EEG SIGNAL COMPRESSION WITH ADPCM SUBBAND CODING

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ABSTRACT

An EEG signal compression method that combines both octave-band filter bank frequency decomposition and coding in subbands using adaptive differential pulse code modulation (ADPCM) is proposed. Besides being computationally effective, this compression method in its simplest form yields 70 % data reductions with very little distortion. Higher compression rate are obtained by increasing the order of the predictor used.

1. INTRODUCTION

Medical signal processing is a fast growing field of research that is producing increasingly sophisticated applications in today's high-tech medicine. In the field of neurology common signals of interest are electrical potentials caused by firings of millions of neurons in the human brain during various mental activities. The recordings of these signals are obtained by electrodes placed on the head and are called electroencephalogram (EEG) signals.

A typical digital EEG recording session requires a large amount of memory. As an example, consider a 12 hour session in which EEG data is recorded from 24 channels, where each channel is sampled at 256Hz with 8 bits per sample. The memory required for the storage of this session is 264MB. It is expected that the rise in the amount of recorded files will create the need for data compression.

Recently, Hinrichs [1] used compression method based on adaptive differential pulse code modulation (ADPCM) coding that was applied on full-band EEG signal. To further reduce the memory requirements for EEG recording a coding method should be employed that exploits the properties of EEG signal in order to achieve a greater compression. One such technique is subband coding. Here, a signal of interest is split into frequency subbands by some multirate filter bank structure and then coded in subbands using any of the known coding techniques. Reconstruction of the signal is performed after decoding and interpolation in the synthesis part of the filter bank.

Multirate filter banks can operate within different transform domains. Here we used the Fourier domain such that our filter bank performs time-frequency decomposition of EEG signal. To the best of our knowledge, this approach combined with ADPCM coding has not been applied to EEG signals.

2. FILTER BANK STRUCTURE

The multirate filter bank that would decompose the EEG signal into frequency subbands can be based on basic EEG frequencies called Alpha, Beta, Delta and Theta rhythms

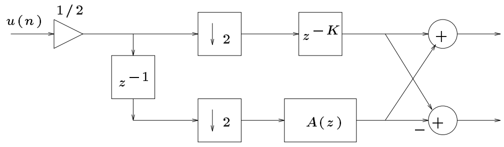

Figure 1. IIR QMF analysis building block

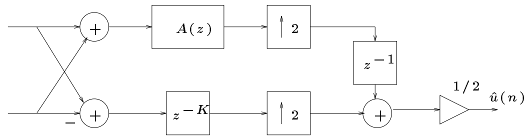

Figure 2. IIR QMF synthesis building block

[2]. The same filter bank can be used for identification of these rhythms as well as for compression. If one uses the power of two sampling, then octave band splitting closely approximates characteristic EEG rhythms. For these purposes we apply a simple tree-structured octave-band filter bank [3]. Elements that perform split and merge of subbands are specific two-band filter banks. Due to the time delay caused by filter bank, delay elements must be included in the synthesis part. This delay depends on filter types used in the split block.

The fundamental component in our coding system is a two-channel, allpass based, IIR QMF filter bank. If we cascade these components, then the analysis part splits the signal into two equal frequency bands and downsamples it to half the sampling rate, while the synthesis part upsamples and reconstructs the signal. The same allpass filter, $A(z)$, with transfer function

$$A(z) = \frac{a_N + \cdots + a_1 z^{-(N-1)} + a_0 z^{-N}}{a_0 + a_1 z^{-1} + \cdots + a_N z^{-N}} \tag{1}$$

is used is each structure. The details of its design can be found in [4]. We used an eighth-order allpass filter ($N = 8$) that in turn requires the same order for a delay element inside the block. The analysis block performs highpass and lowpass half-band filtering that is characterized with linear phase in the passbands and stopband attenuation of 71.2 dB. This filtering block requires only eight multipliers. The synthesis block has the same features. The analysis and synthesis blocks are shown in Figures 1. and 2.

To complete the tree-structured design one needs to calculate or estimate the exact delays produced by the filter banks in each branch of a tree. This can be done by observing unit step responses. The entire analysis/synthesis structure is shown in Figure 3.

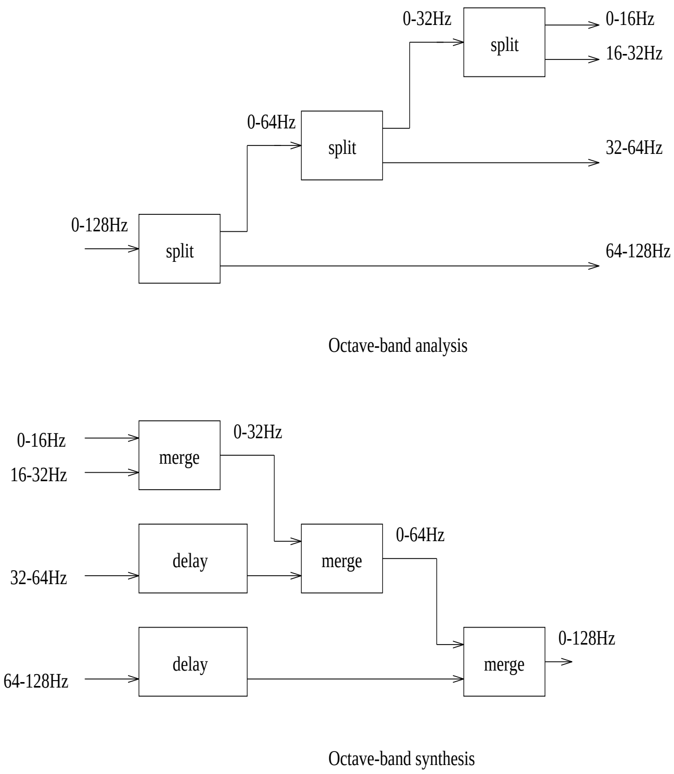

Figure 3. Octave-band filter bank with three levels for EEG frequency decomposition

The subbands that our octave band decomposition produces are: 0-4Hz, 4-8Hz, 8-16Hz, 16-32Hz, 32-64Hz; 64-128Hz. Since Delta rhythm is less than 4Hz, Theta between 4 and $7\frac{3}{4}$ Hz, Alpha between 8 and 13Hz and Beta is greater than 15Hz, first four frequency bands of the filter bank are approximately those of the characteristic EEG patterns.

3. SUBBAND ADPCM CODING

Standard block coding, represented in Fourier or cosine transforms for example, are unsuitable for EEG signals due to their non stationarity, especially spike appearances. Sample-by-sample methods can produce better results when faced with EEG nonstationarities.

One such method is adaptive differential pulse code modulation technique or ADPCM. The main blocks of this scheme are predictor and quantizer, shown in Figure 4.

The p-th order predictor estimates the actual sample using a linear combination of p preceding samples according to

$$\hat{s}_i = \sum_{j=1}^{p} a_j s_{i-j} . \quad (2)$$

The resulting decorrelated prediction error

$$e_i = s_i - \hat{s}_i \quad (3)$$

is then fed into the quantizer.

Passing the quantizer, e_i is mapped to e_i' which deviates from e_i by the quantization error $q_i = e_i - e_i'$ (see Figure 4.).

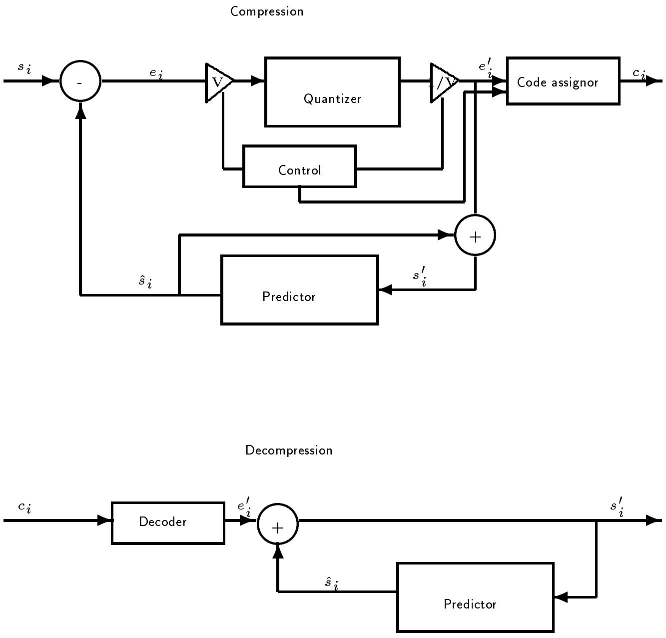

Figure 4. ADPCM coder with variable range quantizer and fixed predictor

Therefore, instead of the original samples, the quantized prediction errors are recorded and transmitted. For correlated signals these errors have smaller variance than the original signal and can be coded with fewer bits.

However, for non stationary signals prediction, error variance changes as well. Certain EEG features, like spikes, lead to sudden changes of variance of the prediction error. Therefore the application of a quantizer with a variable range is necessary to cope with these variabilities. In addition, a varying predictor has to be considered with respect to varying signal statistics. This procedures represent the adaptive part of ADPCM coding scheme.

Combined with filter bank subband decomposition of signal, ADPCM coding can be now employed into each or some of subbands. Since signal decomposition further decreases the prediction error in subbands, fewer bits are required for a unique representation of a subband. Also, subbands that have a small energy content can be directly coded using a clever bit allocation or completely discarded.

4. RESULTS

In experiments, an original file from a commercially available EEG machine is used. The sampling rate was 256Hz with 8 bits for A/D conversion. By reconfiguring the filter bank tree it was straightforward to obtain the sampling rates of 128Hz and 64Hz. In our experiments we used standard electrode positioning, the so called Queen's Square montage [2].

In general, the experiments conducted led us to conclude that substantial memory savings for storage of EEG signal can be achieved just by using filter bank decomposition and elimination of high frequency bands. Further, for compression purposes, the analysis of signals in subbands is necessary for successful subband coding, where one has to choose how many bits/sample should be assigned for every subband. Often it is the energy in subbands that dictates the bits/sample ratio.

Having a tool like the previously described filter bank structure we can now apply it to EEG signal compression by using some standard coding technique in subbands. An efficient, although lossy, coding method is ADPCM, which was already being applied for compression of full-band EEG signal in [4]. A first order predictor was used to achieve computational efficiency. The quantizer was being updated every 5 samples, which somewhat burdened computation and added extra data.

With the proposed filter bank tree, the method from [4] can be extended, simplified and made more efficient. For example, in our experiments, we first completely avoided updating the quantizer. This was made possible because subband decomposition spreads the signal energy over the subbands. Adjusting the quantizers beforehand to accommodate for different signal amplitudes we were able to avoid quantizer updating. This procedure produced a very fast algorithm, applicable for on-line compression. The results of this approach are given in Figure 5.

On the Figure 5, an EEG signal with spike-wave patterns was selected due to its high variability. On Figure 5 (b), the subbands were coded, starting from the highest to the lowest frequency subbands, with 2, 6, 8 and 8 bits/sample, resulting in an average of 4.5 bits/sample, while on Figure 5 (c), the subbands were coded, following the same direction, with 1, 1, 8, and 8 bits/sample, resulting in an average of 2.75 bits/sample [1].

For comparison purposes, we also applied the algorithm from [1] to a full-band EEG. As before, adaptive adjustment of quantizer levels was excluded. This resulted in severe distortions even for compression rates of 4 bits/sample, as shown on Figure 6.

Introduction of predictors of higher orders further decreased differential error and thus reduced distortions caused by the quantizer. This approach, however, introduced more time consuming operations and required more complicated prediction algorithms [5]. In our case the decrease of distortion together with reduction of data rate was not dramatic since our original signal was highly oversampled.

5. CONCLUSION

To summarize, we presented a simple coding method for EEG signals based on octave-band filter bank subband decomposition. Although, some distortion in reconstruction of signal is present due to rounding errors, it was shown that filter bank tree is a very a powerful processing tool and can even be used alone for EEG data reduction. Further, we combined the filter bank tree with the ADPCM coding method and applied it to the compression of EEG signals. We showed that a filter bank in conjunction with a simplified but computationally efficient ADPCM coding can easily obtain satisfactory data reductions of 70 % or higher. The application of more advanced, although computationally expensive, predictors obtained even better results with lesser amount of distortion.

[1] Note that it is difficult to directly compare our results and those in [1] since our original data was quantized with 8 bits/sample while Hinrichs [1] had 12 bits/sample.

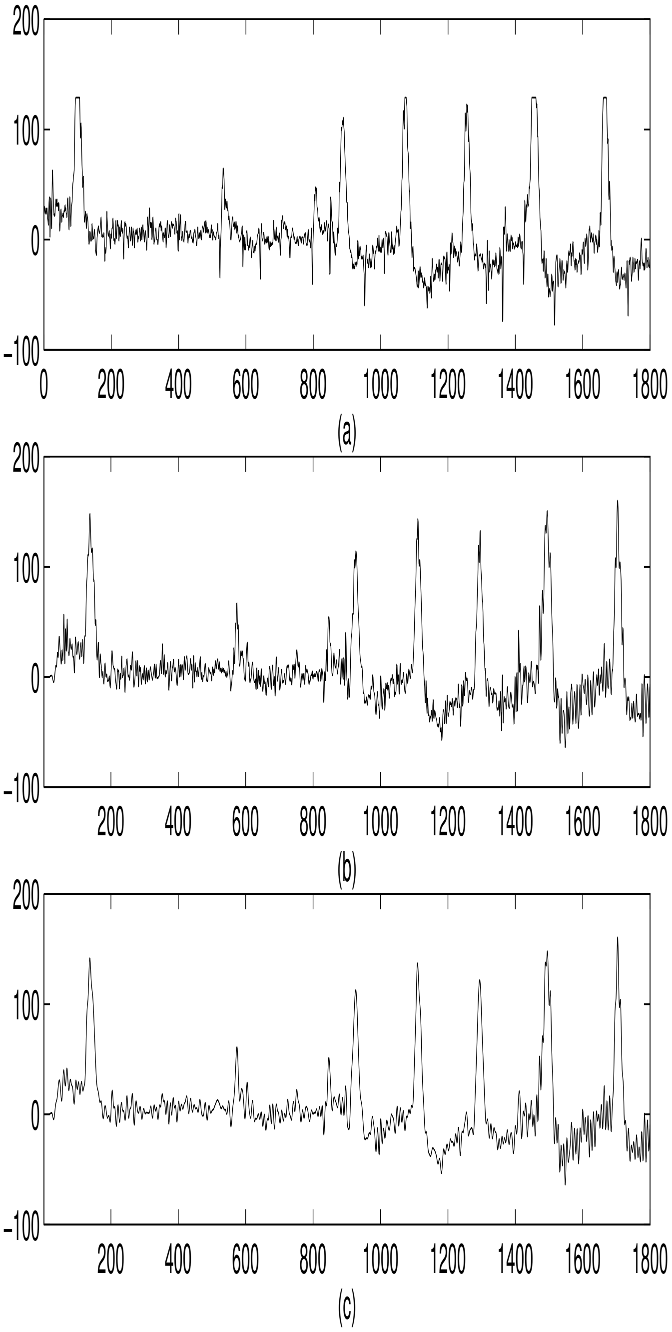

Figure 5. Application of Subband ADPCM coding: (a) Original signal, 8 bits/sample, (b) subband ADPCM with 4.5 bits/sample and (c) subband ADPCM with 2.75 bit/sample.

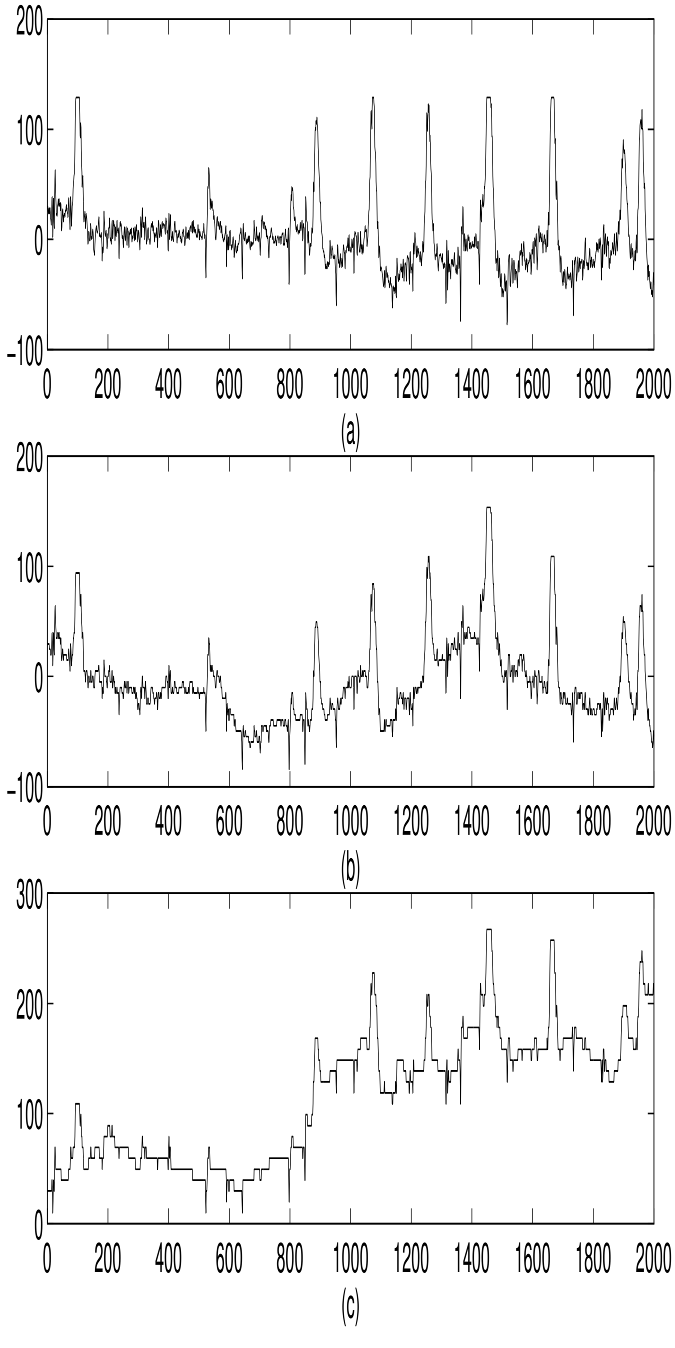

Figure 6. Application of ADPCM on full-band EEG signal: (a) Original signal 8 bits/sample, (b) ADPCM coded/decoded with 4 bits/sample, (c) ADPCM coded/decoded with 3 bits/sample.

REFERENCES

[1] H. Hinrichs, "EEG Data Compression with Source Coding Techniques", *Journal of Biomedical Engineering,* vol. 13, pp. 417-423, Sept. 1991.

[2] F. M. Dyro, "The EEG Handbook", *Little, Brown and Company,* Boston MA, 1989.

[3] Z. Sijerčić and G. Agarwal, "Tree Structured Filter Bank for Time-Frequency Decomposition of EEG Signals", *in Proc. of 17th Annual Int. Conference IEEE Engineering in Medicine and Biology Society,* Montreal, Canada, September, 1995.

[4] T.Q. Nguyen, T.I.Laakso and R.D.Koilpillai, "Eigenfilter Approach for the Design of Allpass Filters Approximating a Given Phase Response," *IEEE Transactions on Signal Processing,* vol. 42, pp. 2257-2263, Sept. 1994.

[5] L. Ljung, "System Identification Toolbox User's Guide," *The MathWorks, Inc.,* Natick MA, 1995.