

## VIDEO QUALITY IMPROVEMENT USING LOCAL CHANNEL ENCODER AND MIXED PREDICTOR BY WAVELET, NEURAL NETWORK AND GENETIC ALGORITHM

H. Farsi P. Etezadifar

*Department of Electronics and Communications Engineering, University of Birjand, Birjand, Iran  
hfarsi@birjand.ac.ir, p.etezadifar@birjand.ac.ir*

**Abstract-** In recent decades, progress of technology causes that video transmission via communication channels has met high demands. Therefore, many methods have been proposed to improve the quality of video under channel errors. The aim of this paper is to increase PSNR for reconstructed video at receiver. This is achieved by increasing channel encoder rate but in constant transmission rate. Since the motion compensation block is unable to minimum the variance of frame information, accurately, the proposed method is applied after this block. We proposed the secondary predictor by combination of wavelet transform, neural network and Genetic Algorithm (GA). This predictor reduces the variance of frame information considerably, and therefore lower bits are required for transmission. Thus, it is possible to apply the local channel encoder whose rate depends on variance reduction. The proposed method is able to increase channel coding rate without increasing the amount of information for any frame. This method provides more robustness for video frames against channel errors. In order to evaluate the performance of the proposed method, it is tested for different source coding rates and several SNRs for channel and the obtained results are compared with a new method.

**Keywords:** Video Coding, Variable Length Coding (VLC), Channel Coding, Wavelet Prediction, Artificial Neural Network, Genetic Algorithm (GA).

### I. INTRODUCTION

Today's, increasing grow of using wireless channels and mobile telecommunication and tend to serving the achievements of modern telecommunications for real-time multimedia data transmission such as video involves advance research in this area. Different methods for video coding with lower bit rate and higher quality have been proposed so far [1]. In wireless and mobile telecommunication due to nature of wireless environment and channel errors, the quality of video is affected at the receiver. Several methods have been proposed to reduce the errors caused by the channel [2]. Channel coding is an efficient method which is generally used to detect and

correct the errors. However, channel coding was performed without considering source coding rate. In recent years, different methods as combination of channel and source coding have been introduced. These methods are known as Joint Source and Channel Coding (JSCC). JSCC is generally based on channel estimation [3, 4, 5, 6]. This means that in order to have higher PSNR, higher channel coding rate is required.

In this paper, we propose a new JSCC method which increases channel coding rate but total transmission rate is maintained still constant. In fact by increasing channel coding rate, more robustness of video frames can be provided. One of the most important problems in general JSCC methods is that the channel information is required for modification of source and channel coding rates. However, in the proposed method the channel information is not needed. In the other word, the proposed method is applied on every source coding rate, independently to improve the quality of the reconstructed video frames at receiver. In this paper, Peak Signal to Noise Ratio (PSNR) and also Bit Error Rate (BER) are used as measures to indicate the quality of reconstructed video. This paper has been organized as follow:

In section II, video frame encoding using MPEG-4 part 2 is introduced for both Intra-picture (I-picture) and Predicted-Picture (P-picture). Then, in section III, the channel coding is briefly indicated. In section IV, combination of wavelet prediction, neural network and GA is indicated. Next, in section V, the proposed method is introduced and explained step by step. Finally, the performance of the proposed method is evaluated and compared with a new method indicated in [7].

### II. VIDEO FRAMES CODING OVERVIEW

In this section, a group of pictures (GOP) is shortly introduced. Then video frames encoding and decoding are explained using MPEG-4. In the MPEG standard, three types of pictures are introduced by I, P and B briefly. The GOP is made by combination of these pictures. A picture which is coded by its information is called I-picture. Thus JPEG compression algorithm is used for coding I-pictures [8].

The picture which is estimated by the nearest previous I or P pictures is called P-picture. For P-picture, motion compensation predictor is also used. The picture which is estimated by the nearest previous and next, I or P pictures is called B-picture. For B-picture, motion compensation predictor is also used, similar to P-picture. We use two P-pictures and I-pictures from GOP for simulation, because using B-pictures (Bidirectional-pictures) leads to a delay in the system. The delay is created in B-pictures because prediction is performed on the next and previous frames. Thus, system is forced to wait until the next frames to be entered and buffered such that the estimation of B-picture is possible to be performed.

**A. Overview of the MPEG-4 Encoding Method**

In this paper, we use the MPEG-4 part 2. In following section some of the operations used in the encoder and decoder are described.

**A.1. Reorder**

After quantization of DCT coefficients, due to having many zero values, the obtained coefficients are arranged in zigzag form. Therefore, the values related to different frequencies are grouped [9].

**A.2. Data Packetization**

Occurring errors in data transmission may cause loss of synchronization between transmitted video frames. To overcome to this problem, a resynchronized mechanism is required. One method for synchronization is to use data packetization. Thus, a number of markers are inserted in each packet for synchronization. Encoder divides each frame into packets and inserts markers related for Synchronization at the beginning of each packet. Encoder and decoder perform resynchronization when the markers located in the beginning of each packet are received. This procedure causes synchronization between video frames and avoids additive errors [10].

**A.3. Data Partitioning**

In this model, the data packet is divided into two areas. The main idea is to separate data with higher values (DC coefficients of DCT matrix, coding model information and motion vectors) from data with lower values (AC coefficients of DCT matrix and residual errors). For I-picture transmission, the first area contains source coding information and DC coefficients. However, the second area containing lower values has AC coefficients. For P-picture transmission, the first area contains source coding information and motion vectors whereas the second area contains the DCT data (texture, DC and AC coefficients). In data partitioning the markers (which are unique) are used for synchronization.

**A.4. Motion Compensation Block**

Motion compensation algorithm is based on the Minimum Mean Square Matching Error (MMSE) between two successive frames that are based on the following equations:

$$MSE(d_x, d_y) = \frac{1}{M_1 N_1} \sum_{(m,n) \in W} (b[m,n,k] - b[m-d_y, n-d_x, k-1])^2 \tag{1}$$

$$(d_x \ d_y)^T = \arg \min_{(d_x, d_y)} MSE(d_x, d_y) \tag{2}$$

In Equation (1), the third parameter, *b*, including, *k*, and, *k* - 1, calls the pixels of two consecutive frames, and two variables, *m*, and, *n*, show the pixel location in a two dimensional space for each frame. The variable, *W*, represents the search window, and motion prediction vector is obtained by searching at this range. *W* = 15 is considered in this paper. This means that the window size is 16×16 for searching. Also, prediction window size is considered to be 8×8. In other words, two consecutive frames with in the search window are compared with each other, and the least value is stored as a member of motion prediction matrix.

Next, after prediction of a frame completely and construction of the motion prediction matrix, the predicted values are subtracted from the previous frame. Finally, the obtained values are applied to DCT block. After passing through the DCT block, the obtained values are quantized by 64 levels. Finally, DCT blocks are coded by Variable Length Coding (VLC) block for transmission [11]. At the receiver, firstly, the received data is decoded by VLC block and dequantized. Then, the dequantized information is passed through IDCT block. Finally, using motion prediction matrix, input frame is added to previous frame and therefore the transmitted frame is reconstructed.

**A.5. Variable Length Coding (VLC)**

Variable Length Coding is final step before transmission of information on each frame. The VLC tries to reduce the transmission rate. This type of coding is performed in two different procedures:

In first procedure, due to existing many one values after quantization of DCT matrix as shown in Figure1, for instance, a few bits are considered for transmission. For example, only 3 bits are assigned to send the number of one value which represents as 11s. This code, 11s, is determined using specified table for the MPEG standard [11]. The value of *s* is determined by the sign of one value. If the sign of the one value is positive (+1) then *s* = 0 and otherwise, *s* = 1. On the hand, since the occurrence probability of the sample values except ± 1 is low, the number of allocated bits for these samples is higher than those bits allocated for ± 1. For example, for sending the sample value of 7, 11 bits are required [11].

The second procedure in VLC uses this reality that the number of zero values in quantized DCT matrix is quite high, as shown in Figure 1, for instance. After arrangement of DCT matrix elements in zigzag form, this procedure searches the next element whether is zero. If this element is zero, the next element value is checked. In each stage, the number of zeros is counted until reaching to a non-zero element or the entire DCT matrix elements are ended. After determining number of repetition of zero elements between two non-zero elements, a set of binary code is given by established table for standard MPEG [11].

1	-7	-8	0	2	0	0	0
3	4	-1	-3	-3	-1	0	0
-5	1	4	3	2	-1	-1	0
-2	-1	1	0	1	2	1	0
-1	0	-1	-1	-1	-1	0	0
1	1	0	0	0	0	0	0
3	-1	1	0	-1	0	0	0
0	-2	1	1	-1	0	0	0

Figure1. DCT matrix coefficients quantized

**III. CHANNEL CODING METHOD OVERVIEW**

For channel coding, Reed-Solomon coding is used in this article which is a class of cyclic BCH code [12, 13], linear and non-binary and it is made by  $GF(q)$  field [14]. The reason for using this type of coding is non-binary characteristics. Therefore, the information of each block can be directly coded without converting to binary. Thus, speed of video frame coding increases. In this method, assuming that the encoded message,  $c(X)$ , is transmitted and the noisy message,  $r(X)$ , is received, then:

$$r(X) = r_0 + r_1X + \dots + r_{n-1}X^{n-1} \tag{3}$$

Also, the error polynomial,  $e(X)$ , and its relation with transmitted and received messages is given as:

$$e(X) = e_0 + e_1X + \dots + e_{n-1}X^{n-1} \tag{4}$$

$$r(X) = c(X) + e(X)$$

Thus, according to Equation (4), it can be shown that  $r(\alpha^i) = c(\alpha^i) + e(\alpha^i) = e(\alpha^i)$ .

In this type of coding, both error and value location error are calculated. Due to having variable of  $t$  and  $\beta_i = \alpha^{ji}$ , the location and error value are obtained using Equation (5). This type of coding is able to correct any pattern of errors with length of,  $t$ , or less than,  $t$ , [14].

Channel coding rate is given by:

$$R_c = \frac{k}{n} \tag{5}$$

In Equation (5),  $k$  is the number of coded bits related to frame information (main information),  $n$ , is length of stream which contains the frame information and added information by channel encoder. The channel encoder is able to detect and correct,  $\frac{n-k}{2}$ , errors. Therefore, more  $n-k$ , bits (the number of parity bits) results in more correction of message bits by channel encoder.

**IV. COMBINATION OF WAVELET PREDICTION, NEURAL NETWORK AND GENETIC ALGORITHM**

Wavelet transform and Neural Networks (WNN) are used for estimating a curve by non-linear functions instead  $f$  using common prediction function. This transform is focused in both time and frequency domains and this method is developed for nonlinear fitting problem [15, 16].

There are two major problems during the designing process of WNN are as follows: First, Structure of the WNN and second learning algorithm which can be used for training neural networks [17]. These problems are solved to determine the optimal WNN structure and wavelet function.

The output of the NN is obtained by sum of weighted basis functions. Normally basic functions are: Gaussian radial basis functions, B-spline basis functions, wavelet basis functions and some of neuro-fuzzy basis functions [18]. Genetic Algorithm, (GA), is a searching algorithm in computer science that is employed to find approximate solution for optimization and searching problems. GA is special kind of evolutionally algorithm which uses biology techniques such as: inheritance and Mutation [19].

GA is based of problem solution using optimized fitness function. In GA, searching process is randomly and using mutation and crossover to create new generation of chromosome according to previous generation is called parents. During creation of new chromosomes, the participate probability of chromosomes belonging to previous generation is become higher in which the fitness function can be more optimal compared to other chromosomes [20, 21]. Time-series prediction is an important research and application area. Sometime series models are classified into three main models:

Linear, non-linear and mixed linear and non-linear models, moving average (MA), autoregressive integrated moving average (ARIMA) are some important example of linear models and neural network (NN) and fuzzy system indicate some non-linear models [22]. Finally system is obtained by combination of linear and non-linear models [23]. In this paper, we use combination of wavelet prediction, neural network and GA for having better prediction from a specific signal. For description of prediction method, in following section wavelet transform and GA are briefly indicated and finally the used prediction in this article is described.

**A. Wavelet Transform Neural Network**

In terms of wavelet transformation theory, wavelets in the following equations:

$$\Psi = \left\{ \Psi_i = \left| a \right|^{-\frac{1}{2}} \psi \left( \frac{x-b_i}{a_i} \right); a_i, b_i \in R^n, i \in Z \right\} \tag{6}$$

$$X = (x_1, x_2, \dots, x_n)$$

$$a_i = (a_{i1}, a_{i2}, \dots, a_{in})$$

$$b_i = (b_{i1}, b_{i2}, \dots, b_{in})$$

As shown in Equation (6), wavelet basis transforms are a set of functions constructed by a single function  $\psi(x)$  using delay and shift operators focused in both time and frequency domains.  $\psi(x)$  is named a mother wavelet and the parameters  $a_i$  and  $b_i$  are the scale and translation parameters, respectively. The variable,  $x$ , is WNN input. The standard form of WNN is:

$$f(x) = \sum_{i=1}^M \omega_i \Psi_i(x) = \sum_{i=1}^M \omega_i \left| a_i \right|^{-\frac{1}{2}} \psi \left( \frac{x-b_i}{a_i} \right) \tag{7}$$

where  $\Psi_i$  represents the  $i$ th activation function in the hidden layer and  $\omega_i$  represents the  $i$ th weight connecting the hidden layer to the output layer.

For using the capacity of wavelet basis functions, a type of WNN has been proposed and is given by Equation (8). The structure of the proposed WNN is shown in Figure 2. Also, Equation (8) shows the outputs of the neural network using wavelet transform:

$$y = \sum_{i=1}^M (\omega_{i0} + \omega_{i1}x_1 + \dots + \omega_{in}x_n) \Psi_i(x) = \sum_{i=1}^M (\omega_{i0} + \omega_{i1}x_1 + \dots + \omega_{in}x_n) |a_i|^{-\frac{1}{2}} \psi\left(\frac{x-b_i}{a_i}\right) \quad (8)$$

$$x = [x_1, x_2, \dots, x_n]$$

According to Equation (8), it can be observed that the weights of neural network are a linear combination of several weights instead of single weight. Equation (9) represents the linear combination of several weights:

$$v_i = \omega_{i0} + \omega_{i1}x_1 + \dots + \omega_{in}x_n \quad (9)$$

In constructed neural network, the weights consist of  $v_i$  instead of  $\omega_i$  as shown in Figure 2.

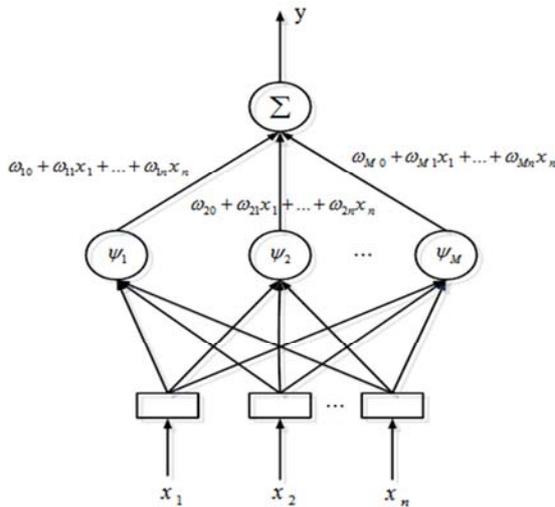


Figure 2. Neural network architecture used in this paper, using wavelet transform

**B. Combination of Neural Networks and GA**

GA is used in neural network to train the neural network. The motivations for introducing models using GA into a WNN are firstly, local linear models have been studied in some neuro-fuzzy systems and shown good performances [24, 25] and secondly, the use of searching algorithms for interpolation provides great savings, when modeling samples are sparse.

**C. The Proposed Algorithm Predictor**

In proposed predictor, basis function of neural network hidden layer neuron is given by:

$$\psi(x) = -x \cdot \exp\left(-\frac{x^2}{2}\right) \quad (10)$$

This function is a wavelet basis functions and shown in Figure 3. It should be noted that transfer function shown in Equation (10) is selected using Leave-One-Out method [26, 27] after testing on several signal obtained from several non-similar frames. The basis of the proposed predictor is as follow:

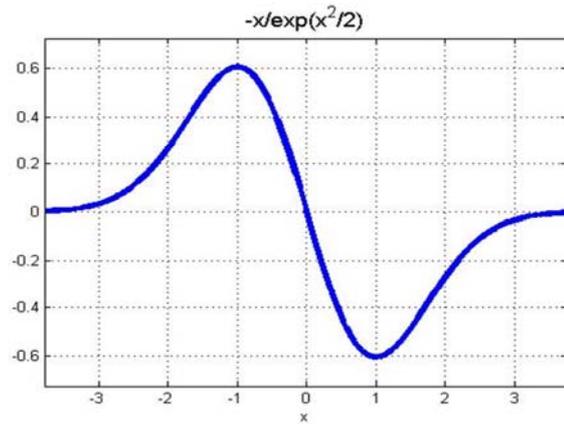


Figure 3. Wavelet basis function defined in Equation (10)

- Step 1: After applying input and output to the neural network, weights are optimized using the GA algorithm so that the error between the output data and the target data is minimized.
- Step 2: If the number of iterations of the algorithm is less than 100 times and error rate is not less than  $10^{-8}$ , then operations will be transferred back to step 1. Otherwise, step 3 is performed.
- Step 3: If the number of iterations of the algorithm is equal to 100 times, weights are selected for the network to provide the least error in 100 iterations of the algorithm. Then, in this stage, the network weights are determined. Also, error rate is calculated using Equation (11):

$$MSE = \frac{1}{Q} \sum_{k=1}^Q (y_1^i - y_2^i)^2 \quad (11)$$

In Equation (11),  $y_1^i$  and  $y_2^i$  represent the output and the target values, respectively.

**V. PROPOSED ALGORITHM**

The motion compensation block in MPEG-4 standard tries to remove the correlation between the values of the two consecutive frames pixels, but the correlation cannot be removed, completely. Because when the correlation is removed, the characteristics of the obtained signal should be as same as pseudo noise whereas the output signal of the motion compensation block still contains correlation as shown in Figure 4.

Thus, the secondary predictor is used to remove residual correlation After difference between two consecutive frames from each other using motion compensation predictor, the generated signal has higher correlated compared to signal generated by secondary predictor. This means that the output signal of the secondary predictor has low variance compared to the output signal of the motion compensation predictor.

Therefore, by using secondary predictor presented in section IV and subtracting the estimated values from the output signal of motion compensation predictor, the correlation significantly reduces. Secondary predictor used in this paper is detailed as follows:

Suppose  $X = [x_1, x_2, \dots, x_n]$  as input and  $Y = [y_1, y_2, \dots, y_n]$  as output. In this paper, the inputs of the prediction model are selected as  $x_{m-4}, y_{m-1}$  and the output is  $y_m$ . If the neural network consists of 'm' nodes in hidden layer, n, input and, d, output, then the total number of parameters for representation of NN is  $(n + 1) \times m \times d$ . In the proposed algorithm, the number of inputs are 2 and the number of output is 1, therefore the number of neural networks weights are  $3 \times m$ . Also, in this paper, the number of neurons in the hidden layer neuron is assumed to be equal to 20. Thus, the number of weights during training stage is 60.

After obtaining the difference between two consecutive frames, the secondary predictor is applied on differential signal and the estimated values are subtracted from the differential signal. Since the obtained signal has lower variance, fewer numbers of bits are required for signal transmission. Thus, the data frames are encoded with lower number of bits. Figure 4 shows the values of the autocorrelation function for two signals estimated by using secondary predictor and motion compensation predictor. It is observed that the autocorrelation function values of the output signal using secondary predictor has lower values compared to those values using motion compensation predictor. Therefore, the characteristics of the output signal using the secondary predictor is more likely random noise. Also, Figure 5 typically shows the output signal obtained from the motion compensation predictor (up signal) and from the secondary predictor (down signal). It is observed that output signal using secondary predictor has lower variance.

In the next stage, the local channel encoder, re-encodes bit sequences. The rate of this encoder is derived from the following equation:

$$R'_c = \frac{B_{T1} - B'_{T2}}{B_{T1}} \tag{12}$$

where  $B_{T1}$  and  $B'_{T2}$  represent the number of bits produced by VLC block for encoded signal using motion compensation predictor and secondary predictor, respectively. Therefore, channel coding rate increases and the frame information are more robust against noise and channel errors.

**A. The Proposed Algorithm Encoder**

The flowchart of proposed encoder algorithm is shown in Figure 6. The proposed algorithm is detailed as follows:

- Step 1: After passing the signal through motion compensation predictor block and subtraction of the estimated values from the original signal, the signal information are quantized. The quantized signal is passed through VLC block, placed in a buffer until the data representing a frame is completely entered into buffer. Then, the number of bits,  $B_{T1}$ , is calculated.
- Step 2: The obtained signal by subtraction of the estimated signal using motion compensation predictor from original signal is dequantized and after performing IDCT block on frame data, all frame data is buffered. In this stage, similar to step 1, this information is stored in buffer until the data related to a frame is completely entered into buffer.

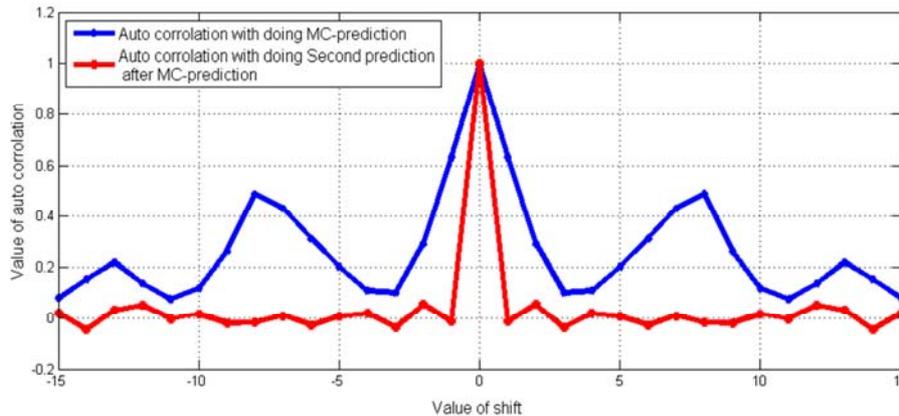


Figure 4. Autocorrelation function for two estimated signals using motion compensation predictor and secondary predictor

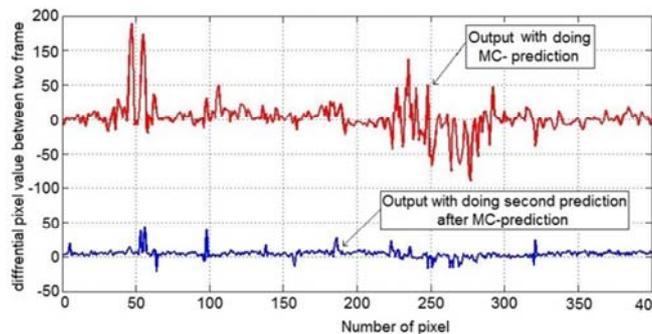


Figure 5. Output signal obtained from motion compensation predictor (up signal) and output signal obtained from secondary predictor (down signal)

Then, secondary predictor (described in section IV) estimates the information related to a frame and estimated values are subtracted from signal stored in the buffer.

- Step 3: The obtained signal after the secondary predictor is divided into blocks of  $8 \times 8$  and neural network weights (60 weights) are coded as a separate block.
- Step 4: The  $8 \times 8$  blocks after passing through DCT block and quantization are entered into VLC block. Similar to Step 1, the data is stored into buffer. Then, the number of bits,  $B'_{T2}$ , is calculated.
- Step 5: In this stage, the encoding rate used for the secondary channel encoder is calculated using Equation (12). Thus, the  $8 \times 8$  blocks constructed in step 4 are encoded using local channel encoder with coding rate

equal to  $R'_c$ . After reordering and packetization, the frame information is encoded again using the global channel encoder which is a part of the MPEG-4 encoder standard and after passing through the VLC block it is transmitted to the receiver. Also  $R'_c$  with neural network weights obtained in step 3 are encoded and transmitted.

It should be noted that although the proposed method has 3 buffers and it seems that there is a delay between any two consecutive frames. The proposed algorithm operates parallel and buffering operations occur simultaneously. Therefore, the system has only one frame delay and there is not delay between two consecutive frames.

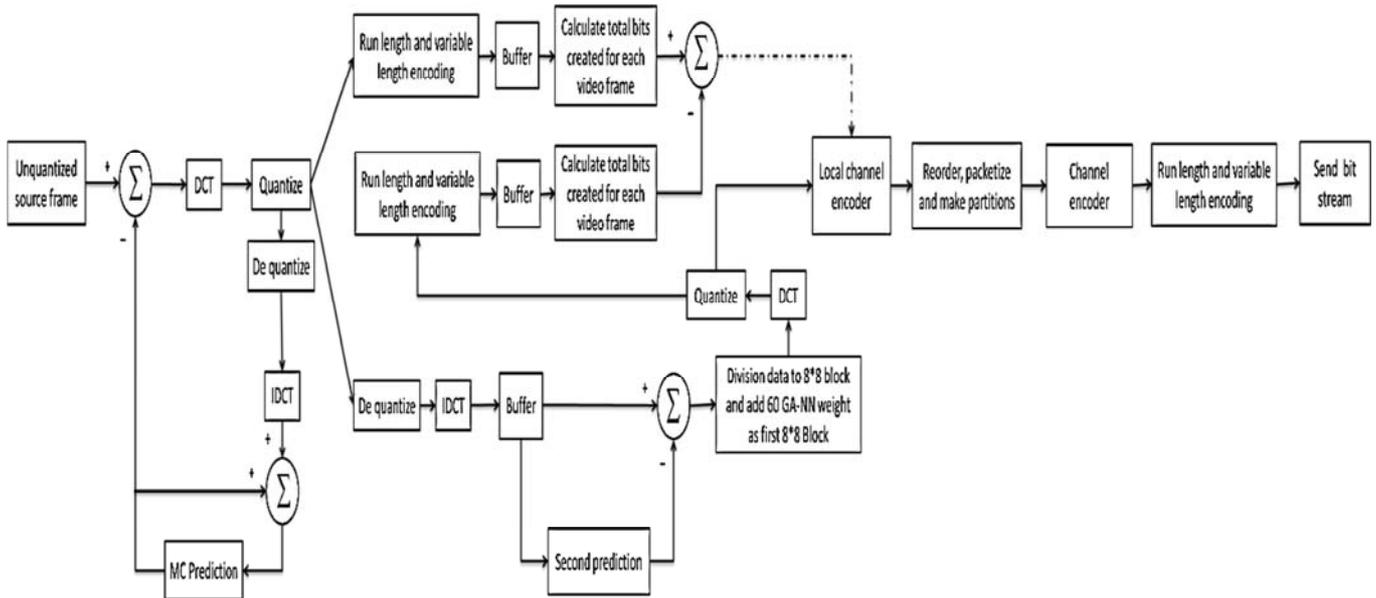


Figure 6. Proposed method encoder flowchart

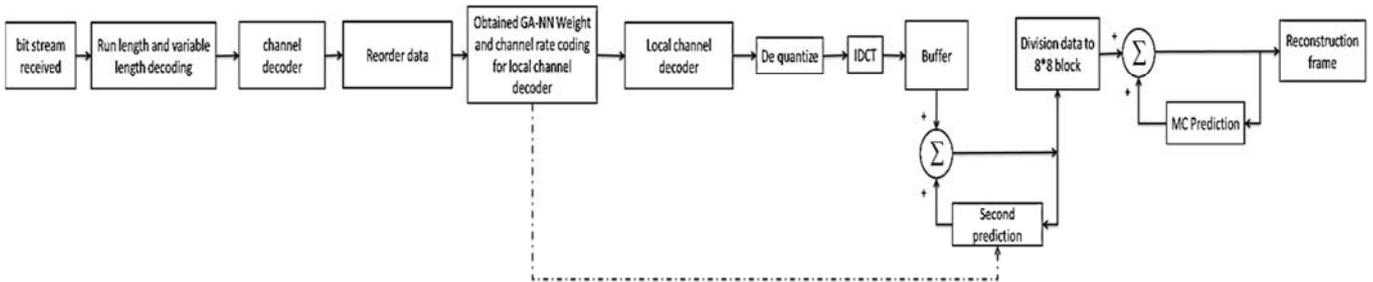


Figure 7. Proposed method decoder flowchart

### B. The Proposed Algorithm Decoder

The flowchart of proposed decoder is shown in Figure 7. The procedure of the proposed algorithm is detailed as follows:

- Step 1: After receiving the information of bit stream and cross through global channel encoder, secondary channel encoding rate and neural network weights are calculated from the encoded bit stream. Then, secondary channel decoder decodes the information using calculated rate.

- Step 2: Received information is placed in a buffer, and buffering is performed until the data related to a frame is completely entered into the buffer. Secondary predictor estimates data and then obtained data is added with the data stored in the buffer.
- Step 3: At this stage, the data data stored in the buffer is divided into  $8 \times 8$  blocks and entered to motion compensation predictor block.
- Step 4: In motion compensation predictor block, after motion compensation, the frame is reconstructed.

**VI. EVALUATION OF THE PROPOSED METHOD**

In order to evaluate the proposed method, we firstly introduce several parameters used in our simulation and then we compare the experimental results of the proposed method with the results of a new method indicated by Farooq in [7]. In this article, the channel is Binary Symmetric Channel (BSC), the occurred error is Additive White Gaussian Noise (AWGN) and fading has Rayleigh PDF. We have used two video files named Foreman and Walk. The number of Foreman video frames is 123 in which 3 frames are coded as I-picture and the rest frames are divided into three parts.

Each part has 40 frames which are coded and transmitted as P-pictures. It should be noted that 3 frames coded by I-picture, at the beginning of each part, are transmitted. The numbers of Walk video frames are 105 in which 5 frames are coded as I-picture, and the rest frames are divided into five parts. Each part contains 20 frames which are coded as P-picture. It should be noted that 5 frames coded by I-picture, at the beginning of each part are transmitted. Frame size is equal to 288x352. Also, frame transmission rate is 25 frames per second. This simulation has been performed for three source transmission rates. Since the number of transmission frames is constant for a fixed source coding rate, higher transmission rate corresponds to higher channel coding rate and therefore results in more robustness against noise for video frames. Channel encoder rate,  $R_c$ , also the secondary channel encoder rate,  $R'_c$ , for two video files with different rates are shown in Table 1.

As shown in Table 1, due to reduction of transmitted data variance, the secondary channel coding rate increases. This means that the number of bits used in the secondary channel encoder increases. Therefore, the transmitted frames are significantly more robust against channel errors.

After simulation of the proposed method, the obtained results have been compared with the results of Farooq method in [7]. Figures 8, 9, 10, show the obtained PSNR by applying the proposed method in comparison with Farooq method for all 123 frames of Foreman. The only difference in Figures 8, 9, 10 is source coding rate. This means that if source coding rate increases then the transmission of the frames is robust against channel noise and distortion.

Table 1. Channel encoder rate  $R_c$  and also  $R'_c$  secondary channel encoder rate for two video sequences Foreman and Walk with difference transmission rates

Source transmission rate	Type of video sequence which transmit	$R_c$	$R'_c$
384 Kbps	Foreman	0.466	0.781
768 Kbps	Foreman	0.2	0.656
2 Mbps	Foreman	0.0645	0.531
256 Kbps	Walk	0.6	0.834
512 Kbps	Walk	0.33	0.718
1.5 Mbps	Walk	0.129	0.539

Also, Figures 11, 12, 13, show the obtained PSNR by applying the proposed method in comparison with Farooq method for all 105 frames of Walk. As observed, for low BERs, for instance  $BER = 10^{-8}$ , the difference of resulting PSNRs between the proposed method and Farooq method [7] is negligible. This is due to having low noise channel. However, when BER increases or in other words, for higher noisy channel, the resulting PSNRs of the proposed method considerably increases. Therefore, the proposed method through embedding channel encoder bits in transmitted frames is able to increase channel encoding rate without increasing the transmission bit rate. Therefore, the quality of the reconstructed frames has considerably increased.

**VII. CONCLUSIONS**

In this paper, we proposed a novel method in which the received video frames contain higher PSNR in comparison with the new method indicated by Farooq. Secondary predictor used in the proposed method causes that the correlation of output signal obtained motion compensation predictor and consequently the signal variance reduces, significantly. Therefore, the number of required bits for transmission of frame information reduces. This means that in constant bit rates, the remaining bits are accessible and could be used by local channel encoder. In the proposed method, frame information obtained from secondary predictor is encoded using local channel encoder. Thus, the quality of the reconstructed video frames increases in comparison with the new methods. The proposed method by increasing the channel coding rate and therefore more protection of transmitted bits and maintaining overall transmission rate constant, simultaneously could improve the quality of the reconstructed video frames.

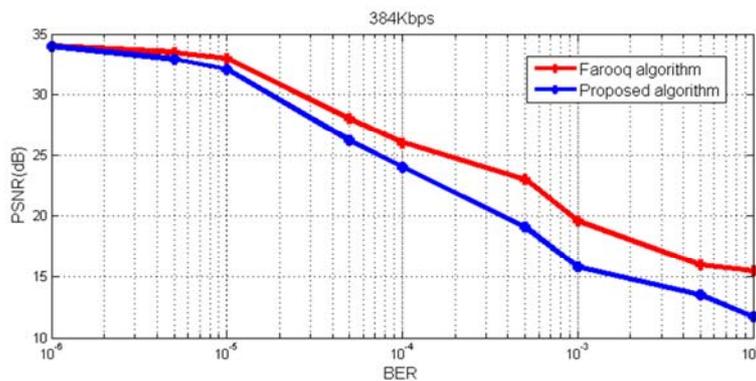


Figure 8. Comparison of PSNR for 123 frames of Foreman between proposed method and Farooq algorithm [7], with source coding rate = 384 Kbps

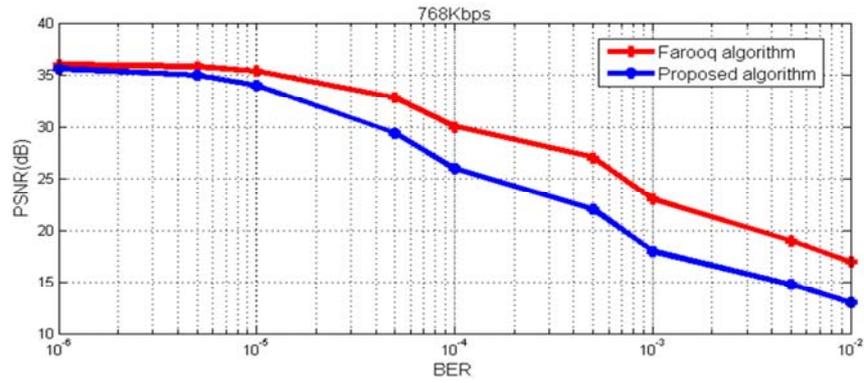


Figure 9. Comparison of PSNR for 123 frames of Foreman between proposed method and Farooq algorithm [7], with source coding rate = 768 Kbps

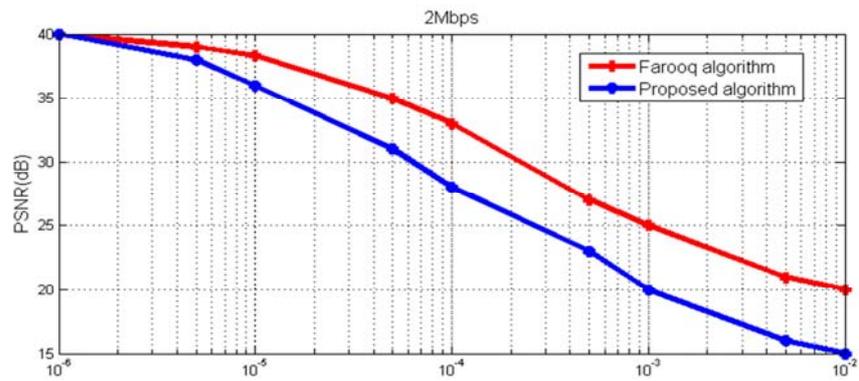


Figure 10. Comparison of PSNR for 123 frames of Foreman between proposed method and Farooq algorithm [7], with source coding rate = 2 Mbps

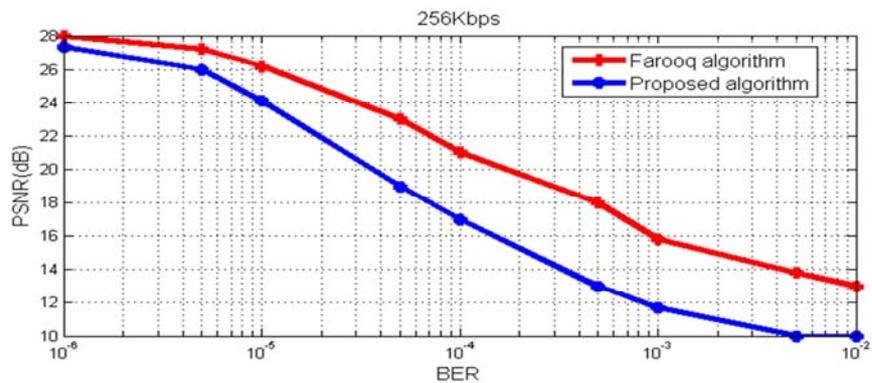


Figure 11. Comparison of PSNR for 105 frames of Walk between the proposed method and Farooq algorithm [7], with source coding rate = 256 Kbps

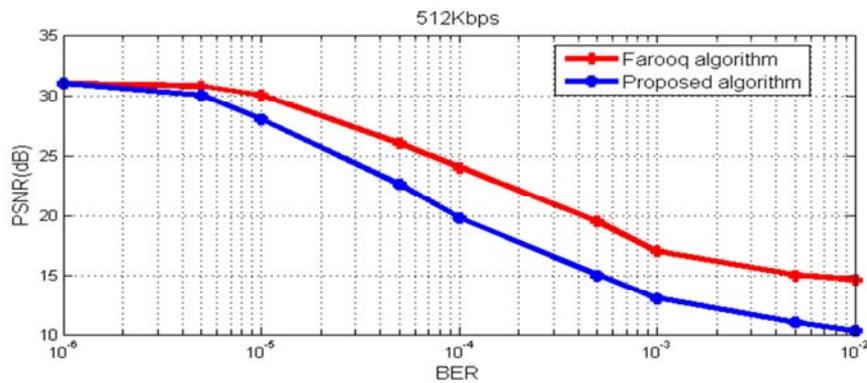


Figure 12. Comparison of PSNR for 105 frames of Walk between the proposed method and Farooq algorithm [7], with source coding rate = 512 Kbps

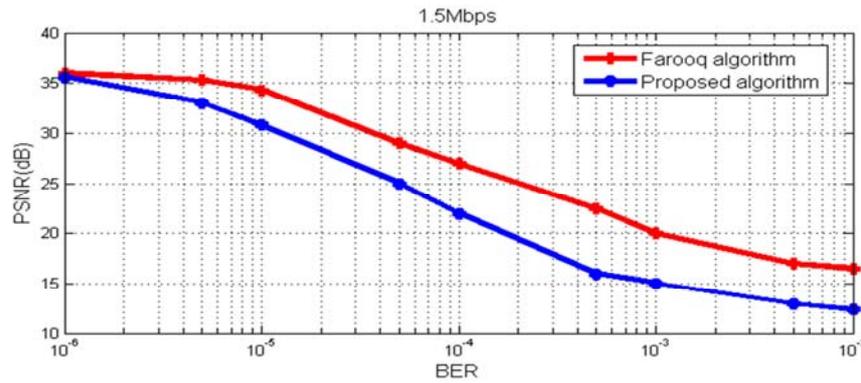


Figure 13. Comparison of PSNR for 105 frames of Walk between the proposed method and Farooq algorithm [7], with source coding rate = 1.5 Mbps

**REFERENCES**

[1] M. Flierl, B. Girod, "Video Coding with Superimposed Motion-Compensated Signals", Kluwer Academic, 2004.

[2] J.C. Moreira, P.G. Farrell, "Essentials of Error Control Coding", Wiley Ltd, USA, 2006.

[3] Q. Zhang, "Joint Source-Channel Sate-Distortion Optimization Algorithm for Multiview Video Coding", 12th IEEE Computer and Information Technology Conference, pp. 101-108, Sichuan, China, Oct. 2012.

[4] H. Kim, P.C. Cosman, L.B. Milstein, "Motion Compensation Scalable Video Transmission over MIMO Wireless Channels", IEEE Trans. on Circuits and Systems for Video Technology, Vol. 23, No. 1, pp. 116-127, 2013.

[5] H. Chen, E. Steinbach, W.C. Chang, "Joint Source-Channel Rate Control for Pixel-Domain Distributed Video Coding", IEEE Acoustics, Speech and Signal Processing Conference, pp. 732-736, Prague, Czech Republic, May 2011.

[6] J. Meng, Y. Zhang, Z. Zhao, W. Shao, "Power Rate Distortion Analysis for H.264 Video Coding Over Wireless Channel", IEEE Intelligent computing and intelligent systems Conference, pp. 538-541, Shanghai, China, Nov. 2009.

[7] M. Farooq, W. Robert, J. Heath, C.B. Alan, "Joint Source-Channel Distortion Modeling for MPEG-4 Video", IEEE Trans. on Image Processing, Vol. 18, No. 1, pp. 90-105, 2009.

[8] D. Solomon, "Data Compression", Third Edition, Springer, 2004.

[9] I.E.G. Richardson, "H.264 and MPEG-4 Video Compression, Video Coding for Next Generation Multimedia", Wiley, USA, 2003.

[10] Y.Q. Shi, H. Sun, "Image and Video Compression for Multimedia Engineering, Fundamentals, Algorithms, and Standards", CRC Press, 2000.

[11] H. Zhihua, M. Bystrom, "Improve Conversion From DCT Blocks to Integer Cosine Transform Blocks in H.364/AVC", IEEE Trans. on Circuit and Systems for video technology, Vol. 18, No. 6, pp. 851-857, 2008.

[12] A. Katariya, A. Yadav, N. Jain, G. Tomar, "BER Comparison of RS and BCH Coded FDM Transmission Over Noisy Channel", IEEE Computational Intelligence and Communication Systems Conference, Gwalior, India, pp. 227-232, Oct. 2011.

[13] B. Carlson, "Communication Systems: An Introduction to Signals and Noise in Electrical Communication", 3rd Edition, McGraw-Hill, New York, 1986.

[14] D. Declercq, "Non-Binary Decoder Diversity for Dense or Locally-Dense Parity-Check Codes", IEEE Trans. on Communications, Vol. 59, No. 3, pp. 729-739, 2011.

[15] A. Ghadamyari, A.A. Safavi, "Self Organization Map (SOM) Neural Network Based on Novel Fuzzy Wavelet for Nonlinear Function Approximation", The 2nd Control, Instrumentation, and Automation Conference, Shiraz, Iran, pp. 699-704, Dec. 2011.

[16] X.X. Jian, T. Ying, "Nonlinear Adaptive Wavelet Control Using Constructive Wavelet Networks", IEEE Trans. on Neural Networks, Vol. 18, No. 1, pp. 115-127, 2007.

[17] L. Ying, S. Mengqing, "Wavelet Based Evolving Neural Network Forecasting Algorithm and its Application", IEEE Computer Application and System Modeling Conference, Taiyuan, China, pp. 350-353, May 2010.

[18] J.L. Cheng, H.C. Cheng, T.L. Chin, "Efficient Self Evolving Evolutionary Learning for Neurofuzzy Inference System", IEEE Trans. on Fuzzy System, Vol. 16, pp. 1476-1490, 2008.

[19] J.F. Douglas, "Evolution", Sinauer Associates, Sunderland, 2005.

[20] D.E. Goldberg, "Genetic Algorithms in Search, Optimization and Machine Learning", Addison-Wesley, New York, 1989.

[21] L. Davis, "Handbook of Genetic Algorithms", Van Nostrand Reinhold, New York, 1991.

[22] Z. Jin, Z. Huajun, "The Application of Fuzzy Neural Network in Ship Course Control System", The 7th Fuzzy Systems and Knowledge Discovery Conference, Yantai, China, pp. 752-756, Aug. 2010.

[23] N. Qazi, H. Hoi, "Estimation of Weights to Combine Trained Neural Networks Using Linear Estimation Techniques", The 14th Multitopic Conference, SSUST Campus, Karachi, pp. 1-7, Dec. 2011.

[24] M.A. Balafar, "Review of Noise Reducing Algorithms for Brain MRI Images", International Journal on Technical and Physical Problems of Engineering (IJTPE), Issue 13, Vol. 4, No. 4, pp. 54-59, December 2012.

[25] M.A. Mazinan, N. Sadati, N.H. Ahmadi, "A Case Study for Fuzzy Adaptive Multiple Models Predictive Control Strategy", IEEE Industrial Electronics Symposium, Seoul, Korea, pp. 1172-1177, July 2009.

[26] S. Zollanvari, N.U.M. Braga, E.R. Dougherty, "Joint Sampling Distribution between Actual and Estimated Classification Errors for Linear Discriminant Analysis", IEEE Trans. on Information Theory, Vol. 56, No. 2, pp. 784-804, 2010.

[27] H. Shayeghi, H.A. Shayanfar, "PSO Based Neuro-Fuzzy Controller for LFC Design Including Communication on Time Delays", International Journal on Technical and Physical Problems of Engineering (IJTPE), Vol. 2, No. 2, pp. 28-36, June 2010.

### **BIOGRAPHIES**



**Hassan Farsi** received the B.Sc. and M.Sc. degrees from Sharif University of Technology, Tehran, Iran, in 1992 and 1995, respectively. In 1995, he was employed in University of Birjand, Birjand, Iran as a lecturer in the Department of Electronics and Communications Engineering. Since

2000, he started his Ph.D. in the Centre of Communications Systems Research (CCSR), University of Surrey, Guildford, UK, in the field of speech quality improvement provided by low bit rate speech coders and received the Ph.D. degree in 2004. Currently, he is a member of editorial team in IJETSP, Majlesi Journal of Multimedia Processing and IJCIT journals. Now, he works as an Associate Professor in University of Birjand, Birjand, Iran.



**Pouriya Etehadifar** was born in Neyshabour, Iran, in 1989. He received B.Sc. degree from University of Birjand, Birjand, Iran in Communication Engineering in 2011. He works in video, image processing and intelligent systems. Currently, he is a M.Sc. student in Communication Engineering, Department of Electrical and Computer Engineering, University of Birjand, Birjand, Iran.