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An error prediction Model with cross layer architecture to enhance QOS for multimedia communication over wireless network

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ABSTRACT

Wireless multimedia applications demand is increasing rapidly which have boosted the development of video transmission technologies with the design of cross layer architecture, to optimize the Quality of Service (QoS) of the end user. In this paper, cross layer architecture is proposed by using the adaptive selection of the layer and error prediction strategy in the wireless transmission. Detailed algorithms with corresponding theoretical reasoning are provided. Since forward error corrections (FEC) techniques are widely implemented in modern wireless communication systems, the proposed scheme is further extended to the systems with the error prediction scheme. Simulation is carried out using MATLAB tool and performance of proposed model is measured in terms of PSNR, End-to-End delay and packet retransmission rate. Proposed model shows the better performance results when compared to existing system.

INTRODUCTION

Multimedia is term which use in sense of interactive communication process of media which consisted of video, text, animation, graphic and sound. The objective of multimedia to designs the interactive media material for consumer devices and delivers this material to consumers with good QoS. In telecommunication systems generally quality of service (QoS) considered the network architecture and resources such as limit of bandwidth, jitter and packet loss. Now a day's, distributed business applications on internet are increasing and quality of services for these applications depended on many factors such as coding and compression techniques, data recovery algorithm [1] for delivering the acceptable quality of services to end users. The current rate of grading scale of multimedia showing that network quality is not only way for assessing the user perceived quality of multimedia applications.

Watching videos online has become a popular form of entertainment in recent years. Online VoD services enable users to watch video content such as user generated videos, movies, TV shows, music videos, and live streams. In the past few years, it has been observed that video streaming has accounted for most of the traffic carried over the Internet and it continues to steadily increase every year. The global video streaming traffic is expected to account for 69% of the consumer traffic on the Internet by 2017, up from 57% in 2012 [1]. YouTube, Netflix, and Hulu are the most popular online video streaming service providers. According to [2], in 2013 Netflix accounted for 32.25% of the total downstream traffic during peak periods in North America, followed by YouTube with 17.1%, and Hulu at 2.41%.

The increasing dominance of video traffic over other Internet traffic could soon make it a prime online service for the average user. In order to meet the satisfaction of the user with these services, it is necessary to be able to measure the performance of these services. Traditionally, the Quality of Service (QoS) metrics have been used to study the performance of online services and networked elements. QoS metrics, which are more suitable to measure the performance and reliability of the network elements, however, do not capture the actual experience of the user. Quality of Service (QoS) as defined by ITU [3] reflects the performance of the network and its components. It measures the network's ability to satisfy the needs of the service and is thus, a network-centric metric. The common QoS metrics used are throughput, bandwidth, packet loss, delay, or jitter.

The service received by the user, however, is affected and influenced not just by the network components but also by several other factors. These are end-to end factors that include the effects of the service infrastructure, terminal, client, and network. On the other hand, Quality of Experience (QoE) [4], [5] is a user-centric metric that captures the overall acceptability of the service and includes the end to- end factors. It has also been defined as the degree of delight or annoyance of the user of an application or service [6]. QoE measures the performance as subjectively perceived by the user. However, QoE and QoS are not mutually exclusive; rather, QoE is an extension to QoS. The user-perceived quality of video streaming services is influenced by four different categories of influence factors [7].

• System Level factors encompass the effect of the factors that work at the technical level. These factors include the factors related to the network (packet-loss, delay), end-devices (system hardware, screen size), and also the application layer (video buffering strategies, browser).

• Context Level considers the environmental factors associated with the user such as the user's location, purpose of using the service such as entertainment, education, etc.

• User Level considers the psychological factors such as the expectations of the user, browsing history, and hour of the day.

• Content Level considers the defining characteristics of the video file such as the encoding rate, format, resolution, playback duration, quality of the video, age of the video etc.

Due to the varied influence factors that affect the QoE, it presents a unique challenge to measure it. In this survey, we present the different tools and methodologies that have been developed to measure the QoE of online video-on-demand (VoD) streaming services for stored videos. Most video services are not provided with any special provisioning for end users and hence, need to compete with other services to be able to maintain a satisfactory service. In contrast, broadcast television and cable television have dedicated infrastructures and do not typically share network resources with other services. The Internet Protocol Television (IPTV) provides television services over a managed IP network and ensures a superior entertainment experience [8], [9]. IPTV services utilize networks that guarantee a Quality of Service, which differentiates them from other online streaming services such

In short it can be stated that, delivery of delay sensitive data on wireless networks put forth variations in the physical layer, medium access control layer (MAC) and the routing layer. Apart from these variations observed it is also critical to establish a balance between the data delivery and the QoS provided. Providing QoS at the cost of delay in delivery of data has no meaning in real time data transmission. Work is still going on this issues research purposed a cross layer architecture physical or MAC layer for video transmission.

Relation exist between QoS and end to end data delivery is directly proportional in video transmission i.e. QoS is high if delay is high it proved in [2]. To address these issues in this paper author give a new model for wireless video transmission reducing distortion, increases the quality of video and reduce the delay name EQIVST model adopting a cross layer optimization approach. SVC encoding video transmission scheme considered in this EQIVST model. Based on condition of physical layer two parameter defined Quality prediction specifier (γ) and a parameter which contain the knowledge about physical layer name physical layer specifier (δ)[4]. (γ) and (δ) are utilized in SVC video encoding scheme at the MAC layer.

Packet constructed at the MAC layer routed to next hop based on the value of (γ) , (δ) and pending packet on that node. This concept is applied for every intermediate hop node. The EQIVST model proposed is designed to address the tradeoff between QoS provisioning and delivery of the delay bound multimedia data. The cross layer optimization adopted in the EQIVST model provides adaptability to achieve better QoS in wireless networks and ensures the essential delay bound multimedia data delivery and reduce distortion by the use of (δ) in adaptive manner

RELATED WORK

Traditional layered design cannot provide QoS for mobile multimedia because of its limited adaptation to the dynamic wireless channels and interaction between layers. The goal of a cross-layer design is to improve the overall performance of the mobile multimedia applications, including the quality of video and power consumption. Crosslayer design jointly adjusts the parameters of different network layers, but the computation is complicated. Cross-layer optimization is very complex since it requires the optimization of multiple parameters across the network layers. One of the challenges of cross-layer design is the difficulty to model the complex cross-layer interactions among the parameters at different network layers. In general, there is a trade-off between the performance and complexity in the cross-layer optimization. A low-complexity cross-layer design is desired.

A cross-layer design enhances the performance of the application by jointly considering the mechanisms at multiple network layers. For example, modulation and coding scheme at the physical layer, scheduling and admission control at the MAC layer, routing at the network layer, congestion control and rate control at the transport layer, and source coding, traffic shaping, scheduling, and rate control at the application layer. Cross-layer QoS mechanisms proposed for 802.11 WLAN can be divided into different categories according to the layers involved.

• Application-PHY layer

Joint source-channel coding at the application layer has been extensively studied [10-11]. Argyriou provided a methodology for joint setting of the parameters of source and channel coding based on an analytical model of the overall system. It employs joint source and application-layer channel coding and rate adaptation at the wireless physical layer.

• Application-Transport layer- MAC/PHY layer

Zhu, Zeng and Li [12] proposed a joint design of source rate control and congestion control for video steaming over the Internet. A virtual network buffer management mechanism was introduced and the QoS of the application was translated into the constraints of the source rate and the sending rate. At the transport layer, a QoS-aware congestion control mechanism was proposed to meet the sending rate requirement derived from the virtual buffer. The joint optimization of parameters in [13] was designed to minimize the expected end-to-end video distortion constrained by a given video playback delay. It includes video coding at the application layer, packet sending rates at the transport layer, and the modulation and coding scheme at the physical layer. A cross-layer design is proposed in [14] that incorporates source rate control at the application layer, congestion control at the transport layer, and wireless loss ratio from the MAC layer.

• Application-MAC layer

Van Der Schaar and Turaga [15] developed a joint application-layer adaptive packetization and prioritized scheduling and MAC-layer retransmission strategy, where the application and the MAC layers jointly decide the optimal packet size and retransmission limits. Cross-layer design in [16] utilized the data partitioning technique at the application layer and QoS mapping technique at the EDCA-based MAC layer of the 802.11e network. Chilamkurti et al. [17] proposed a cross-layer design for 802.11e which maps video packets at the application layer to the appropriate access categories of 802.11e EDCA at the MAC layer according to the significance of the video data. The approach proposed for IEEE 802.11e HCCA WLAN in [48] consists of admission control and resource allocation at the MAC layer and video adaptation at the application layer.

• Application-MAC-PHY layer

Van Der Schaar, Andreopoulos and Hu [10] proposed an optimization over Application-MAC-PHY layer for scalable video over IEEE 802.11 HCCA. It maximizes the number of admitted stations by creating multiple sub-flows from one global video flow. Shankar and Van Der Schaar [49] proposed an integrated system view of admission control and scheduling for both content and poll-based access of IEEE 802.11e MAC protocol. The scheme in [1] set parameters at three layers: application, link, and physical layers. It is designed to optimize the video quality of all streams given different power levels and channel conditions of the wireless stations. Wu, Song and Wang [12] proposed a cross-layer optimization framework for delivering video summaries over wireless networks. It jointly optimizes the source coding at the application layer, allowable retransmission at the data link layer, and adaptive modulation and coding at the physical layer within a rate-distortion theoretical framework.

PROPOSED MODEL

Source Model

In this section we present the proposed model for video coding to improvise the QoS (Quality of service) of the system. As discussed earlier, in conventional methods of video coding, to construct the high quality reference base layer and enhancement layer is used which causes error in the transmission. To overcome the issue of the transmission error, prediction based method is used in the proposed system which is shown in fig 1.

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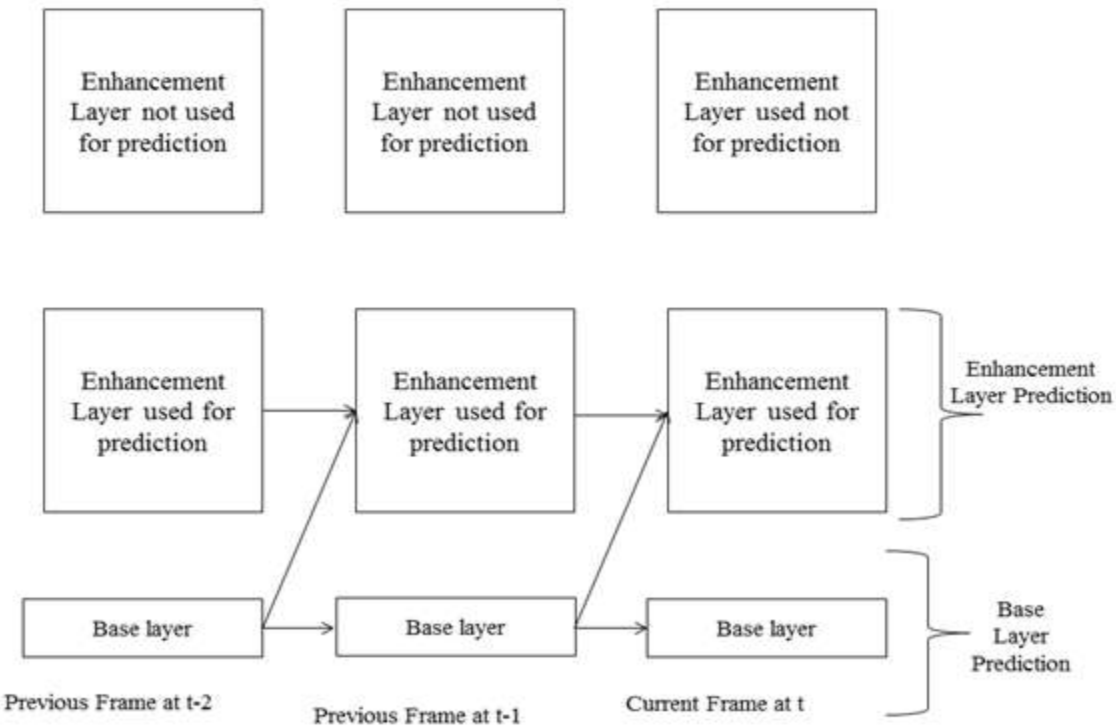

Fig.1. Video coding architecture

Prediction parameter $\alpha \in [0, 1]$ is used for the scaling of the enhancement layer. At the encoder stage, at a time , reference for the enhancement layer is computed as the weighted sum of the previous base layer L_{t-1}^{BL} and enhancement layer L_{t-1}^{EL} or it can be represented as eq. (1)

$$L_t^{EL,encd} = L_{t-1}^{BL} + \alpha L_{t-1}^{BL} + \alpha L_{t-1}^{EL}$$

In the above given equation, video coding technique depends on the prediction parameter value. Proposed model ehibits the method of reducing the progation error by fixing the value of prediction parameter.

If partial prediction is used for the video coding, in that scenario amount of the enhancement layer is used to reconstruct the reference frame in the motion compensation. Selection of the partial prediction is done based on the bandwidth of the channel. In this work, we use arbitrary number of symbols in the enhancement layer for motion compensation error.
Partial prediction in this model is defined as β which is the total number of symbols in the enhancement layer and the maximum number of symbols in the enhancement layer symbols for the particular frame so equation 1 can be represented as

$$L_t^{EL,encd,\beta} = L_{t-1}^{BL} + \alpha L_{t-1}^{BL} + \alpha L_{t-1}^{EL,\beta}$$

$L_t^{EL,encd,\beta}$ presents the reference for the current enhancement layer for the prediction and $L_{t-1}^{EL,\beta}$ represents the partial reconstructed frame using proposed model..

ERROR PROTECTION SCHEME WITH FEC

In this section we discuss about the error and distortion minimizing scheme for the video coding to improve the QoS by using FEC (Forward Error Correction) scheme and suitable code rate for the model.

Here we have considered that the size of packet is N for the transmission in the enhancement layer and packet length is considered as equal to the duration of the symbols m For ith packet, channel code rate is denoted by r_i and operation mode is denoted by ϕ_i . for this ith packet, packet, packet error rate is given by $p_i(r_i, \phi_i)$.
Receiving probability of first k packets and probability of first packet error is given by

$$\pi_k \triangleq P_r\text{ (first packet error occurs at packet } k+1)$$
$$= \begin{cases} p_1(r_1, \Phi_1, \quad k=0 \\ \prod_{j=1}^{k} (1 - p_j(r_j, \phi_j)\, p_{k+1}(r_{k+1}, \phi_{k+1}) \quad 0 < k < N \\ \prod_{j=1}^{N} \left(1 - p_j(r_j, \phi_j)\right), k = N \end{cases}$$

Successfully received packets are given by k, total number of bits $\sum_{j=1}^{k} r_j s_j$, s_j denotes the total number of the symbols in the packets.

Average Distortion of the system can be computed as

$$AD_{\alpha,\beta} = \wp(0, \alpha, \beta)\pi_0 + \sum_{k=1}^{N} \wp\,(\sum_{j=1}^{k} r_j s_j, \alpha, \beta)\, \pi_k$$

α, β denotes the partial prediction parameters, distortion is given by $\wp(0, \alpha, \beta)$ when the entire bits of the base layer and enhancement layer is received successfully.

At this stage, we consider the selection of code rate $r = [r_1, r_2, \dots r_N]$ and selection of the mode $\phi = [\phi_1, \phi_2, \dots \phi_N]$, to minimize the distortion of the data. Rate allocation and distortion minimization can be achieved by using the proposed distortion minimization algorithm in the enhancement layer when codeword is fixed for the transmission.
In order to perform this, an effective code rate is defined C_r which is the ratio of information symbols to the maximum transmitted symbol in the packet.

PROPOSED ALGORITHM FOR DISTORTION MINIMIZATION

In this section we consider a channel coder which is given as $\mathbb{C} = [\mathbb{C}_1, \mathbb{C}_2, .. \mathbb{C}_m$ and joint source-channel coding system. These channels have the capability to detect and correct the error. Size of all the channels is considered same for all $\mathcal{L}$ and corresponding code rate is denoted by $\Re = [r_1, r_2, \dots . r_m]$.

According to this method, frames are transformed into successive blocks i.e. $\{\mathbb{b}_1, \mathbb{b}_2, \dots, \mathbb{b}_n\}$ and then later it is converted into sequence of codewords $\{\, \mathcal{C}(\mathbb{b}_1), \mathcal{C}(\mathbb{b}_2) \dots, \mathcal{C}(\mathbb{b}_m)$. Number of bits protected in a frame is given by $v(r_j) = \lfloor I_{rj} \rfloor$.
During the transmission, the packets $\{\, \mathcal{C}(\mathbb{b}_1), \mathcal{C}(\mathbb{b}_2) \dots, \mathcal{C}(\mathbb{b}_m)$ are transmitted over a noisy channel, as the decoding error is detected, decoding of the video data is stopped and video frames are reconstructed from the fully decoded data.
Probability of no error in decoding is given by

$$P_i(R) = p(r_{k+1}) \prod_{j=1}^{i} \left(1 - p\left(r_{k_j}\right)\right)$$

The above given equation shows the probability of no error in decoding in the first packet. Probability of error occuring is given by $P_0(R) = p\left(r_{k_j}\right)$, this is the error probaility in the first packet. Similarly no error in decoding probability for N packets is give as $P_N(R) \prod_{j=1}^{N} \left(1 - p\left(r_{k_j}\right)\right)$.

Selection of the prediction parameters

In this section we discuss about the selection of the prediction parameters for the decoding which plays an important role for reconstruction, as we have discussed earlier. Selection of these parameters is done based on the available channel information.

Error propagation with prediction parameter

In this model for video transmission base layer is defined as

$$\mathbb{R}_{BL}(n,i) = L(n,i) - L_{BL}(n-1,j)$$

Motion compensation of the reference frame is given by $L_{BL}(n-1,j)$, $\mathbb{R}_{BL}(n,i)$ and motion vector of the video are compressed in the base layer for the reconstruction of the video frame. The reconstruction of the video frame is given by

$$L_{BL}(n-1,j) = L_{BL}(n-1,j) + \mathbb{R}_{BL}(n,i)$$

this reconstructed frame is stored in the buffer of the network, which is later used for the encoding of the next frame. Prediction is performed in the base layer from the data which is stored into the buffer based on this the enhancement layer data can be written as

$$\begin{aligned} L_{EL}(n,i) &= L(n,i) - L_{BL}(n,i) \\ &= L(n,i) - L_{BL}(n-1,i) - \mathbb{R}_{BL}(n,i) \end{aligned}$$

Data present in the enhancement layer is compressed as

$$e_{EL}(n,i) = L_{BL}(n,i)$$

During the transmission high quality reference frame is introduced based on the enhancement layer factor α^{el} which is given as

$$g_{EL}(n,i) = L_{BL}(n,i) + \alpha^{el} L_{EL}^{p}(n,i)$$

Enhancement layer factor range is defined 0 to 1 which is $\alpha^{el} \in [0,1]$. In order to improve the robustness and error reduction capacity of the system, partial data of the reference frame is used as the reference frame in the enhancement layer.

Redundancy in the enhancement layer can be removed by substracting the base layer data $L(n,i)$ and enhancement layer $g_{EL}(n,i)$ data which is achieved based on the reference frame. It is represented as

$$e_{EL}(n,i) = L(n,i) - L_{BL}(n-1,j) + \alpha_{n-1}{}^{el} L_{EL}^{p}(n-1,i)$$

During this process similar motion vectors in the base layer are applied to the enhancement layer to avoid the delay caused by the motion estimation process in the base layer and enhancement layer.
Simplification of error presented in the enhancement layer can be written as

$$e_{EL}(n,i) = L(n,i) - \alpha_{n-1}{}^{el} L_{EL}^{p}(n-1,i)$$

Similarly the enhancement layer encoder can be written as

$$e_{EL}(n,i) = \alpha_{n-1}{}^{el} L_{EL}^{p}(n-1,i) + L(n,i)$$

It is stored in the buffer to encode the next frame in the enhancement layer. The error between base original frame and reconstructed frame at the encoder end and decoder end respectively can be written as

$$L(n,i) - L(n-1,i) = e_{n}^{EL} - e_{EL}(n,i) + L_{EL}^{enc} - L_{EL}^{dec}$$

From the above discussion the distortion can be written as

$$\begin{aligned} AD_{(v,\alpha,\beta)} &= E\big(L(n,i) - L_{BL}(n-1,j)\big)^2 \\ &= E\left(L(n,i) - L_{BL}(n-1,j) + \alpha_{n-1}{}^{el} L_{EL}^{p}(n-1,i)\right)^2 \end{aligned}$$

v is the vector of throughtputs for all the frames

According to the proposed model for QoS improvement, below given figure describes the working of the proposed model. In this model initial step is to check the packet status which is the video frames. Frames are converted into bit format for the transmission. Each packet is assigned with fixed time slot, if it crosses the time, then packet is considered as expired packet and if retains the time while transmission, the packet is considered alive packet. In the first case when time is expired, packet is also considered as expired and this packet is counted as dropped packet and if packet is alive then it is transmitted through a channel after encoding process. Next stage is to compute the delay and error performance after performing the decoding at the transmitter end. Based on this performance QoS (Quality of service) is computed for the system, it is achieved according to the required QoS then the transmission of data continues else the data is recollected and updated in the table to improve the QoS by removing the errors and reducing the delay by selecting the predictive layers as discussed above. In another case, if received QoS is better than the required QoS then the comparison table of QoS is updated based on the received QoS.

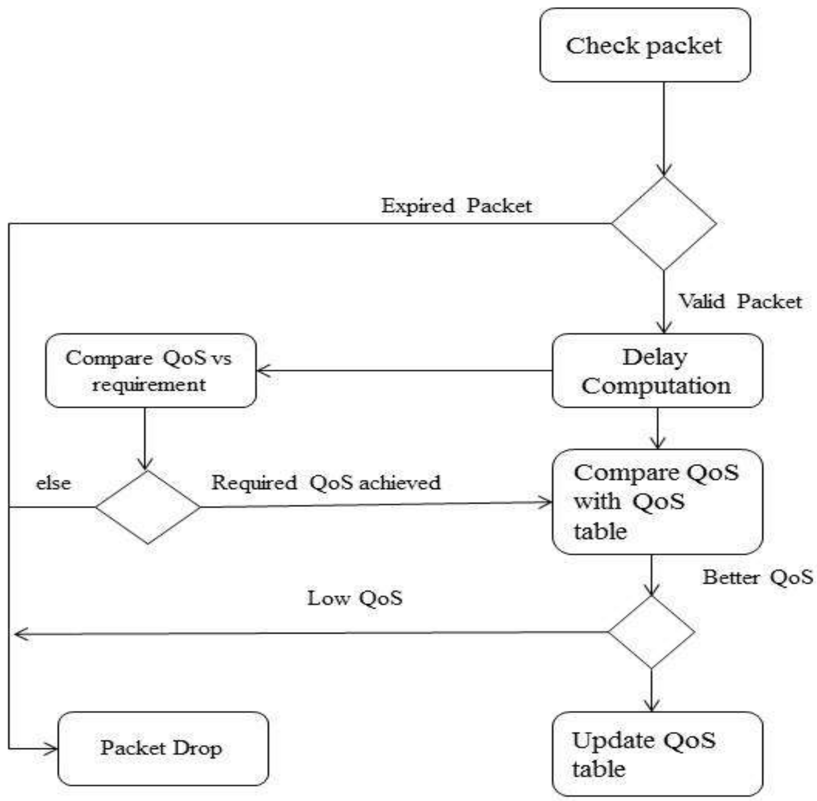

Fig. 2

The selection procedure of the adaptive layer in the proposed model is done by using predictive coefficient. In the scenario of video transmission over a channel by using base layer and enhancement layer, if received packets are lesser than enhancement layer then mismatch in the encoded data and decoded data is caused. To avoid this mismatch in the data we use transmitter to estimate the throughput for the encoded frame which is used for the selection of β. We assume the channel is varying slowly and it correlated with the multiple frames.

EXPERIMENTAL STUDY

In this section we test and discuss the performance of the proposed model and comparative study with existing algorithm. The video sequence "Foreman", "Bus" and "Stefan" with length of 50 s (CIF resolution, 50 Hz frame-rate) is compressed by an H.264/AVC codec with target bit rate of 1.5 Mbit/s. At PHY, we consider the 802.11a standard, with the symbol rate varies from 333 to 500 KB/s. Convolutional codes are employed to perform forward error correction (FEC). Table I summarizes the parameters used at each layer. The simulation platform was developed using Matlab and C++. An area i.e. $\mathcal{A} = 100 \times 100$ m was considered to deploy 100 wireless nodes.

Table I. Simulation parameters

Packet length	1000 bytes
Frame Interval	20 ms
GOP length	15
Tx OP duration	500 μs
Time slot	2 ms
FEC rate	½, 2/3,3/4

Table II. Average Utility Performance

SNR (dB)	Average Utility(ES)	Average Utility(PS)
0	0.777	0.817
10	0.811	0.892
20	0.892	0.930
30	0.945	0.972

The tests sequences that are used for simulations in this paper are summarized in TABLE I and TABLE II. All sequences are in YUV 4:2:0 color formats, in which the two chroma components are downsampled by a factor of two in each spatial direction. The tables specify the maximum spatial and temporal resolution of the sequences. Sequences with a lower temporal resolution are obtained by frame skipping, and sequences with a lower spatial resolution are obtained by downsampling.

To evaluate the performance of the proposed algorithm the authors have considered multipath video transmission scenarios over wireless nodes in a simulation platform. The simulation platform was developed using Matlab and C++. An area i.e. $\mathcal{A} = 100 \times 100$ m was considered to deploy 100 wireless nodes. The bandwidth assigned to each wireless node is 1 Mbps . The wireless channels are modelled using the Additive white Gaussian noise based noise model. The physical layer developed for the simulation environment is in accordance to the IEEE 802.11g standard. Orthogonal Frequency Division Multiplexing is considered for transmitting and receiving of the video data. The video data to be transmitted is initially encoded using a codec [17]. The encoded video data is transmitted via a multipath channel.

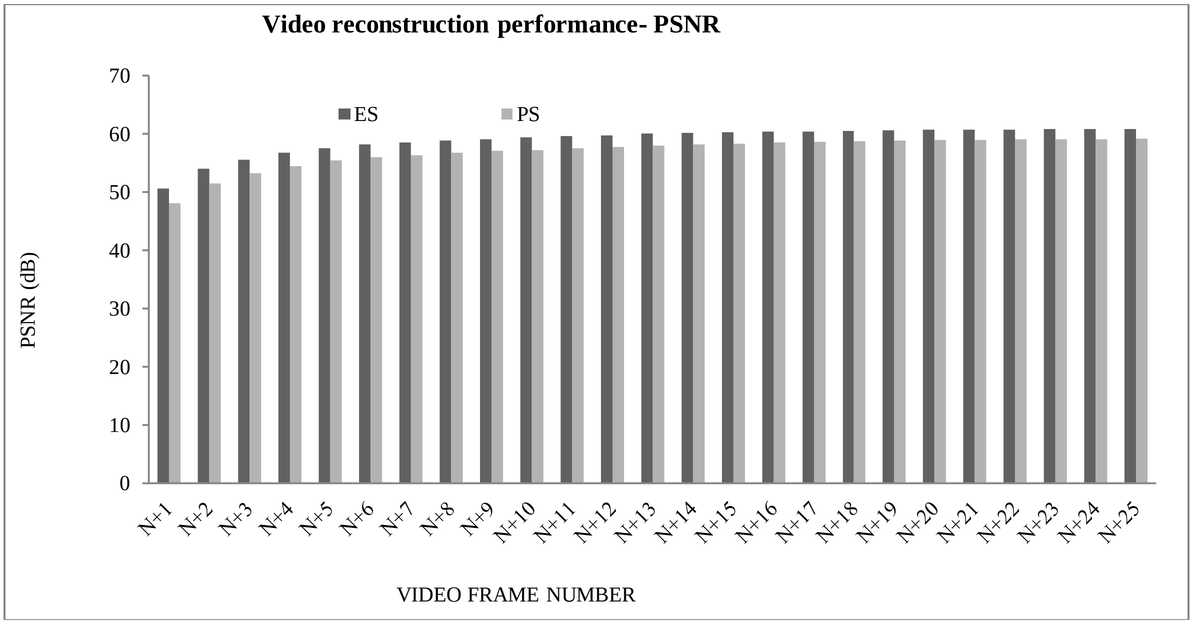

Fig. 3. Video frame reconstruction at the destination based on PSNR

The video reconstruction results obtained considering the ES and the PS scheme is shown in Fig. 3.of this paper. Considering the PS the highest PSNR value observed was found to be 59.15 dB which is in concurrence to the results published in reference. From Fig. 3, An average of 2 dB improvement is observed in the PS when compared to the ES.

The ED is measured in accordance to Equation (1). The end to end delay is measured considering the frame reconstruction at the D node and the results obtained is shown in Fig. 4. of this paper. From Fig. 4, It is clear that the ED shown considering ES and PS are exponential in nature shown by a dotted trend line in the Fig. 4. The cumulative per hop delays are used to obtain the end to end delay i.e. ED . Curve fitting was incorporated to derive the end to end delays definitions.

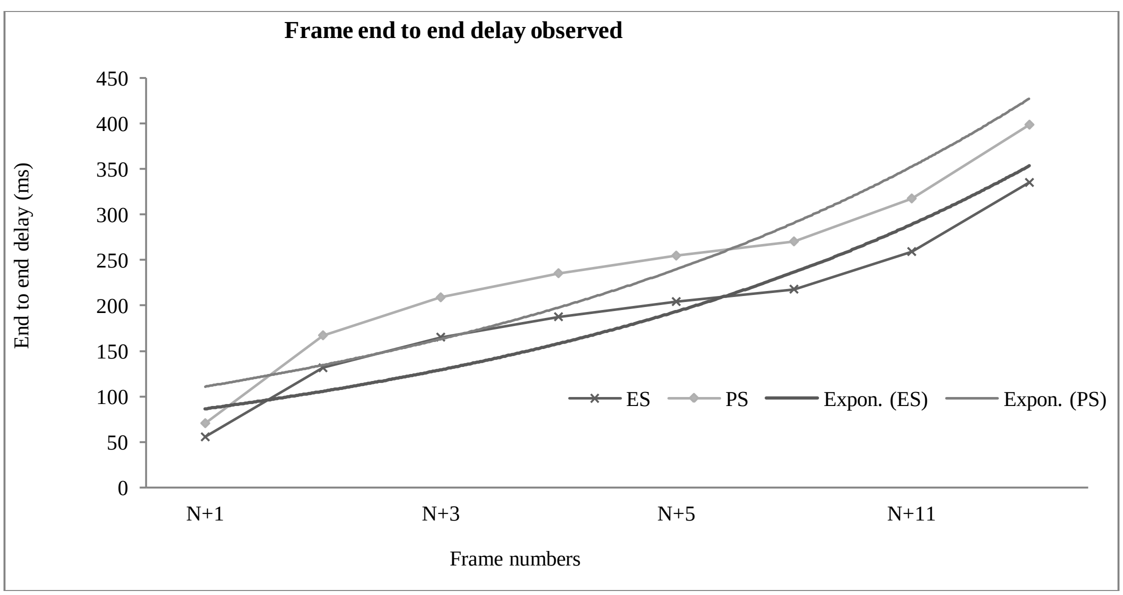

Fig. 4. Frame based end to end delay measure the destination considering ECDDAM and SPMLD

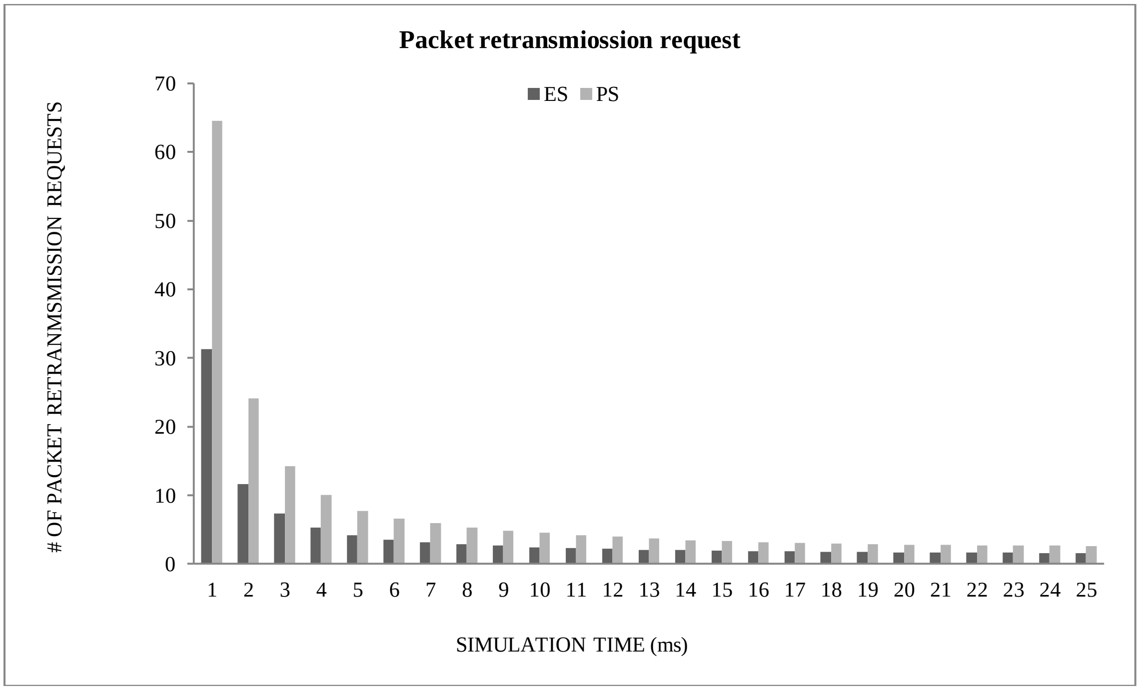

Fig. 5. Re transmission requests considering Existing system and proposed system

The number of retransmission requests observed by all the nodes in the Tree constructed is noted and the results obtained are graphically shown in Fig. 5. From the figure it was observed that the retransmission requests are reduced by about 47.86% considering the ES when compared to the PS algorithm.

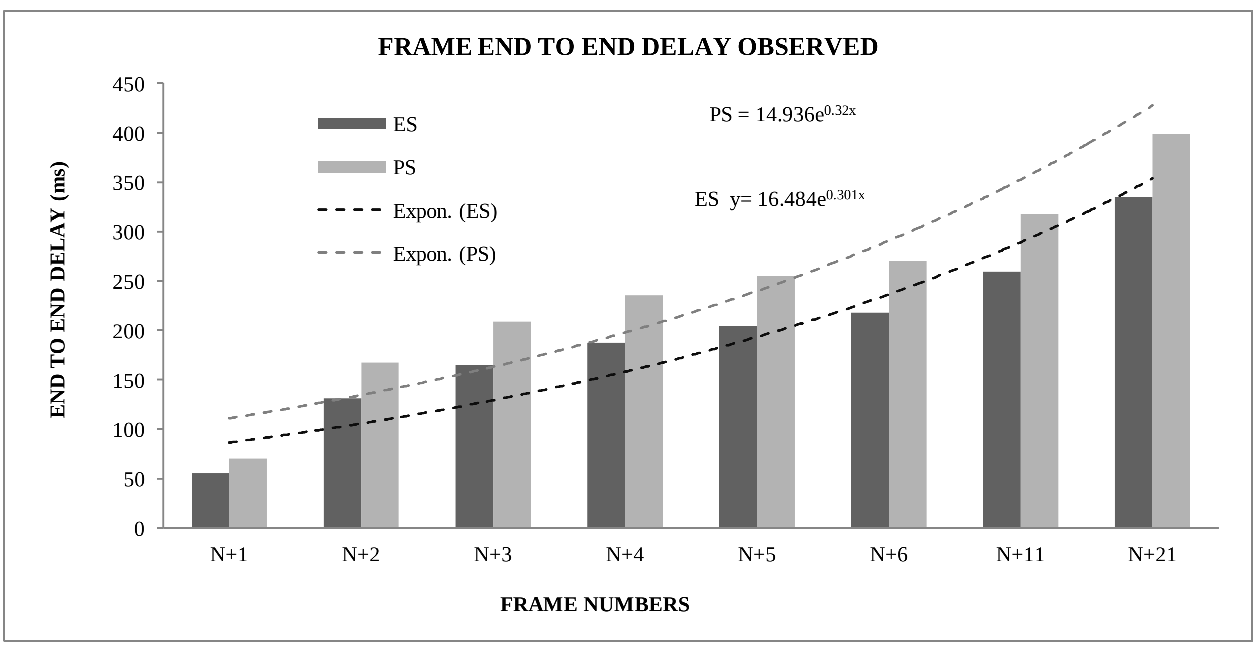

Fig. 6. End to end delay performance of the existing system and proposed system.

CONCLUSION

In this work, a simulation study is carried out for video transmission for wireless channel network. We propose a new approach for efficient video transmission. We have introduced enhancement layer based video transmission by using error prediction scheme. This scheme is used to minimize the error in the transmission for video frames in the enhancement layer. Average PSNR performance is computed for varying channels. Simulation results shows that the proposed scheme enhance the system performance significantly. Furthermore we have used adaptive selection of base layer and enhancement layer for fading channels. According to this approach if the information of the channel is available at the transmitter end, the adaptive layer selection parameters can be chosen based on the throughput. Simulation results show that the adaptive selection of the layer parameters is useful if the variation in the channel is slow which allows transmitter to attain the information. In simulation we have studied about the PSNR performance, average delay and packet retransmission rate.

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