Understanding the effect of network-coding and video-encoding on multimedia streaming for peer-to-peer (P2P) systems in wireless networks

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Abstract - Although great advances have been realized in wireless networking over the last ten (10) years, the wireless transport medium is not ideal for multimedia video applications. This paper proposes the basis for a study in quantitatively understanding the effect in deploying various combinations of video encoding and network coding approaches on peer-to-peer streaming systems in a wireless environment. A simulation model will be implemented and used for analyzing the video traffic and its flow through the network. The PSNR (peak signal-to-noise ratio) will be used in understanding the quality of video for the various cases. The PSNR is a common quality metric used that is based on the MSE (mean-squared-error) of two images. The study will also use the coding-bit-rate, packet-loss-ratio and video-characteristics as key factors in quantitatively comparing various coding schemes [19]. In all cases, the video packets will be multicasted and a retransmission-based error-control method will be used.

Keywords: Wireless networks, P2P, Multimedia streaming, compression, network coding

1 Introduction

Streaming video traffic over a wireless peer-to-peer system is becoming more and more in demand. Applications of video streaming over wireless are ranging from multimedia applications in mobile devices to high data-rate video applications in home networks. Although great advances have been realized in wireless networking over the last ten (10) years, the wireless transport medium is not ideal for multimedia video applications due to fragile and dynamically changing links, and contention issues with limited resources. There is also an effect of interference due to devices operating in the same frequency range.

Video and multimedia streaming is a continuous process that is delay sensitive such that, any packets not received by a certain timeframe are dropped. Traditional approaches to wireless transmission are not sufficient for good quality-of-service of wireless video streaming.

2 Key Issues

Some of the key issues that must be considered and included in the study pertain to resource allocation, video coding or compression, transport layer reliability, and network coding.

2.1 Resource Allocation

Resource allocation is one of the main challenges in wireless video streaming due to the erratic behavior of the medium. Implementation of a suitable video coding algorithm and reliable transporting of coded packets to receivers are the other aspects that need to be taken into account in wireless video streaming. Novel cross-layer design frameworks propose approaches to resource allocation, routing and rate allocation [2, 21-22]. Due to continuously changing link qualities, routing is a significant concern in wireless mesh networks. Routing solutions are needed to prevent low throughput in multi hop links which occurs with the contention among adjacent links [1]. In cross-layer design frameworks, two or more layers are jointly optimized for increasing the streaming performance. In the cross-layer design study of Wang et al., network coding was also used to adopt reliability of end-to-end transmission at the application layer [2].

2.2 Video Coding

Video encoding or compression is essential before transmission because of the large bit rates of a digital video and limited bandwidth of a wireless channel. In addition, compression methods are very important in terms of achieving optimal QoS by aiming to send only relevant data [18]. There are three sources of redundancies which are reduced by encoding or compression. One such redundancy is spatial redundancy. Spatial redundancy is having neighboring pixels similar in a single frame. The other redundancy is called temporal redundancy, where adjacent frames are correlated. And as a result of reducing these redundancies, a third one occurs in the stream of output symbols. Intra-frame coding, inter-frame coding and variable length coding are techniques used for eliminating the redundancies [16, 17]. H.26X is the family of video coding standards. Some of the mainstreamed standards are Advanced Video Coding
Bandwidth limitations and packet losses are the main issues in multicasting video packets over the Internet. Aiming for more reliable data communications by reducing the number of packet losses, the retransmission process at the link-layer is provided by IEEE 802.11 standard MAC protocol. However, link-layer retransmission is not effective in the presence of heavy network traffic loads. As the retransmission limit increases, end-to-end delay increases and as a result, it reduces the quality of streaming video over multi-hop wireless mesh networks. If there is TCP traffic along with video streaming, both video and TCP performances are not satisfactory with retransmissions [6]. Retransmission based error control methods are widely studied for video transmission over wireless networks and will be implemented in the study [7-10].

Some parameter settings need to be adjusted in reducing TCP delays, such as; disabling Nagle’s algorithm which causes transmission delays by limiting the transmission of small segments; enabling SACK; and using larger receiver buffers for delay sensitive applications. It is also possible to reduce the delay by using packet splitting and having multiple parallel connections with the video flow [23].

2.4 Network Coding

Network coding makes sending more information in one transmission possible by combining packets from different flows into a single packet. In the study of Seferoglu and Markopoulou, video-aware opportunistic network coding scheme increased the throughput and as a result enhanced the video quality [15].

In lossy networks, network coding reduced the number of retransmission of the packets thus increasing the reliability of the network. When network coding was compared to link-by-link ARQ and end-to-end FEC error-control techniques, the results showed that network coding outperformed the other techniques both in lossy links and links where loss-packet probability was lower [3].

Rate adaptation at the application layer avoids link congestion, maintains continuous stream of video by meeting the deadlines of the packets and on the other hand reduces video quality [12].

As it is mentioned before, video streaming requires a particular amount of bandwidth in satisfying the consistency condition. Video servers need to provide high quality video to receivers even in peak traffic times. There are also other concerns like cost and storage in such client-server and P2P systems. Multicasting proposes a solution to these concerns by replicating the packets in intermediate nodes in network [12].

Multicasting reduces the cost by proposing a multicast tree alternative to the client-server systems; however, routers which can manage multicasting, are not widely deployed. Peer-to-peer (P2P) systems with overlay structures do not need a central server like conventional client-server systems [11, 12]. Network coding can also be used with peer-to-peer (P2P) streaming systems. One of the main issues in using P2P streaming is the degree of heterogeneity of the receivers in terms of bandwidth, CPU capacity and screen resolution; and limited upload capacity of the peers.

(H.264/AVC), Scalable Video Coding (H.264/SVC), and Multiview Video Coding (H.264/MVC). A next generation coding standard is H.265. Another emerging concept is Distributed Video Coding (DVC) which is popular for video surveillance applications where the decoder is more complex than the encoder. On the other hand, in traditional H.26X coding systems, the encoder has more complexity with motion estimation. Traditional coding systems are more suitable for wireless video streaming applications like Video-on-Demand (VoD) and applications where the video is compressed once and decoded more than once [14]. Source coding techniques and channel coding solutions are widely proposed for successful video streaming in wireless networks.

Scalable video coding (SVC) is the scalable extension of H.264/AVC standard. SVC offers scalability by having video which is coded into more than one layer. With a base layer and enhancement layers, SVC provides a wide range of video quality, resolution, and picture rate in a heterogeneous network. Bit errors and channel errors are the main causes of packet losses in video transmission channels. Using the user datagram protocol which discards packets with errors is another issue which decreases the video quality. It is likely to have a transmission error propagate temporally and spatially, unless error control methods are applied. Error-resilient coding is an error-control approach which adds redundancy to data to get minimum distortion. The redundancy can provide concealment of errors, detection of data losses and preventing of error propagation. [20].

2.3 Transport Layer

TCP (Transmission Control Protocol) and UDP (User Datagram Protocol) are currently the major protocols used for video transmission over the Internet. Stream Control Transmission Protocol (SCTP) is an up-and-coming transport protocol that is message-oriented like UDP and ensures reliable, in-sequence transport of messages with congestion control like TCP. TCP is not preferred for video streaming applications due to stringent delay requirement which is hard to satisfy because TCP uses a back-off procedure and retransmission process [4]. A traditional TCP approach can be better used by having multiple TCP connections for one streaming application. With the multi-TCP design, it overcomes the short-term limited bandwidth concern and rate control insufficiency of TCP. This approach is implemented at the application layer by giving control of the sending rate to the receiver during congestion [5]. Another study that uses multipath TCP-based streaming, proposes a design named Dynamic MPath Streaming (DMP). DMP model shows that it is possible to achieve satisfactory performance with TCP in more stable networks like wired networks [4].

In regards to UDP, Real Time Protocol (RTP) and Real Time Control Protocol (RTCP) run on top of the UDP. These protocols are suitable for media streaming and IPTV with the properties provided like security, media codec control and error protection. These protocols also provide basic RTP media encapsulation and synchronization, quality-of-service signaling and multi-party communication coordination [24].
3 Conclusion

It is expected that the simulation study findings will indicate improved performance with appropriate combinations of video encoding and network coding approaches deployed on peer-to-peer streaming systems in a wireless environment. One expects the scalable video coding approach to effectively deal with the heterogeneous receiver problem for P2P systems. One also expects network coding to provide higher throughput to the system [13]. OpNET's Modeler networking simulation environment will be used in modeling the problem and conducting the study.

4 References

