Applying Multi-Connection Video Streaming to WiMAX Broadband Wireless

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***Abstract*—Streaming a single video over multiple TCP-Friendly Rate Control (TFRC) connections is a way of separately coping with both wireless channel losses and traffic congestion, without the need for cross-layer intervention or retransmission delay at the data-link layer. At the same time, the wireless channel is properly utilized, as throughput improves with an increasing number of connections. Nevertheless over IEEE 802.16e (mobile WiMAX), tuning is needed to select the number of connections and the Time Division Duplex (TDD) frame size. The paper assesses the impact on video quality of packet drops due both to channel loss over a WiMAX access link and router buffer overflow across an all-IP network, consisting of broadband wireless access and core network. The paper also considers end-to-end delay and start-up delay when employing several connections. Results show that provided the TDD frame size is selected appropriately then using multiple connections preserves video quality and improves wireless channel utilization, with a minimal impact on end-to-end delay. As a trade-off, there is an increase in start-up delay arising from the need to avoid possible buffer underflow**.

***Index Terms*—Multi-connections, TFRC, video streaming, WiMAX**

I. INTRODUCTION

In this paper, we consider multi-connection TCP-friendly Rate Control (TFRC) [1] video streaming across a network consisting of a wireless access network serving an all-IP core network [2]. We assume IEEE 802.16e (mobile WiMAX) [3] broadband access, though the results will broadly be applicable to other packet-based wireless solutions. In such a network, a video stream is subject to packet loss due to wireless channel conditions *and* from traffic congestion on the wired network. We also model traffic congestion on the access network.

Recent proposals for congestion control of video streaming across broadband networks [4][5][6] (further examined in Section II.B) consider single-connection TFRC combined with cross-layer intervention for the wireless network. Our contribution is to adapt the multi-connection approach for wireless LANs [1] to broadband networks. In this new environment, we show that packet re-ordering at the receiver, multiplexing of the video stream at the sender, and selecting an appropriate wireless configuration cannot be taken for granted. Therefore, our contribution is to take the multi-connection system beyond the proof of concept stage and show the advantages and limitations of the approach. We would expect other contributions to follow in which different forms of multi-connection streaming are explored, just as the cross-layer approach has resulted in a number of variants, each of which thoroughly explores this research space.

In multi-connection TFRC video streaming, a *single* video source is multiplexed onto several connections across the wireless link in order to increase the throughput, thereby improving wireless channel utilization. By multiplexing a video stream across multiple connections in this way it is hoped that the impact of packet loss on one or more of these connections will be mitigated by the datarate across the remaining connections. TFRC's main role when congestion occurs across the network path is to reduce the video streaming data-rate across the wired portion of the concatenated network. It does this in response to packet drops at intermediate routers, which signal the presence of contending traffic. Unfortunately, TFRC can misinterpret as congestion packet losses due to wireless interference and noise, leading to a reduction in wireless channel utilization. Though cross-layer approaches to avoid misinterpretation are possible, these are complex to implement and inflexible. In fact, cross-layer approaches are most appropriate when a network has a fixed application, not one in which multimedia streaming is mixed in with other types of traffic.

In pioneering work on multi-connection TFRC, MULTTFRC [1] improved video quality by increasing the quantity of video data that could be sent over the multiple connections. Of course, increased video data implies a lower compression ratio and, hence, higher quality delivered video, provided the rise in packet losses across the wireless channel does not degrade the quality. If burst errors occur then during the time that they occur all connections are affected, leading to a rise in packet losses, which was countered in [1] by means of application-layer forward error correction. Unfortunately, if the number of connections varies, as it does in [1], then sending rate oscillations can occur. If the compression ratio is varied at the source to adapt to the sending rate oscillations then there is the risk of

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disconcerting changes in the displayed video quality at the user's display. (The compression ratio can be varied either by changing the quantization parameter at the encoder for live video or through a bit-rate transcoder for pre-encoded video.)

However, we show that, by keeping the number of connections constant, delivered video quality can be maintained without the need to dynamically change the compression ratio. This is because, with multiple TFRC connections, TFRC is better able to control its sending rate. In fact, TFRC was originally designed [7] with a high number of streams in mind, as may arise from a Video-on-Demand server, and special measures are recommended [7] if the number of contending flows is *not* large enough.

Our results show that provided the WiMAX Time Division Duplex (TDD) frame size is selected appropriately then using multiple connections preserves video quality, as a result of the differential effect of packet loss patterns. Wireless channel utilization is considerably improved, with a minimal impact on end-to-end delay. As a trade-off, there is an increase in start-up delay arising from the need to avoid possible buffer underflow, though this is smaller compared to TCP-based streaming, when large buffers are normally employed [8] to cater for the possibility of repeated retransmissions. For example, in [9] a buffer size of 10 s was required to counter wireless burst errors. However, for a WiMAX uplink effective streaming can only be achieved if the TDD frame length is tuned to avoid queue servicing scheduling delays. The frame length is significant as a longer frame reduces delay at a WiMAX subscriber station, thus permitting more data to be removed from queues when the subscriber station's queues are polled.

It has been suggested [4] that in Long Term Evolution (LTE) [10] packet loss can be virtually eradicated by retransmission at the data-link layer. However, that approach has the potential to introduce unbounded delay across the wireless link, apart from the drop in throughput that results. The approach also reintroduces the problems that led to the search for an alternative to TCP transport for multimedia streaming. There is also the overhead of maintaining state at the evolved node B (an LTE radio head) and the delay arising if retransmissions are still going on when a handoff occurs. Therefore, we consider that further investigation of multiple TFRC connections is a way forward for multimedia streaming over broadband networks.

II. RELATED WORK

A. WiMAX background

IEEE 802.16 (known as WiMAX) [1] [11] allows rapid deployment of multimedia services in areas in the world unlikely to benefit from extensions to both 3G systems such as High Speed Downlink Packet Access (HSDPA) and 4G UMTS such as LTE. WiMAX's current uplink capacity is likely to exceed that of HSPDA's 384 kbps, though not LTE's potential bandwidth. In Brazil, mobile WiMAX is the basis of a digital TV (DTV) service but there is also interest in exploitation of uplink interactive services [12], which could involve video streaming. WiBRO in Korea is also now harmonized with WiMAX and interactive multimedia services are also proposed [13]. Uplink video streaming is a value-added addition to a broadcast IPTV service. As such it is attractive to commercial operators, as it could be used for mobile video conferencing and video telephony, as well as the exchange of personal video clips.

All-IP [2] or Next Generation Networks are being widely developed (for instance in the UK in BT's 21CN, in the Netherlands by KPN) as an economically-effective replacement for traditional telephony networks. In video streaming across an all-IP network unreliable UDP transport serves to reduce delay at the expense of some packet loss, while application-layer TCP emulation [14], such as TFRC, acts as a form of cooperative congestion control (assuming most other traffic is carried through TCP transport). TCP emulation is introduced to avoid upsetting the cooperative form of congestion control exercised by TCP, which remains the dominant transport protocol within the Internet. However, TCP emulation by the application is *not* the same as TCP. TCP itself is unsuitable for delay-variation 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 exaggerated 'saw-tooth'-like rate fluctuations that arise from TCP's aggressive congestion control algorithms, which cause disconcerting quality fluctuations at an end-user's display.

B. TFRC over wireless

The research in [1] originally proposed MULTTFRC as a form of downlink control. Misinterpretation of channel loss as congestion is the source of under-utilization if a single TFRC connection were to be used. Any single TFRC connection responds to packet loss by reducing its output rate by increasing the inter-packet gap and reducing its throughput. In MULTFRC, other connections not affected by the packet loss can balance out the drop in throughput over the wireless part of the path. MULTTFRC represents a lightweight way to retain TFRC for the Internet path but avoid more complex means of suppressing channel loss feedback to TFRC over the wireless link. In the most definitive account of MULTTFRC so far, the work in [15], there is no explanation of how a single video stream is multiplexed onto the multiple connections using dynamic scheduling or what the resulting video quality is in quantitative terms. Only generic packet loss and delay statistics are reported, even though the type of error pattern is known to change the video quality (PSNR) by several dBs. Other work on MULTTFRC and its variants such as [16] appears largely confined to analysis of a generic link without other traffic.

Research in [4] applied single connection TFRC over the emerging LTE 4G cellular wireless system. TFRC was combined with the Scalable Video Coding (SVC) extension to H.264, in such a way that the layering was adapted to the bitrate. The results are encouraging in terms of reduced packet losses, reduction of streaming interruptions and end-to-end delay, and buffering levels compared to not using TFRC. However, at the time of writing, these results for LTE rely on a stand-alone emulator, which does not include the effect of transport across the core network. Another issue is to what extent this approach is dictated by the need to avoid the loss of key frame packets from the base layer of SVC, after which it becomes difficult to reconstruct the stream at the decoder.

It is also possible to modify TFRC to detect packet losses occurring due to wireless channel conditions. In [16], a reassembly failure at the Radio Link Control (RLC) layer signals signaled such losses to a 3G SS. Feedback packets contained an estimate of the wireless channel packet loss rate. However, this approach [16] assumes an absence of congestion at the BS and requires cross-layer interaction. It is also reliant on the safe arrival of the feedback packets.

The Data Congestion Control Protocol (DCCP) [17] includes a TFRC-like component. In [6], SVC is again used in conjunction with DCCP in a scenario in which there are wireless access links to the core network at the sender and receiver sites. Therefore, that work [6] investigated uplink as well as downlink streaming. Similarly to [16], lower layer information was employed to modify TFRC's estimate of packet loss rates. In this case, the PHY layer ARQ information was used. The work in both [16] and [6] appears to have been anticipated in [18], though TFRC was not used in that work. Both [6] [16] are cross-layer modifications and, hence, suffer from the need to accord TFRC special treatment compared to other traffic.

In [19], multiple connections over an IEEE 802.16e link were used to send different layers of a scalable video stream. Unfortunately, the scheme was only tested for two layers and video quality evaluation does not appear to have accounted for lost packets. As mentioned previously, this is an important issue in H.264/SVC, 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.

Alternatively, many methods [20] have been investigated to modify the behavior of wireless routers or base stations when dealing with TCP. It is possible that these suggestions could be extended to TFRC or DCCP. In [21], the wireless router sends an arrival acknowledgment to the source so that the source is able to judge that packet loss over the wireless channel has occurred if the normal TCP acknowledgment from the mobile destination does not arrive. Interestingly, in this early work [21], the idea of repeated transmissions by the wireless router is rejected because of the extra state at the router in terms of connection tables and retransmission timers that must be re-established if the MS moves from one cell to another. In [21] the possibility of packet loss at the wireless link is made explicit, whereas in the approach exemplified by Snoop [20], 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. By suppressing channel loss reporting, Snoop allows a congestion controller to respond only to packet loss by congestion.

III. Scenario Investigated

The scenario tested in this paper is shown in Fig. 1. The following describes the WiMAX part and this description is followed by a description of the inset in the Figure, showing traffic sources and sinks within the core IP network.

A. *WiMAX system*

In Fig. 1, once a Base Station (BS) has allocated bandwidth to each subscriber station (SS), each SS must manage its queue according to the data arrival rate from user applications. 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 as this discipline is the default. Queue sizes in tests 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 at the buffer's memory. Access to the SS service class queues was round-robin.

The physical layer (PHY) settings selected for WiMAX simulation are given in Table I. The antenna is modeled for comparison purposes as a half-wavelength dipole. The Gilbert-Elliott 'bursty' channel model is further explained in Section III.C. The TDD frame length was varied in experiments, because it has an important effect on the service rate at an SS. Current implementations have apparently mostly opted for a fixed 5 ms TDD frame size, though this is difficult to verify. The uplink(UL)/downlink (DL) ratio is adaptable at the BS and was set to favor the UL for the purposes of our tests.

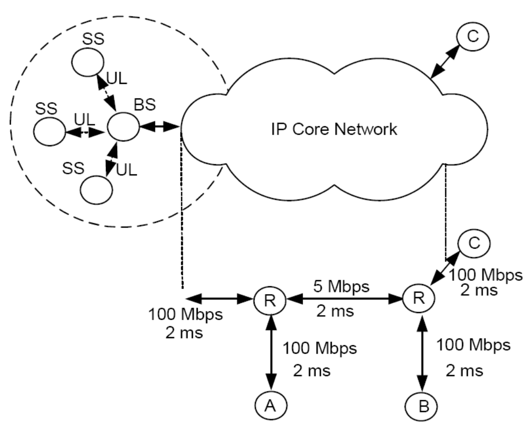

Figure 1. Network scenario with inset showing routing across the core network, A, B and C being sources and sinks, and R = router

TABLE I. SIMULATED WIMAX SETTINGS

Parameter	*Value*
PHY	OFDMA
Frequency band	5 GHz
Duplexing mode	TDD
Frame length	5 to 20 ms
Max. packet length	1024 B
Raw data rate	10.67 Mbps
IFFT size	1024
Modulation	16-QAM 1/2
Guard band ratio	1/8
DL/UL ratio	1:3
DL length	1.25 to 5 ms
UL length	3.75 to 15 ms
Channel model	Gilbert-Elliott
MS transmit power	250 mW
BS transmit power	20 W
Approx. range to SS	0.7 km
Antenna type	Omni-directional
Antenna gains	0 dBD
MS antenna height	1.5 m
BS antenna height	32 m

B. *WiMAX traffic characteristics*

There were three SSs communicating to the BS, with one of the SS sending a VBR video sequence encoded with H.264/AVC (Advanced Video Coding) [22] and split between the multiple TFRC connections. The other SSs are introduced as sources of contending 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 the well-known network simulator ns-2 and packet losses recorded in the output. The output served to calculate the PSNR. Video quality comparisons were made under the EvalVid (version 2) environment [23]. Data points are an average of fifteen runs. The output allowed calculation of the Peak-Signal-to-Noise ratio (PSNR) as an objective measure of video quality. As a test, we used the *Paris* sequence (VBR)-encoded at 30 frame/s in Common Intermediate Format (CIF) (352×288 pixel/frame) with initial quantization parameter set to 26 (from a range 0 to 51). *Paris* consists of two figures seated around a table in a TV studio setting, with high spatial coding complexity. The Intra-refresh rate was every 15 frames with IPBB…I coding structure, i.e. the Group of Pictures (GOP) size was 15. 1063 frames were transmitted. Previous Frame Replacement (PFR) was set for error concealment at the decoder for comparison with coding results, which assume PFR. The slice size was fixed at the codec as 900 B. In selecting codec determination of slice size, packet segmentation is avoided, which improves video quality, as slices are not separated from their resynchronization headers.

Table II records the simulated traffic characteristics for the three SSs communication with the BS. Network Adaptation Layer units (NALUs) from the H.264/AVC codec were encapsulated with Real Time Protocol (RTP) headers. After the addition of IP headers, these in turn formed a single WiMAX MAC Packet Data Unit (MPDU), which are variable-sized WiMAX packets. For simplicity, a WiMAX MPDU is now referred to as a packet. Coexisting rtPS queue CBR sources were all sent at 1500 kbps, i.e. at a similar rate to the video source. The inter-packet gap was 0.03 s for the CBR traffic. The FTP applications, which continuously supplied data according to available bandwidth, 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 was selected to fully occupy the DL link capacity.

For TFRC, the inter-packet sending time gap was varied according to the TFRC equation [7], not the simplified version reported in [15]. TFRC [7] 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. A throughput equation models TCP New Reno to find the sending rate:

$$TFRC\ (t_{rtt}, t_{rto}, s, p) = \frac{s}{t_{rtt}\sqrt{\dfrac{2bp}{3}} + t_{rto}\ \min\left(1, 3\sqrt{\dfrac{3bp}{8}}\right) p\,(1 + 32\,p^2)} \qquad (1)$$

where t_{rtt} is the round-trip time, t_{rto} is TCP's retransmission timeout, s is the segment size (TCP's unit of output) (herein set to the packet size), p is the normalized packet loss rate, w_m is the maximum window size, and b is the number of packets acknowledged by each ACK. b is normally set to one and $t_{rto} = 4t_{rtt}$. It is important to notice that t_{rto} comes to dominate TFRC's behavior in high packet loss regimes, which is why it is unwise to use a simplified form of (1). General inspection of (1) indicates that if the round-trip time and/or the packet loss rate increase then the throughput reduces as terms containing these parameters exist in the denominator.

In our variant to standard TFRC, the packet size, s, in the TFRC equation (1) was dynamically altered according to EvalVid-created trace file sizes. This variant makes for more responsive control rather than the mean packet length employed in the reference TFRC formulation [7]. TFRC was originally intended for video-on-demand applications, when it is feasible to calculate the mean packet length from the stored video. Setting a mean packet length is inappropriate for interactive multimedia applications. The underlying TFRC transport protocol was set to UDP, as is normal for real-time applications. Though (1) appears to represent a considerably computational task that could impede real-time performance, it is possible to extract a term parameterized by p, the packet loss rate. Therefore, a look-up table indexed by p represents a practical way to speed up calculations.

TABLE II. SIMULATED WIMAX TRAFFIC CHARACTERISTICS

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

C. Channel model

A Gilbert-Elliott two-state, discrete-time, ergodic Markov chain [24] modeled the wireless-channel error characteristics at the ns-2 simulator physical layer. The result of applying this model is that burst errors typical of known wireless channel conditions during fast fading appear. 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 and the bad state default was 0.05.

D. Core network traffic characteristics

In Fig. 1, all links except a bottleneck link within the core network are set to 100 Mbps to easily accommodate the traffic flows entering and leaving the network. The link delays are minimal (2 ms) to avoid confusing propagation delay with re-ordering delay in the results. A bottleneck link with capacity set to 5 Mbps is set up between the two routers. The buffer size in each router was set to 50 packets. This arrangement is not meant to physically correspond to a network layout but to represent the type of bottleneck that commonly lies at the network edge.

Node A sources to node B a CBR stream at 1.5 Mbps with packet size 1 kB and sinks a continuous TCP FTP flow sourced at node B. Node B also sources an FTP flow to the BS and a CBR stream at 1.5 Mbps with packet size 1 kB (see the Table II downlink section). Other SS sources apart from the video connections do not pass over the core network shown but are assumed to be routed elsewhere after passing the WiMAX BS. Node C in Fig. 1 is the sink for the TFRC multiple connections.

E. Management of connections

To systematically test the effect of multiple TFRC connections the number of TFRC connections was incrementally stepped up in successive experiments. In MULTTFRC [1] [15] the number of connections was changed over time according to the average round-trip time of all the connections, but this procedure hides the interpretability of results. It is also unclear from [15] how a single video stream would be apportioned between a variable number of connections. In our experiments, a single queue was segmented into GOPs (one GOP = 15 frames). Each connection was statically allocated its GOPs, which are taken in interleaved manner from the video sequence. This assumes that a re-ordering buffer is available at the receiver, the size of which is discussed in Section IV.

IV. EVALUATION

A. Initial investigations

Initial investigations considered the WiMAX link alone in Fig. 1. Table III shows the average data-rate when transmitting the Paris clip over one or more connections, for two different WiMAX frame lengths: 5 ms and 20 ms. Allowable frame lengths are specified in the Standard [3], ranging from 2.5 to 20 ms. 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 sharply reduces its overall sending rate in response to packet loss. Because the sending period for one connection with the shorter frame duration is more than the display period of the *Paris* clip, the longer frame length is preferable if only one connection were to be used. However, with more than one connection, throughput and, hence, wireless channel utilization by the congestion-controlled video streams increases significantly. There is a marked difference if the larger frame length is used whether with one or four connections. As smaller frame lengths than 20 ms are generally employed for WiMAX, this is an important observation. In fact, the UL proportion of the frame length, that is 15 ms, is more than the total 5 ms frame length that appears to be normally implemented.

B. Including the core network

In respect to the longer frame length of 20 ms, an interesting comparison is with the throughput when the core network is included. In Table IV, there is a similar pattern to the throughputs in Table III but the rates are reduced to when streaming only over the WiMAX link. We interpret this effect as not being due to TFRC's response to packet loss but being due to its response to the increased round trip time caused by queuing delay in the buffer prior to the bottleneck link in Fig. 1. Notice that TFRC uses reliable TCP to return ACKs, which will tend to add to the round-trip time. Recall also that from equation (1) round-trip-time is one of the parameters determining TFRC's sending rate. This interpretation is confirmed by the increase in per slice/packet end-to-end delay as more connections are added. In effect, the packets from other connections intervene in the router buffers causing an increase in latency. However, even though the delay is larger for four connections, the mean is still less than 100 ms for this scenario.

More significant than end-to-end delay for reconstruction of the video stream is GOP arrival ordering, as this ordering has the potential to introduce interruptions to the display. GOP arrival ordering for four connections is shown in Fig. 2. Notice that the first H.264/AVC GOP contains parameters that are fixed throughout the sequence [22]. Therefore, this GOP is transported more quickly. To avoid a sudden injection of

TABLE III. SENDING PERIODS AND THROUGHPUTS WHEN STREAMING FROM A MOBILE SS TO THE WIMAX BS

No. of connections	*SS to BS (s) (frame length 5 ms)*	*Throughput (kbps)*	*SS to BS (s) (frame length 20 ms)*	*Throughput (kbps)*
1-conn	71.4	217	33.5	467
2-conn	35.8	437	20.5	754
3-conn.	23.3	663	17.7	874
4-conn.	17.4	889	14.6	1059

TABLE IV. STREAMING PERIODS, THROUGHPUTS AND MEAN PACKET END-TO-END DELAYS FROM MOBILE SS TO NODE C IN FIGURE 1 (FRAME LENGTH 20 MS)

No. of connections	*Sending Period (s)*	*Throughput (kbps)*	*Mean end-to-end delay (s)*
1-conn	35.2	444	0.035
2-conn	22.4	690	0.036
3-conn.	21.6	716	0.039
4-conn.	15.6	991	0.062

traffic into the network, connection starting times were offset by 0.5 s. In respect to the general findings, a noticeable feature of this Figure is the lengthier start-up periods in sending initial GOPs on each of the connections. We attribute this to the loss of packets at an early stage, which causes TFRC to sharply reduce its rate in a similar manner to TCP's slow-start mechanism. This does mean that about 6 s of frames (amounting to 90 frames) should be stored in the reordering buffer, to avoid the possibility of subsequent underflow in the decoder's playout buffer. As the destination is on the fixed network the reorder buffer is not expected to be a drain on energy resources, as it might be on an SS. Of course, data are not physically reordered in the buffer but accessed through pointers. 6 s is longer than a typical start-up time of around 2 s but is not too large to be objectionable to the user.

Returning to the effect of frame length, video quality (PSNR) and mean packet end-to-end delay were found for a range of WiMAX frame lengths. However, the standard deviation (stdv) over the runs is relatively large (but similar to those reported in [15]). This is explained by the strong effect resulting from the position of error bursts. From Table V, video quality is generally 'good', as there is an approximate equivalence of PSNR's over 31 dB and above to the ITU's subjective scale. Again the larger TDD frame size results in better and in the mean surprisingly in this instance improves with an increasing number of connections. However, we take this to signify that using four connections produces equivalent video quality at the destination to using one connection, provided the larger frame size is employed. A 5 ms frame size consistently reduces the quality by one or two dB, which on a logarithmic scale is significant.

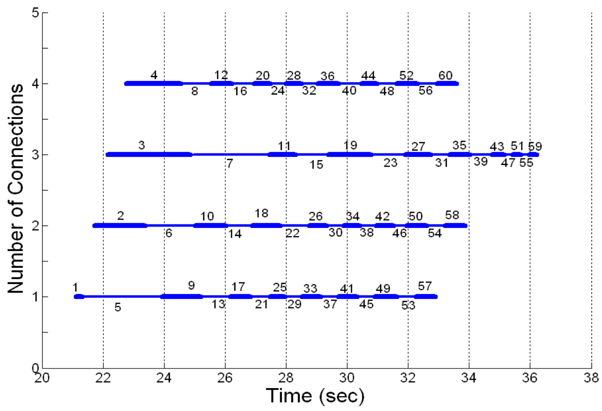

Figure 2. Example GoP arrival sequence at the receiver (node C in Fig. 1) showing the start and end times of each GoP

Examination of the total packet losses (congestion and channel loss), Fig. 3, shows that losses also are generally higher for a 5 ms TDD frame length than a 20 ms frame length. However, between the connections, it is *not* the case that mean PSNR is a direct reflection of mean packet loss. As might be expected, employing four connections leads to an increase in congestion loss and also channel loss (because error bursts affect more than one connection). Examining the relative breakdown between frame types, shows that anchor frames (I-frames) and reference frames (I- and P-frames) are evenly affected whatever the number of connections. Therefore, we conclude that the differences in the mean PSNRs are explained by the relatively low number of packet losses when using congestion control and possibly the volatility in the pattern of packet losses when burst errors occur.

From Fig. 3, packet loss is particularly high for three connections. More generally, the advantages of using four connections in terms of improved wireless utilization and video quality equivalent to one connection are offset by the increased mean end-to-end packet delay, Table V1. However, as remarked earlier, the mean is still below 100 ms in this scenario.

C. Detailed time-wise analysis

Fig. 4 is a time-wise plot of throughput resulting from TFRC for a sample four connections with two different WiMAX frame sizes. The plots are the

TABLE V. MEAN PSNR FOR A RANGE OF FRAME LENGTHS WHEN STREAMING FROM A MOBILE SS TO NODE C IN FIGURE 1

	1-conn.		*2-conn.*		*3-conn.*		*4-conn.*	
	PSNR (dB)							
Frame length:	*Mean*	*stdv*	*Mean*	*stdv*	*Mean*	*stdv*	*Mean*	*stdv*
5 ms	29.95	2.90	28.88	3.07	29.54	3.25	28.12	3.11
8 ms	32.85	3.32	31.28	3.81	31.07	3.04	31.92	3.51
10 ms	32.44	3.45	32.17	3.38	31.83	3.14	31.20	3.45
12.5ms	33.22	3.45	33.07	3.47	30.79	3.25	33.31	3.80
20 ms	31.84	3.78	32.34	3.49	33.15	3.68	33.34	3.74

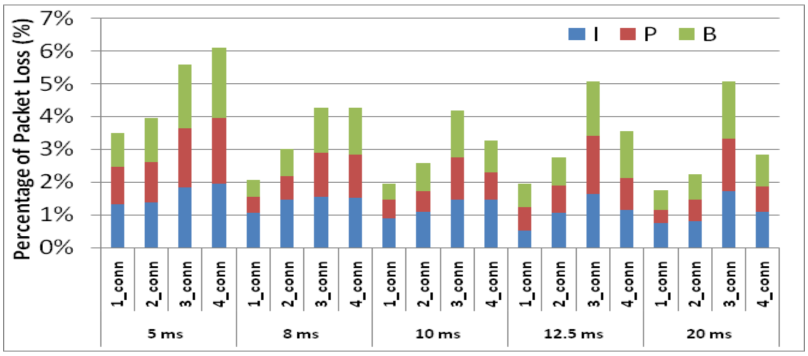

Figure 3. Mean percentage packet loss broken down according to frame type, by connection and TDD frame length, I = Intra-coded frame, P = Predictively-coded frame, B = Bi-predictively coded frame.

TABLE VI. MEAN PACKET END-TO-END DELAY FOR A RANGE OF FRAME LENGTHS WHEN STREAMING FROM A MOBILE SS TO NODE C IN FIGURE 1

	1-conn.	*2-conn.*	*3-conn.*	*4-conn.*
Frame length:	*Mean end-to-end delay (s)*			
5 ms	0.020	0.045	0.073	0.098
8 ms	0.017	0.029	0.024	0.069
10 ms	0.017	0.029	0.028	0.067
12.5ms	0.020	0.027	0.023	0.073
20 ms	0.035	0.036	0.039	0.062

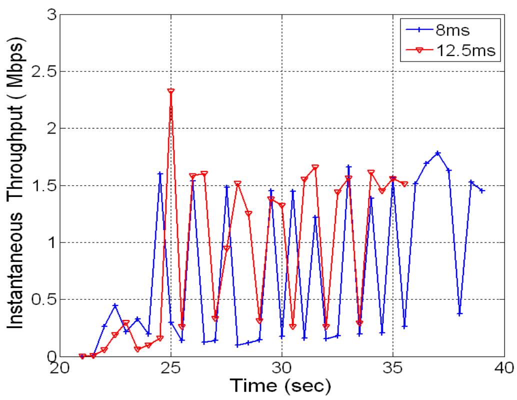

Figure 4. Throughput over time for two selected frame sizes with aggregate instantaneous throughput for four connections.

instantaneous aggregate throughput of all four connections. From the Figure it is apparent that one consequence of multi-connection TFRC is that the oscillations in throughput are deeper with four connections than with one, given that single stream TFRC is designed to be smoother than TCP. Another feature is the evident synchronization of the four connections throughput. However, despite the oscillations, the presence of the reordering buffer will restore the smoothness of the overall video stream.

The two inputs to TFRC (refer to equation (1)), end-to-end delay and packet loss are shown in Figs. 5 and 6. It should not be surprising that, given the oscillations in throughput in Fig. 4, in these Figures end-to-end delay and to some extent packet loss also oscillates over time. Notice also the high packet losses that occur at the beginning of the streaming period

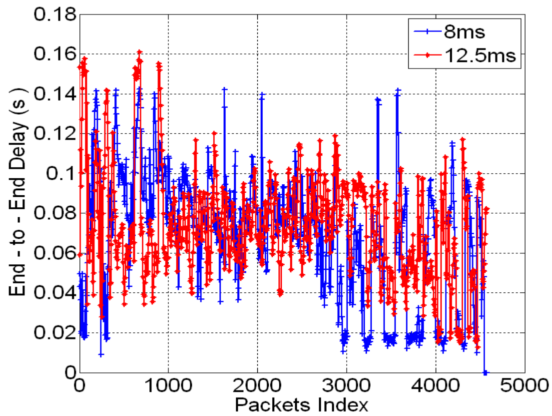

Figure 5. Per-packet end-to-end delay for two selected frame sizes for four connections.

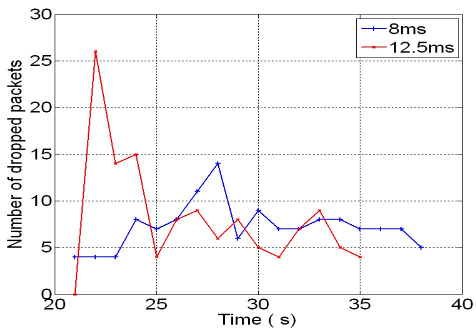

Figure 6. Packet drops over time for two selected frame sizes for four connections.

leading to a staunching of the rate across the TFRC connections for the 12.5 ms frame size.

The variation of performance with different SNR, actually the probability of packet loss in the Gilbert-Elliott bad state, P_B, (refer to Section III.C) as there is no easy translation from this metric to SNR, is given by the change in packet loss rate in Fig. 7. This did not present any significant variations between the number of connections and performance increases in an approximate linear fashion.

D. Including a downlink stage

The system of Fig. 1 can be extended to take into account the effect of downlink wireless streaming, as illustrated in Fig. 8. In the Figure, an additional

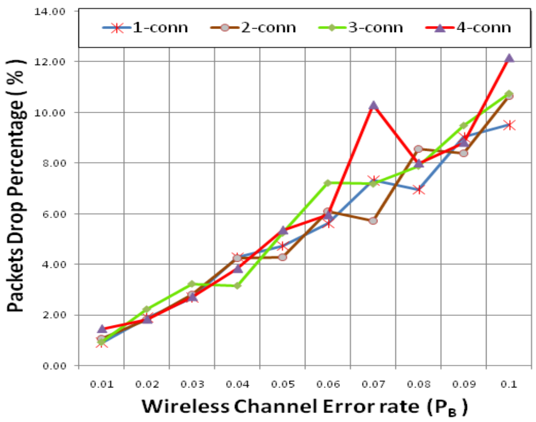

Figure 7. Packet drop rate with increasing probability of error in the Gilbert-Elliott bad state (P_B).

WiMAX connection replaces the wired node C in Fig. 1. Therefore, because of this extra wireless link contributing additional channel losses, it is necessary to increase the number of connections to maintain throughput and, hence, wireless channel utilization. Results for this configuration are given in Table VII. Mean end-to-end delay increases as a result of the extra wireless stage. However, this delay is less than it might have been, because, from experiments, it was realised that multiplexing on the basis of GOPs introduces excessive delay over a longer path. Therefore, a lower granularity of multiplexing was introduced by multiplexing between connections at the slice level.

For one and four connections, the video quality is lower than when there is no extra wireless link but it is still of acceptable quality. However, the sending period for one connection is much longer than the display time of the clip and, hence, is unacceptable. Increasing the number of connections beyond four connections decreases the sending period, at some cost in delivered video quality.

E. Comparisons

Compared to single-connection TFRC over an LTE network [4], if extra data-link layer delay is introduced then we anticipate the scheme of [4] will perform over WiMAX as in the TFRC results for one connection included in this paper but with extra end-to-end delay. Other TFRC schemes with cross-layer assist reviewed in Section II are not comparable to ours as they require extra implementation complexity to disguise channel losses from the TFRC congestion controller. It is also possible that a different congestion controller could outperform TFRC but this would be to neglect the advantage conferred upon TFRC by virtue of its standardization as DCCP [17], rendering it compatible across different platforms. Example alternative research congestion controllers are TEAR [25] or a congestion controller designed with wireless in mind such as LDA++ [26]. We also checked the performance of employing UDP without benefit of congestion control in single and multiple connection form.

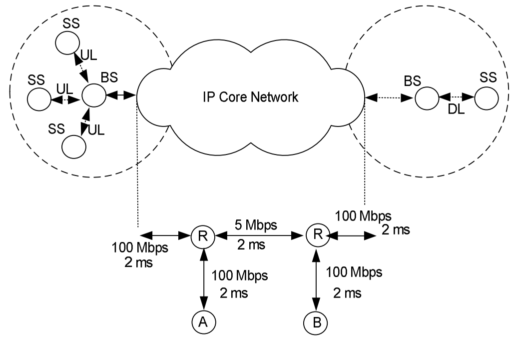

Figure 8. Network extension of Figure 1, with additional wireless downlink to mobile user.

TABLE VII. MEAN PERFORMANCE FOR DOWNLINK AS WELL AS UPLINK STREAMING (20 MS FRAME SIZE)

No. of connections	*Sending Period (s)*	*Throughput (kbps)*	*End-to-end delay (s)*	*PSNR (dB)*
1-conn	179	100	0.139	23.51
4-conn	60	466	0.081	28.99
6-conn.	46	591	0.077	28.54
8-conn.	39	706	0.076	28.50

TABLE VIII. MEAN PERFORMANCE FOR UDP TRANSPORT (8 MS FRAME SIZE)

No. of connections		*Sending Period (s)*	*Thru-put (kbps)*	*Delay (s)*	*PSNR (dB)*
To node C only	1-conn.	35	445	0.062	31.73
	4-conn.	10	1645	0.195	17.14
User-to-user	1-conn.	137	214	0.016	30.10
	4-conn.	34	714	0.025	24.60

Table VIII shows mean performance for an 8 ms frame size. For uplink communication to node C over one connection, UDP's performance is not that different to TFRC's (for 20 ms frames). Nevertheless, due to the lack of congestion control, resulting in more packet losses, UDP's video quality is less than TFRC's. When four UDP connections are applied then UDP's packet losses accelerate, with a serious decline in quality. We assume that the end-to-end delay increases because the packets that survive are those that have already queued at the routers in the path to node C, thus increasing the average end-to-end delay.

When the path length increases for a user-to-user connection then again for one connection the sending period increases. In this set of tests, the mean PSNR was found to actually be better than TFRC (for 20 ms frames). However, this is irrelevant as the sending period is far longer than the duration of the video clip. Therefore, streaming cannot be accomplished. When four connections are used instead then the video quality deteriorates by several dB below that of TFRC, even though the sending period is reasonable. End-to-end delay is now considerably reduced because of slice multiplexing and because UDP is not constrained by the presence of other traffic, allowing its packets to seize the path and dominate coexisting traffic.

IV. CONCLUSION

Multi-connection congestion control adapts existing congestion controllers to all-IP networks that include a broadband wireless access link. In effect, they allow the congestion controller to accommodate wireless channel losses but still respond to congestion with the network edge and possibly the core. This in turn leads to improved wireless channel utilization, whereas previous observers have noticed a marked drop in utilization if congestion controllers are employed.

However, for any wireless technology there still remain issues about how many connections should be used if the disadvantages of multi-connections are to be avoided. This study has found that though there is a small percentage increase in packet loss with four connections over just one, video quality remains equivalent because of the differential effect of packet loss patterns when burst errors are present. There was also a small (in practical terms) increase in packet end-to-end delay.

An important observation is that a longer WiMAX TDD frame size is favorable to video transport, though this may not be apparent unless tests are conducted across the whole of a network path and not just the wireless link. An advantage of the multi-connection method of congestion control is the reduction in state when it comes to handoff in a cellular WiMAX, which is important for a delay-intolerant application. Another advantage of the multi-connection method is that a portion of the additional throughput that results is available for error protection, either application layer FEC, or more promising, in terms of compatibility with existing physical layer FEC, the use of source-coded error resilience.

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