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Multiple TFRC Streaming in a WiMAX Environment

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***Abstract*-Employing multiple TFRC connections has emerged as a promising lightweight way of coping with wireless channel losses in a congestion-controlled tandem network. This work extends multiple TFRC connection streaming to broadband uplink streaming. This study differs from earlier work for a number of reasons: it reports on a specific wireless technology, IEEE 802.16 (WiMAX); it considers the impact on buffer management at the subscriber station rather than at the Internet end; and it considers actual delivered video quality and not simply network statistics. In terms of video quality, it is reported that multiple TFRC connection streaming has a role to play provided there are only a few connections opened at any one time. Therefore, the work demonstrates that multimedia services can be achieved over broadband wireless by multiple TFRC connection streaming.**

I. INTRODUCTION

In peer-to-peer file download and HTTP, greedy acquisition of multiple TCP connections is already a well-known technique to increase throughput. In [1] for web cache download, multiple TCP connections were consolidated in a single one with an N-fold increase in throughput over a single connection. MULTTFRC has also been proposed [2] as a multi-connection version of the standardized TCP-Friendly Rate Control (TFRC) [3]. In tandem networks with Internet path and wireless link combined, whatever path the multiple connections take over the Internet, they are assumed to cross a single link before reaching their destination. MULTTFRC potentially 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.

In this paper, the potential gain from using multiple connection TFRC transport is demonstrated for an IEEE 802.16d,e (WiMAX) broadband wireless link [4]. Unfortunately, in the contributions to MULTTFRC [5; 6] including the most definitive to date [7], there appears to be no account of the impact upon video quality. We ask what would occur if multiple TFRC connections were opened in the uplink (UL) from WiMAX subscriber station (SS) to base station (BS), as might occur in a mobile interactive video service. In this situation, congestion will occur at the WiMAX real-time polling service (rtPS) queue. Therefore, the paper concentrates on the effect of congestion at the SS. By and large our results are favorable to the MULTTFRC approach, provided only a few connections are opened in a session.

TFRC in its single connection form was originally intended for the wired Internet, controlling an IP/UDP stream to transport real-time services. TFRC is intended to provide less aggressive congestion control than TCP's 'sawtooth' response to network congestion, resulting in a smoother sending rate more suitable for video streaming. As TFRC responds to packet loss, as well as round-trip time and packet size, it will change its rate if packets are actively dropped from an SS queue. The advantage of sending multiple TFRC flows through a WiMAX buffer is that the loss rate is spread across all the streams, enabling TFRC to better control its output. In fact, in the original formulation of TFRC [3] it is stated that TFRC is well suited to "an application in which the sender is a large server handling many concurrent connections". A single SS cannot be described as a large server but it can still create a situation in which there are multiple connections, albeit for the same video stream.

Multiple TFRC connections 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 [8], 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. Therefore, this paper proposes multiple connection video streaming for WiMAX uplinks.

II. WIMAX ENVIRONMENT

Fixed WiMAX allows rapid deployment of video services in rural districts, where ADSL and cable are uneconomic. Mobile WiMAX allows rapid deployment of video services in areas in the world unlikely to benefit from extensions to either 3G or UMTS. WiMAX's uplink capacity is likely to exceed that of High Speed Downlink Packet Access's 384 kbps, though not a deployed Long Term Evolution service's speeds. In Brazil, mobile WiMAX is the basis of a digital TV (DTV) service [9] but there is also interest in exploitation of UL interactive services. Likewise in Korea, an interactive extension of the WiBro (now harmonized with WiMAX) DTV service [10] is proposed.

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

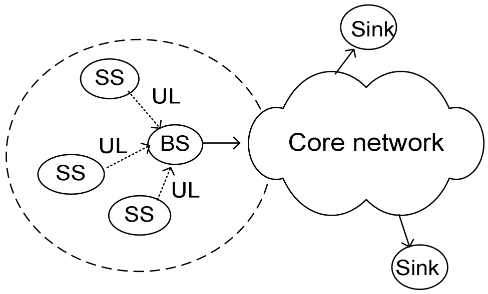

Figure 1. IEEE 802.16 uplink service architecture.

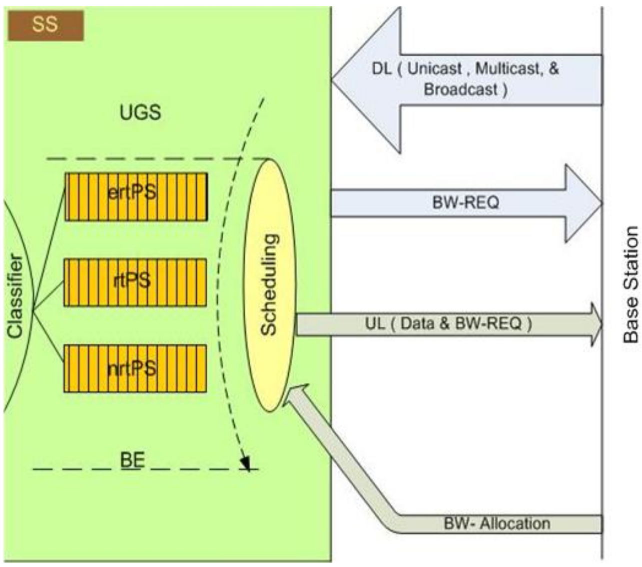

Figure 2. IEEE 802.16 uplink scheduling system.

(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. In particular, these are Unsolicited Grant Service (UGS), extended real-time Polling Service (ertPS) (for voice with silence suppression), real-time Polling Service (rtPS), non-real-time Polling Service (nrtPS), and Best effort (BE). 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. Therefore, the rtPS class is the focus of our study.

Fig. 2 shows how a WiMAX SS communicates with a BS and how the UL packet scheduling works across different service classes. An initial request by the SS results in a bandwidth allocation by the BS to that SS. The SS packet scheduler is able dynamically to request a modification to its bandwidth allocation in a piggybacked requested to the BS. However, for simplicity of analysis, in this paper it is assumed that the allocation remains static. As an SS may not have data available, it is polled by the BS. Polling may be on a group basis or in unicast fashion. As group polling may result in access delay at the SS, this paper assumes unicast polling, though there are UL sub-frame utilisation implications. The

TABLE I
SIMULATED WiMAX SETTINGS.

Parameter	Value
PHY	OFDMA
Duplexing mode	TDD
Frame length	5 ms
Bandwidth	10 MHz
FFT size	1024
DL/UL ratio	1:3
Fragmentation	Yes
Range	1 km

BS passes video and other data to the SS over the downlink (DL) sub-frame and the UL sub-frame accommodates its traffic according to service class and burst slot availability.

The UGS is allocated bandwidth as of priority, while by default SS servicing of polled queues takes place in round-robin fashion (though many other possibilities exist). In Fig. 2 the queues are serviced by the packet scheduler in the order indicated with BE traffic serviced last. We assume an exhaustive service with the default round-robin queue service by the SS packet scheduler. At the three queues, default drop-tail or FIFO queue management was modelled.

In the simulations the queue sizes were all set to fifty packets. This value was selected as it seems appropriate tomobile, 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.

B. Simulation environment

The WiMAX system operating in PMP mode was simulated by well-known ns-2 simulator (v. 2.29 used) augmented by a WiMAX module [11] that has proved to be an effective way of modeling WiMAX's behavior. The PHY settings selected for WiMAX simulation are given in Table I, with the Medium Access Control (MAC) settings defaulted from [11]. The DL/UL ratio in the Table is not intended to be realistic but to aid in testing MULTTFRC, as in practice the DL would be allocated the majority of the bandwidth. The range was set to only 1 km to reduce the propagation delay component of latency, to better judge the queue waiting time.

There were three SS communicating to the BS, with one of the SS sending a Variable Bitrate (VBR) video sequence encoded with the H.264/Advanced Video Codec (AVC) and split between the MULTTFRC connections. A trace file was input to ns-2 and packet losses recorded in the output. The output served to calculate the PSNR. Video quality comparisons were made under the EvalVid environment [12].

The reference 'Paris' clip was encoded at 15 and 30 Hz (frame/s) at Common Intermediate Format (CIF) (352×288 pixel/frame) with 4:2:0 sampling and with initial quantization parameter set to 30. Paris consists of two figures seated around a table in TV studio format, 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

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	

every 30 frames with IPPP… structure. 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.

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 packets, causing slice de-synchronization markers to be missing from some packets. In this sense, our video quality findings are worst case.

For TFRC, the inter-packet interval was varied according to the original TFRC equation [3], not the simplified version of the equation reported in [7]. As described in [3], 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 [3]. 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. Of course, the underlying transport protocol for the FTP streams was TCP. Therefore, including FTP streams in the uplink load allows the response to TCP traffic to be judged.

C. Management of connections

To systematically test the effect of multiple TFRC streams, rather than apply MULTTFRC's connection switching mechanism [2; 7], 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 [2; 7] how a single video stream would be apportioned between a varying number of connections. We suggest one possibility is that a single queue is segmented into GOPs. Each connection takes from the queue according to the most recent GOP available. This would require a queue locking mechanism to prevent simultaneous access to the same GOP.

However, to also aid interpretation, in this paper each connection is statically allocated its GOPs, which are taken in interleaved manner, from the video sequence, leaving adaptive connection access to future work. A reordering buffer is required at the receiver to re-sequence arriving GOPs. This buffer will obviously have an impact upon latency which might be addressed by accessing the video stream at a finer granularity, such as a frame-by-frame basis. However, this aspect of multiple TFRC streaming, which was not considered in [2; 7] is also reserved for future work.

III. RESULTS

In Fig. 3, the gain in aggregate throughput across the WiMAX UL from employing extra connections is illustrated. From the four connection example especially, it is apparent that TFRC acts to staunch the initial over optimistic sending rates. Table III charts the reduction in sending period as the number of connections is increased, which is matched by an increase in throughput. Unfortunately, though the overall sending period is reduced, the per-packet delay on a per-connection basis is not reduced and, in fact, it increases with the number of connections, Table IV. This is explained by the response of TFRC controlling one connection to increased congestion arising from the other congestions. TFRC then reduces the packet sending rate. On average (arithmetic mean) per-packet jitter was in the micro-second range but its maximum value could be high, Table V. Again high jitter is a consequence of TFRC responding to the onset of congestion by varying the inter-packet gap. The impact on delay and jitter have implications for buffer dimensioning at the destination or sink of the video stream and may lead to lack of synchronization with voice or unacceptable delay if there is two-way communication.

The results of streaming multiple connections upon queue length are shown in Fig. 4. Fig. 4 is not intended for exact interpretation but to show the effect of turning on streaming at 20s. Buffer fullness before that time is due to the other traffic sources listed in Table II. From Fig. 4, all multiple connections except just one connection result in packet

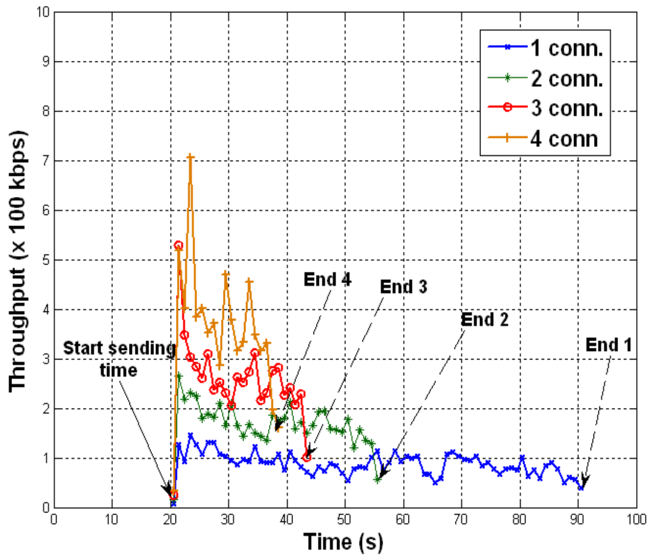

Figure 3. Throughput at 15 Hz for sending the same video sequence with differing numbers of connections.

TABLE III

OVERALL THROUGHPUT

	15 Hz	30 Hz	15 Hz	30 Hz	15 Hz	30 Hz
	Sending period (s)		Maximum (kbps)		Average (kbps)	
1	70.47	32.97	258.63	872.58	120.83	477.43
2	35.47	18.97	505.73	877.00	237.65	451.98
3	23.47	18.47	718.42	876.43	353.91	462.87
4	17.94	17.97	800.13	889.04	457.13	450.67

TABLE IV

PER-PACKET DELAY

	15 Hz	30 Hz	15 Hz	30 Hz
	Average (s)		Maximum (s)	
1	0.006	0.022	0.073	0.119
2	0.013	0.066	0.131	0.194
3	0.020	0.085	0.182	0.192
4	0.028	0.089	0.192	0.204

TABLE V

PER-PACKET JITTER (MAX.)

	15 Hz	30 Hz
	Maximum (s)	
1	0.00458	0.00458
2	0.07195	0.00864
3	0.08567	0.10055
4	0.11864	0.13500

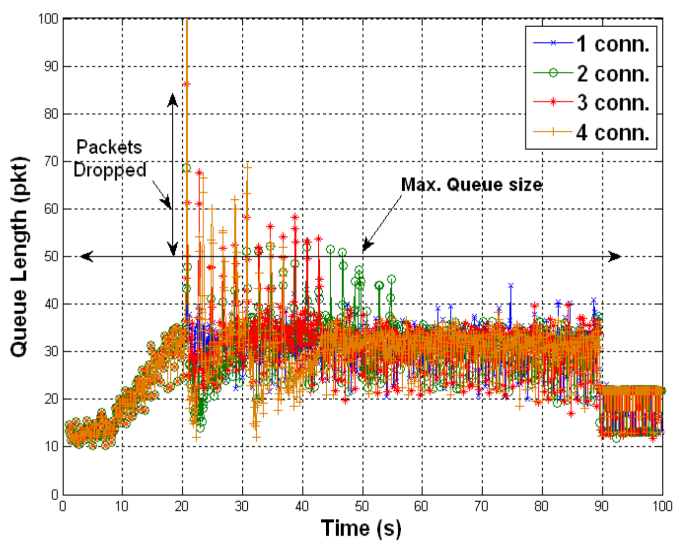

Figure 4. Queue length at 15 Hz for sending with differing numbers of connections.

overflow at the buffer, even at the reduced rate of 15 Hz. Of course, the initial sending rate of four-connection streaming could be restrained by adaptation or staggered start times. However, this would either reduce the throughput considerable or result in a start-up delay of around 20 s. For four connections, this would approximately double the sending time for this particular video sequence. In fact, both these consequences could arise, delay and reduced throughput. Moreover, when turning to Fig. 5 for 30 Hz streaming, packet losses can occur some way into the streaming session (for one and two connections) due to the coincidence of VBR sending bursts and the background WiMAX traffic, including other rtPS queue traffic.

PSNR for one connection at 15 Hz is without packet loss and, therefore, represents the maximum achievable quality after application of the codec. In fact, video quality is good at all connection levels for 15 Hz, Table VI. Unfortunately, when a voice signal is synchronized with video, the jerkiness of the display at 15 Hz begins to disturb the Quality-of-Experience (QoE). However, at 30 Hz video quality is reduced and becomes barely acceptable with two connections.

We have applied no error resilience protection [13] to the Paris sequence and it is likely that the quality could be improved by application of error concealment, possibly combined with enhanced error concealment (other than previous frame replacement). This reduce coding efficiency due to the relative overhead arising from error resilience, particularly for low bit rate video. However, in view of the

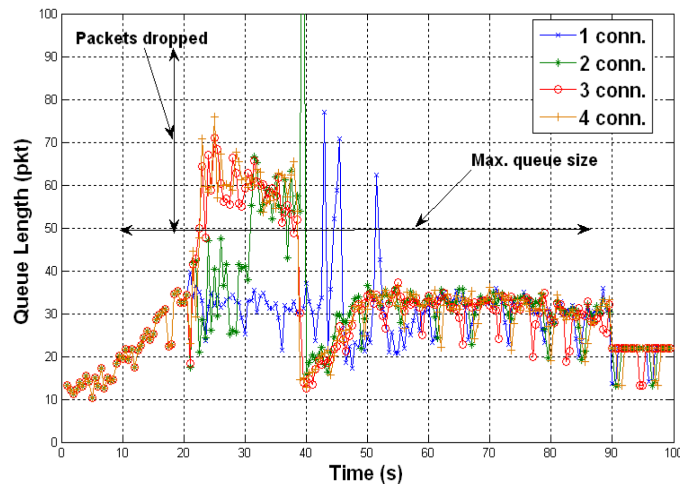

Figure 5. Queue length at 30 Hz for sending the same video sequence with different numbers of connections.

TABLE VI

VIDEO QUALITY

	15 Hz	30 Hz
	PSNR (dB)	
1	36.02	34.13
2	33.21	21.92
3	30.86	18.57
4	28.13	14.92

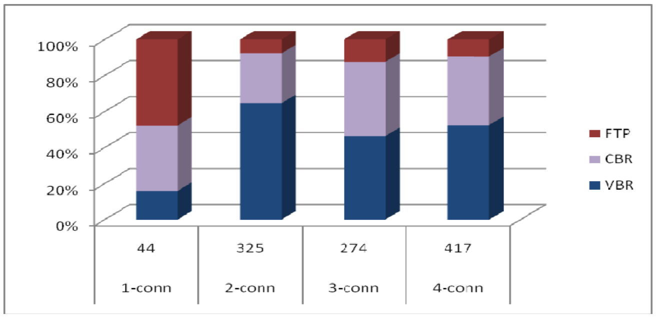

Figure 6. Packet loss distribution for 30 Hz streaming

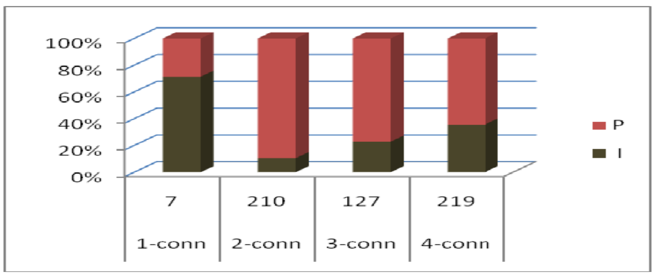

Figure 7. Packet loss distribution for video packets by frame type for 30 Hz

increased throughput available from MULTTFRC and because error resilience provision is advisable across a wireless channel, this would be the next step in investigations.

Of course, channel errors will also lead to deterioration of video quality with or without error resilience. In [7], Additive White Gaussian Noise was modeled, but this type of noise represents the most favorable conditions that could exist for MULTTFRC streaming over wireless, while 'bursty' error conditions from fast and slow fading are likely to occur for mobile applications.

The distribution of packet losses between traffic types is shown in Figs. 6 for 30 Hz video. The number of lost packets over the period of the test session concerned is shown underneath the bars. Though no detailed findings can be read into the results, there does seem to be a trend towards balancing losses between the three types of traffic source and their relative weighting in the throughput, as the video throughput increases.

An interesting feature of these results is that at 30 Hz streaming more video packets are lost with two connections than with one (and in fact more total packets are lost). However, in Table V1 the video quality with two connections is on average superior to that of three connections. This is explained by the similar numbers of I-frame packets lost in both cases (refer to Fig. 7) as these frames have a strong impact on the overall video quality. This pattern of behavior was confirmed by us through additional experiments.

IV. CONCLUSIONS

As manufacturers opt for all-IP networks, there is a need to engineer the transition across underlying heterogeneous networking technologies. A combination of best effort Internet with wireless broadband is a case in point. In this paper, the issue of effective congestion control has been examined with the video streaming source lying at the wireless subscriber station rather than an Internet server. In fact, it is more logical to use multiple-connection streaming in interactive applications, because an Internet server will *already* benefit from multiple connections, simply because it will have multiple clients. We have found that a single client employing multiple connections for a single stream will gain if there are a small numbers of connections. Future work will assess active queue management, other than drop tail and will also include the effect of realistic channel conditions on the MULTFTRC solution.

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