Supporting Video/Image Applications in a Mobile Multihop Radio Environment Using Route Diversity*

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Abstract

This paper investigates the need for Multiple Path Transport (MPT) of video and image information in a multihop mobile radio network. The video and image information is encoded non-hierarchically into a multiple description coding (MDC) with the following objectives. The received picture quality should be acceptable even if only one description is received and every additional received description contributes to enhanced picture quality. Typical applications will need a higher bandwidth/higher reliability connection than that provided by current mobile networks. For supporting these applications a mobile node may need to set up and use multiple paths to the desired destination, either simply because of the lack of raw bandwidth on a single channel or because of its poor error characteristics, which reduces its effective throughput. In the context of this work, the principal reasons for considering such an architecture are providing high bandwidth and more robust end-to-end connections. We describe a protocol architecture that addresses this need and, with the help of simulations, we demonstrate the feasibility of this system and compare the performance of the MDC-MPT scheme to that of a hierarchical coding scheme.

1 Introduction

Multiple path[1] transport schemes have been proposed in the past for wired networks for increased connection capacity as well as for reliability [1, 2, 3]. The earliest reference to multiple path transport, referred to as *dispersity routing*, is in Maxemchuck's Ph.D. dissertation [1]. End nodes might communicate with each other using multiple parallel paths/routes constituting a single virtual circuit for various reasons. For example, a channel coding scheme using multiple parallel paths was considered in [2], which improved the fault tolerance of digital communication networks while traffic was spread on multiple paths in [3] to increase the end-to-end throughput. We feel that multiple path transport has more potential in wireless networks where individual physical links may not have adequate capacity to support a high bandwidth service. There are several ways to set up multiple paths or links for a single virtual connection in a wireless network. In a single hop wireless network, a station would need to establish channels to multiple base stations instead of one. This is already done in "soft" hand-off systems during the hand-off phase. In a multihop wireless network where each station has router-like functionality, each station needs to establish multiple disjoint paths with another wireless station or with the wired network. To achieve this, each mobile must be able to support multiple channels simultaneously. This can be done by having multiple transceivers in each mobile [4, 5], and using either receiver oriented or link oriented codes, or a code for each transmitter-receiver pair. The ability to communicate with multiple neighbors (base-stations), instead of having a higher bandwidth connection to a single base-station, for example by using multiple codes [18], allows for better adaptability to the varying radio channel quality, hand-offs and alternate routing in the case of a route failure.

In Part I of the work on MPT we considered non-real time data applications [14] in a mobile multihop radio environment with the underlying Transmission Control Protocol (TCP) based transport. In this work we concentrate on the video/image applications with a Real-time Transport Protocol (RTP) based transport. Most of the coding and transmission schemes proposed for video transport over wireless channels employ layered coding with unequal error protection [19, 20, 21, 22]. With this scheme, a signal is split into a base layer and one or more enhancement layers. The base layer is transmitted with a high priority and with strong error protection (including the use of ARQ), while the enhancement layer is transmitted with fewer error control bits and is simply discarded in the case of congestion. These methods can tolerate a certain degree of burst errors. However, they will break down if the channel carrying the most important layer fails. Although, following a path failure, one could switch over to an alternative route, this may take an unacceptably long period of time. Thus, hierarchical coding schemes assume at least one path with high bandwidth and quality.

Here, we develop a MDC and MPT based system to enhance the system robustness while increasing the usable bandwidth for an end-to-end connection beyond that of a physical link. With MDC, several descriptions are generated for a given signal, so that a better signal reproduction is achieved with more descriptions, and the quality of the decoded signal is acceptable even with only one description. The signal is first decomposed into multiple descriptions and then each description is coded independently. The decomposition is *non-hierarchical* so that the reconstructed signal from any one description is acceptable under a prescribed criterion. Although hierarchical decomposition can lead to greater compression gains, it requires that the channel carrying the most important subsignal be essentially error-free. This may be hard to guarantee given the real-time constraint on video signals and the presence of unpredictable path impairments in a radio environment. The motivation for using MDC is to introduce redundancy at the source coder to combat

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[1]In this work, we use path and route interchangeably. Hence, in our definition of path/route diversity, two communicating entities make use of more than one paths/routes to send information to each other.

these types of channel errors. A key to the success of the proposed system is the close interaction between the source coder and the network interface controller. By carefully allocating packets from different coded descriptions among the available paths, one can ensure, with high probability, the correct and timely delivery of at least one description for any given spatial/frequency location of the source signal, thereby guaranteeing a minimally acceptable quality. By using dynamic path selection and bandwidth allocation on the network control side, scalable (in bit rate and quality) coding of each description on the source coding side, and with close interaction between the two processes, the system can also adapt quickly to changes in link-level connectivity and bandwidth.

In this paper, we address the resequencing and protocol issues associated with transporting video to a desired destination using MPT. We present simulation results obtained when the video is coded using a MDC coder as well as a layered coder. With simple simulation models we show the feasibility of such an architecture from a protocol and resequencing viewpoint and compare the end-to-end performance of an MPT system under different video coding schemes (MDC vs. Layered). In the next section we describe the system and associated protocol model. Section 4 describes the simulation model in detail. The results are presented and discussed in section 5. Finally, conclusions and ongoing work is outlined in Section 6.

2 System and Protocol Model

The network scenario we assume is that of a CDMA network, with each mobile equipped with multiple transceivers, operating in a multihop packet radio environment. Although we assume a multihop radio network scenario, in a conventional cellular network this corresponds to a mobile node capable of communicating with either more than one base station or the same base station using multiple codes. Thus the maximum bandwidth available to the application is the basic channel rate times the number of transceivers (codes). Analogously, in a FDMA or a TDMA based system, a mobile could talk to its neighbors using multiple frequency channels or timeslots.

We use a multiple path transport scheme in conjunction with a multiple description coding scheme for providing robust image/video transfer in an environment with varying channel quality in terms of bit error rate and channel availability. The system schematic of the MDC-MPT communications system is shown in Figure 1. In

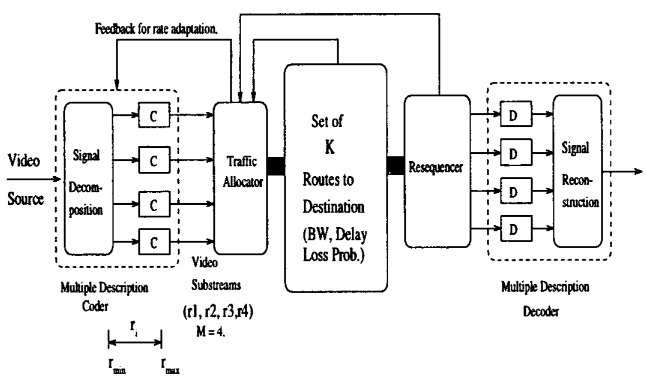

Figure 1: System schematic for the proposed MDC-MPT communications system.

the multiple description coding scheme, on the sender side, a coder breaks the signal into M subsignals, each subsignal is then coded, packetized and sent on different paths through the multihop radio network. A coded subsignal constitutes a description. We assume the rate of each description is between r_{min} and r_{max}. The decomposition process is non-hierarchical in the sense that each subsignal contains sufficient information for the decoder to reconstruct an acceptable quality signal. Each description is divided into slices so that each slice is carried in a single transport packet. To enable graceful degradation and source rate adaptation, we assume each description is coded in layers with different importance. When there is insufficient path capacity to carry all slices in a description, slices carrying bits from less important layers can be dropped. At the receiver, the slices arriving from all the paths are put into a resequencing buffer where they are reassembled into M descriptions after a preset time-out period. All or some of the slices allocated to a path may be lost because of the errors on the path or because of path breakdown. Some slices may arrive late and will also be considered lost. The decoder will attempt to recover the original frame from the received slices in separate descriptions. In the following, we describe the proposed transport model and mechanisms for mapping the coded slices to paths.

2.1 Transport Control Issues and Options

Given that image and video transport can tolerate some amount of loss and may have real time delivery constraints, we consider the Real-time Transport Protocol (RTP) as the transport layer protocol entity [8]. RTP

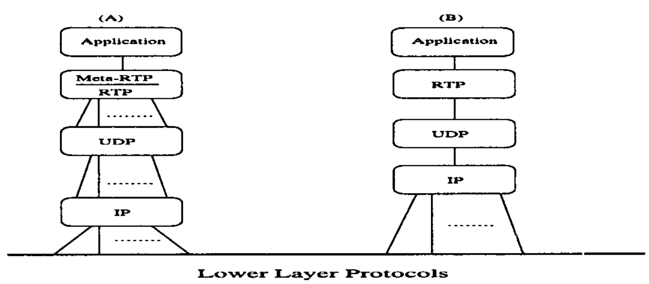

Figure 2: Layered Protocol Model.

is complemented by a control protocol called RTP Control Protocol (RTCP) which tackles issues such as quality of service, mechanisms to disperse QoS and membership information, membership control and identification. RTP provides time-stamping, sequencing and delivery monitoring services to the application. Typically, RTP is implemented as an application level protocol which makes use of underlying transport/network (for example UDP/IP) layer services. In general, traffic could be split at any layer in the protocol stack. We consider two such options as shown in Figure 2. In option B, traffic is distributed at the IP layer. In option A, we introduce a layer called Meta-RTP, which is on top of RTP, and is responsible for traffic distribution at the sender and resequencing at the receiver. We split the traffic at the Meta-RTP layer in the protocol stack and not at a lower layer (e.g. IP) because splitting traffic at lower layers may not be helpful in being able to exploit the QoS (path quality) information associated with each path. In general, the application would be in the best position to decide or act on slice losses, slice resequencing, slice retransmissions and rate adaptation.

The quality of each path is continuously updated based on the feedback from the intermediate nodes (routers, gateways) and that from the destination node. A destination node participating in a RTP session generates Receiver Reports (RR) at regular intervals. The Receiver

Reports give the sender useful information on path quality such as packet losses, delay and delay jitter. In order to get timely feedback, at the cost of increased traffic, we send these reports on all the paths. Thus the information on any impaired path can be received on any remaining unimpaired path. Each report will contain information regarding all K paths packed into one or more RR packets. A path could go down in the middle of an ongoing session. This could be conveyed to the sender by the routing protocol, or the sender itself can conclude that a route has become unusable based on the RRs. We believe that the latter may be a faster way of determining the route usability and hence more pertinent to a real time traffic adaptation. The portion of traffic carried by each path will be dynamically adjusted based on the feedback. Also, based on the feedback, if needed, the encoder rate also could be adaptively changed over the range $[r_{min}, r_{max}]$.

2.2 Traffic Allocation and Description-to-Path Mapping

Based on the total bandwidth available and the overhead to be added in the transport layer (including headers and FEC), the total encoding rate R is then determined. Here, we consider a coder that codes each frame independently, without making use of temporal prediction. For each frame, the encoder generates M coded descriptions, each with a rate of $r_m, m = 1, 2, \ldots, M$. To guarantee the delivery of the most important information, the coded descriptions are organized so that each description contains L layers, $L \geq 1$, with the first layer being most important, the next layer less important, and so on. Each layer is further partitioned into slices, so that the bits in each slice are decodable by itself (i.e. if its previous and/or following slices are corrupted, this slice is still decodable). Each description has a dynamic range (r_{min} to r_{max}) over which its rate can change. So, if the number of descriptions is M, then the range over which the offered load from this video source can change is, $r_m = r_{min} + \alpha(r_{max} - r_{min})$, $R = \sum_{m=1}^{M} r_m$. The variable α in the above equation is initially chosen based on the available bandwidth and is updated based on the path quality feedback from intermediate nodes and the destination. The total network bandwidth required to carry the load is $BW = R/\beta$, where β is a factor we initially set at 0.85, the purpose of which is to prevent overloading the paths. In the MPT scheme the required bandwidth can then be found for a set of paths to the given destination.

There are a number of ways traffic could be sent on a set of routes with a given proportion. For example, one could simply consider random routing, weighted round robin and its variants. We perform this mapping taking into consideration the following two criteria: i) The allocation granularity is one "slice"; and ii) The portions of descriptions which overlap (i.e. carry the information about nearby samples in the subband domain or in the original signal domain) should not be assigned to the same path. This ensures that recovery process is not adversely affected even if we lose a path completely. We have developed a description to path mapping algorithm that minimizes the probability of simultaneously losing slices corresponding to the same spatial/frequency content of the source signal in different descriptions, so as to help the error concealment task in the receiver. The details of the algorithm [15] are not presented here for conciseness.

3 The Multiple Description Coding Scheme

In MDC, multiple bitstreams (called descriptions) are generated for the same source signal, with the premise that the probability that all the descriptions experience transmission losses simultaneously is minimal. The decoder operates in different modes depending on which descriptions are available. The coder/decoder (codec) is designed so that a high quality reconstruction of the original signal can be obtained from all the descriptions, and a lower but still acceptable quality is achievable even when only one description is available. This is accomplished by introducing correlation among separate descriptions so that the lost description can be estimated from the received ones. With MDC, the correlation among the descriptions will reduce the coding efficiency, however, in exchange for improved reconstruction quality. As described in Section 4, we use a network simulation model where a slice is either delivered correctly or lost. In the simulation results shown in Section 5, we only considered the non-layered implementation within each description. A detailed description of the MDC, the Layered Coder/Decoder and error concealment schemes can be found in [11, 12, 13].

4 Simulation Model for the MDC-MPT Scheme

We developed a simple model for the MDC-MPT system using the OPNET simulation and modeling tool [17]. We developed the required subset of functions of the RTP/RTCP layer in OPNET, to run on the UDP layer. Initial simulation results for the MPT scheme for file transfer/non-real time data transfer using TCP as the transport layer and Meta-TCP as the traffic allocator and resequencing layer are reported in [14]. An MPT system with two paths was simulated. In general, the paths could consist of several wireless hops. In the simulation, the two paths are identical in terms of available bandwidth and the number of hops. Each path has three wireless hops. In order to study the effect of the wireless channel, we have set the parameters such that the losses are due to channel impairments. The video source application process opens multiple UDP (datagram) socket (where the number of sockets equals the number of paths) connections to the destination application process. At the source, traffic is distributed according to the description-path-mapping algorithm at the the Meta-RTP layer. The slices that reach the destination are resequenced at the Meta-RTP layer and delivered in order to the application.

4.1 Source Model

The source is a short video sequence consisting of 25 frames. This sequence is sent repetitively ten times at 25 frames/second. The video is in SIF resolution with 352×240 pixels per frame. Only the luminance pixels are coded. The frames are individually decomposed, quantized, and coded so as to generate two descriptions for the MDC case and two layers for the Layered coder case. The quantization factors were adjusted such that the average number of bits per frame is the same for both coding schemes. There are 31 slices per frame for each description in the MDC coder and 32 slices per frame for each layer in the layered coder. The temporal duration of each slice is therefore $1/(25 * 31)$ seconds. For this video sequence the average number of bits/slice is about 1870 for each description in the MDC case, and 1910 and 1747 for the base layer and enhancement layer for the layered coder case. As mentioned earlier, each slice is sent as a single RTP/UDP packet. The average video source rate (without the header overhead of RTP, UDP and IP layer) is 1.45 Mbps for each of the descriptions, whereas the average video source rate for the layered coder is 1.53 Mbps for the base layer and 1.40 Mbps for the enhancement layer. In a lossless scenario, that is if all the slices are received within the deadline, the PSNR for the Layered coder is higher than that of MDC.

4.2 Channel Model

There is extensive literature on wireless channel modeling based on theory as well as measurements, both in the indoor and outdoor (urban, suburban and rural) environments [9, 10]. As our primary focus is to study Multiple Path Transport schemes, we chose to model the bursty error nature of the wireless channel as described below. We model the radio link by a two state Markov model with the two states corresponding to the link being in either a "good" state or a "bad" state. We assume the presence of some power control and error correcting mechanisms at the data link layer which take care of the short term link impairments, so the longer term channel impairments can be qualitatively modeled by the two state Markov model. In the good state we assume that the packet is lost with some low loss probability p (the bit error rate is so low that most errors are corrected at the link layer), whereas in the bad state the packet is lost with a high loss probability q. For most of the simulations, we use a value of 0.005 for p and 1 for q. For simplicity, we assume that the effect of FEC is already reflected by the uncorrectable error probabilities associated with the channel states. We consider a radio channel operating at 2.0 Mbps. Typically, the effective bandwidth seen at the application layer is much less than 2.0 Mbps due to packet losses at the the link layer and additional overhead due to ARQ retransmissions, if ARQ is employed.

4.3 Resequencer

At the receiver all the slices coming from both the paths are stored in a buffer. The application process retrieves and displays the next frame in sequence at regular intervals. The slices of the frame that are received past its display time are considered useless. Currently, we assume that the resequencing buffer size is not a constraint.

5 Results and Discussion

We compare the end-to-end performance of a layered coding scheme and that of MDC scheme when both use the MPT as the transport mechanism. We vary the channel error characteristics by appropriately controlling the channel *good* and *bad* durations and compare the peak signal to noise ratios (PSNRs) of the received video stream under both schemes.

First, purely for comparison purposes, we show results obtained when the two paths have the same error characteristics. Figure 4 shows the PSNR values of reconstructed video frames at different error rates. It can be seen that, when the error rate is very low (below a number in the range 0.82 % to 2.27%, for the video sequence considered), the layered coder performed better than the MDC coder. This result is as expected because, at low error rates, the overall reconstruction quality is dominated by those blocks which do not experience transmission loss, and the layered coder yields better performance because it has a lower coding distortion than the MDC coder. At higher error rates, when the reconstruction quality of damaged blocks becomes the deciding factor, the MDC coder is superior. Figure 3 shows the distribution of the PSNRs in individual frames for a MDC coder for a loss rate of 12.81 %.

The above comparison between the MDC and layered coding may not be considered fair, because the layered coder generally assumes that a reliable path exists for transporting the base layer. Figure 5 compares the PSNRs of the three layered coders (Layered I, II and III – that use different error concealment schemes with increasing complexity, namely direct inverse, mean reconstruction and maximally smooth reconstruction [11], [13]) with the MDC coder when the average loss rate on the two paths are the same, but for the layered coder the two available paths are asymmetric in terms of their error characteristics. We see that both the Layered III and the Layered II coders are better than the MDC coder when the average loss rate is lower than a number in the range 12.53% to 13.50% (corresponding to a base layer loss rate of 2.4% to 3.55%). This break point is lower in the Layered II coder case than in the Layered III case. When such low loss rates cannot be guaranteed on the base layer path, then it is better to use the MDC coder. Note that to guarantee a low loss rate on the path carrying the base layer, extra bandwidth/power is needed on this path, with added complexity in transport level control. On the other hand MDC requires a more complex error concealment algorithm at the decoder. Also, for this study, we do not use ARQ for the base layer packets. We feel that even after implementing ARQ for the base layer packets there will still be some losses given the real time nature of the information. The tradeoff between the overhead introduced by performing link layer ARQ and the gain in the performance due to a more reliable base-layer is a subject for further study.

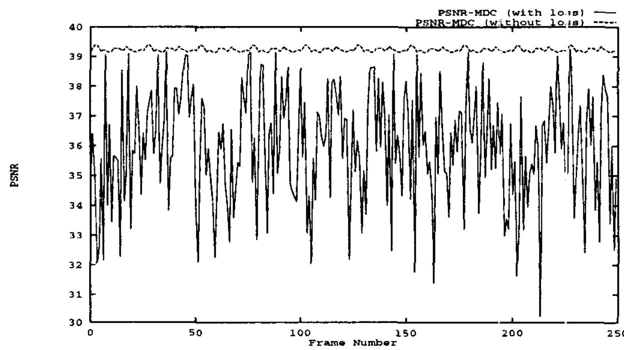

Figure 3: MDC: Frame PSNRs for a radio channel with average "good" period 100 ms and average "bad" period 3 ms (This corresponds to an average end-to-end loss probability of 12.81%).

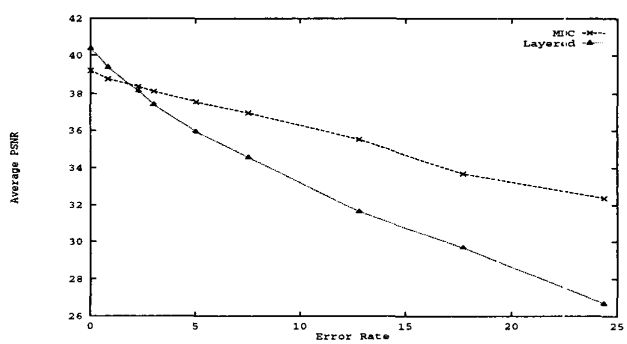

Figure 4: PSNR for different loss rates for a MDC coder and a layered coder. Both the paths are symmetric in terms of available bandwidth and error characteristics.

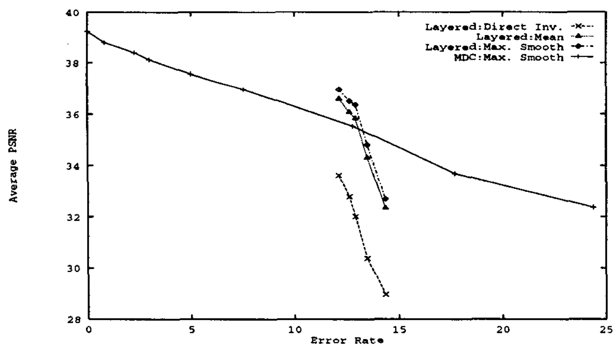

Figure 5: Average Loss Rate vs. PSNR for a layered coder in an asymmetric network scenario and a MDC coder in symmetric scenario.

Performance with link outages on longer time-scales

In general, in a multihop wireless radio network scenario, wireless link impairments and outages can occur on different time-scales. This may happen, for example, either in case of a slow fade or if a path goes down completely when nodes move out of range. Typically, in the latter case the routing algorithm will detect the loss of a link and compute the new topology and routes after exchanging routing PDUs. We simulate such a system and show that an alternate, possibly quicker, method of doing traffic adaptation on these time-scales is using the Receiver Reports of the RTCP. By appropriately adjusting the frequency of these RRs, we have a trade-off between the rapidity in reacting to the network changes and the additional overhead due to the RRs. As mentioned earlier, we send the RRs on all the paths back to the sender so that the sender will get a copy on any unimpaired path. The receiver generates a report once per frame. This translates to an overhead of about 3.2 % in terms of packets in the network. The overhead is much lower in terms of bits in the network because, in our simple model, the bit size of RRs is much less than that of a slice. The sender will conclude that a path has gone down only after receiving a certain number of RRs (on the unimpaired path) indicating such an event. We have set this number to 3 for this study.

To demonstrate this scheme we simulate a network scenario with three available paths to a destination with given error rates. Initially, the two best paths, in terms of error rate, are used to carry a description each, and the third path is left idle. In the middle of the session, when a path goes down completely, the sender reacts to this and switches one description to the alternate path.

In Figure 7, we see three distinct phases. The first is one in which the destination receives both the descriptions, with of course some slices lost due to link errors. The second (Frames 101-104) is in which one path has gone down, and the destination is receiving slices only from a single path. The third region is the one after the sender starts using the alternate path. Note that now the number of slices received improves, but is still worse than the first region. This is because the new path is poorer than the old one, in terms of link errors. We can get some idea as to how the picture quality will be during the second phase, from the frame shown in Figure 6(b), which shows reconstructed image when only one description is completely received. Figure 6(a) shows the reconstructed image when both the descriptions are completely received.

(a)

(b)

Figure 6: Reconstructed MDC images for a selected frame: (a) is obtained when both the descriptions are received without loss, (b) is obtained when only one description is received and the other is completely lost, and using the maximally smooth recovery method. The PSNRs for the images are 39.06 and 31.95 dB respectively.

6 Conclusions and Ongoing work

We described a framework for video transport over an unreliable network using MPT and MDC. We presented a Meta-RTP protocol for transport control. The MDC coder is capable of producing multiple descriptions and each description in general can be arranged in layers of different priority. The rate of each description can be adapted in response to the failures or RTCP receiver reports. We simulated a simple case system with two paths having the same capacity and error characteristics. Each description has a single layer and all the descriptions have similar bit rates. As a comparison, we also used a layered coder, and examined its performance when the two paths carrying the base and enhancement layers have symmetrical and asymmetrical error characteristics. The conclusion from our simulation results is that when the error rate on the path carrying the base layer is relatively low (below a number in the range 0.82% to 2.27% in our limited test case), the layered coder gives a better overall performance. Crossing this break point, the MDC coder becomes more effective. Considering the high error rates typical for a wireless hop, which become even more dominant in a multihop scenario, the proposed MDC-MPT system appears to be a more attractive approach than the layered coding approach. In summary, the proposed MDC-MPT scheme for video and image transport serves

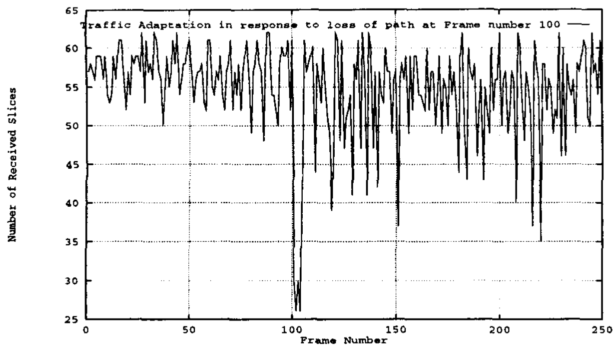

Figure 7: Traffic adaptation, based on the receiver reports, in response to loss of a path at Frame # 100. The MDC coder is expected to provide acceptable quality video signal during this time. Note that the number of slices/frame = 62, since each description has 31 slices.

three purposes. First, it enables real time video services, the bandwidths of which may exceed that of a single wireless link. Second, it enhances the system robustness in the presence of channel errors, especially long burst errors, and link failures. Finally, it enhances adaptation to changes in the network connectivity and bandwidth.

We are in the process of extending this work in the following ways. Firstly, we will simulate the system with heterogeneous paths in terms of error rates and available bandwidth. The effect of the background traffic on the number of the slices that miss the deadline at the destination will be studied. We will further extend our work on RTCP Receiver Reports and source rate adaptation and adaptive routing in response to these reports. Finally we will compare the performance of the layered coding and MDC-MPT with "limited" link layer or end-to-end ARQ, at least for the base-layer slices. The *limited* nature is imposed by controlling the number of times a packet is retransmitted. Given the real-time nature of the video information there is no point in retransmitting a packet if it is not expected to reach the destination within its time constraint.

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