stream. However, the scheme was only tested for two layers and quality testing does not appear to have accounted for lost packets. This is an important issue in the Scalable Video Coding extension to the H.264 codec, because of the complex inter-dependencies between the layers. If packets are lost then many other encoded video-bearing packets that are dependent on them have to be abandoned as well. IPTV is one form of multimedia service that is attractive for WiMAX but proposed solutions have been at the physical layer [10] rather than the application layer, as in this paper. The paper now considers the WiMAX background.

II. WiMAX ENVIRONMENT

A. WiMAX system

In Fig. 1, once a BS has allocated bandwidth to each SS, each SS must manage its queue according to the data arrival rate from user applications. In WiMAX Point-to-Multipoint (PMP) mode, there is no SS-to-SS communication unless it is via the BS. WiMAX networks support multiple service classes to accommodate heterogeneous traffic with varying requirements. WiMAX's rtPS is most suitable for real-time video services, particularly for Variable Bitrate Video (VBR), which is employed to maintain delivered video quality but may lead to 'bursty' arrival rates. Other congesting traffic is assumed to enter the non-real-time Polling Service (nrtPS) queue at the SS. In our experiments for both queues, a drop-tail queuing discipline was simulated. Queue sizes were all set to fifty packets. This value was selected as it seems appropriate to mobile, real-time applications for which larger buffer sizes might lead both to increased delay and also greater active and passive energy consumption from the buffer's memory.

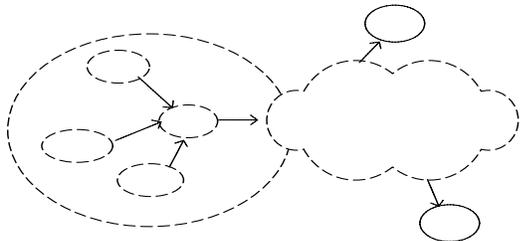


Figure 1. IEEE 802.16 uplink service architecture

B. Simulation settings

The WiMAX system operating in PMP mode was simulated by well-known ns-2 simulator (v. 2.29) augmented by a WiMAX module [11]. The PHY settings selected for WiMAX simulation are given in Table I. The DL/UL ratio is not intended to be realistic but to aid in testing multiple-connection TFRC, as in practice the DL would be allocated the majority of the bandwidth capacity. The range was set to 0.7 km to reduce the propagation delay component of latency, to better judge the queue waiting time.

There were three SSs communicating to the BS, with one of the SS sending a VBR video sequence encoded with the

H.264/Advanced Video Codec (AVC) and split between the multiple TFRC connections. The other SSs are simply introduced as sources of competing traffic across the wireless link and do not indicate the likely size of a WiMAX network, which obviously could be larger. A trace file was input to ns-2 and packet losses recorded in the output. The output serves to calculate the PSNR. Video quality comparisons were made under the EvalVid environment [12].

As a test, we used the 'Paris' clip H.264 VBR-encoded at 30 Hz (frame/s) at Common Intermediate Format (CIF) (352 × 288 pixel/frame) with initial quantization parameter set to 30. Paris consists of two figures seated around a table in a TV studio setting, with high spatial coding complexity. H.264's Baseline profile was selected, as this is more easily supported by mobile devices because of its reduced computational overhead. The Intra-refresh rate was every 30 frames with IPPP...I structure, i.e the Group of Picture (GOP) size was 30. 1000 frames were transmitted. Table II records the simulated traffic characteristics for the three SSs communication with the BS. Network Adaptation Layer Units (NALUs) from the H.264 codec were encapsulated with Real Time Protocol (RTP) headers on a single slice per frame basis. After the addition of IP headers, these in turn formed a single WiMAX MAC Protocol Data Unit (MPDU), which are variable-sized WiMAX packets.

TABLE I SIMULATED WiMAX SETTINGS

Parameter	Value
PHY, duplexing mode	OFDMA, TDD
Frequency, Modulation	5 GHz, 16 QAM 1/2
Frame length	5 ms
Max. packet length	1024 B
Raw data rate	10.67 Mbps
FFT size, Guard band ratio	1024, 1/8
DL/UL ratio	1:3
Fragmentation	Yes
SS/BS transmit power	250 mW, 20 W
Range	0.7 km
SS/BS antenna heights	1.5/32 m
Antenna type, gain	Omni-directional, 0 dBD

TABLE II SIMULATED TRAFFIC CHARACTERISTICS

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

The packet trace created by EvalVid from the H.264 output ensured packets did not exceed 1000 B. This implies that there is some risk of decoder de-synchronization if a slice is split into several MPDUs, causing slice de-synchronization markers to be missing from the payload of some MPDUs. In this sense, our video quality findings are worst case. For simplicity, a WiMAX MPDU is now referred to as a packet.

For TFRC, the inter-packet sending time gap was varied according to the TFRC equation [4], not the simplified version reported in [8]. As described in [4], TFRC is a receiver-based system in which the packet loss rate is found at the receiver and fed-back to the sender in acknowledgment messages. The sender calculates the round-trip time from the acknowledgment messages and updates the packet sending rate. An equation that models TCP New Reno is employed to find the sending rate. In our variant to standard TFRC, the packet size in the TFRC equation was dynamically altered according to the EvalVid-created trace file sizes. This variant makes for more responsive control rather than the mean packet length employed in the original TFRC formulation [4]. TFRC was originally intended for video-on-demand applications, when it is feasible to calculate the mean packet length. Setting a mean packet length is inappropriate for interactive multimedia applications. The underlying TFRC transport protocol was set to UDP, as is normal.

Coexisting rtPS queue CBR sources were all sent at 1500 kbps, i.e. the same target rate as the video source. The inter-packet gap was 0.03 s for the CBR traffic. The FTP applications were set up out of convenience as a way of occupying the nrtPS queues; otherwise a BE queue might be more appropriate. Likewise, the DL traffic is simply selected to fully occupy the DL link capacity.

C. Channel model

A Gilbert-Elliott two-state, discrete-time, ergodic Markov chain modeled the wireless channel error characteristics at the ns-2 physical layer. The probability of remaining in the good state was set to 0.95 and of remaining in the bad state was 0.94, with both states modeled by a Uniform distribution. The packet loss probability in the good state was fixed at 0.01 but was varied in the bad state as [0.01, 0.02, ..., 0.1]. In this way, we were able to judge the effect of worsening burst error channel conditions.

D. Management of connections

To systematically test the effect of multiple TFRC connections, rather than apply MULTTFRC's connection switching mechanism [8], the number of TFRC connections was incrementally stepped up. In MULTTFRC itself, the number of connections is changed over time according to the average round-trip time of all the connections, but this hides the interpretability of results. It is unclear from [3, 8] how a single video stream would be apportioned between a varying number of connections. For test purposes, we used a form of progressive download [13], which, despite the name, is a form of streaming media on demand. In progressive download one 'chunk' of video is sent, while the receiver plays back a previous chunk. A single queue was segmented into GOPs (30 frames). Each connection was statically allocated its GOPs, which are taken in interleaved manner from the video sequence. Of course, this assumes that a re-ordering buffer is available at the receiver.

III. EVALUATION

Table III shows the average data-rate over time when transmitting Paris over multiple connections. Clearly, TFRC is

able to multiplex more data onto a link as the number of connections increases, though observation of a time-wise plot of throughput shows that during transmission TFRC quenches its overall sending rate in response to packet loss. Mean results were taken over ten simulation runs to provide converged results. The resulting throughputs over time and simulation runs are plotted in Fig. 2. There is a gradual decline in throughput as channel error conditions worsen.

Fig. 3 plots the video stream packet drop rate relative to channel packet error rate. Included in the percentages in Fig. 3 are any additional packet losses arising from buffer overflow at the SS caused by the SS packet scheduler being otherwise occupied servicing the rtPS queues in the three SSs. As will be observed, no strong effects result from increasing the number of connections. Moreover, for all but the highest error rates the packet loss rate is below 10%. This is encouraging because subjective testing of video streaming to a variety of mobile devices [14] suggests that packet loss rates below 14% still result in an acceptable quality-of-experience. Packet delay levels were very low, a few milliseconds in Fig. 4. Latency declines with increasing channel error, presumably because TFRC reduces its sending rate and consequently reduces buffer occupancy and waiting time.

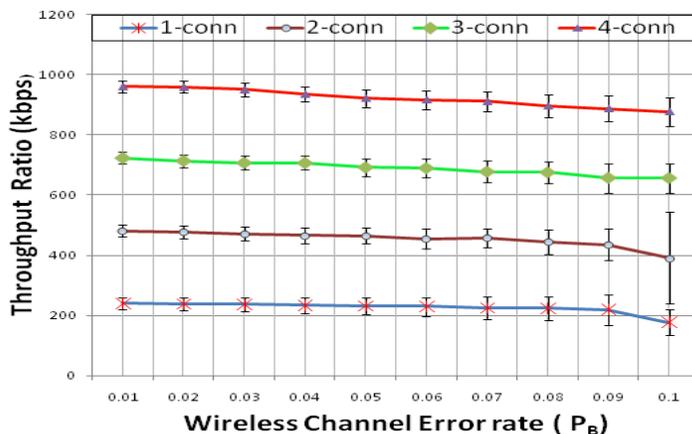


Figure 2. Average throughput for an increasing number of connections, according to channel error rate, showing 95%

TABLE III MULTIPLE CONNECTION S (1-4) TFRC THROUGHPUT

	Sending period (s)	Average datarate (kbps)
1	71.4	216.69
2	35.8	437.45
3	23.3	662.77
4	17.4	889.49

In terms of PSNR, Fig. 5 averaged over ten runs tells a surprising story which might not have been deduced from the raw drop rates of Fig. 3, as reconstructed video quality for a four connection session is markedly below that of a single connection. This results from the packet drop pattern for a four connection, as consecutive packet losses are difficult to

recover from with a GOP. While users are likely to tolerate PSNR of between 25 and 30 dB in the knowledge of the difficulty of transmitting over a mobile network [15], PSNR levels below 20 dB result in unacceptable video quality.

IV. DISCUSSION

Multi-connection video streaming under TFRC congestion control is essentially a compromise between the number of connections and the resulting video quality. The technique is feasible, provided it is confined to a few connections and provided channel conditions are not too severe, as increasing the number of connections leads to unfavorable video quality. The frame length of 5 ms is at the short end of WiMAX's range. If required, increasing the duration to the maximum of 20 ms will reduce queuing and, hence, delay at the SSs. Further work will investigate whether error resilience techniques such as H.264's Flexible Macroblock Ordering will lead to a further improvement in delivered video quality. This work should also clarify remaining issues of video synchronization at the receiver.

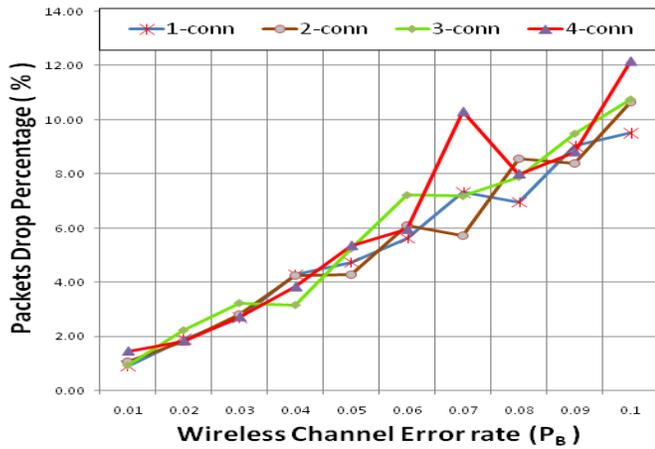


Figure 3. Average packet drop rate for an increasing number of connections, according to channel error rate.

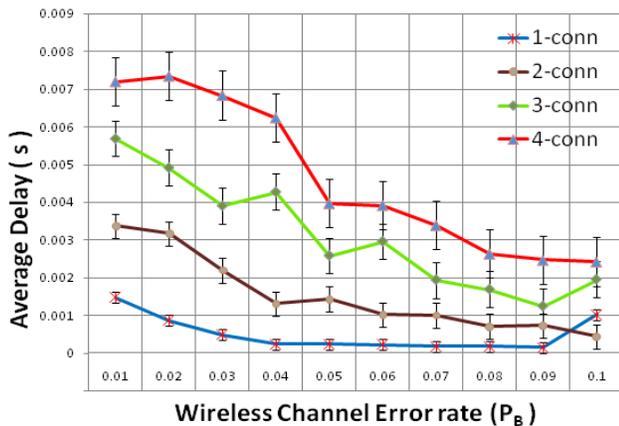


Figure 4. Average per packet delay for an increasing number of connections, according to channel error, with 95% conf. intervals.

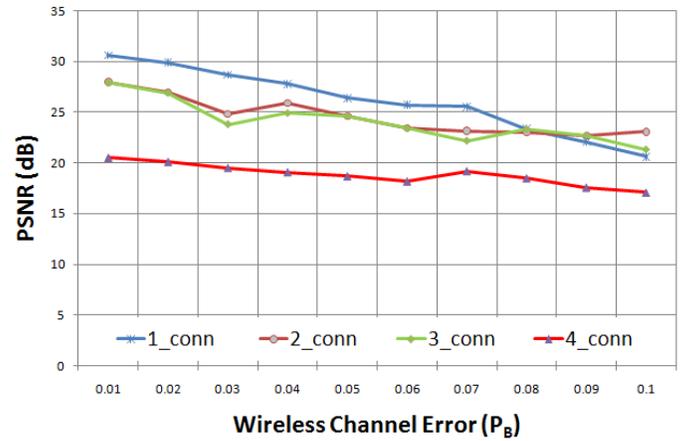


Figure 5. Video quality for an increasing number of connections, according to channel error rate.

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