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Distortion-Resistant Routing Framework for Video Traffic in Wireless Networks

Dr.G.Kavitha, M.Agalya AP,S.Gayathri AP,S.K.Murugaraja AP,
M.E.,Ph.D.,Principal,
Sri Shanmugha college of engineering and Technology,Pullipalayam,Sankari(D.T)
Tamil nadu,India-637304,principal@shanmugha.edu.in

Department of CSE,Sri Shanmugha college of engineering and Technology,
Pullipalayam,Sankari(D.T),Tamilnadu,India-637304.
agalkalai@gmail.com,gayathris3012@gmail.com,murugaraja2003@gmail.com.

***Abstract*—Traditional routing metrics designed for wireless networks are application-agnostic. In this paper, we consider a wireless network where the application flows consist of video traffic. From a user perspective, reducing the level of video distortion is critical. Popular link-quality-based routing metrics do not account for dependence across the links of a path; as a result, they can cause video flows to converge onto a few paths that cause high video distortion. To account for the evolution of the video frame loss process, we construct an analytical framework .The framework allows us to formulate a routing policy for minimizing distortion, based on which we design a protocol for routing video traffic. We find via simulations and test bed experiments that our protocol is efficient in reducing video distortion and minimizing the user experience degradation.**

***Index Terms*— Routing, video communications, video distortion minimization, wireless networks.**

I. INTRODUCTION

WITH the advent of smart phones, video traffic has become very popular in wireless networks. In tactical networks or disaster recovery, one can envision the transfer of video clips to facilitate mission management. From a user perspective, maintaining a good quality of the transferred video is critical. The video quality is affected by: 1) the distortion due to compression at the source, and 2) the distortion due to both wireless channel induced errors and interference.

Video encoding standards, like MPEG-4 define groups of I-, P-, and B-type frames that provide different levels of encoding and, thus, protection against transmission losses. In particular, the different levels of encoding refer to: 1) either information encoded

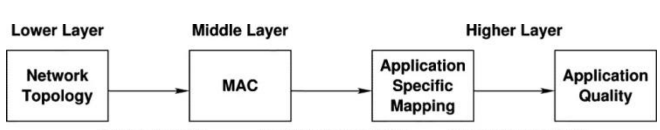

Fig. 1. Multilayer approach.

independently, in the case of I-frames, or 2) encoding relative to the information encoded within other frames, as is the case for P- and B-frames. This Group of Pictures (GOP) allows for the mapping of frame losses into a distortion metric that can be used to assess the application-level performance of video transmissions.

One of the critical functionalities that is often neglected, but affects the end-to-end quality of a video flow, is routing. Typical routing protocols, designed for wireless multi hop settings, are application-agnostic and do not account for correlation of losses on the links that compose a route from a source to a destination node. Furthermore, since flows are considered independently, they can converge onto certain links that then become heavily loaded while others are significantly underutilized. The decisions made by such routing protocols are based on only network parameters.

In this paper, the user-perceived video quality can be significantly improved by accounting for application requirements, and specifically the video distortion experienced by a flow, end-to-end. Typically, the schemes used to encode a video clip can accommodate a certain number of packet losses per frame. However, if the number of lost packets in a frame exceeds a certain threshold, the frame cannot be decoded correctly. A frame loss will result in some amount of distortion. The value of distortion at a hop along the path from the source to the destination depends on the positions of the unrecoverable video frames in the GOP, at that hop. As one of our main contributions, we construct an analytical model to characterize the dynamic behavior of the process that describes the evolution of frame losses in the GOP as video is delivered on an end-to-end path. Specifically, with our model, we capture how the choice of path for an end-to-end flow affects the performance of a flow in terms of video distortion. The packet-loss probability on a link is mapped to the probability of a frame loss in the GOP. The frame-loss probability is then directly associated with the video distortion metric. By using the above mapping from the network-specific property (i.e., packet-loss probability) to the application-specific quality metric (i.e., video distortion), we pose the

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problem of routing as an optimization problem where the objective is to find the path from the source to the destination that minimizes the end-to-end distortion.

In our formulation, we explicitly take into account the history of losses in the GOP along the path. This is in stark contrast with traditional routing metrics wherein the links are treated independently. Our solution to the problem is based on a dynamic programming approach that effectively captures the evolution of the frame-loss process. We then design a practical routing protocol, based on the above solution, to minimize routing distortion. In a nutshell, since the loss of the longer I-frames that carry fine-grained information affects the distortion metric more, our approach ensures that these frames are carried on the paths that experience the least congestion; the latter frames in a GOP are sent out on relatively more congested paths. Our routing scheme is optimized for transferring video clips on wireless networks with minimum video distortion. Since optimizing for video streaming is not an objective of our scheme, constraints relating to time (such as jitter) are not directly taken into account in the design.
Specifically, our contributions in this paper are as follows.

- *Developing an analytical framework to capture the impact of routing on video distortion*: As our primary contribution, we develop an analytical framework that captures the impact of routing on the end-to-end video quality in terms of distortion. Specifically, the framework facilitates the computation of routes that are optimal in terms of achieving the minimum distortion. The model takes into account the joint impact of the PHY and MAC layers and the application semantics on the video quality.
- *Design of a practical routing protocol for distortion-resilient video delivery*: Based on our analysis, we design a practical routing protocol for a network that primarily carries wireless video. The practical protocol allows a source to collect distortion information on the links in the network and distribute traffic across the different paths in accordance to: 1) the distortion, and 2) the position of a frame in the GOP.
- *Evaluationsviaextensiveexperiments*: We demonstrate via extensive simulations and real test bed experiments on a multi hop 802.11a test bed that our protocol is extremely effective in reducing the end-to-end video distortion and keeping the user experience degradation to a minimum. In particular, the use of the protocol increases the peak signal to-noise ratio (PSNR) of video flows by as much as 20%, producing flows with a mean opinion score (MOS) that is on the average 2–3 times higher compared to the case when traditional routing schemes are used. These PSNR and MOS gains project significant improvements in the perceived video quality at the destination of a flow . We also evaluate our protocol with respect to various system parameters.

Organization: The paper is organized as follows. Related work is presented in Section II. Our analytical models are in Section III, followed by the problem formulation in Section IV. In Section V, we discuss how our framework can be used to route video flows in practice. Section VI contains results from our simulations and testbed experiments. We conclude in Section VII.

II. RELATED WORK

The plethora of recommendations from the standardization bodies regarding the encoding and transmission of video indicates the significance of video communications. Different approaches exist in handling such an encoding and transmission. The Multiple Description Coding (MDC) technique fragments the initial video clip into a number of sub streams called descriptions. The descriptions are transmitted on the network over disjoint paths. These descriptions are equivalent in the sense that any one of them is sufficient for the decoding process to be successful; however the quality improves with the number of decoded sub streams. Layered Coding (LC) produces a base layer and multiple enhancement layers. The enhancement layers serve only to refine the base-layer quality and are not useful on their own. Therefore, the base layer represents the most critical part of the encoded signal .

Standards like the MPEG-4 and the H.264/AVC provide guidelines on how a video clip should be encoded for a transmission over a communication system based on layered coding. Typically, the initial video clip is separated into a sequence of frames of different importance with respect to quality and, hence, different levels of encoding. The frames are called I-, P-, and B-frames, and groups of such frames constitute a structure named the GOP. In each such GOP, the first frame is an I-frame that can be decoded independently of any other information carried within the same GOP. After the I-frame, a sequence of P- and possibly B-frames follows. The P- and B-frames use the I-frame as a reference to encode information. However, note that the P-frames can also be used as references for other frames.

There has been a body of work on packet-loss-resilient video coding in the signal processing research community . In the video stream is split into high- and low-priority partitions, and FEC is used to protect the high-priority data. To account for temporal and spatial error propagation due to quantization and packet losses, an algorithm is proposed in [8] to produce estimates of the overall video distortion that can be used for switching between inter- and intracoding modes per macroblock, achieving higher PSNR. In an enhancement to the transmission robustness of the coded bit stream is achieved through the introduction of inter/intracoding with redundant macroblocks. The coding parameters are determined by a rate-distortion optimization scheme. These schemes are evaluated using simulation where the effect of the network transmission is represented by a constant packet-loss rate, and therefore fails to capture the idiosyncrasies of real-world systems.

In an analytical framework is developed to model the effects of wireless channel fading on video distortion. The model is, however, only valid for single-hop communication. In the authors examine the effects of packet-loss patterns and specifically the length of error bursts on the distortion of compressed video. The work, although on a single link, showcases the importance of accounting for the correlation of errors across frames. Finally, a recursion model is derived in [13] to relate the average transmission distortion across successive P-frames. None of these efforts considers the impact of routing on video distortion.

There have also been studies on the performance of video transmissions over 4G wireless networks that have been designed to support high QoS for multimedia applications. In an assessment of the recently defined video coding scheme (H.264/SVC) is performed over mobile WiMAX. Metrics such as the PSNR and the MOS are used to represent the quality of experience perceived by the end-user. The results show that the performance is sensitive to the different encoding options in the protocols and responds differently to the loss of data in the network. Again, these are single-link wireless networks, and routing is not a factor.

Cross-layer optimization and QoS routing is not new. An extensive body of research exists on routing algorithms for wireless ad hoc and mesh networks. Furthermore, the survey in provides various ways of classifying QoS routing schemes based on protocol evaluation metrics (transport/application, network- and MAC-layer metrics). However, none of the routing schemes presented in these surveys takes into account performance metrics defined for an application and specifically for video transfers. Even when a QoS routing is defined as application-aware, the applications need to specify throughput and delay constraints. This is in contrast to our approach, where an application-related performance metric, namely the video distortion, is directly incorporated into the route selection mechanism.

Prior work on routing for video communications focuses on Multiple Description Coding (MDC). In multipath routing schemes are considered to improve the quality of video transfer. In an extension to the Dynamic Source Routing is proposed to support multipath video communications. The basic idea is to use the information collected at the destination node to compute nearly disjoint paths. A rate-distortion model is defined and used in an optimization problem where the objective is to minimize the overall video distortion by properly selecting routing paths. Due to the complexity of the optimization problem, a genetic-algorithm-based heuristic approach is used to compute the routes. To achieve good traffic engineering, the scheme relies on maximally disjoint paths. However, this work does not consider distortion as a user-perceived metric. It simply aims to reduce the latency of video transmissions, and thus, its objective is different from what we consider here.

The work in proposes a scheme for energy-efficient video communications with minimum QoS degradation for LC. However, the routing scheme is based on a hierarchical model. To support such a hierarchy, the nodes need to be grouped in clusters, and a process of electing a cluster head has to be executed periodically, increasing the processing and data communication load of the network. In contrast, our proposed scheme assumes a flat model where all nodes in the network are equivalent and perform the same set of tasks.

In the source routing scheme, the routing decisions are made at the source node ahead of time and before the packet enters the network. Therefore, source routing is an open-loop control problem where all decisions have to be made in the beginning. The decisions are taken sequentially; a decision at a stage corresponds to the choice of the next-hop node at the node corresponding to the stage. The source node cannot know exactly the state of the selection process because of the randomness of the second component of the state. It has to estimate at each stage the value of c_k and use this estimate to make a decision for that stage.

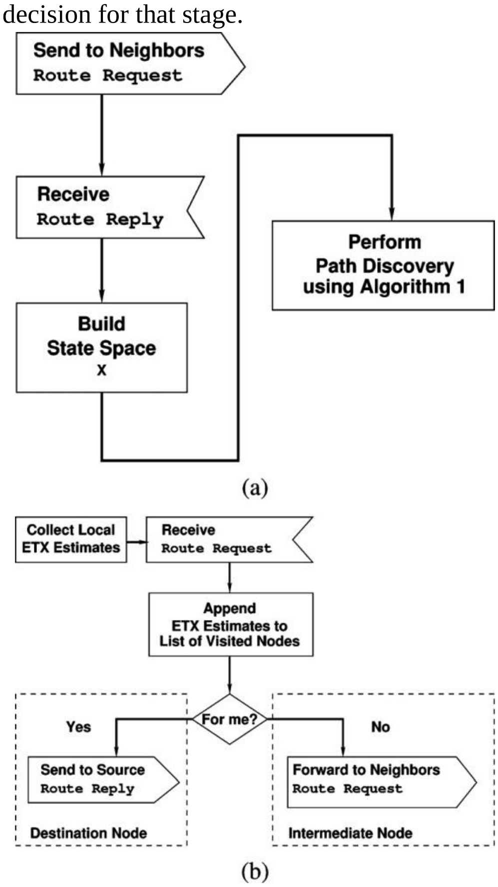

Fig. 2. Flowchart for application-aware routing. (a) Source node. (b) Intermediate and destination node.

performance metrics such as the PSNR and the MOS [34]. To adapt the EvalVid to the ns-2 simulator, we follow the procedure. Specifically, for each simulated video flow between two nodes in the network, we need to produce a sequence of files. We start with the initial uncompressed video file that consists of a sequence of YUV frames [36]. Using the EvalVid toolset, we transform the YUV format first to the MP4 and then to the MPEG4 format, which contains hints of how the video file should be transmitted over a network. When we do this, we do not constrain the GOP size to be the same from GOP to GOP, but rather, we let the tool decide the appropriate size for each GOP based on the video clip content.

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We then need to capture a log from an attempted transmission over a real network. This log indicates which frame and at what time instance was transmitted over the network. The log is fed as an input to the ns-2 simulation, which plays back the video transmission, producing at the end two sets of statistics regarding the transmission, one for the sender and one for the receiver. By applying the EvalVid toolset on this sequence of files, we can reconstruct the video file as it is received by the destination and compare it to the initial video file. The comparison provides a measure of the video quality degradation due to the transmissions over the network.

A. Simulation Results

To evaluate the performance of the MDR protocol, we compare it against the minimum ETX routing scheme. We consider a wireless multihop network that covers an area of 1000 $\times$ 1000 m^2. The nodes are distributed over this area according to a Poisson random field. The pair of nodes that constitute the

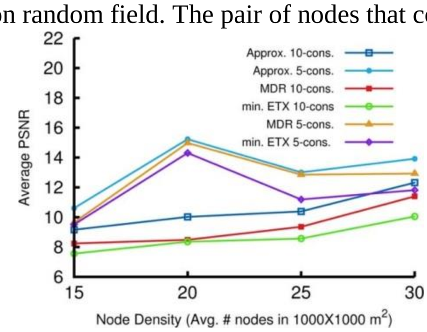

Fig. 3. Average PSNR for 5 and 10 video connections (Set-I).

source and destination in each case are selected at random. If they happen to be neighbors, we discard that pair and repeat the process until we select a source and destination that are more than one hop apart. Each node uses the IEEE 802.11b protocol where the propagation model is the Two Ray Ground, yielding a communication range of about 250 m. Each set of experiments is repeated 10 times, and the average value is reported in each case.

In Table I, three sets of values are defined for the video encoding parameters. We vary the GOP size and the frame rate and thus, effectively, the video encoding rate. We keep the frame size constant as per the QCIF standard (176 $\times$ 144 pixels) and set the maximum packet size to 1024 B. Our simulation experiments focus on three metrics: 1) the PSNR, which is an objective quality measure; 2) the MOS, which is a subjective quality metric; and 3) the delay experienced by each video connection. The effect of the node density on the PSNR is shown in Fig. 3. We plot the average PSNR for 5 and 10 concurrent video connections for different node densities and for Set-I of the video encoding parameters of Table I. We also plot the performance of our proposed scheme (MDR) when instead of estimating the per-link packet-loss probabilities through the ETX metric, we use the model in Section III-A to do so. In this case, we assume full knowledge of the network topology, and so the state space where we solve the optimal control problem of Section IV is a superset of the state space when we collect the local estimates of ETX through the network.

We then fix the number of nodes to 20 (distributed as described earlier) and compute the PSNR of each video connection when: 1) the network serves four concurrent connections, and 2) when the number of concurrent connections is 8. In each case, the source–destination pairs are chosen uniformly from among the nodes in the network. We define the tail distribution of PSNR as the probability $P\{\text{PSNR} > x\}$ and plot it in Fig. 4 for the different traffic loads. The tail distribution of PSNR that corresponds to Set-II of the video encoding parameters is shown in Fig. 4(a). For both the light and heavy traffic loads (four and eight concurrent connections, respectively), the MDR protocol performs better, providing a higher percentage of paths that have a given PSNR value. As expected, a performance degradation is observed for both schemes when the traffic load increases. This is due to the fact that under heavier traffic conditions in the network, the interference becomes more prevalent; furthermore, interference across adjacent links can be correlated in some cases. Under such network conditions, the benefits from the distortion-based optimization have a greater impact on the path selection process for the different types of frames in a video GOP as discussed earlier. The I-frames are sent on relatively uncongested paths. With fur concurrent connections, the median of PSNR is 17 for the minimum ETX policy and 18 for the MDR protocol. The median decreases when the traffic load increases, and it is 9.5 and 10 for the minimum ETX and the application-aware schemes, respectively. The tail distribution of PSNR that corresponds to the parameters of Set-III is shown in Fig. 4. As is the case for Set-II, a large GOP size results in a denser state space, and therefore a better performance for the MDR protocol. In the case of the light traffic loads (four concurrent connections), the median for the PSNR is 15 for the minimum ETX scheme and 17 for MDR. Under heavier traffic loads (eight concurrent connections), Pthe median for the PSNR is 9 for the minimum ETX scheme and 10.5 for the MDR protocol.

Although the PSNR is the most widespread objective metric to measure the digital video quality, it does not always capture user experience. A subjective quality measure that tries to capture human impression regarding the video quality is the MOS.

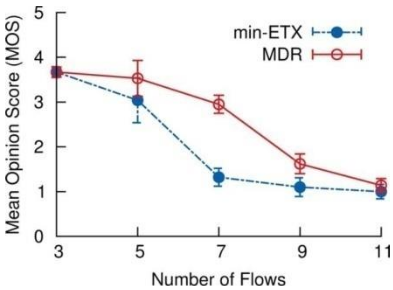

Fig. 4. Average mean opinion score.

The metric uses a scale from 1 (worst) to 5 (best) to represent user satisfaction when watching a video clip [34].

To evaluate the MOS with the MDR and ETX-based routing, we consider the wireless multihop network with the average number of nodes equal to 20 (distributed as discussed earlier). The initial raw video is processed using the H.264 encoder with a maximum GOP size of 30 frames and a sampling frequency of 30 frames per second. Fig. 6 shows the average MOS as the number of concurrent video flows in the network increases. When the number of connections is three, the traffic load is low, and so both the ETX-based routing and MDR provide similar user experience regarding video quality. As the traffic load increases, the distortion-based routing distributes the load across the network, causing the I-frames to avoid highly congested areas. When a moderate number of video flows are concurrently active in the network, there is a significant gap in video quality in favor of MDR. However, no significant gains are possible with MDR when congestion is high (more than nine concurrent video flows are active). In such cases, there are no congestion-free routes available to be used by MDR. This results in higher MOS values, which translates to a better user experience.

The delay characteristics of the two routing schemes are shown in Fig. 7 for Set-II of the video encoding parameters. The nodes are again randomly distributed according to a Poisson random field with varying density with values 14, 16 , and 18. The traffic load corresponds to five concurrent video connections. We compute and plot the mean and variance of the end-to-end delay for the five connections along with the 95% confidence intervals. As seen in Fig. 7, for all three different node densities, the MDR protocol produces routes that exhibit less variability compared to the routes computed by the minimum ETX scheme. Smaller variability implies less jitter, which in turn suggests a better video quality as perceived by the end-user. Moreover, because of the smaller variability, the required sizes of buffers at the intermediate nodes is smaller.

The primary reason for this reduction in the delay is that the distortion-aware approach tries to avoid paths that are congested; ETX, on the other hand, results in convergence of flows onto a few good paths. For both routing schemes, the mean and variance of the delay increase with the average number of nodes in the network. As the network becomes denser, the effect of interference becomes more profound, increasing the number of retransmissions and, thus, the delay. In contrast, a sparser network topology provides a smaller number of "good" routes, and thus it is more difficult to separate flows and cope with congestion. It is in the moderate density regions, where the MDR protocol provides the most benefits in terms of delay and jitter.

We evaluate the MOS of slow- and fast-motion video flows when the MDR routing scheme is used. We consider a wireless multihop network with an average number of nodes equal to 20 (distributed as discussed above). Fig. 7(a) shows the average MOS for Set-II, and Fig. 7(b) shows the MOS in the case of Set-III. In both cases, the slow-motion flows experience slightly lower distortion compared to the fast-motion videos and, thus, higher MOS. This is the result of the fact that in the slow-motion video clips, the I-frames carry most of the information. Due to rapid changes in the content of a fast-motion clip, the P-frames are larger and contain more information than the P-frames for slow-motion video flows. The MDR routing scheme protects the I-frames by routing the corresponding packets through less congested paths. The P-frames are packed together on congested paths and could be lost. As evident from Fig. 6, such losses affect fast-motion video to a greater extent. However, as we increase the traffic to extremely high levels (11 flows), the performance of slow- and fast-motion videos is similar due to high frame losses.

Next, we compare the behavior of MDR against a routing protocol that chooses routes so as to minimize the overall expected transmission time (ETT) . The ETT is a function of the loss rate and the bandwidth of the link. Therefore, it can capture delays due to transmissions in multirate settings, unlike ETX, which only estimates the packet-loss ratio at the base rate.

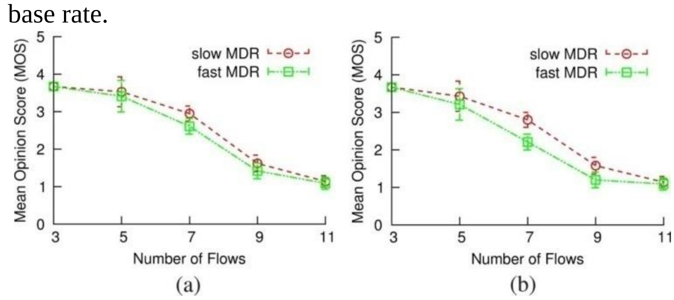

Fig. 5. Average value of the MOS for slow- and fast-motion video flows. (a) Set-II. (b) Set-III.

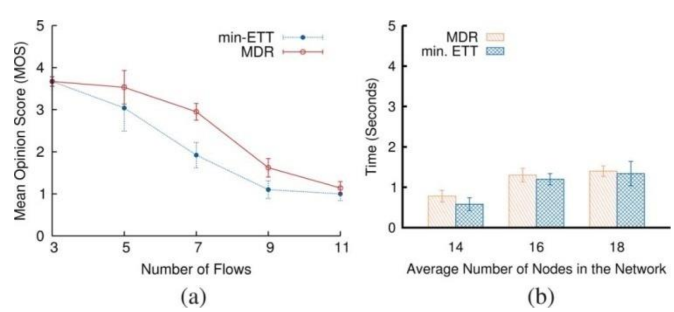

Fig. 6. Comparison between the MDR and the ETT-based routing scheme. (a) Mean opinion score. (b) Mean delay.

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In Fig. 6, the comparison between MDR and the ETT-based scheme is shown. The mean opinion score is shown in Fig. 6(a), where we observe a behavior similar to the one shown in Fig. 6. The average end-to-end delay is shown in Fig. 6(b). In contrast to what happens when the ETX is used, the routing mechanism that minimizes the total ETT on the path from the source to the destination yields smaller delays. However, the delays with MDR are comparable to those with ETT-based routing; in other words, the video quality is improved with minimum impact on delay with MDR.

B. Testbed Experiments

Next, we evaluate the MDR protocol on a wireless indoor testbed composed of 41 nodes [40]. The nodes are based on the Soekris net5501 hardware configuration and run a Debian Linux distribution. Each node is equipped with 500 MHz CPU, 512 MB of RAM, and a WN-CM9 wireless mini-PCI card, which carries the AR5213 Atheros main chip. Each node uses IEEE 802.11a to avoid interference from co-located campus networks. To further minimize interference from these other networks, all experiments were performed at night. The network topology of the testbed is shown in Fig.7.

The experiment setup consists of an initial raw video processed using the H.264 encoder with a maximum GOP size of 30 frames. The traffic load ranges from 2 to 12 concurrent video flows, where the sender and receiver pairs are randomly selected. Each scenario is repeated five times.

To capture the effect of the ETX-based and MDR routing schemes on the user experience, we measure the average MOS as the number of concurrent video flows in the network increases. Fig.7 shows that as the number of video connections in the network increases, the average MOS for both schemes decreases. However, when the traffic load increases, the MDR protocol computes multiple paths between the source and the

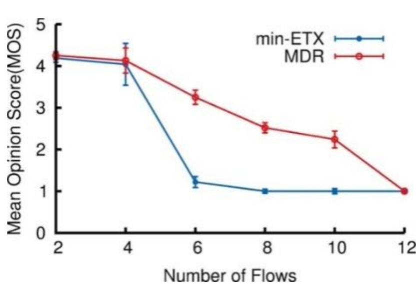

Fig. 7. Average value of MOS for a different number of concurrent video flows.

destination nodes and is better in distributing the load across the network such that the frames at the beginning of a GOP avoid congestion. On the other hand, the shorter paths computed through the ETX-based scheme become quickly congested, resulting in heavy packet losses. As discussed, we observe that this primarily has a negative impact on correctly decoding the relatively longer (but more important) I-frames, resulting in a worse user experience.

A visual comparison between Figs. 6 and 7 immediately shows the similarity in behaviors between our simulations and real experiments, thereby validating the realism of our simulations. Fig. 12 shows snapshots from video clips transmitted over the testbed under different traffic conditions for both the ETX-based and the MDR protocols. As shown in Fig. 11, when there are two connections in the network, the MOS for both routing schemes is the same. This is reflected in Fig. 12(a) and (b), where both snapshots are of very similar quality; in this case, the traffic load is fairly low, and congestion is not a big issue (the flows do not cause high levels of interference to each other). When there are eight concurrent video connections (and interference across connections is more prevalent), the MDR protocol achieves a higher MOS compared to the ETX-based scheme. The snapshot in the case of MDR is much clearer than the noisy snapshot form the ETX-based protocol. Specifically, our protocol distributes the I-frames across diverse paths with low interference; P-frames that are toward the end of GOPs are relatively packed together onto more congested paths. The ETX

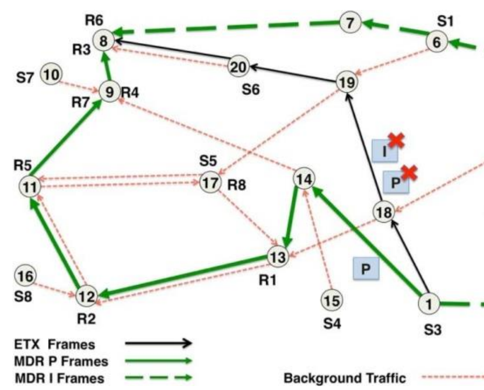

Fig. 8. Routes for I- and P-frames.

metric, which is agnostic to video semantics, does not distinguish between frames and packs them together, causing high distortion. It is difficult to explicitly prove that I- and P-frames follow somewhat disjoint paths due to the stochastic nature of the process. The intuition, however, is based on the fact that the sensitivities and of the I- and P-frames, respectively, are, in general, different. This has as a consequence that the frame-loss probability for an I-frame is different from that of a P-frame, resulting in their choosing different routes.

VII. Conclusion

In this paper, we argue that a routing policy that is application-aware is likely to provide benefits in terms of user-perceived performance. Specifically, we consider a network that primarily carries video flows. We seek to understand the impact of routing on the end-to-end distortion of video flows. Toward this, we construct an analytical model that ties video distortion to the underlying packet-loss probabilities. Using this model, we find the optimal route (in terms of distortion) between a source and a destination node using a dynamic programming approach. Unlike traditional metrics such as ETX, our approach takes into account correlation across packet losses that influence video distortion. Based on our approach, we design a practical routing scheme that we then evaluate via extensive simulations and testbed experiments. Our simulation study shows that the distortion (in terms of PSNR) is decreased by 20% compared to ETX-based routing. Moreover, the user experience degradation due to increased traffic load in the network is kept to a minimum.

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