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An Efficient Packet Loss Resilience Method Using Closed-Form Solution for Unequal FEC Assignment

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Abstract: This paper deals with an effective video transmission in communication networks by maintaining the quality parameters of the transmitted video file within the desired value. In this work, the Forward Error Correction (FEC) method is proposed to reduce the distortion in video transmission with H.264 video encoder. Initially, the packet distortion model is derived based on the error concealment property and error propagation effect in H.264 encoder. The transmission medium induces various noises which are transmitted to the transmission medium induces various noises which are transmitted to the receiver. Hence, the Channel Pattern Integration (CPI) algorithm is employed to find the best channel which is free of noises, for transmitting the selected video. Then, the Hierarchical Sequence Pattern (HSP) algorithm is used to validate the errors present in the received packets. The simulation results show that the proposed FEC scheme gives the substantial increase in PSNR performance. Due to the reduced computational complexity compared to the previous scheme, the proposed CPI technique offers an increased flexibility of dynamic video transfer over packet-lossy environments.

Keywords: Peak Signal to Noise Ratio (PSNR), Forward Error Correction (FEC), Channel Pattern Integration (CPI), H.264 Encoder, Packet Distortion Model, Packet lossy, Hierarchical Sequence Pattern (HSP), Parallel Processing.

1 Introduction

Video compression is done to reduce redundancy in video data. Some popular video compression standards are M-JPEG, MPEG-1, MPEG-2, MPEG-4, MPEG-7, MPEG-21, DV, H.261, H.263, H.264. Out of all video compression standards, the most popular standards are MPEG-4 and H.264 [1, 2]. In the field of communication system, the process of video transfer has been increased day by day. The video file is comprised of frames with moving objects. Applications like video calling, video conferencing and other communication technologies based on video, requires good quality at reception. It is achieved by transmitting compressed video files in the communication channel. Due to their smaller size, compressed files can be transferred faster in the channel. This increases the reliability of the communication system. Video compression consists of various types such as image, audio and motion compression. Two modes of video compression such as lossy and lossless compression are done. Mostly the lossy mode of compression is preferred. The highly compressed video file has high pixelation. Especially in continuously changing frames like explosion, scenery panning, flames and fast moving objects, the encoder has to compress larger pixels which further reduces the picture quality.

H.264 is an Advanced Video Coding (AVC) standard which is popular in recent years. It offers good quality video output with varying bit rates and also it offers flexibility in several applications. The H.264 encoding standards are used in DVD mastering, broadcasting and mobile communications. The H.264 codec is used to create BluRay Discs, various video streaming services like vimeo, Youtube, iTunes, Adobe Flash player and satellite broadcasting receivers. H.264 is patent protected. Usually the H.264 encoder is refereed to as x264 and MP4 which is incorrect. The x264 is a free encoder for H.264 format. On the other hand, MP4 is a container like AVI, MKV which uses H.264 and other such encoders. The H.264 encoder offers better video compression than

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other compression methods [3]. It offers DVD quality video with bit rate less than 1mbps. This is sufficient to transfer or stream online video.

The video files are transmitted over the communication network as a part of data sharing. To achieve fast data transfer, the video file is compressed. The transmission network contains noise parameters in various forms. The compressed data gets deteriorated due to induction of noise and some packets may get lost. Traditional method of data rectification like Error Correction Code (ECC) was used to reduce packet loss at the receiving end. Cyclic Redundancy Check (CRC) method was executed at the receiving end to correct the error bit. This method is highly complex and consumes more time. The previous ECC algorithm blocks any erroneous data and the transmitter has to resend the entire data. Hence the PSNR at the receiving is reduced. This is the drawback for the existing technique.

The proposed method uses H.264 to compress the video file. The transmission channel contains noises. These noises deviates the properties of the transmitted data. Hence the best channel for transmitting the compressed video file is selected by proposed novel algorithm. The compressed video packets are encoded and then transmitted through the selected channel. The encoded video file is decrypted at the receiving end. H.264 decoder is used to decompress the received data and transmit it to the user. The structure similarity matrices of the compressed file before and after transmission are compared for any parameter deviations. In case of any deviation, the parameters of the novel channel-selecting algorithm is modified to get the required result.

Section 2 surveys the existing techniques of channel optimization for quality data transfer. Section 3 provides the detailed description of the proposed channel selection technique. Then the performance of the proposed technique is evaluated in Section 4. Finally the paper is concluded and the enhancement that will be implemented in future is stated in Section 5.

2 Literature Survey

This section furnishes the overview of the existing concepts which led to this proposal. Several existing techniques are discussed as follows:

2.1 Wireless Video Streaming

High Frame Rate (HFR) video transfer gained popularity in multimedia applications. To achieve smooth viewing experience for the end users, the modeling-based approach for optimizing the HFR video transmission in the wireless network was proposed [1]. A novel Joint Frame Selection and FEC Coding (JASCO) was implied to focus on transmission performance of H.264 streaming. JASCO achieved higher frame rate at reception and maintained same PSNR. The disadvantage of the work is that it is unable to transfer multiple HFR content with varying quality and frame rate. There existed a challenging task to deliver HFR video in real time using Transmission Control Protocol (TCP) through a noisy wireless network. TCP execution caused fluctuations in throughput and desired reception quality was not achieved. Hence, an application layer FEC framework on HFR video over TCP was proposed [2]. Initially a mathematical model was developed to analyze the distortion of the video at its frame level. To reduce the total sum of distortion in the HFR video at the reception, a novel raptor coding with joint approximate distortion estimation proposal was adopted. This scheme achieves good video PSNR and good frame rate through TCP. Improvement was required in QoS metric of the existing ROCHET technique.

An algorithm to reduce the distortion of video at the reception was proposed [3]. The Inter Layered FEC (IL-FEC) coding is used to transmit layered video through a Truncated Hybrid Automatic Repeat request (THARQ) system: the major problem as the THARQ scheme limits the retransmission possibilities. Hence an Adaptive Truncated Hybrid Automatic Repeat request (ATHARQ) found an appropriate scheduling of IL-FEC [4]. The PSNR value was increased than conventional methods. The work required improvement in throughput.

Maximum throughput and minimum transmission delay of the video packets at the receiving end was prior requirement in wireless data transmission [5]. A model for predictive control approach to optimally combine buffer space and throughput predictions in order to enhance the QoE of user was implemented [6]. The limitation of the design was the need of accurate algorithm. A novel FEC coding scheme is introduced to meet the high throughput demand in wireless networks for reliable cloud gaming video streaming [7]. Cloud gaming was a demanding service with the growth of 4G networks. The frame rate reduction of the gaming video was not acceptable by the users. Hence an adaptive Source FEC Coding over TCP (ESCOT) is introduced. The TCP characteristics was leveraged to provide overall video quality improvement in real-time gaming. The work required improvement in the optimization of the video stalling (by improving the delay performances for the user input in gaming) and distortion on wireless network. To ensure good quality of experience in internet video, a better bitrate adoption technique was required. There was resolution loss in the selected video at the time of starting and mid-section. A CS2P frame work to analyze the throughput of the network is suggested [8].

Predicting incoming signal was done and corresponding bit rate adjustments in the client side was altered to achieve good QoE. The frame work needed improvement in the accuracy of predicting algorithm. High-definition video transmission over wireless network gained popularity. The services were focused on streaming H.264 encoded video on mobile devices. FEC scheme dubbed as Priority Aware TCP Protocol Oriented Coding (PATON) was developed to minimize video distortion by leveraging TCP [9, 10]. End-to-end delay was reduced, frame loss was reduced and throughput was improved. PSNR should be increased and delay at reception has to be reduced.

The maximum video quality was maintained by adjusting the coding rate in the channel loss. Threshold set in the presence of all disturbing channel conditions. The analytical method to model FEC code rate decision schemes was complex and costly. Hence a joint model in a simple closed-form was proposed [11]. It contained small number of scene dependent parameters. This method estimated optimal FEC Code rate with less computation complexity. Wireless video streaming was a challenging task with varying video quality in the network due to changes in the network parameters. Various adaptations were made by transmitting and receiving ends to compensate loss and to maintain video quality. A standardized scalable video coding technique with iterative steps to allot bitrate for various incoming video streams is recommended [12]. It maximized the weighted sum of video qualities associated with different streams. The work lacked spatial scalability. An optimized technique for Bit level IL-FEC coded scalable video transmission over wireless network is proposed [13, 14]. The PSNR value was increased than the conventional Unequal Error Protection (UEP) methods. The algorithm had to accompany advanced FEC Code such as LDPC, BICM and TCM etc.transmitting un-coded video over wireless communication network was a challenging task. Most data was lost in the transmission system. Hence improvement in the transmission efficiency is necessary.

A real-time adaptive algorithm for video streaming over multiple wireless networks is designated [15]. The proposed work gave smooth and high quality video with low start up latency. The proposed work could not predict bandwidth requirement and can't allot load to links appropriately. The unsuccessful decoding of the Base Layer (BL) instructed the video decoder to discard all the Enhancement Layers (ELs) depending on it, regardless of whether they were successfully decoded. Naturally, this course of action wasted the transmit power assigned to the dependent layers. A bit-level Interlayer FEC (IL-FEC) scheme that embedded the BL into the FEC coded ELs to improve the reception of the BL with the aid of soft decoded EL is suggested [16]. Inefficient packet scheduling was the limitation of the system.

Novel Robust Bandwidth Aggregation (ROBBIA) method for streaming video in integrated heterogeneous wireless networks was proposed [17]. Problems in FEC redundancy adaption, transmission rate assignment and path interleaving was analysed. The proposed technique could exploit the path diversity in heterogeneous wireless networks for optimizing transmission scheduling and video streaming quality was improved. The drawback of the scheme was that it can't interleave uneven paths. Interactive screen was designed and used in many self-service booths like ATM, Ticket reservation systems, etc. The screen with better image presentation gave better user response. Hence efficient video, image compression and data encoding technique were modelled.

2.2 Packet loss

A Cognitive Go Back-N-HARQ to achieve the need is recommended [4]. FEC was used to protect transmitted data over the communication network. The performance of the algorithm was affected when burst packet loss occurred. This resulted in degraded video quality at reception. A scheme termed Trading Delay for Distortion (TELFORD) facilitated packet data scheduling of multiple destination [5]. The packet losses were eliminated by estimation of the properties of the path before data transmission. The method was unable to provide lossless mobile video streaming and enhanced two-way transmissions in video conferencing. Quality of experience was always a factor of demand in internet video applications. For better video at the receiver equipment, an algorithm in the player has to be executed to adjust the bit rate setting.

For effective adaptation of the algorithm to achieve universal solution for the bit rate issue was proposed [6]. The increased bandwidth for the users gave rise to the usage of video calls. The rate control and video quality of Skype video calls under different network conditions [18] is analysed. The measurement results showed that Skype video responded quickly when the network was overloaded. Further, the Skype adjusted data transfer rates according to the available bandwidth and also adjusted the frame rate to compensate packet loss and delayed data propagation. The Skype was TCP friendly. Based on the results, rate control model, FEC model was developed for Skype. FEC model required further optimization.

2.3 Noise reduction techniques

An optimal FEC procedure to determine the optimal and channel coding rate to transfer video packets with minimum loss in a noise-prone network was proposed [10]. A motion compensated temporal filtering (MCTF) and de-noising techniques at the transmitting and receiving terminals respectively to increasing the transmission efficiency is anticipated [14]. The de-noising algorithm at the receiving end reduces random noise in the video. The work required optimal improvement to explore other possible signal processing techniques. As a result, video streaming gained importance. The mobile devices had many interfaces for accessing wireless networks. Choices were made to select the best network to achieve better video streaming. High frame rate screen video coding for screen sharing application was developed [19] by using Frame level selection algorithm in which content updation was provided. Dynamic video capturing contained high picture details in addition to the noise levels. Joint non-Gaussian de-noising and super solving of raw high frame rate videos is introduced [20]. While processing a video to get more details out of it, this resulted in a grainy noisy video output. The proposed technique implied local linearization methods to achieve noise-free video at the reception. The techniques required optimal improvement to reduce noise in the transmitted data.

3 Proposed work

In this section, the clear description of an efficient packet loss resilience method using closed-form solution for unequal FEC assignment based on a new packet distortion model is proposed. The objective of the work is to deliver error-free video at the receiving end by adopting novel CPI and HSP algorithm based on FEC. Each frame of the video to be transmitted is initially compressed by using H.264 Encoder. These packets undergo HSP-FEC encoding for ensuring secure file transferring. The Structure Similarity Index Metrics (SSIM) of the encoded video is updated. At this stage the best channel has to be selected. The channels possess various disturbances called noise, they are thermal noise, flicker noise, Additive White Gaussian Noise (AWGN) and multiplicative noise. These noises interfere with the transmitting data and corrupt it. Corruption of a video file includes degraded video at reception, smudge effect on the video, etc. Hence all the available channels are analyzed by the CPI algorithm and a noise-free channel is selected for the video file transmission. At the receiving end the HSP-FEC decoding of the received encrypted video file is done to obtain error-free decoded video packets. Using the H.264 decoder the data packets are reconstructed at receiving end. The SSIM of the decoded video is noted. Now both SSIM are compared to verify the presence of distortion. There is no notable distortion in the video file at the receiving terminal, this enabling the user to view the high quality undistorted video. The work flow of the proposed technique is given in Fig. 1.

The experiment is evaluated using MATLAB 2017 (b). H.264 delivers MPEG-4 quality with a frame size up to four times greater. It can also provide MPEG-2 quality at a reduced data rate, requiring as little as one third the original bandwidths.

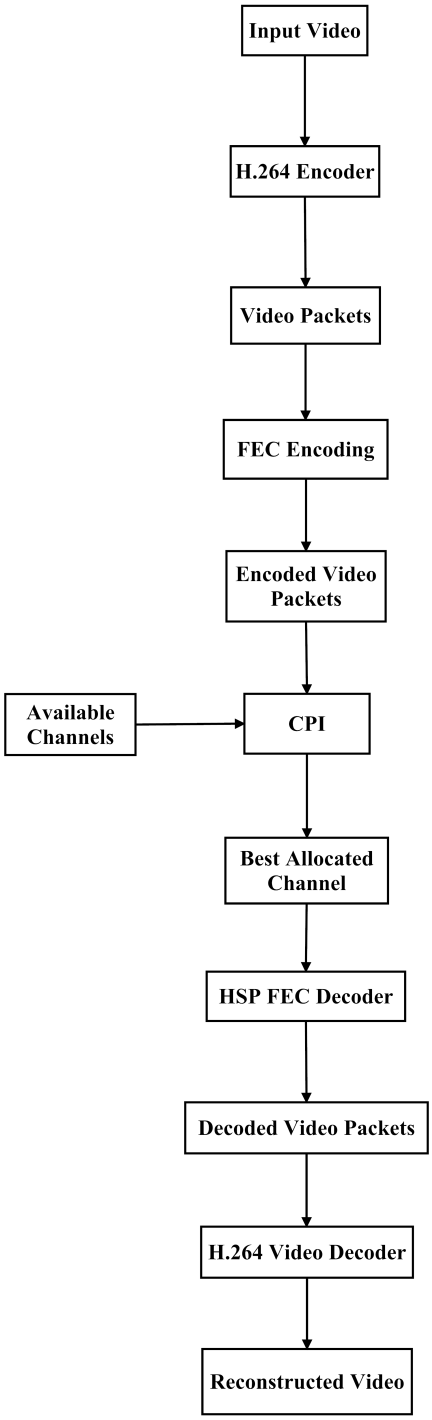

Fig. 1: Proposed work flow

3.1 Forward Error Coding

FEC coding technique is used to control the occurrence of error in the transmitted data over noisy channel. A suitable error correcting algorithm is used to perform FEC. The algorithm adds unwanted codes to the data to be transmitted. FEC transmits each character more than twice. The receiver inspects every instances of each character. The data is accepted at the reception only if the confirmation is true in two instance checking. If the data does not abide to the protocol then it will be rejected. An underscore or blank space will be displayed in that position. It facilitates the user to self-correct the errors without requesting to resend the data.

3.2 Channel Pattern Integration (CPI)

This is the process in which the best channel for data transmission is selected. The model is formed in which all the noises are simulated with desired decibel level. The generated noises are added into the channel

3.3 Thermal noise

It is also called as Johnson or Nyquist noise. Noise occurs when the signals move in between the transmitting and receiving end. The magnitude of the noise is directly proportional to the temperature of the circuit. When a thermal noise remains in the communication system, then the system is modeled as an additive white gaussian noise channel. The Root Mean Square (RMS) value of the voltage due to the thermal noise is given by

$$v_n = \sqrt{4k_n TR\Delta f} \tag{1}$$

where, k_n—Boltzmann's constant (J/K)
T—temperature of the resistance of the circuit path (K)
R—resistance of the path (Ω)
Δf—bandwidth in Hz

The signal power is measured in decibels (dB) in milliwatts. Thus the noise power in a resistive element is given by

$$p_{\mathrm{dBm}} = 10\log_{10}(k_b T\Delta f * 1000) \tag{2}$$

On further simplification we get

$$p_{\mathrm{dBm}} = -174 + 10\log_{10}\Delta f. \tag{3}$$

By the use of the above relation the noise power for various bandwidths can be calculated.

3.4 Additive white Gaussian noise channel

It is implied to imitate various factor that affect the signal propagation channels. It is termed as additive, because it could sum-up with any intrinsic content that affects the actual signal. Since its power is equally spread across the system bandwidth, it is termed as white. Its average time domain distribution is zero and hence it is called as Gaussian. AWGN is applied for background noise simulation to the investigating channel in addition to multipath, terrain blocking, interference, ground clutter and self-interference that modern communication system faces.

3.4.1 Flicker noise

1/f is the other term for flicker noise. It is called as pink noise as well. The noises are high-frequency noise with pink spectrum and hence the name pink noise. This noise is accompanied by reactions such as recombination of particles in the electronic circuits of the communication system.

3.4.2 Multiplication noise

This noise multiplies with the relevant signal while capturing, transmitting and other processing of data in a network. The noises are simulated into the channel and the PSNR values of all the added noises are calculated. Best channel with higher PSNR value is selected for data transfer. The structure similarity metrics are used to determine the similarities of the original and the transmitted video file. The components under consideration are luminance, contrast and structure. The algorithm for selecting the best channel for data transmission is given as follows.

Algorithm 1: Channel Pattern Integration Algorithm (CPI)

Input : Encoded result Encode$_{res}$
Output : Best Channel
Procedure:
Step 1: Generate various noise in the channel. Create Thermal noise and add to signal with the parameter
NF = 5, Sample$_{rate}$ $= 10^6$
where, NF is the Noise figure which is the measure of degradation of SNR
Sample$_{rate}$ is defines as the number of samples per second
Step 2: Convert the Encode$_{res}$ to packet formation
Pack = Encode$_{res}$(:)
Step 3: Perform QAM modulation to the input packet
Modulation$_{data}$ $= qammod(Pack)$
Step 4: Add the Thermal noise to Modulation$_{data}$ to obtain noisy data. Threm$_{data}$ = step(Thermal noise, Modulation$_{data}$)
Step 5: Perform Flicker noise and add to signal
Initialize SNR = 20 dB
$\mathrm{SP} = 10 * \log_{10}(std(Pack)^2)$
// SP-Signal Power
std–Standard Deviation $\mathrm{NP} = 10^{(SP+SNR)/10}$,
// NP–Noise Power Noise$_{RMS}$ $= \sqrt{Np}$

Algorithm 1 continued. . .

Step 6: Perform AWGN noise and add to signal
Initialize $NSD = \frac{\sum Pack^2/length(Pack)}{SNR}$
// NSD–Noise Spectral Density
Step 7: Create AWGN noise
$AWGN_{noise} = \sqrt{NSD/2} * randn(1length(Pack))$
Add the AWGN noise to input Pack,
$AWGN_{data} = Pack + AWGN_{noise}$
Step 8: Perform Multiplicative noise and add to signal
$Multi_{noise} = imnoise(Pack^{'}, speckle^{'}V)$ // V-variance is set to 0.04
Step 9: Estimate Peak Signal to Noise ratio for all added noise result
$PSNR = 20 * \log_{10}(\frac{255}{\sqrt{MSE}})$
// MSE–Mean SquareError
Step 10: Select the best channel by estimating maximum PSNR
$[PSNRmaxChannel_{id}] = \max(PSNR)$
Based on the $Channel_{id}$ select best channel among available channel
When the PSNR gets reduced for a given channel bandwidth, the number of bits gets increased. By this the best channel is selected. This can be represented as,
Maximum number of bits/sec $= Hlog2(1 + PSNR)$
Step 11: Quality of the video which is transmitted via best channel by Distortion of Structure Similarity
$a = Encode_{res}$, $b = Flicker_{data}$
// $Flicker_{data}$—data which is transferred via best channel
$lum(a,b) = \frac{2*\mu_a*\mu_b+\alpha}{\mu_a^2+\mu_b^2+\alpha}$
$cont(a,b) = \frac{2*\sigma_a*\sigma_b+\beta}{\sigma_a^2+\sigma_b^2+\beta}$
$stru(a,b) = \frac{\sigma_{ab}+\gamma}{\sigma_a\sigma_b+\gamma}$
$DSSIM(a,b) = lum(a,b) * cont(a,b) * stru(a,b)$
where
μ_a—mean of a, μ_b—mean of b
σ_a—variance of a, σ_b—variance of b
σ_{ab}—covariance of a & b
α—1, β—0.5, γ—0.6

3.5 Hierarchical Sequence Pattern (HSP)

It is an encoding and decoding algorithm executed at the transmitting and the receiving terminals. The video to be transmitted is compressed by the H.264 coder. The compressed file is in the form of combination of video packets. These packets undergo the encoding process.

4 Result and Discussions

In this section, the performance results of both existing and proposed techniques are analyzed and evaluated based on PSNR value for various video streaming rates and video frame index, cumulative distribution function for various end-to-end delay rates. To prove the superiority of the proposed system, it is compared with some of the existing techniques like PATON, CLOSET,

Algorithm 2: Hierarchical Sequence Pattern Algorithm (HSP)

Input : Compress Image $Compress_{img}$
Output : Encode Result $Encode_{img}$
Procedure:
Step 1: Initialize input bits
$Bit_1 = 0$, $Bit_2 = 0$, $Bit_3 = 0$, $Bit_4 = 0$
Step 2: Perform Shift operation $Bit_4 = Bit_3$, $Bit_3 = Bit_2$, $Bit_2 = Bit_1$
$Bit_1 = Compress_{img}(ij)$
$\forall 1 < i < size(Compress_{img}, 1) \& 1 < j < size(Compress_{img}, 2)$
Step 3: Generate reference message bit which is going to be encoded with $Compress_{img}$
$Ref_{msg}(1) = 0$, $Ref_{msg}(2) = 1$, For $i = 1:7$
$Ref_{msg}(2+i) = bitshift\left(Ref_{msg}(2+i-1), 1\right)$;
End
Step 4: Consequently create rest of the bits with the size of $Compress_{img}$ by bitwise ExOR operaton
$Y = length(Ref_{msg})$,
initialize random values
$x1 = 2, x2 = 4, x3 = 5, x4 = 6$
For $j = 1 : (size(Compress_{img}, 1) - length(Ref_{msg}))$
$Temp = bitxor(Ref_{msg}(x1)Ref_{msg}(x2))$;
$Temp1 = bitxor(Ref_{msg}(x3);\ Ref_{msg}(x4))$;
$Ref_{msg}(Y+i) = bitxor(Temp, temp1)$;
$x1 = x1+1, x2 = x2+1, x3 = x3+1, x4 = x4+1$
End
Step 5: Image intensity value should be 1 to 256
Generate random value between1 to 256 for 17 bits
$Rnd = randint(1, 17, [x1256])$
For $j = 2 : length(Rnd)$
For $i = 1 : 256$
If $Rnd(j) == Ref_{msg}(i)Intensity_{val}(j-1) = i-2$
End
End
End
Step 6: Calculate $Encode_{img}$ with the Ref_{msg}
$Out_{res}(1:16) = 0$;
For $dc = 1 : size(Compress_{img}, 1)$
$Tempmsg = bitxor(Out_{res}(1)Compress_{img}(dc))$;
For $k = 1 : size(Compress_{img}, 1)$
If $(Temp == Ref_{msg}(k))$
Tempmsg$= k-2$;
End
End
$Temp1 = 0$;
$Register_{inp} = 2 + \mod((Intensity_{val} + Tempmsg), 255)$
For $j = 15 : -1 : 1$
$Tt = Out_{res}(j)$;
If $Tempmsg == -1$
$Out_{res}(j) = bitxor(0, Temp1)$;
Else
$Out_{res}(j) = bitxor(Ref_{msg}(Register_{inp}(j)Temp1)$;
End
$Temp1 = Tt$;
End
End

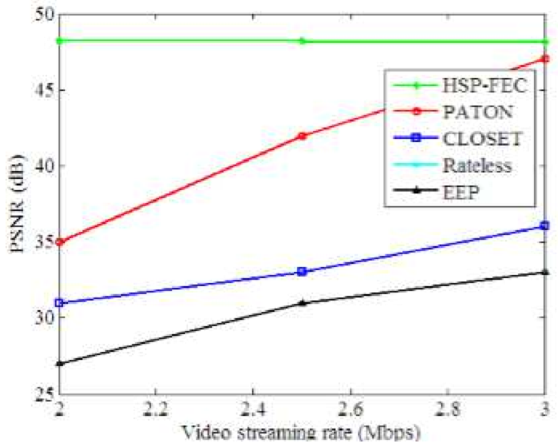

Fig. 2: PSNR vs. video streaming rate

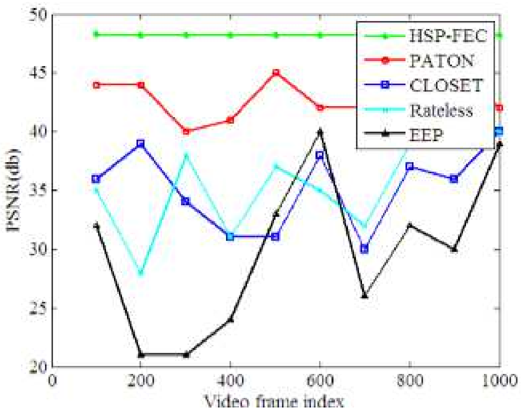

Fig. 3: PSNR vs. video frame index

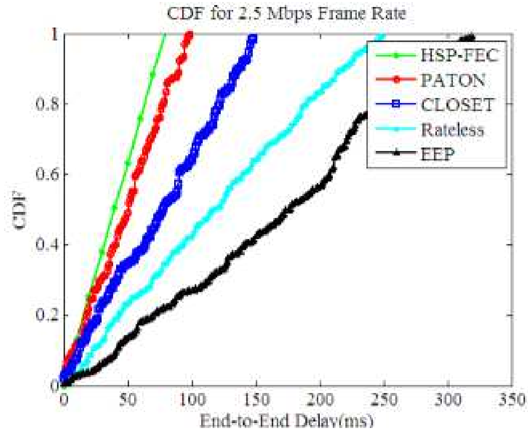

Fig. 4: CDF vs. end-to-end delay

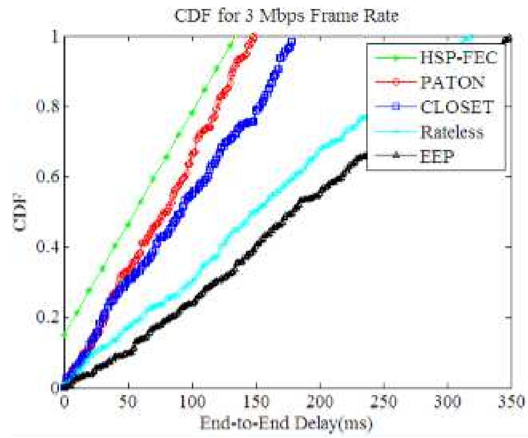

Fig. 5: CDF vs. end-to-end delay for 3 Mbps

Rateless and EEP. The throughput is said to be number of bits through which the destination receives it successfully.

4.1 PSNR (Peak Signal to Noise Ratio)

PSNR is the ratio between maximum possible power of a signal and the power of corrupting noise that affects the signal. It is used to measure the quality of reconstructed data.

Fig. 2 gives the graph plotted between PSNR value of the channel and video streaming rate given in Mbps. The PSNR value attained through the proposed technique is higher than the results obtained from conventional methods. Hence by applying the proposed method for channel selection and video transmission, a high quality undistorted video at reception is achieved.

Fig. 3 is a plot between PSRN and video frame index. The results confirm that the compressed video packets are transferred with high PSNR value in the proposed techniques than the conventional PATON, CLOSET, Rateless and EEP methods. The reception has distortion less data

4.2 CDF (Cumulative Distribution Function) for 2.5 Mbps frame rate

CDF is used to specify the distribution of multivariate random variables.

Fig. 4 represents plots response of CDF to end-to-end delays for 2.5 Mbps. There is a considerable delay of video packet transmission between sending and receiving end. With increase in the end-to-end delay the corresponding variation of CDF is steeper than other conventional methods, which proves the proposed technique is better than the techniques used earlier.

In Fig. 5 the response of CDF is plotted against end-to-end delays of 3 Mbps. With increase in the end-to-end delay the corresponding variation of CDF is steeper than other conventional methods, which proves that the proposed technique is better in all varying speed conditions than the techniques used earlier.

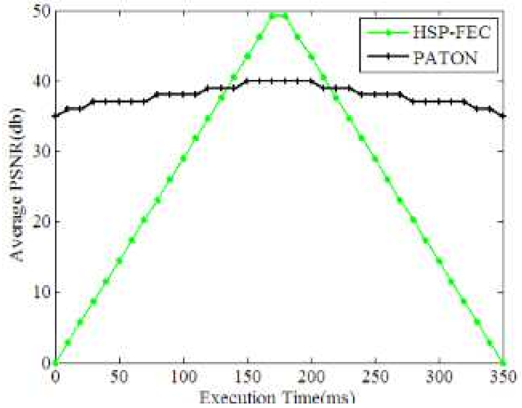

Fig. 6: PSNR vs. execution time

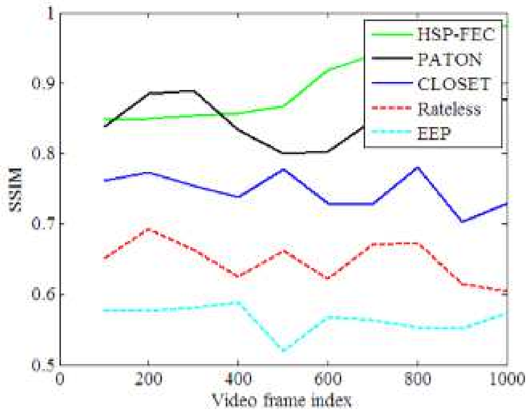

Fig. 7: SSIM vs. video frame index

4.3 Average PSNR for Execution time

The PSNR value for execution time is plotted in Fig. 6 The PSNR slope increases continuously and is stable for certain duration and then decreases. If the execution time increases, then it ends up adding noise to the video packet. The PSNR value is maintained until the video file is transmitted in the channel. The proposed scheme keeps the PSNR value to the value of 50db whereas the conventional methods have had a less PSNR value of 40 db. This proves the proposed technique is better than earlier methods.

4.4 SSIM

Structure similarity metrics are the collection of video characteristics such as luminance, contrast and structure. Fig. 7 shows the SSIM for different video frame transmission. For every frame the metrics are calculated which gives optimum improvements when compared to conventional techniques.

Table 1: Bit error rate

	Number of frames					
BER	100	200	300	400	500	600
	0.0996	0.0989	0.0996	0.0994	0.0997	0.0992

Table 2: Compression ratio

	Number of frames					
CR	100	200	300	400	500	600
	80.3027	79.0366	79.2863	80.1660	81.0755	81.6105

4.5 Bit Rate Error (BER)

Before transmitting the video data to the receiver, it is compressed using H.264 encoder. The compressed data may contain certain losses. It is necessary to estimate the presence of error in all video frames. Table. 1 shows the Bit Error Rate for the corresponding video frame index

In the proposed method, for every frame index, the BER is 0.1, which is the best rate.

4.6 Compression Ratio

Compression ratio is the ratio between the uncompressed size and compressed size. Table. 2 shows the video compression ratio for every video frame index given. 79–81% compression of each video frame index is achieved.

5 Conclusion

The growth in the communication networks with reduced cost services increases the demand of video sharing. The channels selected for data transfer contains noise which distorts the transmitted data. Due to this, the receivers experience data packet loss and increased noise level in the video. To rectify the data packet loss in communication networks a novel HSP algorithm based on FEC is adopted for encoding and decoding the compressed video packets. The FEC technique is used for controlling errors in transferring data in a noisy channel. The video file is compressed by H.264 encoder before transmission. The output of the encoder is a combination of video packets. These video packets undergo the HSP encoding and are ready to be transmitted. The CPI algorithm is executed to find the best available channel for video packet transmission. In the model environment, the channel noises like thermal noise, flicker noise, AWG noise and multiplicative noise are simulated into the channels at the desired levels. The PSNR value of all the channels are read and updated. The best channel is selected based on the higher PSNR values. The channel ID corresponding to the highest PSNR value is chosen for transmission. At the receiving terminal, the HSP algorithm is implied for decoding the received data packets. After decoding, the video packets undergo H.264 decoding. By this the data is uncompressed. At this instant the structure similarity metric of the received video packet is compared with the previously-updated metric. The distortion in the received video is not noticeable. The results obtained from the proposed technique are better than those obtained from conventional methods. Thus the proposed technique of selecting the best channel for encoded video transfer is proved to be optimum.

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