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RESEARCH ARTICLE

Secure and Energy-Efficient Cooperative Video Distribution over Wireless Networks

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Abstract— For real-time video broadcast where multiple users are interested in the same content, mobile-to-mobile cooperation can be utilized to improve delivery efficiency and reduce network utilization. Under such cooperation, however, real-time video transmission requires end-to-end delay bounds. Due to the inherently stochastic nature of wireless fading channels, deterministic delay bounds are prohibitively difficult to guarantee. For a scalable video structure, an alternative is to provide statistical guarantees using the concept of effective capacity/bandwidth by deriving quality of service exponents for each video layer. Using this concept, we formulate the resource allocation problem for general multi hop multicast network flows and derive the optimal solution that minimizes the total energy consumption while guaranteeing a statistical end-to-end delay bound on each network path.

Indexed Terms: - wireless fading channels, multi hop network, mobile-to-mobile cooperation.

I. INTRODUCTION

The real-time nature of video broadcast demands quality of service (QoS) guarantees such as delay bounds for end-user satisfaction. Given the bit rate requirements of such services, delivery efficiency is another key objective. The basic level of quality is supported by the base layer and incremental improvements are provided by the enhancement layers. Deterministic delay bounds are prohibitively expensive to guarantee over wireless networks. Consequently, to provide a realistic and accurate model for quality of service, statistical guarantees are considered as a design guideline by defining constraints in terms of the delay-bound violation probability. For scalable video transmission, a set of QoS exponents for each video layer are obtained by applying the effective bandwidth/capacity analyses on the incoming video stream to characterize the delay requirement. The problem of providing statistical delay bounds for layered video transmission over single hop unicast and multicast links was considered in this system.

For general multi hop multicast network scenarios, it is inefficient to allocate resources independently among network links since the variation in the supported service rates among different links affects the end-to-end transport capability in the network. Cooperation among mobile devices in wireless networks has the potential to provide notable performance gains in terms of increasing the network throughput, extending the network coverage, decreasing the end-user communication cost, and decreasing the energy consumption.

We model the queuing behavior of the cooperative network according to the effective capacity link layer model. Based on this model, we formulate and solve the flow resource allocation problem to minimize the total energy consumption subject to end-to-end delay bounds on each network path.

II. REVIEW OF SOME VIDEO DISTRIBUTION MECHANISMS AND PITFALLS

In this section we will review some of the existing mechanisms used for video distribution and also discuss some of their weaknesses.

2.1 Video Coding Schemes over Wireless Networks

The challenge is to deliver video over a wireless channel with widely varying packet delay, loss, and throughput, in a way that simultaneously maximizes the display quality at the receiver, meets bit-rate limitations, and satisfies latency constraints. To achieve this requirement, the system requires efficient compression, some form of rate scalability, and error-resiliency techniques.

2.1.1 Layered Coding

Scalable video codecs that provide layered embedded bit-streams that are decode-able at different bitrates, with degrading quality provide a means to achieve rate scalability. Layered representations for internet streaming have been widely studied and have become part of established video coding standards, such as MPEG and H.263 [1], [2]. The idea is to transmit the more important layers with higher QoS guarantees, and the less important layers with fewer or no QoS guarantees.

Layered representation of video signals consists of a base layer and multiple enhancement layers. The base layer provides a basic level of quality and can be decoded independently of the enhancement layers, thus it represents the most critical part of the scalable representation. On the other hand, the enhancement layers serve only to refine the base layer quality and are not useful alone. The first enhancement layer depends on the base layer and each enhancement layer i+1 depends on its subordinate layer i, thus can only be applied if i was already applied. Media streams using the layered approach are interrupted whenever the base layer is missing and, as a consequence, the data of the respective enhancement layers is rendered useless. The same applies for missing enhancement layers. In general, this implies that in lossy networks the quality of a media stream is not proportional to the amount of correctly received data.

2.1.2 Multiple Description Coding

A popular alternative for layered coding is multiple description coding. It fragments a single media stream into 'n' independent substreams ($n \geq 2$) referred to as descriptions. The packets of each description are routed over multiple, (partially) disjoint paths. In order to decode the media stream, any description can be used. Each description alone can guarantee a basic level of reconstruction quality of the source, and every additional description can further improve that quality. The idea of MDC is to provide error resilience to media streams. Since an arbitrary subset of descriptions can be used to decode the original stream, network congestion or packet loss will not interrupt the stream but only cause a (temporary) loss of quality. The quality of a stream can be expected to be roughly proportional to data rate sustained by the receiver. In [3], an extensive study is presented on the MD model, the information theory behind channel splitting, and its applications.

2.1.3 Performance Comparison of LC and MDC Schemes

Some literature provides performance comparisons between LC and MDC [1] [14]. It is shown that specific implementations affect the relative performance between multiple descriptions coding and layered coding according to 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, while the opposite is true for scenarios where the packet schedules are not rate-distortion optimized. In this work, we formulate our problem assuming layered video transmission. The layered coding scenario is more generic, and allows further optimization of resource allocation among different layers with degrading quality.

2.2 Video Transmission Schemes over Wireless Networks

Some literature provides [4] [5] the problem of transporting layered video over erroneous multi-hop wireless networks. Proposed system is comprised of Distributed Control (DC), Distributed Buffer (DB) scheme.

2.2.1 Distributed Control Scheme

A subset of the nodes along the streaming path from the server to the client is chosen dynamically as the set of distributed control nodes. The DC nodes provide control functions. The DC nodes are used to control network bandwidth usage and QoS of the streaming application and are set up by a self-learning scheme. Each intermediate node periodically checks its wireless link condition. When a node detects severe fading, the node becomes the DC node. The node will keep serving as the control node until the wireless link condition goes back to normal. Each DC node makes decision on whether to drop the recently received video packet or continue to transmit the packet to the destination. When the packets current delay plus DC node delay threshold exceeds a preset end-to-end delay, the DC node early-drops that packet. Otherwise, the DC node forwards it to the next hop. Without these DC nodes, the outdated video packets are still forwarded to the next hop even though they may be no longer useful for the receiver. Precious wireless bandwidth is wasted. The early drop scheme saves the wireless bandwidth so that the video playback quality at the receiver side can be improved.

Inaccurate delay threshold can either unnecessarily drop packets or push some outdated packets into the forwarding queue, resulting in a situation where the receiver either fails to receive video packets or receives outdated video packets. In either case, video playback quality is lowered at the receiver side. The delay threshold (maximum delay) of each DC node can be updated by packet timestamps. Before video streaming starts, all intermediate nodes along the path need to be synchronized. Each DC node time-stamps the packet which is located at the head of its forwarding queue. In every t second interval, from the most-recently received video packet, the receiver retrieves all the timestamps marked by intermediate DC nodes. It compares them against its current clock to calculate the delay thresholds for each DC node.

2.2.2 Distributed Buffer Scheme

The video streaming delay jitter can be reduced by pre-buffering a large number of video frames at the receiver side. However, in mobile wireless networks, there is no guarantee that a mobile device has enough buffering capacity. In this case, overflow occurs at the receiver buffer and video packets are dropped. In this scheme, we pre-select some intermediate nodes along the video streaming route which have relatively larger buffering and computing capacity as the distributed buffer nodes (DB nodes). At the beginning of video streaming, each DB node fills up its buffer to some extents (say, 50% of the full capacity). Since each DB node has pre-buffered video data, delay jitter can be reduced at each distributed buffer node. From the receiver point of view, the end-to-end jitter is reduced step-by-step by the collaborative work of all the DB nodes.

2.2.3 Disadvantages of DC and DB Scheme

- No Band width optimization
- Round Robin Techniques are applied for Flow Selection
- No Cryptography techniques are applied for Encryption and Decryption
- The user cannot understand the system due to the Simulation experimental set up

2.3 Statistical QoS Provisions for Layered Video Transmission

The theory on statistical delay QoS guarantees proved to be a valuable tool for analysing the queuing behaviour for time-varying arrival and service processes at the source and channel respectively [7]. A rich body of literature is focused on providing statistical QoS guarantees for multi-layered video streams. Due to the inherently stochastic nature of wireless channels, it is often difficult to provide deterministic QoS guarantees while taking into account most impurities of the channel such as packet loss, delay, jitter, etc. For instance, instead of enforcing a nominal delay-bound on the video packets, it is more practical to enforce a delay-bound violation probability as an approach to QoS modelling over wireless fading channels. This approach applies specifically to modern multi-layered video coding standards such as MPEG or H.264/AVC.

2.3.1 MIMO Technology

Multiple antenna systems are typically known as Multiple Input, Multiple Output systems (MIMO).

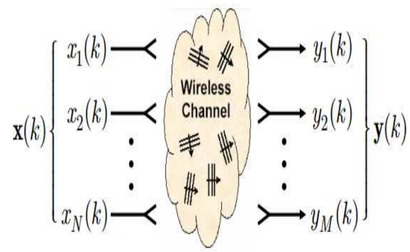

Fig 2.1: MIMO System

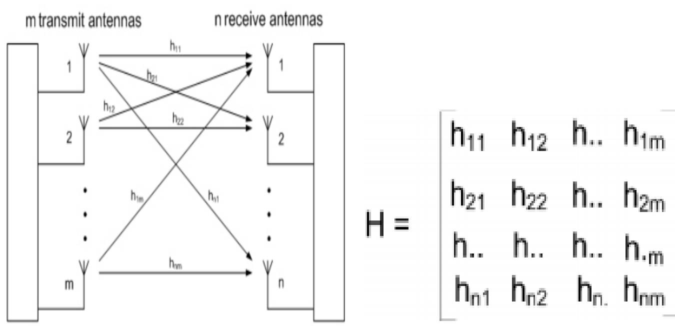

Fig 2.2: Working of MIMO

A MIMO system typically consists of m transmit and n receive antennas (Figure 2.2). By using the same channel, every antenna receives not only the direct components intended for it, but also the indirect components intended for the other antennas [8]. A time-independent, narrowband channel is assumed. The direct connection from antenna 1 to 1 is specified with h11, etc., while the indirect connection from antenna 1 to 2 is identified as cross component h21, etc. From this is obtained transmission matrix H with the dimensions n x m. Data to be transmitted is divided into independent data streams. The number of streams M is always less than or equal to the number of antennas. Disadvantages include:

- More congestion will be occurred due to MIMO Technique
- Quality of Service(QOS) is very less due to single flow selection
- No Cryptography techniques are applied for Encryption and Decryption
- This system supports only less size video files.

III. PROPOSED SCHEME

The proposed system transfers the video file from base station to the remote user and enhances the Quality of Service by introducing the number of video layers. The security has been enhanced by ECC technique. We formulate the resource allocation problem for general multi hop multicast network flows and derive the optimal solution that minimizes the total energy consumption while guaranteeing a statistical end-to-end delay bound on each network path. We also propose low complexity approximation algorithms for energy-efficient flow selection.

3.1 Advantages

- The Proposed System provides Energy Efficient Resource Allocation and Flow Selection
- End-to-End Delay Bounds on Network Paths
- Queuing Network Model for Multi hop Layered Video Transmission

- The video stream generated by the scalable video codec consists of 'N' video layers. Each layer maintains a separate queue at each node and has specific QoS requirements according to its relevance in the decoding process.
- Approximation Algorithms For Flow Selections
- Elliptic Curve Cryptography for security

IV. Conclusion

In this paper, we have presented weaknesses of some of the previous video distribution schemes. To overcome the identified problems we proposed an enhanced video distribution scheme.

V. Future Work

In future, we can apply ECC for Security. We can also maintain the events of energy and user details in the base station via mobile streaming. We can also generate the alerts about the users via hand held mobile devices.

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