A NEW APPROACH FOR DISTORTION-RESISTANT ROUTING FRAMEWORK FOR VIDEO TRAFFIC IN WIRELESS

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***Abstract*— Broadband and wireless communication systems in today's world are more robust and ubiquitous than they used to be earlier. Video traffic has become a problem nowadays due to the increase in the use of wireless networks. Maintaining a good quality of video is very important. The video quality is affected by: 1) the distortion due to compression at the source and 2) distortion due to both wireless channel induced errors and interference. Here, we are working to reduce distortion in video traffic flowing over a wireless network. Today's users demand high quality videos to be delivered seamlessly on their devices. In this paper, we discuss routing policies to reduce video distortion on an end to end basis. Conventional and popular link based routing metrics such as ETX cause high video distortion as they do not account for dependence across the links of a path. Hence, video traffic merges onto few paths causing distortion. To reduce the distortion in videos and report frame loss in videos, we build an analytical framework. A routing protocol for reducing distortion in videos is designed based on the framework's routing policy. Simulations are done to show the protocol designed is efficient in minimizing video distortion.**

Index Terms*— video distortion, distortion minimization, routing protocol.

I. Introduction

Video quality can be improved by accounting for application requirements. The schemes used to encode a video clip can accommodate a certain number of packet losses per frame. If the number of lost packets exceeds a threshold value then the frame cannot be decoded correctly. Thus, resulting a distortion. The value of distortion depends on position of unrecoverable video frames in the GOP (Group of Pictures). So, we construct an analytical model to view the behaviour of the process that describes the evolution of frame losses in the GOP. Using this we capture how the choice of path for an end-to-end flow affect the performance of a flow in terms of video distortion. Our model is built based on a multilayer on approach as shown in fig1.The packet-loss probability on a link is mapped to the probability of a frame loss in the GOP and the frame loss probability is then directly associated with the video distortion Metric. Using the above mapping from the network specific property to the application-specific quality metric, we indicate the problem of routing as an optimization problem where we can find the path from the source to the destination that can minimizes the end-to-end distortion. The solution for this problem is based on a dynamic programming approach that effectively captures the evolution of the frame –loss. After this we design a practical routing protocol, based on the above solution, to minimize routing distortion. Television) and VOIP (Voice over Internet Protocol) which have high bit – rate multimedia content and high QOS (Quality of Service) are being delivered to users due to increase in bandwidths of broadband year after year. Providing broadband access is still a challenge in rural and mountainous regions because of technical and/or economic reasons due to which people living in such regions cannot benefit from the advantages offered by broadband access . 802.11 WLANs have limited coverage and one-hop wireless networks such as 3G and licensed WiMAX are costly and usually require licences for channel. Multihop broadband wireless networks is a solution which provides broadband access along with much needed QoS . Multihop wireless networks have one or many intermediate nodes which independently communicate among themselves along the route and send or receive packets using wireless links. Multihop networks can perform routing in a self-made manner, since they don't rely on any past framework base. Research interest has been increasing in wireless networks to deliver multimedia services as multimedia is expected to be a major traffic source over next – generation wireless networks . Multimedia traffic is becoming very popular in wireless networks with the coming of smart phones. Transfer of video clips, pictures and voice data in areas of natural calamities, disaster recovery, drought hit areas, etc. to facilitate mission management by government agencies and NGO's has come as a hope to people in distress. Under such extreme scenarios maintaining a good quality of the video which is transferred is demanding from the user's prospect. The quality of video sent over wireless network is influenced by: 1) the use of compression techniques during which noise or distortion is added at the source and 2) both, errors entering in wireless channel and tampering also causes distortion in video . Transmission losses can prevented by using different levels of encoding described in video encoding standards like MPEG-4 or H.264/AVC. I-type, P-type and Btype frames are groups of frame types which are defined in these encoding standards. In case of I-type frames data is encoded independently. In case of P-type and B-type frames encoding is performed based on the data encoded within other frames. Application-level performance of video transmissions can be derived using Group of Pictures (GoP) which allows for the matching of frame losses into a distortion metric . Routing is the most often neglected critical functionality which affects the end-to-end video quality. There is a correlation between losses on the links that constitute routes from a source node to a destination node but most routing protocols which are designed for wireless multihop networks are application specific. Sometimes, few links can become heavily loaded with traffic which results in video distortion and while other links are less utilized as network traffic is independent. Network parameters and not application parameters are the only basis on which most of the routing protocols make their decisions to route the traffic.

II. PROPOSED SYSTEM

In this paper, our thesis is that 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 (simply referred to as 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 (instead of just focusing on a network quality metric such as the packet-loss probability) 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. Our model is built based on a multilayer approach.

A. ADVANTAGES OF PROPOSED SYSTEM:

Our solution to the problem is based on a dynamic programming approach that effectively captures the evolution of the frame-loss process.

Minimize routing distortion.

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.

III. LITERATURE SURVEY

A. Overview of the H.264/AVC video coding standard

AUTHORS: T. Wiegand, G. J. Sullivan, G. Bjontegaard, and A. Luthra

H.264/AVC is newest video coding standard of the ITU-T Video Coding Experts Group and the ISO/IEC Moving Picture Experts Group. The main goals of the H.264/AVC standardization effort have been enhanced compression performance and provision of a "network-friendly" video representation addressing "conversational" (video telephony) and "nonconversational" (storage, broadcast, or streaming) applications. H.264/AVC has achieved a significant improvement in rate-distortion efficiency relative to existing standards. This article provides an overview of the technical features of H.264/AVC, describes profiles and applications for the standard, and outlines the history of the standardization process.

B. A high throughput path metric for multi-hop wireless routing

AUTHORS: D. S. J. D. Couto, D. Aguayo, J. Bicket, and R. Morris

This paper presents the *expected transmission count* metric (ETX), which finds high-throughput paths on multi-hop wireless networks. ETX minimizes the expected total number of packet transmissions (including retransmissions) required to successfully deliver a packet to the ultimate destination. The ETX metric incorporates the effects of link loss ratios, asymmetry in the loss ratios between the two directions of each link, and interference among the successive links of a path. In contrast, the minimum hop-count metric chooses arbitrarily among the different paths of the same minimum length, regardless of the often large differences in throughput among those paths, and ignoring the possibility that a longer path might offer higher throughput.This paper describes the design and implementation of ETX as a metric for the DSDV and DSR routing protocols, as well as modifications to DSDV and DSR which allow them to use ETX. Measurements taken from a 29-node 802.11b test-bed demonstrate the poor performance of minimum hop-count, illustrate the causes of that poor performance, and confirm that ETX improves performance. For long paths the throughput improvement is often a factor of two or more, suggesting that ETX will become more useful as networks grow larger and paths become longer.

C. Packet loss resilient transmission of MPEG video over the internet

AUTHORS: J. M. Boyce

A method is proposed to protect MPEG video quality from packet loss for real-time transmission over the Internet. Because MPEG uses inter-frame coding, relatively small packet loss rates in IP transmission can dramatically reduce the quality of the received MPEG video. In the proposed high-priority protection (HiPP) method, the MPEG video stream is split into high- and low-priority partitions, using a technique similar to MPEG-2 data partitioning. Overhead resilient data for the MPEG video stream is created by applying forward error correction coding to only the high-priority portion of the video stream. The high- and low-priority data, and resilient data, are sent over a single channel, using a packetization method that maximizes resistance to burst losses, while minimizing delay and overhead. Because the proposed method has low delay and does not require re-transmission, it is well suited for interactive and multicast applications. Simulations were performed comparing the improvement in video quality using the HiPP method, using experimental Internet packet loss traces with loss rates in the range of 0–8.5%. Overhead resiliency data rates of 0%, 12.5%, 25%, and 37.5% were studied, with different compositions of the overhead data for the 25% and 37.5% overhead rates, in an attempt to find the "best" composition of the overhead data. In the presence of packet loss, the received video quality, as measured by PSNR and the Negsob measure, was significantly improved when the HiPP method was applied.

D. Layered coded vs. multiple description coded video over error-prone networks

AUTHORS: Y.-C. Lee, J. Kim, Y. Altunbasak, and R. M. Mersereau

Layered (LC) and multiple description coding (MDC) have been proposed as source coding techniques that are robust to channel errors for video transmission. LC and MDC have similar characteristics: they both generate multiple sub-bitstreams, and it is permissible to drop some portion of the data from the sub-bitstreams during transmission for both methods. However, they are different in the sense that the sub-bitstreams for LC have different levels of importance while all sub-bitstreams for MDC are equally important. Since these two encoding techniques have similar properties, some performance comparisons between LC and MDC have recently been reported. However, these studies are still not conclusive because several scenarios have not been carefully considered. Furthermore, they have been performed in different environments. In this paper, we further investigate the error-resilience capabilities of these two encoding techniques through extensive experimentation. Although some of our conclusions agree with those in the literature, we believe that this paper provides the most comprehensive performance comparison yet between LC and MDC.

E. Layered coding vs. multiple descriptions for video streaming over multiple paths

AUTHORS: J. Chakareski, S. Han, and B. Girod

In this paper, we examine the performance of specific implementations of multiple description coding and of layered coding for video streaming over error-prone packet switched networks. We compare their performance using different transmission schemes with and without network path diversity. It is shown that, given specific implementations, there is a large variation in relative performance between multiple description coding and layered coding depending on the employed transmission scheme. For scenarios where the packet transmission schedules can be optimized in a rate-distortion sense, layered coding provides a better performance. The converse is true for scenarios where the packet schedules are not rate-distortion optimized.

IV. RELATED WORK

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 baselayer 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 ratedistortion 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.

V. CONCLUSION AND FUTURE WORK

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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