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A SURVEY ON SCALABLE VIDEO CODING APPROACH FOR WIRELESS VIDEO TRANSMISSION

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ABSTRACT

Wireless Ad Hoc Networks are characterized by their ability to communicate amongst one another independently of any existent infrastructure. Such networks could be vital for multiple applications like medical applications, for emergency applications, rescue operations to name a few. Support for multimedia applications over wireless ad hoc networks has gained tremendous momentum in the past decade. The survey presented here is primarily targeted towards realization of real time video communication in wireless ad hoc networks. When any video is transmitted from a transmitter to a receiver in any wireless network, various video packets may be lost due to channel errors, transmission errors or low bandwidth. Active video communications are time critical and to attain the required Quality of Service it is essential to minimize the packet loss. In such a scenario, a client joining the broadcast service only requests the scalable layers, which provide either a signal that the device is capable or chooses to process. From the past industrial result it is seen that the transmission of multiple video signals using SVC is much more efficient in terms of bit-rate compared to simulcast transmission.As the video stream is passed from one node to another, in order to reduce the packet losses in the transmission path, several frames are introduced through which transmissions are to take place. The video streams are made to pass through this frames

KEYWORDS: video stream, wireless network, performance

I. INTRODUCTION

The use of wireless mobiles and tablets has become too common these days. The transmission of video also has become ordinary in wireless networks. But there will be a large number of packet loss in case of video transmission. In such a scenario, a client joining the broadcast service only requests the scalable layers, which provide either a signal that the device is capable or chooses to process. From the past industrial result it is seen that the transmission of multiple video signals using SVC is much more efficient in terms of bit-rate compared to simulcast transmission.

As the video stream is passed from one node to another, in order to reduce the packet losses in the transmission path, several frames are introduced through which transmissions are to take place. The video streams are made to pass through this frames .Each frame is divided into two layers: Base layer and Enhancement layer .The base layer is the primary layer and the enhancement is the added layer. The video bit stream is carried by the base layer and if not then it is carried by the enhancement layer. Depending on the PSNR, part of the video bit stream passes through the base layer and the remaining part is carried by the enhancement layer. Almost maximum of the bits are passed through the base layer. Only the lost packets are retransmitted using the enhancement layer. If the number of packets lost increases then the number of packets to be transmitted through the enhancement layer increases. Thus the base layer and the enhancement layer works in coordination. The packets that are transmitted through the base layer when combined can almost approximately reconstruct back the original video stream at least to some extent. But the packets transferred through the enhancement layer of any frame are random bits that were not passed successfully through the base layer. So these bits when put together can never successfully reconstruct the original bits stream.

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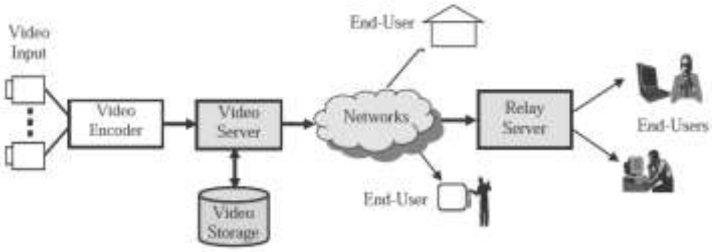
The requirement of the enhancement layer arises only for the lost bits in the transmission done through the base layer.



A typical video streaming system

VIDEO STREAMING METHODS

A) Overview

From decoder side, there are two approaches to view video over networks, which have been extensively investigated in recent years. The first is the downloading-based approach, where the complete video file is downloaded to local storage e before playback. With this approach, the time to download a video increases with the amount of data,which is proportional to the quality of the video and duration of the video. The network bandwidth also plays a significant role in the downloading time as well. The other way to view video is video streaming, where the video is viewed while it is being transmitted.

Video streaming will be the focus of the following discussion.

In video streaming, the end user can start viewing the video almost as soon as it begins downloading with a limited delay. To achieve a seamless playback, the data must be received at a rate that allows the client device to decode and display each frame of the video sequence according to a playback schedule. The video server has two ways to provide the compressed video bitstreams. The first is to select one among multiple non scalable bitstreams and the second is the use of a single bitstream, which is encoded with the scalable video encoding or can be transcoded during the streaming . In the first way, several bitstreams for the same video with different bit-rates, which may also have different temporal or spatial resolutions, have been stored in the video server. The end-user can select the bitstream according to its capability and available bandwidth of the network. The advantages of this method are that the compressed bitstream is optimized to the specified user and is the decoder has lower complexity since it only needs to receive and decode a single layer. he main disadvantage is that the video sever must store multiple bitstreams for the same video, which is redundant and could impose significant memory constraints with very large video repositories. Also, the different versions of the video have to be pre-encoded, which makes real-time applications almost impossible. This approach is also limited in the granularity that it could provide.

The second method of scalable video streaming is implemented with the scalable encoded video bitstreams . In this method the video is encoded once and stored in the video sever. The encoded video bit stream can be truncated in ways such as SNR,temporal and spatial scalability based on the requirements of the end user and network conditions as shown in Figure 9. This method is attractive since it provides more flexibility in getting the desired compromise between granular scalability and coding performance. As mentioned previously, this method has to be evaluated by the market. First, the coding technique must not incur significant loss of coding efficiency compared to single layer coding schemes. With significant loss in coding efficiency, it is likely that the content providers would choose not to adopt the coding format. The other issue is decoder complexity. If the scalable decoder is costly to produce, then there may be limited or no deployment of devices capable of receiving a scalable encoded bitstream. The third method is to use a single encoded bitstreams with higher quality. During the streaming, the bitstreams are converted to match the end userdevice and network conditions with a transcoder. The key advantage of this method is that transcoding techniques could be easily installed on servers to satisfy a very diverse set of network and terminal constraints. The transcoding solution offers a layer of flexibility between the content providers who encode the data and consumers that wish to receive the data. The main drawback compared to the scalable coding solution is that transcoding typically requires more computation than simple bitstream truncation. However, advances in the area of transcoding have pushed the complexity much lower than full re-encoding of video without sacrificing quality.

B) Comparison of Streaming Methods

As described in the previous section, scalable coding specifies the data format at the encoding stage independently of the transmission requirements, while transcoding converts the existing data format to meet the current transmission requirements. With scalable video coding, the video is encoded once, then various qualities, spatial resolutions, and/or temporal resolutions could be extracted. Ideally, this scalable

II. CHALLENGES IN WIRELESS NETWORKS IN VIDEO STREAMING

- Dynamic varying network topology
- Imprecise state information
- Scare resources
- Absence of communication infrastructure
- Lack of centralized control
- Power limitations
- Heterogeneous nodes and networks
- Error-prone shared radio channel
- Hidden terminal problem
- Insecure medium
- Other layers

2.1 Dynamically varying network topology:
Since the nodes in an ad hoc wireless network do not have any restriction on mobility, the network topology changes dynamically. Hence the admitted QoS sessions may suffer due to frequent path breaks, thereby requiring such sessions to be re-established over new paths. The delay incurred in re-establishing a QoS session may cause some of the packets belonging to that session to miss their delay targets/deadlines, which is not acceptable for applications that have stringent QoS requirements.

2.2 Imprecise state information: In most cases, the nodes in an ad hoc wireless network maintain both the link-specific state information and flow-specific state information. The link-specific state information includes bandwidth, delay, delay jitter, loss rate, error rate, stability, cost, and distance values for each link. The flow specific information includes session ID, source address, destination address, and QoS requirements of the flow (such as maximum bandwidth requirement, minimum bandwidth requirement ,maximum delay, and maximum delay jitter).The state information is inherently imprecise due to dynamic changes in network topology and channel characteristics. Hence routing decisions may not be accurate, resulting in some of the real-time packets missing their deadlines.

2.3 Lack of central coordination: Unlike wireless LANs and cellular networks, AWNs do not have central controllers to coordinate the activity of nodes. This further complicates QoS provisioning in AWNs
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2.4 Error prone shared radio channel: The radio channel is a broadcast medium by nature. During propagation through the wireless medium the radio waves suffer from several impairments such as attenuation, multi-path propagation ,and interference (from other wireless devices operating in the vicinity).

2.5 Hidden terminal problem: The hidden terminal problem is inherent in AWNs. This problem occurs when packets originating from two or more sender nodes, which are not within the direct transmission range of each other, collide at a common receiver node. It necessitates re transmission of packets, which may not be acceptable for flows that have stringent QoS requirements.

2.6 Limited resource availability: Resources such as bandwidth, battery life, storage space, and processing capability are limited in AWNs. Out of these, bandwidth and battery life arevery critical resources, the availability of which significantly affects the performance of the QoS provisioning mechanism. Hence efficient resource management mechanisms are required for optimal utilization of these scarce resources.

2.7 Insecure medium: Due to the broadcast nature of the wireless medium, communication through a wireless channel is highly insecure. Hence security is an important issue in AWNs, especially for military and tactical applications.
AWNs are susceptible to attacks such as eavesdropping, spoofing, denial of service, message distortion, and impersonation. Without sophisticated security mechanisms, it is very difficult to provide secure communication guarantees.

III. RELATED WORK

[1]To address critical problems such as performance limitation, reliability of wireless network and throughput, this research proposes an application-layer source-FEC (Forward Error Correction) coding framework dubbed ESCOT (adaptivE Source-FEC COding over TCP). First, optimization problem of joint source-FEC coding to minimize the end-to-end distortion of real-time video communication over TCP is formulated. Second, a heuristic solution for effective loss rate approximation, source rate control, and FEC coding adaptation is developed. ESCOT is distinct from existing source-FEC coding schemes in proactively analyzing and leveraging the TCP characteristics. The proposed solution is able to effectively mitigate both consecutive and sporadic video frame drops caused by congestion and random packet losses. congestion and TTL based packet filtering technique to identify attacker and blacklist that attacker.

[2] 4K-ultra-high definition (UHD) video encoded by high efficiency video coding (HEVC) to investigate its feasibility for 4K-UHD TV broadcasting services, is presented. The subjective quality assessment on the HEVC-encoded (impaired) 4K-UHD video is conducted for the three target bitrates of 18, 23, and 36 Mb/s, the two color formats of YUV420 and YUV444, and the two viewing distances of 0.75 times the height of a display screen (H) and 1.5 H.

[3] consider a group of wireless users who may interact with an FVV by independently switching views. A novel live FVV streaming network where each user pulls a subset of anchors from the server via a primary channel is studied. To enhance anchor availability at each user, a user generates network-coded (NC) packets using some of its anchors and broadcasts them to its direct neighbors via a secondary channel. Given limited primary and secondary channel bandwidths at the devices, author seek to maximize the received video quality (i.e., minimize distortion) by jointly optimizing the set of anchors each device pulls and the anchor combination to generate NC packets. Authors first formulate the problem and show that it is NP-hard. Then propose a scalable and effective algorithm called PAFV (Peer-Assisted Freeview Video). In PAFV, each node collaboratively and distributedly decides on the anchors to pull and NC packets to share so as to minimize video distortion in its neighborhood.

[4] A novel reconfigurable hardware architecture for interpolation filtering in high efficient video coding that adapts to run-time changes of the number of interpolation filter calls and thereby provides a high potential of energy efficiency is proposed in this article. It employs a picture-based prediction scheme to estimate the number of interpolation filter calls at run-time by monitoring the group of pictures history based on video coding structure knowledge. Reconfigurable acceleration engines are developed that can adapt to different filter types. Dynamic composition of different instances of these engines enables different implementation versions with area versus throughput tradeoff.

[5] Currently, dynamic adaptive streaming over HTTP (DASH) standard is used to change the video resolution according to the network capacity of the end user's. However, if the video resolution changes frequently, the attention of the users can be affected decreasing their quality-of-experience (QoE). In this context, this paper proposes a no-reference video quality metric that takes three impairment factors into account, initial buffering delay, temporal interruptions or pauses, and video resolution changes during a video transmission. Also, the temporal locations of pauses and resolution changes are considered. In order to perform this task, extensive subjective tests were conducted.

IV. PROPOSED SURVEY

4.1 Reserved bandwidth approach, which makes difficulty to reutilize bandwidth

4.2 Loss in wired networks is typically caused by excessive congestion that causes packets to be dropped at routers in the network. A negligible amount of data is lost due to corruption during transmission on a wire. A wireless link, however, typically suffers much more loss due to data being corrupted during transmission. One cause of loss in wireless transmission is fading, in which multiple versions of the same signal are received at the destination. If these signals are out of phase with each other or Doppler-shifted, they can interfere with each other. Other types of interference may also cause problems in wireless transmissions. This interference may come from other communication occurring on the same frequency, electrical noise, or possibly even intentional communication jamming.

4.3 Another obstacle in wireless QoS involves propagation delay. Some wireless networks span distances that are measured in kilometers. In these networks, propagation delay can be a tremendous burden to all

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communication, but especially to communication that requires a guarantee on total delay. This problem may exist to some extent in metropolitan area networks (MANs), and it is a significant issue in satellite communications.

V. STREAMING OVER MESH NETWORKS

Video streaming over wireless mesh networks imposes additional challenges introduced by multi-hop transmissions. Cross-layer design and optimization for this problem is a very active area of investigation with many remaining open problems. In the following, a survey of research efforts in joint optimization of multiple protocol layers is presented first, followed by discussions on routing for media streaming, and rate allocation among multiple video streams in mesh networks.

5.1 Multi-layer resource allocation

The flexibility offered by cross-layer design has been exploited in a number of research efforts. Joint optimization of power allocation at the physical layer, link scheduling at the MAC layer, network layer low assignment and transport layer congestion control has been investigated with convex optimization formulations. Our own cross-layer design framework attempts to maintain a layered architecture while exchanging key parameters between adjacent protocol layers. The framework allows enough flexibility for significant performance gains, while keeping protocol design tractable within the layered structure, as demonstrated by the preliminary results exploring adaptive link-layer techniques, joint capacity and low assignment, media-aware packet scheduling and congestion aware video rate allocation.

5.2 Routing for Media Streaming

Routing over wireless mesh networks is a difficult problem due to dynamic link qualities, even when nodes are static . For video streaming, multipath routing has been proposed in combination with multiple description coding, to achieve robust delivery via path diversity. In spite of the high data rates achieved over single-hop wireless transmissions, throughput over a multi-hop wireless path is typically significantly lower, due to contention among adjacent links along the path. Since video packets need to be delivered by their play out deadline, self-inflicted congestion may drastically degrade received video quality over a throughput-limited path . Route selection should therefore minimize network congestion, measured as average per-link delay of all packets. Congestion-minimized routes can be derived from solutions to a classical flow assignment problem, either via centralized computation or with a distributed algorithm

5.3 Multi-Stream Rate Allocation

When multiple streams share a wireless mesh network, their rates need to be jointly optimized to avoid network congestion while maximizing overall received video quality. The joint rate allocation problem can be solved by minimizing the Large gain cost of total video distortion and overall network congestion. For each stream, the optimal allocated rate strikes a balance between minimizing its own video distortion and minimizing its contribution to overall network congestion. This is achieved by a distributed rate allocation protocol, which allows cross-layer information exchange between the video streaming agents at the application layer on the source nodes and the link state monitors at the MAC layer on the relay nodes.

VI. NNETWORK PROTOCOLS FOR SCALABLE VIDEO STREAMING

TCP is the dominant protocol in the Internet for data transfer. In general, TCP could also be used for video streaming over Internet . However, in order to provide reliable and good quality video streaming over TCP, several problems have to be addressed. The first is how to handle the data rate variability. In the Internet, the data rate may have sawtooth behavior, i.e., additive increase and multiplicative decrease. The second is the end to-end delay due to retransmission at same time. However, these problems can be addressed with buffering the data. Therefore, the proper buffer size should be decided considering the impact on various performance metrics such as delay, smoothness of playback and data loss. In general, a small buffer size implies smaller delay since the time between the start of transmission and the first picture being displayed is less with a smaller buffer. With regards to smoothness of playback, a larger buffer size will typically ensure smoother playback since larger variations in the bit-rate and transmission time could be tolerated. Larger buffer sizes will also lead to fewer dropped packets in the receiver due to buffer overflow.

Given these dependencies, being able to analytically model a video streaming system with TCP is necessary .The minimum buffer size requirements for three scenarios have been studied: 1) when TCP throughput matches video encoding rate, 2) when TCP throughput is smaller than the encoding rate, and 3) when TCP throughput is limited by the maximum window size. Another problem with video streaming over TCP is how it handles network layer packet loss. If packets are delayed or damaged, TCP will effectively stop traffic until either the original packets or backup packets arrive. In this sense, TCP is unsuitable for video streaming because TCP handles the packet loss with the method of retransmission which causes further jitter and skew. UDP (User Datagram Protocol) is another network protocol for video streaming . UDP handles the packet loss or delay in different way. It allows packets to drop out if these packets are timeout or damaged. This function introduces the packet loss which user can hear or see video damaged, but the stream will continue. With UDP, the error concealment function may be needed in video decoders.

Another problem with UDP is that many network firewalls block UDP information. In this case, video streaming over TCP is the only choice since it can get around the firewalls using well-known port numbers (e.g., HTTP or RTSP).RTP (Real-time Transport Protocol) is alternative choice for video streaming .RTP is an Internet standard protocol for the transport of real-time data, including audio and video. RTP consists of two parts, a data part and a control part which is called RTCP. The data part of RTP supports real-time transmission for continuous media such as video and audio. It provides timing reconstruction, loss detection, security and content identification. The RTCP (RTP control protocol) part provides source identification and support for gateways like audio and video bridges as well as multicast-to-unicast translators. It offers Quality-of-Service (QoS) feedback from receivers to the multicast group as well as support for the synchronization of different media streams. RTP/RTCP is commonly built on the top of UDP and provides some functionality for media transport. But RTP does not guarantee the QoS, address the reservation and negotiate the media format.

VII. CONCLUDING REMARKS

In this article, an overview of scalable video streaming has been presented. The main components of a scalable video streaming system include video server with storage, video encoder, video transcoder or bitstream truncator, and network protocols which enable the transport of video data to end-users. Several technical aspects related to video encoding, streaming methods and network related issues have been discussed. Since video streaming is an extremely broad area, many special topics such as rate control, transmission from multiple servers, peer-to-peer networking, caching strategies, crosslayer design, QoS and DRM issues have not been covered in this article. Nevertheless, we hope this article provides a useful overview for those readers that might not be familiar with this area and gives some useful links to related works.

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