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Fountain Code Based Encoding Scheme for Wireless Video Streaming

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***Abstract* — Video streaming has been regarded as one of the key applications of multimedia technologies among both network and video coding experts. The fountain codes were developed to achieve efficient transmission in erasure channels with the primary application in multimedia video streaming. The fountain code allow the receiver to recover the message symbols with high probability by receiving as few encoded symbols as possible. The video streaming data is very vulnerable to data losses since they are highly compressed to reduce the amount of data. The challenging problem is to provide a stable video streaming service of high quality over wireless network. The proposed system aims to provide a solution for the efficient transmission of video data over error prone wireless network using fountain code based encoding scheme using Aware Symbol Packetization algorithm which is a robust packetization algorithm that reduces the dependency among data packets.**

Keywords* — *Fountain codes; erasure channels; correlators; orthogonal frequency division multiplexing.

I. INTRODUCTION

Video has been an important media for communications and entertainment for many decades. Initially video was captured and transmitted in analog form. The advent of digital integrated circuits and computers led to the digitization of video, and digital video led to a revolution in the compression and communication of video. Video compression became an important area of research in the late 1980's and 1990's and enabled a variety of applications including video storage on DVD's and Video-CD's, video broadcast over digital cable, satellite and terrestrial (over-the-air) Digital Television (DTV), and video conferencing and videophone over circuit switched networks. The growth and popularity of the Internet in the mid- 1990's motivated video communication over best-effort packet networks. There exist diverse range of different video communication and streaming applications, which have different operating conditions or properties. For example, video communication application may be used for point to- point communication or for multicast or broadcast communication, and video may be pre-encoded (stored) or may be encoded in real-time (e.g. interactive videophone or video conferencing). It is, however, a difficult problem to transmit video streaming traffic over the network because video contains a large amount of data and comes with strict Quality-Of-Service (QoS) requirements. Quality of service comprises requirements on all the aspects of a connection, such as service response time, loss, signal-to-noise ratio, cross-talk, echo, interrupts, frequency response, loudness levels, and so on. As a subset of telephony, QoS is Grade Of Service (GoS) requirements, which comprises aspects of a connection relating to capacity and coverage of a network, for example guaranteed maximum blocking probability and outage probability. The probability that an outage will occur within a specified time period is called outage probability. To reduce the amount of data, it is indispensable to employ effective video compression algorithms.

It is a more challenging problem to efficiently provide a stable video streaming service of high quality over wireless network. Large-scale path loss, small-scale fading, multi-path, and interference cause random variations in the received SNR in wireless links. Path loss models describe the signal attenuation between a transmitter and a receiver as a function of the propagation distance and other parameters. In wireless communications, fading is the deviation of the attenuation affecting a signal over certain propagation media. In wireless telecommunications, multi-path is the propagation phenomenon that results in data/signal reaching the receiving antenna by two or more paths. Interference (in communication), is anything which alters, modifies, or disrupts a message as it travels along a channel. The various types of interferences are Electromagnetic Interference (EMI), Co-Channel Interference (CCI), also known as crosstalk, Adjacent-Channel Interference (ACI) (i.e. interference caused by extraneous power from a signal in an adjacent channel), andIntersymbol Interference (ISI) (i.e. distortion of a signal in which one symbol interferes with subsequent symbols).All these variations increase the Bit Error Rate (BER) because it is more difficult for the modulation scheme to decode the received signal in the case of lower SNR. To mitigate transmission errors and losses over wireless network, Automatic Repeat Request (ARQ) and Forward Error Correction (FEC) are widely used. Automatic Repeat Request (ARQ), also known as Automatic Repeat Query, is an error-control method for data transmission that uses acknowledgements (messages sent by the receiver indicating that it has correctly received a data frame or packet) and timeouts (specified periods of time allowed to elapse before an acknowledgment is to be received) to achieve reliable data transmission over an unreliable service. If the sender does not receive an acknowledgment before the timeout, it usually re-transmits the frame/packet until the sender receives an acknowledgment or exceeds a predefined number of re-transmissions. In telecommunication, information theory, and coding theory, FEC or channel coding is a technique used for controlling errors in data transmission over unreliable or noisy communication channels. ARQ increases delay because it has to retransmit lost data after receiving feedback information. Thus, ARQ may not be suitable for delay sensitive video streaming. On the other hand, FEC requires some redundant data to compensate for errors and losses without any feedback information over the network. This feature is generally appropriate for delay sensitive video streaming. So, we need to adopt some other methods and algorithms to enable secure video streaming. Fountain codes (also known as rateless erasure codes) are a class of erasure codes with the property that a potentially limitless sequence of encoding symbols can be generated from a given set of source symbols such that the original source symbols can ideally be recovered from any subset of the encoding symbols of size equal to or only slightly larger than the number of source symbols. The term ‘fountain’ or ‘rateless’ refers to the fact that these codes do not exhibit a fixed code rate. Fountain codes such as Luby transform (LT), Raptor, and Online codes are emerging erasure codes and block-based FEC schemes. Luby transform codes (LT codes) are the first class of practical fountain codes that are near-optimal erasure correcting codes. They are erasure correcting codes because they can be used to transmit digital data reliably on an erasure channel. The next generation beyond LT codes is raptor codes, which have linear time encoding and decoding. Raptor codes use two encoding stages for encoding, where the second stage is an LT encoding. The characteristics such as high coding efficiency, low encoding/decoding processing time, and flexibility are very useful for transmitting delay-sensitive data over error-prone wireless network. Here, we propose a fountain code based encoding scheme for wireless video streaming applications. The goal of the proposed algorithm is to effectively tra LT encoded symbols over error-prone wireless network and thereby, to support video streaming service over wireless network.

II. RELATED WORK

Rateless codes were initially developed to achieve efficient transmission in erasure channels. The LT encoded symbols are
generated[1] in accordance with a specific degree distribution and there is potentially an unlimited number of encoded symbols. These symbols are generated on the fly and broadcasted to the receiver until an acknowledgment (ACK) response of successful decoding is received at the transmitter.the design of rateless codes for wireless channels, such as Binary Symmetric Channels(BSC), Additive White Gaussian Noise (AWGN) channels and fading channels, has attractedsignificant attention. Rateless codes have a wide spectrum of
applications in various modern wireless communication networks such as to enhance the transmission efficiency in IEEE *ad-hoc* 802.11b wireless network and cooperative relay
networks and to control the peak-to-average power ratio in OFDM systems. Rateless codes are particularly beneficial to wireless transmissions because, in contrast to traditionalfixed-rate coding schemes, the transmitter potentiallydoes not need to know the channel state information beforesending its encoded symbols, and the receiver can retain aresilient decoding performance.

There are two categories of physical-layer ratelesscodes for wireless channels: systematic and nonsystematic. For systematic rateless codes,the systematic message symbols are first transmitted to thereceiver, followed by a number of encoded symbols. For nonsystematicrateless codes, only encoded symbols are broadcasted.The receiver uses the classic belief propagation (BP) algorithm for decoding. At the transmitter, the encoderof the existing rateless codes uses coding schemes that areakin to the low-density generator matrix (LDGM) codes of linear complexity.ShuangTian*et al.* [1] proposed a new physical layer ratelss code for wireless channels and the BER performance of the proposed code outperforms the existing rateless codes in AWGN channels, particularly at low BER regions.Donghyeok*et al.*[2] proposed a cost effective video streaming system over wireless networks which uses a fountain encoder and packetizer with a rate controller unit.The proposed system was implemented on Java and C/C++ and tested over real WiFi networks.. This system provided a high quality video streaming service with minimum networking cost. Dongju*et al.*[3] proposed a new robust LT encoding scheme with symbol packetization algorithm for wireless video streaming. The relationship among Luby transform encoded symbols was analyzed based onLuby transform encoding pattern, and the proposed packetizationalgorithm is designed to minimize packet loss effects by reducingthe dependency among packets conveying Luby transform encoded symbols. The basic concepts underlying the above mentioned works forms the integral part of our work. Our goal is to implement the proposed scheme in [3] over an error prone network using the architecture mentioned in the system proposed by Donghyeoket al. to provide a secure video streaming service.

III. PROPOSED SCHEME

We propose a robust fountain code based encoding scheme to support video streaming service over wireless network. As a first step, the video input is converted into bit stream, which is LT encoded for robust transmission through the wireless network. During the encoding process, the relationship among LT encoded symbols is analyzed based on LT encoding pattern. Based on the analysis, LT encoded symbols are packetized to reduce the correlation among packets so that the effect of a lost packet is locally limited. Then, the data packets are transmitted over the designed Binary Erasure Channel. Now, the transmitted packets are depacketized into symbols, which are decoded using LT dec
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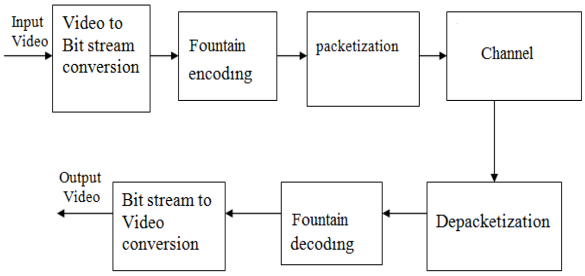

Fig1. Basic block diagram of the proposed system.

A. LT encoding and LT decoding.

When applying LT codes for transmitting data over wireless channels contaminated by fading and inter-symbol interference (ISI), the packets may become contaminated, which may result in catastrophic inter-packet error propagation during LT decoding. LT codes were originally designed for error freeBEC channels.Since the decoder can recover thedata from nearly the minimal number of encoding symbolspossible, this implies that LT codes are near optimal withrespect to any erasure channel.

B. Aware Symbol Packetization

A packet is a basic unit of communication over a digital network. A packet is also called a datagram, a segment, a block, a cell or a frame, depending on the protocol. When data has to be transmitted, it is broken down into similar structures of data, which are later reassembled to the original data once they reach their destination. The grouping of packets is needed when the length of the channel code, hence the number of packets, has to be modest for low decoding complexity. In the case of streaming videos, packetization has become a strategy of choice to alleviate the impact of packet loss. Packetisation method includes an encoded symbol and target packet selection step of deciding a first source symbol and selecting an unpacketized first encoded symbol and a target packet into which the unpacketized first encoded symbol is inserted if there is the unpacketized first encoded symbol of at least one first encoded symbol. There is a packetization step of generating a second source symbol based on at least one unpacketized first encoded symbol by the use of the AND-OR tree structure, generating at least one second encoded symbol based on the second source symbol by the use of the AND-OR tree structure, and packetizing at least one of the second encoded symbols into the target packet along with the first encoded symbol. The proposed packetization algorithm works better than the conventional packetization algorithm. But, the proposed packetization algorithm requires a slightly higher processing time than the conventional packetization algorithm.

C. Control Unit

The video streaming data are very vulnerable to data losses since they are highly compressed to reduce the amount of data. Hence, a small amount of lost data can incur serious video quality degradationThevideo streaming data are very vulnerable to data losses since they are highly compressed to reduce the amount of data. Hence, a small amount of lost data can incur serious video quality degradation.To reduce this data loss, we are trying to design a control unit part to change the fountain code rate to deliver video of better quality. Thus the proposed scheme behaves as if it is an adaptive system which improves the video quality by changing the code rate to match the best code rate slection for meeting the required video quality.

IV. RESULTS

We applied the proposed encoding and packetization algorithm on the input video and transmitted it through a binary erasure channel. At the receiver we performed the depacketization and decoding. The input video had a frame rate of 23 frames per second. The output video had almost the same specifications as that of the input. In order to prove the usefulness of the proposed encoding and packetizing algorithm, we applied the algorithm to a collection of bits and transmitted it through a binary erasure channel. At the receiver we performed the depacketization and decoding. We could easily retrieve the input bits. The performance analysis graph showing the erasure probability vs. bit loss rate is shown in FIG 2From the graph we can infer that the bit loss rate is logarithmically proportional to the erasure probability.MATLAB (Matrix Laboratory) is the programming environment chosen for running our algorithms.thebasic video format supported by MATLAB is AVI format (audio-video interleave). The .avi files support multiplestreaming audio and video. The avi format, which stands for audio video interleave, was developed by Microsoft. It stores data that can be encoded in a number of different codec's and can contain both audio and video data. The AVI format usually uses less compression than some similar formats and is a very popular format amongst internet users. During processing, video is created as an .avi file, which is the commonly used format in MATLAB. The performance analysis graph obtained is shown here which confirms the performance improvement using the proposed scheme.

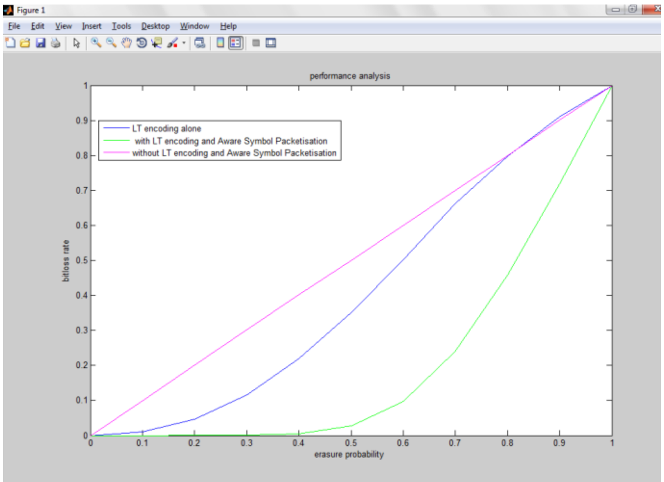

FIG2. PERFORMANCE ANALYSIS GRAPH

V. CONCLUSION

The video streaming data are very vulnerable to data losses since they are highly compressed to reduce the amount of dataTo avoid disastrous loss, the relationship among LT encoded symbols is analyzed based on LT encoding pattern. Based on the analysis, the proposed packetization algorithm is designed to reduce the correlations among packets while increasing those among encoded symbols in each packet. As a result, the LT decoding success rate can be improved at the receiver by minimizing the effect of a lost packet. In other words, we can achieve the same LT decoding success rate with a smaller amount of overhead using the proposed scheme. Thus we can conclude that our developed technique ensures, streaming features like live streaming, video streaming with outstanding quality.

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