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Bandwidth Compression of the Digitized High-Definition Television Images for Transmission via Satellites

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This paper investigates a subband coding scheme to reduce the transmission bandwidth of the digitized high-definition television (HDTV) images. The HDTV signals are decomposed into seven bands. Each band is then independently encoded. The base band is DPCM encoded and the high bands are encoded by using nonuniform Laplacian quantizers with a dead zone. By selecting the dead zone on the basis of energy in the high bands, an acceptable image quality is achieved at an average rate of 45 Mbits/s (Mbps). This rate is comparable to some very hardware intensive schemes of transform compression or vector quantization proposed in the literature. The subband coding scheme used in this study is considered to be of medium complexity. The 45-Mbps rate is suitable for transmission of HDTV signals via satellites.

I. Introduction

THE next major breakthrough in the television industry will be in the areas of production, distribution and reception of high-definition television (HDTV) images. The revenue possibilities are in the billions of dollars. Many companies in different parts of the world are working to obtain a major share of this market. Yet very few U.S. organizations have addressed the problem of HDTV distribution, particularly via communication satellites. Because of their point-to-multipoint nature, communication satellites are the most cost-effective means of video distribution including HDTV. Considering the advantages of digital transmission and the trend toward all digital satellite communications, it is important to develop a cost-effective digital codec for HDTV signals. Digitized HDTV signals require a very large bandwidth. For satellite communications, transmission costs generally increase with the increase in bandwidth requirements. Therefore, for the design of a cost-effective HDTV codec, it is necessary to employ a simple data compression scheme to reduce the transmission bandwidth of the HDTV signals.

Several researchers have proposed data compression schemes for the HDTV signals.[1-3] However, major emphasis has been at the 100–140 Mbits/s (Mbps) data rate that is generally intended for optical fiber networks. Transmission via satellites at this rate is generally not cost effective. An HDTV codec operating at data rates between 27 and 55 Mbps, including the DS3 rate of 45 Mbps will be very desirable. The goal of this research effort is to develop a data compression scheme for the HDTV signals to bring the data rate from around 500 Mbps down to around 45 Mbps. Another goal is to keep the compression scheme simple and easy to implement in hardware.

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II. HDTV Data Description

Data of six different HDTV images were obtained from COMSAT Laboratories in Clarksburg, MD, to develop and test the compression algorithm. The data are in the form of component signals consisting of luminance and chrominance difference signals (YUV) band limited to 20 and 6 MHz, respectively. The luminance signal is sampled at the rate of 54 MHz, and the chrominance difference signals are sampled at the rate of 13.5 MHz each. The samples of each signal are quantized to 8 bits. The U and V signals are vertically filtered and 2:1 decimated (sampling rate reduction by a factor of 2) vertically to remove the excess chroma vertical resolution. Only active pixels are encoded. This results in an uncoded active rate of 41.32 megasamples/s for Y and 5.155 megasamples/s for each of the U and V signals. The algorithm developed in this research is tested only on the luminance component. It is assumed that the same encoding algorithm can be used for compressing the U and V components. For a goal of encoding all three components at a total rate of 45 Mbps, the Y component must be encoded at a rate of 36 Mbps. Thus the entropy requirement for the compression algorithm is 36/41.32 = 0.87 bits per pixel (bpp).

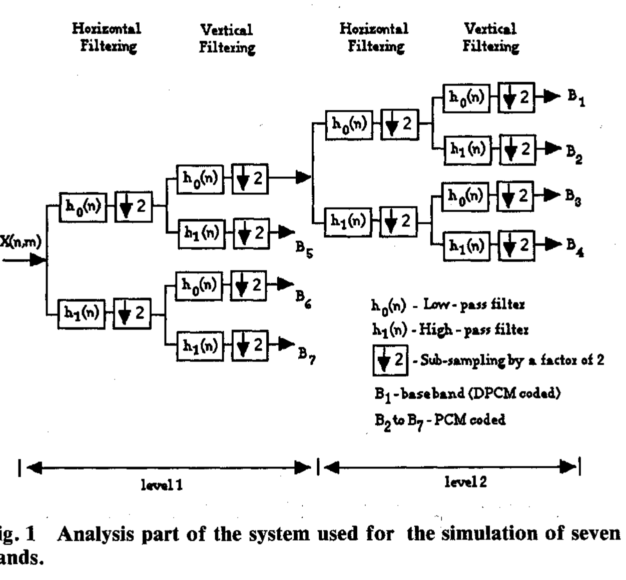

Fig. 1 Analysis part of the system used for the simulation of seven bands.

III. Compression Algorithm

As discussed earlier, the goal of this research was to compress the digitized HDTV images to obtain an entropy of 0.87 bpp with an acceptable picture quality. Although the final measure of the performance of any compression algorithm is the subjective quality of the processed pictures, it is convenient to use a quantitative measure during the development phase of the algorithm. This is especially true when the facility to subjectively evaluate the pictures is not at the same location where the algorithm is being developed. A commonly used quantitative measure for processed images is the peak signal to noise power (SNR) ratio defined as SNR = 10 $\log_{10}$ [255^2/MSE] dB where MSE is the mean squared reconstruction error. From experience, an SNR of 30–32 dB gives good processed picture quality. This SNR range was set as a goal for this research. To keep the implementation simple, it was decided to develop the algorithm on an intraframe basis. The final processed pictures were also subjectively evaluated at the COMSAT Labs.

Several algorithms were explored to achieve our design goals. It was decided to implement subband coding because of its superior performance with simple hardware implementation. There are two main reasons for the superior performance of subband coding.[4–8] First, the reconstruction error variance can be separately controlled in each band; therefore, the shape of the overall reconstruction error can be controlled as a function of frequency. Second, each subband can be encoded separately by using an encoder that is closely matched to the requirements of that band. The subband coding algorithm developed in this research is described next.

A. Subband Filtering and Encoding

A separable quadrature mirror filter (QMF) bank is used to divide the images into seven subbands as described in Ref. 4 and as shown in Fig. 1. One-dimensional finite impulse response (FIR) QMF of length 16, as reported by Johnston[9] as 16a is used. Filtering operations yield one band (base band) with intensity distribution similar to that of the original image and six bands that have distributions close to Laplacian. Figure 2 shows band B_1 as the base band that carries most of the information about the original image and B_2–B_7 as the high bands. Because of the high pixel correlation in band B_1, this band is differentially pulse code modulated (DPCM) encoded using a two-dimensional predictor as reported in Ref. 10 and shown in Fig. 3. The predicted value is given by

$$\hat{x}_A = bx_B + cx_C + dx_D \tag{1}$$

where x_B, x_C, and x_D are previously reconstructed pixels and is x_A the prediction of the present pixel. The weighting factors *b*, *c*, and *d* are 0.5, 0.25, and 0.25, respectively.

1. Fixed-Rate Subband Coding

To meet the requirement of encoding the images at a given rate, a fixed-rate subband coding (FRSBC) algorithm is designed.[11] This is shown in Fig. 4. To implement this algorithm, the variance of the prediction error for the base band and the variance of the high bands is computed. Bits are then allocated to each band based on the variance of the band. For equal band splitting, the bit allocation is given by[11]

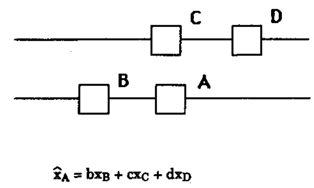

Fig. 3 Two-dimensional prediction algorithm for DPCM coding the base band.

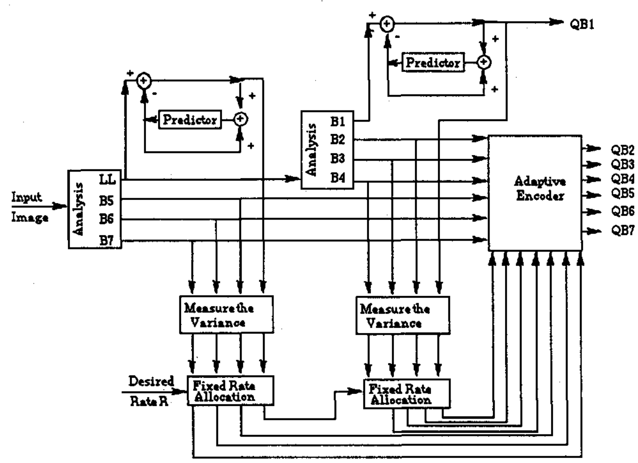

Fig. 4 Fixed-rate subband coding algorithm.

$$R_k = R + \tfrac{1}{2}\log_2\left[\frac{\sigma_{xk}^2}{\left(\prod_{i=1}^{M}\sigma_{xi}^2\right)^{1/M}}\right], \quad k = 1, \ldots, M \tag{2}$$

where

M = number of subbands
R = fixed rate
R_k = bits allocated to the kth band
σ_{xi}^2 = variance of the ith band

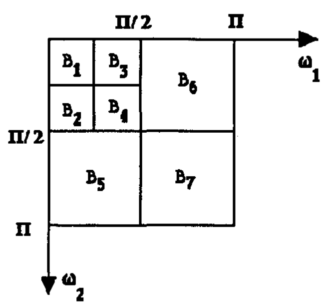

Fig. 2 Frequency domain decomposition.

2. Encoding the Bands

The prediction error signal for the base band B_1 has a Laplacian distribution. The high bands B_2–B_7 also have Laplacian distribution. A 32-level symmetric, nonuniform Laplacian quantizer with dead zone width of 2 is used to quantize the prediction error signal of the band B_1. The quantized values are Huffman encoded for transmission. An adaptive algorithm is developed to quantize the high bands with either an 8-level or a 16-level nonuniform Laplacian quantizer. An 8-level quantizer is used if the energy in the band is less than 12.42, otherwise a 16-level quantizer is used. The quantized levels are run length encoded. Nonzero values are Huffman encoded. Runs of zeros and ones are sent using the B-1 code.[12] An iterative algorithm is used to set the width of the dead zone for each band so that the band is encoded within the allocated bit rate.[11]

A problem in encoding the high bands using the preceding adaptive algorithm was that the Huffman code design depended on whether a band was quantized with an 8-level or a 16-level quantizer. To overcome this problem, it was decided to design Huffman encoders on an average basis. To accom-

Table 1 Results from the four algorithms using QMF 16a filters

Algorithm[a]	Average bpp[b]	Average SNR[b]
1	0.833	31.362
2	0.811	31.229
3	0.916	31.671
4	0.798	31.192

[a]Algorithm 1: adaptive quantizers, adaptive dead zones; algorithm 2: fixed quantizers, adaptive dead zones; algorithm 3: fixed quantizers, fixed dead zones: algorithm 4: fixed quantizers, three fixed dead zones.
[b]Average is for the six images.

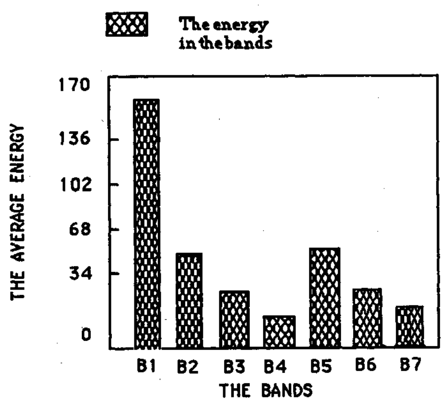

Fig. 5 Energy distribution in the subbands.

plish this, twelve 256×256 windows were constructed from the given data of six images. The adaptive FRSBC algorithm is implemented on these 12 windows, and the dead zone values that meet the allocated bit rates are computed. An average dead zone value is then computed for each of the high bands. Further, average energy for each band of the six images is computed. This is shown in Fig. 5. From this data, it is observed that bands 4 and 7 will be quantized most of the time by an 8-level quantizer and bands 2, 3, 5, and 6 with a 16-level quantizer. By fixing these quantizers and by using the average dead zone values computed, Huffman encoders for 8-level and 16-level quantizers are designed. The FRSBC algorithm is now implemented with these Huffman encoders, and the results are obtained. It is found that, on an average for the six images, a bit rate of 0.8332 bpp results in a average SNR of 31.36 dB. This easily meets our design goal.

One of the major considerations in designing the subband encoding algorithm was to obtain an easily implementable algorithm. The algorithm developed earlier requires computation of energy in a band to decide whether an 8-level or a 16-level quantizer will be used. Also, the dead zones are determined using an iterative algorithm. This requires extensive computations. Results are given for three variations of the algorithm just developed. These require fewer computations.

In algorithm 2, the energy is not computed to decide whether an 8-level or a 16-level quantizer will be used. For bands 4 and 7, an 8-level quantizer is used, and for bands 2, 3, 5, and 6, a 16-level quantizer is used. However, the dead zones are still obtained using the iteration technique. The results obtained from this algorithm are very close to the results from algorithm 1, which uses the energy threshold to decide the quantizers. So algorithm 2 is rated better than algorithm 1. In algorithm 3, the quantizers are fixed as in algorithm 2. In addition, the dead zones are also fixed at the average values computed for designing the Huffman coders. This algorithm gives an average 0.91-bpp rate at an average SNR of 31.67 dB. The bit rate slightly exceeds our goal of 0.87. However, this algorithm is the simplest from the viewpoint of implementation. There are no iterations and no adaptive strategies in this algorithm. To bring the bit rate down to our desired goal, a fourth algorithm was implemented. In this algorithm, instead of using one set of fixed average dead zone values, three fixed values are used. These are called small, medium, and large. Small dead zone values for each band are obtained by computing average dead zones for six images by implementing algorithm 2. The largest value of the dead zone obtained for the six images for each band corresponds to the large dead zone values. The medium dead zone is the average of the small and large values. The algorithm starts with small dead zone values for each band. If the obtained bit rate for any band exceeds the allocated bit rate, the algorithm steps to the next higher dead zone value. This algorithm gives an average bit rate of 0.798 at an average SNR of 31.192 dB, which is well within our design goal. Results for the four algorithms are compiled in Table 1.

IV. Subband Coding With Short Kernel Filters

Since the sample rate of the HDTV images is very high, it is desirable to use short kernel filters for decomposing the images into bands. Results were obtained for two different short kernel filters. The first pair, SKF2/2 filters, has two-tap filters for both low pass and high pass. The second pair, SKF5/3, has five taps for the low-pass filter and three taps for the high-pass filter. Their transfer functions are given as follows[13]:

SKF2/2 pair:

$$H_0(z) = \tfrac{1}{2}(1 + z^{-1}) \tag{3}$$

$$H_1(z) = \tfrac{1}{2}(1 - z^{-1}) \tag{4}$$

SKF5/3 pair:

$$H_0(z) = \tfrac{1}{8}(-1 + 2z^{-1} + 6z^{-2} + 2z^{-3} - z^{-4}) \tag{5}$$

$$H_1(z) = \tfrac{1}{4}(-1 - 2z^{-1} + z^{-2}) \tag{6}$$

Results for these two filter pairs are obtained by implementing algorithms 1 and 4. These results are compared with the results obtained from the 16-tap filters and are given in Table 2 for algorithm 1 and Table 3 for algorithm 4. The results obtained with the SKF2/2 pair meet our design goal. The results obtained with the SKF5/3 pair are better than our design goal and are very close to the results obtained with the 16-tap filters. It is recommended that algorithm 4 be fine tuned with the SKF2/2 or SKF5/3 filter pairs for final implementation. The processed picture frames were viewed on the HDTV monitor at the COMSAT Laboratories by the authors and the COMSAT experts. The processed frames were rated as almost indistinguishable from the original unprocessed frames.

Table 2 Bit rate and SNR performance of algorithm 1 for the HDTV signal using QMF 16A, SKF2/2, and SKF5/3 filters

Bit rate, bpp	QMF 16a	SKF2/2	SKF5/3
Bit rate, bpp	0.8332	0.9172	0.8572
SNR, dB	31.3624	30.9855	31.0381

Table 3 Bit rate and SNR performance of algorithm 4 for the HDTV signal using QMF 16A, SKF2/2, and SKF5/3 filters

Bit rate, bpp	QMF 16a	SKF2/2	SKF5/3
Bit rate, bpp	0.7979	0.8759	0.8245
SNR, dB	31.1924	30.9481	30.8978

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