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Modified Soft Decoding of Huffman Codes and Iterative JSC Decoding With Channel Equalizer Design Based On Weiner Filter

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Abstract - Most source coding standards (voice, audio, image and video) use Variable-Length Codes (VLCs) for compression. However, the VLC decoder is very sensitive to transmission errors in the compressed bit-stream. In this paper, we propose a soft-input VLC decoding method using an a priori knowledge of the lengths of the source symbol sequence and the compressed bit-stream with Maximum A Posteriori (MAP) sequence estimation. Performance in the case of transmission over an Additive White Gaussian Noise (AWGN) channel is evaluated. A new VLC decoding method generating additional information regarding the reliability of the bits of the compressed bit-stream is also proposed. We consider the serial concatenation of a VLC with LCPC channel code and perform iterative decoding. To improve the S/N ratio by reducing the inter symbol interference arise from multipath propagation a channel equalizer design based on Weiner filter and LMS algorithm is added at the decoder end.

***Key Words*:** Soft Decoding, Channel Equalization, LDPC Decoding

1.INTRODUCTION

Next generation communication systems are faced with the competing goals of providing high quality of service for multimedia applications while minimizing power and bandwidth consumption. It is clear that both source coding and channel coding are important components to achieve these goals. Shannon's separation theorem states that these two components can be optimized separately if unconstrained block lengths and unconstrained coding and decoding complexities are allowed. In most practical applications, it is impossible to fulfill these requirements. Joint Source/Channel (JSC) coding and decoding techniques have emerged as a pragmatic approach leading to a convergence of the multimedia and digital communication worlds.

Variable-Length Codes (VLCs) are widely used in state of-the-art image, video and audio compression schemes. Although they provide a reduction in the data rate by exploiting the redundancy in the source-symbol sequence, the VLC encoded data are very sensitive to transmission errors (error propagation phenomenon). Especially when the compressed bit-stream has to be transmitted over a noisy channel, efficient coding or decoding techniques have to be considered to guarantee a good source-symbol sequence reconstruction.

The first contributions in robust entropy coding techniques, add some redundancy in the entropy coding stage to help detect the erroneous bits or prevent error propagation. But a single bit error in the VLC-encoded stream can lead to the loss of synchronization. To overcome this problem a class of VLCs with good synchronization properties where variable length bit-streams from individual blocks are distributed into slots of equal size to improve their synchronization. Recent work exploits the VLC properties to improve its decoding efficiency. These techniques consider trellises that are built using the structure of the VLC codebook to apply well-known algorithms used for channel decoding such as Viterbi and List Viterbi. Other contributions assume that a priori information such as the number of source symbols or the size of the encoded bit-stream is available at the decoder to improve the decoding performance.

Later contributions proposed to exploit the efficiency of the turbo decoding by integrating the VLCs in an iterative decoding process .While Bauer and Hagenauer proposed new VLC trellis representations for Soft-Input Soft- Output (SISO) VLC decoding, Guyader et al introduced a Bayesian network-based decoding approach. Both decoding schemes considered a VLC concatenated with a convolutional code, and substantial gains were obtained with iterative decoding. Iterative decoding of the serial concatenation of a VLC with a turbo code and a Low-Density Parity-Check (LDPC) code were used.

In this paper, the Maximum A Posteriori (MAP) sequence estimation criterion is employed for the maximum likelihood decoding of the VLCs in the case of transmission over a noisy channel. We suppose that the source-symbol sequence length L and the corresponding compressed bit-stream size l are known at the decoder side. The VLC decoding is based on a search of the MAP sequence among the length-valid sequences (bit-streams of length l decoding L symbols). The original contribution of our work lies in the adaptation of the Chase decoder for soft input VLC decoding. In fact, we proposed a Chase-like VLC decoder with reduced complexity and taking into account the bit-streams' length validity. As a second contribution, we propose an iterative JSC decoding scheme based on a VLC concatenated with a channel code. The VLC is used to perform source compression and the channel code performs error correction. After the soft-input soft-output channel decoding, the VLC Chase-like decoder is modified to identify reliable and non-reliable compressed bits and this extra information is provided to the channel decoder at the next iteration. We consider two types of channel codes: a recursive systematic convolutional code (RSCC) and a capacity-approaching low-density parity-check code (LDPC). The source/channel decoding is iterated several times to improve the error-rate performance.

Many communication system suffer from the problem of inter symbol interference which may arise from the common phenomenon of multipath propagation, thus to achieve reliable communication in these situation, channel equalization is necessary.

2. EXISTING SYSTEM

In this system a Maximum A Posteriori (MAP) sequence estimation criterion is employed for the maximum likelihood decoding of the VLCs in the case of transmission over a noisy channel. We suppose that the source-symbol sequence length L and the corresponding compressed bit-stream size l are known at the decoder side. The VLC decoding is based on a search of the MAP sequence among the length-valid sequences (bit-streams of length l decoding L symbols). The VLC is used to perform source compression and the channel code performs error correction. The fig 1 below shows the existing system.

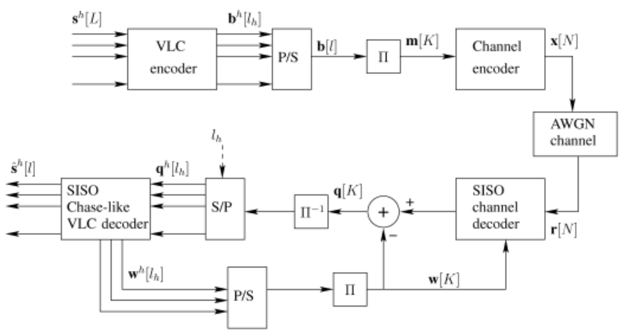

Fig. 1. General block diagram of the proposed concatenated transmission scheme with iterative JSC decoding.

A memoryless source is considered and generates sequences sh = (s1h, s2h, . . . , slh) of L source symbols. A standard VLC encoder encodes the source symbol sequences. The resulting compressed bit streams bh = (b1h, b2h, . . . , bihh) are concatenated to form the binary sequence b = (b1, b2, . . . , bl). The number of concatenated compressed bit-streams will be denoted H. Sequence b is then permuted by a random interleaver Π and, if required, some dummy bits are introduced to obtain the binary sequence m of length K. The latter is encoded by the channel encoder into a codeword x = (x1, x2, . . . , xN), and transmitted through an AWGN channel by means of the BPSK modulation.

At the receiver after demodulation soft-input soft output decoding is done. After the soft-input soft-output channel decoding, the VLC Chase-like decoder is modified to identify reliable and non-reliable compressed bits and this extra information is provided to the channel decoder at the next iteration. Here LDPC decoder based on message-passing iterative decoding algorithms is used as the channel decoder. An LDPC code of length N and rate RLDPC can be represented using a Tanner graph with N variable nodes vn representing the N bits. Each variable node is connected to some of the M parity-check nodes cm. Let Fm,n denote the message from a check node cm to a variable node vn. Similarly, let Tn,m denote the message from a variable node vn to a check node cm. The bipartite graph of the JSC scheme and the messages involved in the iterative decoding are depicted in Fig. 2. The upper part of the figure is the Tanner graph representing the LDPC code. The nodes VLCDh in the lower part denote the VLC Chase-like decoding stages with the proposed compressed bits reliability definition. The iterative JSC decoding starts with the initialization to zero of the a priori information delivered by the VLC decoder (wn = 0 for 1 ≤ n ≤ N). The check to variable messages Fm,n, 1 ≤ n ≤ N, are also initialized to 0. Then, the iterative decoding proceeds as follows:

1) LDPC decoding step: apply ItLDPC iterations of the Belief Propagation (BP) algorithm:

a) BP horizontal step: each variable node vn computes the output messages Tn,m according to:

Tn,m = In + Fn − Fm,n + wn, (1)

where Fn = £m_∈M(n) Fm_,n is the overall information associated with each variable node vn, M(n) is the set of checks connected to vn, In =2/σ2.rn is the intrinsic information delivered by the channel output, and wn is the correction term given by the VLC Chase-like decoder.

b) BP vertical step: every check node cm has to compute the output messages Fm,n using the input messages Tn,m..

c) Horizontal and vertical decoding steps are iterated until a valid LDPC codeword is obtained or the maximum number of iterations ItLDPC is reached. Then, messages qn, 1 ≤ n ≤ N, containing the overall information about the bit bn are updated:

qn=Fn+In (2)

2) Chase-like VLC decoding step:

a) The H sub-streams qh of respective lengths lh, extracted from q, are Chase-like decoded one-by one and the extrinsic information wh is updated as described above.

b) Messages wh are concatenated and the resulting sequence w is then fed back to the LDPC variable nodes for further iterations.

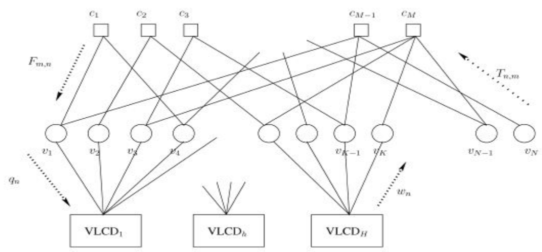

Fig 2: Bipartite graph with message exchange for the proposed JSC iterative decoding

T The algorithm is repeated as from step 1 until a stopping criterion is reached. In the system we are dealing with, both the LDPC and the VLC decoders are able to detect a valid sequence. In fact, the LDPC decoder can detect a valid codeword based on its syndrome. The VLC decoder can detect an erroneous bit-stream if the decoded source symbol sequence is not length-valid. We propose to stop the decoding process if both detections are verified: we have a valid LDPC codeword, and the corresponding H bit-streams are VLC length-valid.

Here Chase-like decoder is used as the VLC decoder. For soft input decoding, we have to specify two constraints: the estimator and the search space. When the described MAP sequence estimator is considered, a length-constrained decoder selects the compressed bit-stream having the best metric Mk (given by (4)) among all length valid sequences in the search space. However, evaluating all these possible bit-streams is infeasible in practice and numerous solutions proposed suboptimal decoding algorithms based on a trellis description of the code. In this section, we propose a Chase-type algorithm to achieve a low-complexity suboptimal soft-input decoding for the VLCs. The Chase algorithm was originally proposed for soft-input block-code decoding. The main idea of this algorithm is to limit the search space to only the Q most probable sequences, based on soft inputs, to reduce the complexity. A classical decoding scheme consists in applying a VLC prefix decoding to binary sequence y = (y1, . . . , yl) obtained by a hard detection on the received values r. The soft input decoder uses the extra information given by r called reliabilities. The reliability of a component yj is defined using the Log Likelihood Ratio (LLR) of the transmitted bit bj.

For a stationary channel, the LLR can be normalized and the reliability of yj is given by λj = |rj |. Then, the Chase like decoding search space is based on binary sequences with different bit combinations at the q least reliable positions. The proposed length-constrained Chase-like decoding of the VLCs is as follows:

1) Evaluate the hard decision vector y and the corresponding reliability vector Λ(y) = (λ1, . . ., λl).

2) determine the positions of the q least reliable binary elements of y based on Λ(y).

3) Form test patterns ti, 0 < i ≤ Q with Q = 2q: all the l-elements binary vectors with elements tji , 0 < j ≤ l having a weight less than or equal to q with "1"s in only the least reliable positions and "0"s in the other positions (all zero test pattern represents the case where no transmission errors have occurred).

4) Form test sequences zi, 0 < i ≤ Q with zij=yj ϴtji for 0 < j ≤ l.

5) Decode all the test sequences zi using the standard VLC

decoding. If a sequence zi decodes exactly L symbols, we compute its metric using (4) and append it to the subset of the competitor valid code sequences Ψ.

6) Finally, the decoded bit-stream corresponds to the sequence having the best metric Mk in the subset Ψ.

The source/channel decoding is iterated several times to improve the error-rate performance. We propose a soft-input VLC decoding method using an a priori knowledge of the lengths of the source symbol sequence and the compressed bit-stream with Maximum A Posteriori (MAP) sequence estimation. Performance in the case of transmission over an Additive White Gaussian Noise (AWGN) channel is evaluated. Simulation results show that the proposed decoding algorithm leads to significant performance gain in comparison with the prefix VLC decoding besides exhibiting very low complexity. A new VLC decoding method generating additional information regarding the reliability of the bits of the compressed bit-stream is also proposed. we consider a concatenation with a low-density parity-check (LDPC) code and it is shown that iterative joint source/channel decoding outperforms tandem decoding and an additional coding gain of 0.25 dB is achieved.

3. PROPOSED SYSTEM

In the proposed system channel equalizer based on Weiner filter is added at the decoder end. The design of a Wiener filter requires a priori information about the statistics of the data to be processed. The filter is optimum only when the statistical characteristics of the input data match the a priori information on which the design of the filter is based. In the proposed method, first the optimal channel weight vector of wiener filter is calculated. The basic concept behind wiener filter is to minimize the difference between filter output and some desired output. This minimization is based on the least mean square error approach which adjusts the filter coefficient to reduce the square of the difference between desired and actual waveform after filtering. Then these weight vectors will be updated by multiplicative neural network using a bisigmoidal activation function so as to get output signals approximately equal to the desired signal.

The existing system is modified by adding an equalizer based on Weiner filter at the decoder end. The fig 4 shows the modified system.

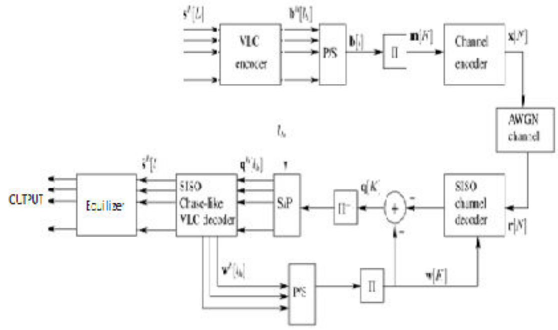

Fig 3 Block Diagram

Here the source symbol sh{L} is variable length coded by using Huffman coding to form compressed bit streams bh = (b1h, b2h, . . . , bihh) are concatenated to form the binary sequence b = (b1, b2, . . . , bl). The number of concatenated compressed bit-streams will be denoted H. Sequence b is then permuted by a random interleaver Π and, if required, some dummy bits are introduced to obtain the binary sequence m of length K. The latter is encoded by the channel encoder into a codeword x = (x1, x2, . . . , xN), and transmitted through an AWGN channel by means of the BPSK modulation.

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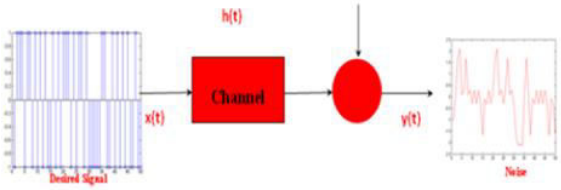
a) Channel Equalization

In communication system transmitter sends information over an RF channel which distorts the transmitting signal before it reaches the receiver. Equalization is the process of recovering the data sequence from the corrupted channel samples. The fig 5 shows the communication system. Equalization reduces Inter Symbol Interference (ISI) as much as possible to maximize the probability of correct decisions. Channel distortion arises in many communication systems and the distortion increases as the data rate compression in time or in space is increased within a fixed bandwidth channel.



$$\underbrace{y(t) = x(t)h_0(t)}_{\text{Desired Signal}} + \underbrace{\sum_{k=1}^{m} x(t-k)h_k(t)}_{\text{ISI}} + \underbrace{n(t)}_{\text{Noise}} \quad (3)$$

b) Channel Equalization Based On Weiner Filter

Wiener filter is the basis of adaptive filter theory. It is the optimal filter that most adaptive filtering algorithms attempt to achieve. The generalization to the complex case is straightforward. Consider the situation in fig. 6 where d(n) is the desired signal and x(n) is the input signal. The input x(n) is processed by a filter so that the output is y(n). The optimal channel weight vector of wiener filter is calculated by the estimation of correlation and cross correlation of input and desired noiseless signal. The basic concept behind wiener filter is to minimize the difference between filter output and some desired output. This minimization is based on the least mean square approach which adjusts the filter coefficient to reduce the square of the difference between desired and actual waveform after filtering. Then these weight vectors will be updated by multiplicative neural network using a bisigmoidal activation function.

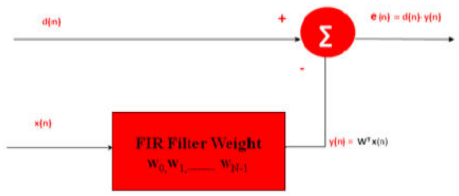


Fig 4 Weiner

4. SIMULATION RESULTS

a) Simulation result of transmitter output.

The transmitter input consists of clk, reset, ready, dc_in which denotes the DC coefficient of quantized image , rl_in which denotes the AC coefficient of quantized image, Scan_type denotes the type of scan whether left to rigt or top to bottom, luma for '1' indicate luminance and '0' indicate chrominance, eob denotes end of block. The output of the transmitter is encoded.

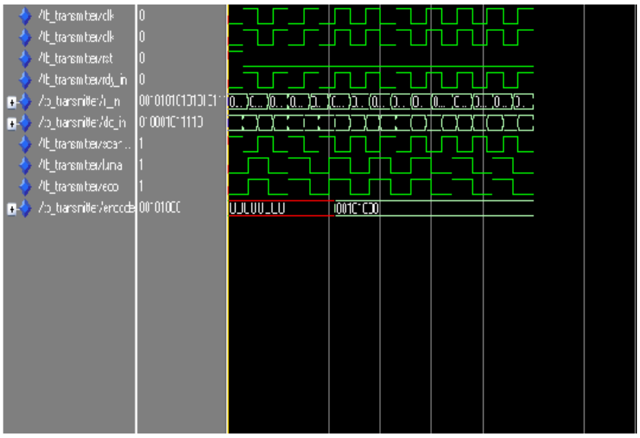


Fig.5. Simulation result for Transmitter output

b) Simulation result of proposed system based on channel equalizer

Clk,rst and transmit_in are the input.The equalizer output is equotput and the decoder output is decoder. When rst=0 and clk=0/1 test bench values given for transmit_in is transmitted and after some propagation delay decoded output and equalizer output is obtained.

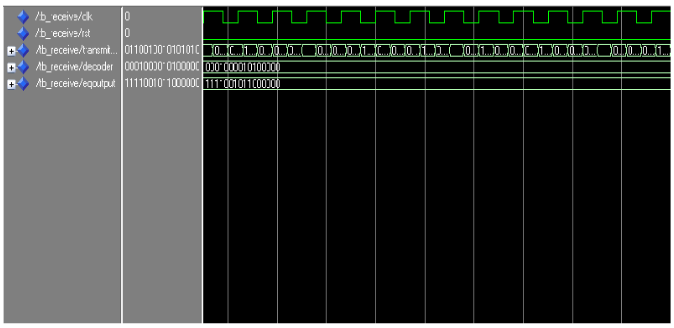


Fig 6. Modified output

5. CONCLUSIONS

In this paper, we have proposed a novel low-complexity soft-input decoder for VLCs. This algorithm is derived from the Chase decoding algorithm initially developed for block codes and has been tested for transmission over an AWGN channel in the case of BPSK signaling. The decoding solution proposed here shows numerous significant advantages. The Chase-like decoding of the VLCs achieves good error correction performance with very low complexity, especially for medium to high values of signal-to-noise ratio and also the proposed decoder can be slightly modified to deliver the reliability of the decoded bits. The Chase-like decoder was concatenated with a SISO channel decoder and iterative decoding was performed. The demand for high data rates has increased the requirement of equalization technique so that the effects of channel may be reduced. Channel equalization is used to improve the received signal quality in telecommunication especially in digital communication system. The simulation results shows that adding channel equalizer at the decoder end there is significant improvement in S/N. Also Bit error rate decreases with increase in the output SNR.

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REFERENCES

1. D. A. Huffman, “A method for the construction of minimum redundancy codes,” Proc. I.R.E., vol. 40, no. 9, pp. 1098–1102, Sep. 1952.

2 Y. Wang and Q. Zhu, “Error control and concealment for video communication: a review,” Proc. IEEE, vol. 86, no. 5, pp. 974–997, May 1998.

3 D. W. Redmill and N. G. Kingsbury, “The EREC: an error resilient technique for coding variable-length blocks of data,” IEEE Trans. Image Process., vol. 5, no. 4, pp. 565–574, Apr. 1996.

4 V. Buttigieg and P. G. Farrell, “A maximum a posteriori (MAP) decoding algorithm for variable length error-correcting codes,” in Proc. 1995 Codes and Cyphers: Cryptography and Coding IV.

5 K. P. Subbalakshmi and J. Vaisey, “On the joint source-channel decoding of variable-length encoded sources: the BSC case,” IEEE Trans. Commun., vol. 49, no. 12, pp. 2052–2055, Dec. 2001.

6 K. Sayood, H. H. Otu, and N. Demir, “Joint source/channel coding for variable length codes,” IEEE Trans. Commun., vol. 48, no. 5, pp. 787– 794, May 2000.

7 R. Bauer and J. Hagenauer, “On variable-length codes for iterative source/channel decoding,” in Proc. 2001 Data Compress. Conf., pp.273–282.

8 K. Lakovic, Tao Tian, and J. Villasenor, “Iterative decoder design for joint source-channel LDPC coding,” in Proc. 2005 EUROCON Int. Conf. Computer as a Tool.

9 L. Pu, Z. Wu, A. Bilgin, M. W. Marcellin, and B. Vasic, “LDPC-based iterative joint source-channel decoding for JPEG2000,” IEEE Trans. Image Process., vol. 16, no. 2, pp. 577–581, Feb. 2007.

10 Low complexity soft decoding of Huffman codes and iterative joint source channel coding

11 Channel equalization using weiner filter,Virendra singh Chauhary

12 A. Zribi, S. Zaibi, R. Pyndiah, and A. Bouallègue, “Chase-like decoding of arithmetic codes with image transmission applications,” in Proc. 2009 IEEE Int. Conf. Signal Image Technol. Internet Based Systems.