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# A Prototype Modelling of Ebers for Video Transmission in Wireless ADHOC Network

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*Abstract-* Provisioning of video streaming over ad hoc wireless networks exhibits challenges associated with high packet loss rates and are delay sensitive. Excessive packet loss can cause significant degradation in quality of video perceived by users of real-time video applications. The recent studies suggest that Forward Error Correction (FEC) is a good technique for decreasing the negative impact of packet loss on video quality in error control scheme. This paper introduces an Estimation based Error Reduction Scheme(EBERS) to support video communication in ad hoc wireless networks. The EBERS considers a frame estimation parameter to support varied bandwidths and attain the delay requirements to support video communication. It is also responsible for improvising the QoS offered. The EBERS considers layered and embodies distortion limiting features owing to which reduced forward error correction is achieved, thus obtaining reduced frame errors, transmission errors and retransmission of frames. Thereby obtaining high degree of quality of service(QoS).The comparative study conducted proves the efficiency of the EBERS scheme over the existing mechanisms.

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# A Prototype Modelling of EBERS for Video Transmission in Wireless ADHOC Network

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Abstract- Provisioning of video streaming over ad hoc wireless networks exhibits challenges associated with high packet loss rates and are delay sensitive. Excessive packet loss can cause significant degradation in quality of video perceived by users of real-time video applications. The recent studies suggest that Forward Error Correction (FEC) is a good technique for decreasing the negative impact of packet loss on video quality in error control scheme. This paper introduces an Estimation based Error Reduction Scheme(EBERS) to support video communication in ad hoc wireless networks. The EBERS considers a frame estimation parameter to support varied bandwidths and attain the delay requirements to support video communication. It is also responsible for improvising the QoS offered. The EBERS considers layered and embodies distortion limiting features owing to which reduced forward error correction is achieved, thus obtaining reduced frame errors, transmission errors and retransmission of frames. Thereby obtaining high degree of quality of service(QoS).The comparative study conducted proves the efficiency of the EBERS scheme over the existing mechanisms.

#### I. INTRODUCTION

Writeless Ad Hoc Networks are characterized by their ability to communicate amongst one another independently sans 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.

There are large variety of exciting multimedia applications over the Internet that could broadly classified into three classes:

Streaming of stored audio and video,

Streaming of live audio and video, and

Real-time interactive audio and video.

The research work 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 band

Author α: Research Scholar, ECE,EPCEW Karnataka, Bangalore. e-mail: ganashreets@yahoo.co.in Author σ: HOD,CSE/ISE,EPCEW Karnataka, Bangalore. e-mail: d prem k@yahoo.com width. Active video communications are time critical and to attain the required Quality of Service (QOS) it is essential to minimise the packet loss [1]. Lower packet loss also results in reduced number of re-transmissions and enables packet delivery as per the required deadline. Considering the ad-hoc networks in which data to the receiver is provided through numerous intermediate nodes the end to end delay increases. To minimize end to end delays it is critical to adopt effective Forward Error Correction (FEC) to reduce the number of retransmissions of lost packets.

Streaming live video is similar to traditional broadcast of radio and television except that transmission takes place over the Internet. Video broadcast can be either unicast or multicast from the server. In a unicast connection, the transmission is replicated by the server for each endpoint user where as in multicast connection as one transmission of same signal to multiple clients over the network is happening. The broadcast from the server to the clients where there are many requests at the same time could be done using live streaming. This paper introduces the EBERS for Scalable high efficiency video coding (HEVC) to support layered video communication in ad-hoc wireless networks.

In this approach we use a 100x100 network. Initially we use 2x2 matrix as the input. The EBERS further encodes the part of Enhancement Layer and the Base Layer of the video using multiple descriptor coding techniques and custom packetization schemes discussed in the third section of this paper to reduce packet losses in the network thus achieve superior QOS.

#### II. LITERATURE REVIEW

Provisioning of QOS to support multimedia streaming has been comprehensively studied by researchers. B Sat and B wah [2] have studied the provisioning of VoIP, providing detailed study about skype and google talk. These mechanisms cannot be directly employed for wireless ad hoc networks due to the delay introduced by the multi hop transmission nature. The data packets transfer from node to node network is given in [3]. To overcome the packet losses which are inherit properties of networks and at the same time to meet the play out time researchers have proposed multiple path establishment between the

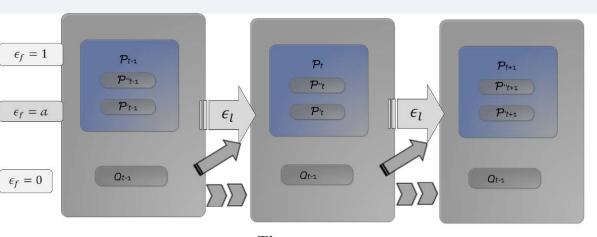
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source node and destination node. In addition to multiple path establishments the use of video's enables to minimize the packet loss, packet delay and improve QOS. The use of multiple description coding scheme [8][9][10] is adopted in the EBERS scheme proposed. Streaming of video packets over the internet is given in [4]. [5] The paper specifies a payload format for generic forward error correction of media encapsulated in Real time protocol (RTP). It uses FEC algorithms based on the parity operation. The arbitrary block lengths and parity scheme is being transmitted using payload format. This allows for the recovery of both the critical RTP header fields and payload. The paper addresses the problem by presenting an end-to-end architecture for transporting MPEG-4 video over the Internet [6].It present a framework for transporting MPEG-4 video, which includes error control, feedback control, packetization and source rate adaptation. The important contributions of this paper are: (1) a feedback control algorithm based on Real Time Protocol and Real Time Control Protocol (RTP/RTCP), (2) an adaptive source encoding algorithm for MPEG-4 video which is able to adjust the output rate of MPEG-4 video to the desired rate, and (3) an efficient and robust packetization algorithm for MPEG video bit-streams at the sync layer for Internet transport. In [7], we propose a reliable highspeed UDP-based media transport with an adaptive FEC (forward error correction) error control. The amount of redundancy by monitoring the network so that it can effectively adapt to the fluctuations of underlying networks is proposed by an adaptive transport scheme controls. The monitored feedbacks of the receiver enable the sender to be aware of current reception status (i.e., rate/type of packet loss and delay change) and to estimate the expected network status. By using this, the proposed media transport attempts to enable reliability by adaptively controlling the amount of both total sending rate and the FEC code ratio. Thus provides increase quality of the video using scalable coding [11].

#### iii. Estimation based Error Reduction Scheme for Scalable HeVC (EBERS)

The EBERS scheme proposed considers a layered encoded video communication streams for transmissions in the network. The video data is encoded using the SHVC into two streams. The important stream namely the Base Layer(Q) and the unimportant stream or the Enhancement Layer( $\mathcal{P}$ ) is generally considered to reconstruct high quality video streams.

Considering both the  $\mathcal{P}$  and  $\mathcal{Q}$  streams can often lead to higher transmission errors. To minimize the error propagation the EBERS adopts a packetization scheme as shown in the Figure 1 where  $\mathcal{P}$  is split into two sub packets  $\mathcal{P}'$  and  $\mathcal{P}''$ .



Time

*Figure 1 :* Packetization Scheme Adopted in EBERS

The enhancement layer packet  $\mathcal{P}'$  is derived by a forward estimation factor  $\epsilon_{l}$  where  $\epsilon_{l} \in [0,1]$ . The video message  $\mathcal{P}$  bits coded  $\mathcal{M}'_{t}^{\mathcal{P},cd}$  at a given time instance t is derived from the preceding  $\mathcal{Q} \mathcal{M}_{t-1}^{Q}$ , preceding  $\mathcal{P}$ ,  $\mathcal{M}_{t-1}^{\mathcal{P}}$ . In the below equation cd stands for coding and is defined as

$$\mathcal{M}_{t}^{'\mathcal{P},\mathrm{cd}} = \left( (1 - \epsilon_{\mathrm{l}}) \mathcal{M}_{t-1}^{\mathcal{Q}} \right) + (\epsilon_{\mathrm{l}} \times \mathcal{M}_{t-1}^{\mathcal{P}}) \tag{1}$$

From the above definition it is clear that if  $\varepsilon_l$  tends towards 0 the  $\mathcal P$  would be completely eliminated and would result in decreased quality of video transmissions in the network. If the  $\varepsilon_l \approx 1$  then the errors induced in the communication would increase and the video quality would significantly improve. The packetization scheme construct  $\mathcal P'$  and  $\mathcal P''$  from  $\mathcal P$  based on the frame estimation factor  $\varepsilon_f$ . The frame estimation factor is an adaptive factor to accommodate

varied modes supported in the scalable high efficiency video coding (SHVC). Incorporating the fram estimation factor the video message  $\mathcal{M}'_{t}^{\mathcal{P}, \mathrm{cd}}$  could be redefined as

$$\mathcal{M}_{t}^{'\mathcal{P}, \mathrm{cd}, \epsilon_{\mathrm{f}}} = \left( (1 - \epsilon_{\mathrm{l}}) \mathcal{M}_{t-1}^{\mathcal{Q}} \right) + \left( \epsilon_{\mathrm{l}} \times \mathcal{M}_{t-1}^{\mathcal{P}, \epsilon_{\mathrm{f}}} \right)$$
(2)

In the ad hoc network considered let  $r_x[a]$  denote the received signal,  $t_x[a]$  denote the transmitted signal and  $n_{awgn}[a]$  denote the additive white Gaussian channel considered in the network. The channel coefficients matrix whose elements are independent and identically distributed variables having a variance defined  $as\sigma_{\hbar}^2$ . All the variables of the channel coefficient matrix are assumed to be zero mean complex Gaussian random variables. If the channel coefficient matrix is represented as  $\mathcal{H}[a]$  then the received signal for the  $a^{th}$  thsymbol is defined as

$$r_x[a] = \mathcal{H}[a]t_x[a] + n_{\text{awgn}}[a] \tag{3}$$

Let  $\varepsilon_{xy}[a]$  denote the channel estimation error for the  $a^{\text{th}}$  symbol transmitted from the  $y^{\text{th}}$  transmitting node in the considered network to the  $x^{\text{th}}$  receiving node in the same network, and then  $\varepsilon_{xy}[a]$  can be considered as a complex Gaussian variable and the variance is defined as

$$\sigma_{\varepsilon}^{2}[a] = 1 - \mathcal{W}^{*}[a]\mathcal{C}\mathcal{M}^{-1}\mathcal{W}[a]$$
<sup>(4)</sup>

Where  $\mathcal{CM}$  represents the auto correlation matrix. The covariance vector between  $\hbar_{xy}[a] \in \mathcal{H}$  and the received samples is represented as  $\mathcal{W}[a]$ . In the EBERS considering that all the channels of the ad hoc nodes are autonomous, the radio layer channel estimation errors can be considered as a matrix whose elements are independent and identically distributed Gaussian variables exhibiting an variance as defined in the above equation.

The  $\mathcal{CM}$  autocorrelation matrix and the received samples  $\mathcal{W}[a]$  are dependent on the modulated carrier wave, the signal to noise ratio(SNR), spreading factors and the frame transmission rate. The EBERS assumes that the SNR of the carrier wave and the frame rate are equivalent to the channel coherence and the data SNR.QPSK modulation scheme is considered in the radio layer of the ad hoc network. The  $\mathcal{P}$ kt packets each having a symbols are allocated to a frame.

Let  $cr_x$  denote the channel rate and  $\rho tx_x$ denote the packet transmission parameter for the  $x^{\text{th}}$  packet. If the error rate experienced by the  $x^{\text{th}}$  packet is represented as  $e_x(cr_x, \rho tx_x)$  and that if zpackets are successfully transmitted and a transmission error occurs at z + 1 packet. The probability Prob( $z \mid \mathcal{P}kt$ ) of such an occurrence in the ad hoc network is defined as

$$Prob(z \mid \mathcal{P}kt) = \begin{cases} e_{1}(cr_{1}, \rho tx_{1}), & z = 0\\ \prod_{\psi=1}^{z} \left(1 - e_{\psi}(cr_{\psi}, \rho tx_{\psi})\right) \times \left(e_{z+1}(cr_{z+1}, \rho tx_{z+1})\right), & 0 < z < Pkt \\ \prod_{\psi=1}^{\mathcal{P}kt} \left(1 - e_{\psi}(cr_{\psi}, \rho tx_{\psi})\right), & z = \mathcal{P}kt \end{cases}$$
(5)

Considering that  $\mathcal{M}_{\psi}$  is the total number of symbols of the  $\psi^{th}$  packet, the data or the bits received at the receiving node of the ad hoc network when if zzpackets are successfully transmitted from the source node is defined as

$$\sum_{\mu=1}^{z} \left( \mathcal{C}\mathcal{F}_{\mu} \times \mathcal{M}_{\mu} \right) \tag{6}$$

Considering the forward estimation factor  $\varepsilon_{\rm l}$  land the frame estimation factor  $\varepsilon_{\rm f},$  the average distortion is defined as

$$\mathcal{A}vg[\mathfrak{D}(\epsilon_{l}, \epsilon_{f})] = \mathfrak{D}(0, \epsilon_{l}, \epsilon_{f})\operatorname{Prob}(0 \mid \mathcal{P}kt) + \sum_{z=1}^{\mathcal{P}kt} \left( \mathfrak{D}\left(\sum_{y=1}^{z} \mathcal{C}r_{y}, \mathcal{M}_{y}, \epsilon_{l}, \epsilon_{f}\right)\operatorname{Prob}(z \mid \mathcal{P}kt) \right)$$
(7)

where the distortion of the bits of the  $\mathcal{Q}$  layer and the  $\mathcal{P}^{tx}$  bits of the  $\mathcal{P}$  layer that are received effectively is represented as  $\mathfrak{D}(\mathcal{P}^{tx}, \epsilon_{l}, \epsilon_{f})$ 

From the above equation it is evident the average distortion observed in the transmission depends on the estimation factor  $\varepsilon_l$  and the frame estimation factor  $\varepsilon_f.$ 

The proposed EBERS considers the bit error rate observed by the frame to compute the total forward error observed in the prorogation of video data in the ad hoc network. The subsequent section of the paper discusses the experimental evaluation of the EBERS and its efficiency over the existing schemes to support real time video communication over ah-hoc wireless networks.

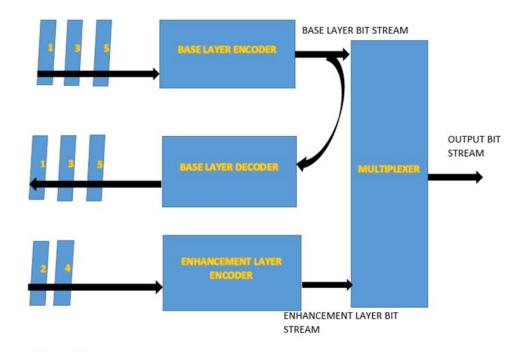


Fig 2: EBERS Architectural diagram

### IV. EXPERIMENTAL STUDY

We have used a 100x100 network with 40 nodes. We are applying 25 frame transmissions and each frame is further divided into 80 bits. All these bits are sent through the baselayerand enhancement layer, depending on the PSNR ratio. We thus activate or deactivate or increase or decrease the estimation factor depending on the PSNR ratio, thus keeping in mind therequired scalability of the required video .We there by obtain low packet loss and high quality of video. The various simulation results as shown below:

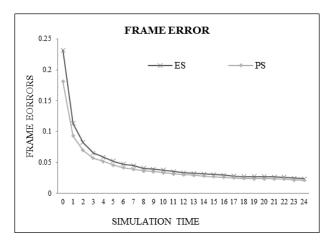


Figure 2 : Frame error versus simulation time

Fig.2 represents a graph indicating the frame error of both proposed system and the existing system

versus the simulation time. In both the systems, as the simulation time increases, the frame error decreases. In the existing system initially the frame error was 0.24 at time t=1 unit of time and gradually reduced to 0.03 at 24 units of time. In the proposed system initially, the frame error was 0.18 which reduced to almost 0.02.

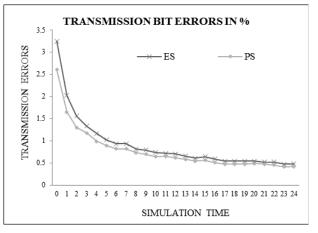




Fig.3 represents a graph indicating the Transmission error of both proposed system and the existing system versus to the simulation time. In both the systems, as the simulation time increases, the transmission error decreases. In the existing system initially the transmission error was 3.3 at t=1 unit of time and gradually reduced to 0.6 at 24 units of time. In the proposed system initially the frame error was 2.6 which reduced to almost 0.05.

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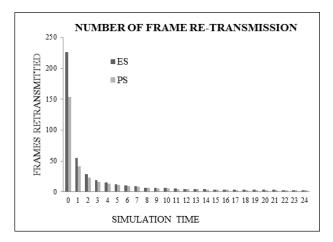


Figure 4 : Frames retransmitted versus simulation time

Fig.4 represents a graph indicating the retransmitted frame versus the simulation time of both proposed system and the existing system. In both the systems, as the simulation time increases, the number of frames to be retransmitted decreases. In the existing system initially the number of frames retransmitted 230 at t=1 unit of time and gradually reduced to 1 at 24 units of time. In the proposed system initially the frame error was 150 which at last reduced to almost 0.

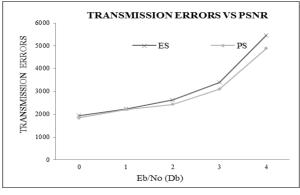




Fig.5 represents a graph indicating the transmission error versus the PSNR (peak-signal to noise ratio) of both proposed system and the existing system. In both the systems, as the PSNR increases, the transmission error increases. In the existing system initially the transmission error was 1900 atPSNR=0 dB and this gradually increased to 5500 at PSNR=4 dB .In the proposed system initially the transmission error was 1900 which at last increased to almost 5000 at 4 dB

# V. Conclusion

In this research paper, we proposed estimation based error reduction scheme (EBERS) with an enhanced and base layer to efficiently support multimedia data transmission over wireless LANs. It

introduces an Estimation based Error Reduction Scheme (EBERS) for Scalable HEVC scheme that not only reduces the transmission bit errors but also reduces the number of retransmission overheads providing the QoS required to support real time video transmissions in wireless ad-hoc networks. The proposed EBERS scheme achieves adaption by incorporating the frame estimation and forward estimation parameters. The EBERS also introduces a novel packetization scheme to reduce the number of retransmissions and yet achieve acceptable video guality in the presence of noisy communication environments. The EBERS discussed in this paper provides support for the transmission of the SHVC standardized by 3GPP. The experimental study discussed in this paper proves that the EBERS is able to achieve an FEC efficiency of about 28% over the existing FEC scheme. The future of the work presented here is considered to evaluate the FEC efficiency in terms of frame error rates and also study the adaptive nature of the EBERS to support varying bit rate transmissions of the SHVC yet achieving video quality over wireless adhoc networks.

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