

NEW RESULTS IN LOW BITRATE AUDIO CODING USING A COMBINED HARMONIC-WAVELET REPRESENTATION

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ABSTRACT

In this paper, we propose a new combined harmonic-wavelet representation for audio where a harmonic analysis-synthesis scheme is used, first, to approximate each audio frame as a sum of several sinusoids. Then, the difference between the original signal and the reconstructed harmonic signal is analyzed using a wavelet filtering scheme. After each step (harmonic analysis & wavelet filtering), parameters are quantized and encoded. Compared to previously proposed methods, our audio coder uses different harmonic analysis-synthesis and wavelet filtering schemes. We use the Total Least Squares (TLS)-Prony algorithm for the harmonic analysis-scheme, and an M-band wavelet transform for analyzing the residual. Altogether, our proposed coder is capable of delivering excellent audio signal quality at encoder bitrates of 60-70 kb/s.

1. INTRODUCTION

Uncompressed audio signals are typically sampled at 44.1 kHz and encoded with 16 bits/sample PCM, resulting in the large bitrate of 705 kb/s per channel. So far, some sophisticated audio coding schemes, such as the ISO/IEC MPEG [1], have been successful in reducing this bitrate to approximately 64 kb/s per channel with near transparent coded signal quality. Ongoing research in low bitrate high quality audio coding continues to occur since further bitrate reduction is desirable for numerous applications. These include Digital Audio Broadcasting (DAB), ISDN, Internet audio, and HDTV, to mention a few.

Among the previous schemes proposed, a number have included Discrete Wavelet Transform (DWT) filtering methods [2]-[5]. In [2], a Wavelet Packet Transform with Cascaded Conjugate Quadrature Filters (CCQF) was investigated. This filter bank contains frequency divisions approximating the critical bands, however, large sidelobes are present in the frequency response of the subband filters. In general, most filter bank designs based on cascade schemes have poor channel separation. Thus compared to traditional filtering schemes such as the Modified Discrete Cosine Transform (MDCT), audio compression methods based solely on CCQF perform poorly at low bitrates for tonal-like audio signals, e.g. violin and clarinet [4].

In contrast to the MDCT, the Wavelet Transform provides a nonuniform time resolution. In wavelet filtering, the impulse response of high frequency filters are short while low frequency filters have a long impulse response. Hence the high frequency energy of the signal is localized to a smaller portion of time. This is useful for preserving the "transient-like" segments in audio signals. In cases where this does not happen, quantization noise produced in the frequency domain is transferred to the time domain. This is referred to, in the time domain, as *pre-echo* noise [6], and is believed to become audible when the filter bank time resolution is much larger than the auditory pre-masking time of approximately a few milliseconds.

Recently, two audio coding schemes which incorporate wavelet filtering into an overall audio coding scheme have been investigated [5], [4]. Although the two schemes are different, the basic philosophy of each approach is to use a wavelet filtering technique for coding non-stationary segments or components. Included in each design is a separate or combined transform with superior frequency selectivity which is used for coding tonal-like segments or components. Thus the Wavelet Transform is mainly included for the purpose of limiting the spread of the high frequency pre-echo noise in the time domain.

In this paper, we propose a low bitrate audio coding scheme similar to that investigated in [4]. Each audio signal is firstly expressed as a sum of several sinusoids. The difference or *residual* between the original signal and the reconstructed harmonic signal is analyzed using a wavelet filtering scheme. Here, we propose an alternative method to [4] for the harmonic analysis-synthesis and wavelet filtering scheme. In section 2, each scheme (harmonic & wavelet) is explained, while the quantization of the respective parameters is detailed in section 3. In section 4, audio coding results are presented, and from which conclusions are drawn in section 5.

2. COMBINED HARMONIC-WAVELET REPRESENTATION

The overall block diagram of the proposed audio encoding and decoding scheme is illustrated in figure 1. The time signal is processed using frame sizes of 512 samples or 12 ms for 44.1 kHz sampled audio signals. A rectangular window is applied to each frame and no overlapping of frames is used here. The encoder starts by performing a harmonic analysis on the input signal frame, s , using the Total Least Squares (TLS)-Prony algorithm (described below). The outputs of

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this algorithm are the frequencies, amplitudes and phases of the major harmonic components which are respectively denoted by f_k , A_k and ϕ_k in figure 1.

A masking model scheme based on the MPEG psychoacoustic model 2 implementation, [1], is then called. The tonality estimate of the masking model is ignored since information about the tonal and noise components of the input signal s is available from the harmonic analysis scheme. The frequencies, amplitudes and phases of the unmasked harmonics are then quantized and encoded, while the masked harmonic components are ignored. Side information about the number of harmonics must be sent to the decoder. The quantization techniques used here are described in section 3.

A harmonic-like signal, h , is then reconstructed from the quantized parameters. This is subtracted from the input signal to give the residual signal, r . An M-band Discrete Wavelet Transform (DWT) described in section 4, is then used to decompose the residual signal into several subbands. In parallel with the DWT, a noise masking model is performed. The output of the noise masking threshold is used as input to a bit allocation algorithm which distributes bits to the DWT subbands. The bit allocation algorithm is similar to the MPEG bit allocation algorithm, [1].

The DWT subbands are then quantized using the number of bits determined from the bit allocation algorithm. The encoded DWT subbands and the bit allocation side information is sent to the encoded signal bitstream. The decoder then unpacks the encoded harmonic parameters, DWT subbands and the respective side information. The harmonic signal, h , is reconstructed from the quantized frequencies, amplitudes and phases of the harmonics. Likewise, an approximate residual signal, $\hat{r}$, is reconstructed from the inverse DWT. The two signals, $\hat{h}$ and $\hat{r}$ are summed to form the reconstructed time signal, $\hat{s}$.

2.1. TLS Prony Algorithm

Prony's method is a technique for modelling sampled data as a linear combination of exponentials. Given N complex data samples, $x[1], \dots, x[N]$, expressed mathematically as, [7],

$$\hat{x}[n] = \sum_{k=1}^p A_k \exp[(\alpha_k + j2\pi f_k)(n-1)T + j\phi_k] \quad (1)$$

where p is the number of complex exponentials in the model, T is the sample interval in seconds, A_k is the amplitude of the complex exponential, α_k is the damping factor in seconds^{-1} , f_k is the sinusoidal frequency in Hz and ϕ_k is the sinusoidal initial phase in radians. For the case of real data samples the complex exponentials must occur in complex conjugate pairs of equal amplitude. For the proposed coder, an undamped sinusoidal model was selected for the harmonic representation, i.e. $\alpha_k = 0, k = 1, \dots, p/2$. Thus for a given model order, p , the frequencies, amplitudes and phases need to be computed. For $N = 2p$, the original sequence $x[n]$ can be represented exactly using a p -th model order as, [7],

$$\hat{x}[n] = \sum_{k=1}^p h_k z_k^{n-1} \quad (2)$$

where

$$h_k = A_k \exp(j\phi_k)$$

$$z_k = \exp[(\alpha_k + j2\pi f_k)(n-1)T]$$

However, in practice an overdetermined case exists where $N > 2p$ and the data sequence can only be approximated. The solution of the overdetermined case is performed by constructing a set of Linear Prediction (LP) equations i.e.

$$\sum_{m=1}^p a[m] x[n-m] = e[n] \quad (3)$$

where

$$\sum_{m=0}^p (z - z_k) = \prod_{k=1}^p (z - z_k) \quad (4)$$

The a_m parameters can be estimated using an Least Squares (LS) algorithm. The frequencies, z_k , can then be derived from the roots of the characteristic equation in equation (4). Then, a Vandermonde matrix of time-index z elements can be formed and used to solve for the unknown h_k parameters using equation (2), and thus giving the amplitudes and phases of the complex sinusoids.

This form of Prony's method degrades for signals containing large amounts of noise or for signals containing closely spaced sinusoids. A slightly different method for solving the LP equations is to use the Total Least Squares (TLS) approach [8]. When this approach is applied to solve the LP equations for frequency estimation [9], the method is generally referred to as the TLS-Prony method. TLS is an extension of the basic LS approach where both the data matrix and observation vector are simultaneously assumed to contain noise. For spectral estimation the TLS-Prony method provides a more accurate estimate of the complex sinusoidal frequencies. In this paper, the method in [9] is used to determine the frequencies. Then, the original approach in equation (2) is performed to determine the amplitudes and phases.

2.2. Wavelet filtering scheme

The residual signal is analyzed using the 4-band Wavelet Transform in figure 2. This configuration was chosen since it roughly approximates the critical band divisions. We use the technique presented in [10] for the design and construction of the Perfect Reconstruction (PR) 4-band filter banks. The filter coefficients selected for the 4-band filters are 32-taps each and have a regularity of $K = 3$. In [10], the impulse response of the 4 filters are designed to be numerically identical except for a shuffling operation on the positions of the coefficients. Overall, we have identified a number of key advantages with the proposed filter bank over the CCQF implementation in [2]. These include :

- Low computational complexity in 4-band filtering operations [10].
- The delay in subband filtering is small since only three stages are used.
- The subband filters are linear phase.

3. QUANTIZATION

3.1. Harmonic Coding

A. Frequencies

Since the harmonics are complex conjugates, $p/2$ harmonic frequencies only need to be encoded where p is the model order. The approach taken for encoding the frequencies was to break the 0-22 kHz range into three frequency regions. The three regions 0-2.2, 2.2-5.5 and 5.5-22 kHz were selected because of the decreasing frequency resolution of human hearing as the frequency increases. Then, for each frequency region, the harmonic frequencies are uniformly quantized using 8 bits. Hence the frequencies in the lower region are effectively quantized more accurately than the frequencies in the middle region and so forth.

B. Amplitudes and Phases

Like the harmonic frequencies, only $p/2$ amplitudes and phases need to be encoded. The harmonic amplitudes are first encoded using a gain value. The gain is computed by finding the power of two which is just greater than the maximum of the absolute value of the amplitude vector. The gain is quantized by computing $\log_2(\text{gain})$. We found that 5 bits were necessary to fully represent the dynamic range of $\log_2(\text{gain})$. Each amplitude is then normalized by dividing by the gain. The normalized amplitudes are then uniformly quantized with 6 bits. Similarly, the phases are uniformly quantized with 6 bits. Further rate reduction is obtained by Huffman coding the quantized normalized amplitudes.

3.2. Wavelet Coding

The technique for encoding the Wavelet Transform coefficients is similar to the MPEG codec [1]. For each subband a *scalefactor* is extracted. Here, the *scalefactor* is computed in the same manner as the harmonic amplitude gain. Similarly, 5 bits are sufficient to represent the dynamic range of each *scalefactor*. Each subband is then divided by its respective *scalefactor*. The normalized coefficients are uniformly quantized using the number of bits determined from the bit allocation algorithm. Again, further rate reduction is obtained by Huffman coding the quantized *scalefactors* and normalized coefficients.

4. AUDIO CODING RESULTS

Comparisons to original source material were made for signals encoded and decoded with the proposed coder at encoding bitrates ranging from 60-70 kb/s. Listed in table 1 are twelve source signals from the European Broadcasting Union SQAM CD that were selected for testing. For each audio signal, the nominal encoding bitrate and the segmental Signal-to-Noise Ratio (SNR) are also given in table 1. In general, higher mean segmental SNRs were recorded for the "tonal-like" signals (e.g. piano and flute), while smaller values were recorded for broadband and transient like signals (e.g. castenets and eddie rabbit).

As well as segmental SNRs, informal listening tests were performed by using an A-B-C-A test sequence. In this sequence, A is original source, while one of B and C is the coded signal and the other is a reference signal. In each test, the reference signal was the MPEG layer III encoded/decoded signal (with an encoding bitrate of 64 kb/s). Six people were selected for listening tests and each test was

conducted with headphones in a quiet office environment. For each test signal, the listener was asked to compare the quality of the two coded signals, B and C. Impairments were not noticeable for the acoustic guitar, eddie rabbit, castenets and the female speech after encoding-decoding with the proposed coder. For these signals, the proposed coder delivered similar quality to the MPEG layer III coder. Slight impairments were identified for the remainder of the test signals and in each case the MPEG layer III coder delivered a slightly better coded signal quality.

5. DISCUSSION

In this paper, we have investigated combined harmonic-wavelet coding of audio signals at low bitrates. Compared to previously proposed methods, different harmonic analysis-synthesis and wavelet filtering schemes have been used here. The proposed coder is capable of delivering excellent signal quality for some of the signals tested at low bitrates (especially "tonal-like" signals). To obtain a more improved coded signal quality for all signals, we believe further research is necessary in identifying the optimum number of bits on a frame-by-frame basis for the harmonic and wavelet coding, the optimum wavelet filtering configuration (i.e. number of subbands) and the filter coefficients.

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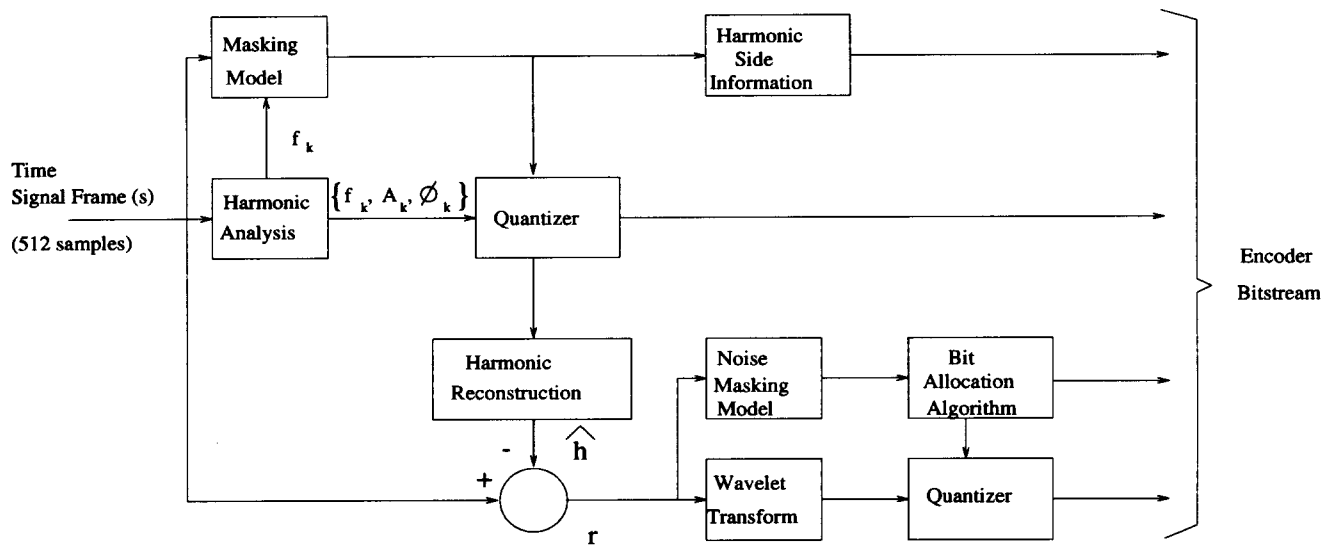


Figure 1. Proposed Audio Encoder

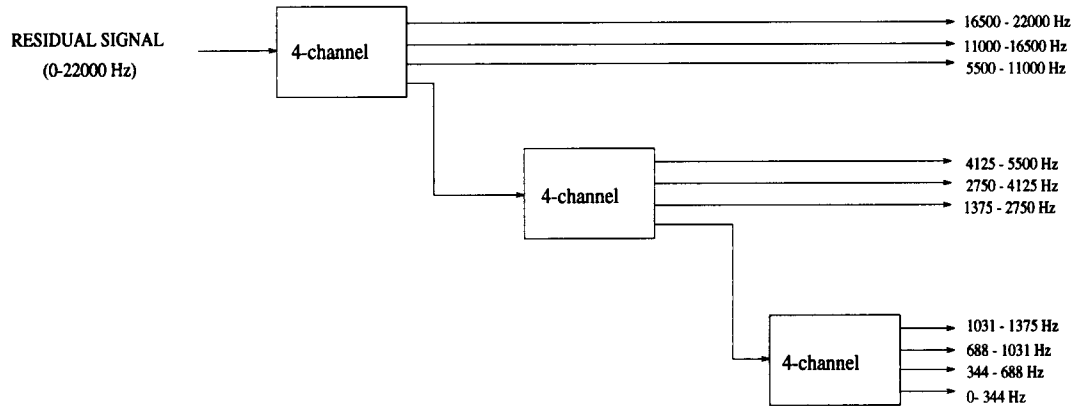


Figure 2. Proposed wavelet filtering scheme

Audio Signal	Bitrate (kb/s)	Segmental SNR (dB)
Eddie Rabbit	67	18
Castenets	70	12
Bass Vocal	60	24
Acoustic Guitar	64	23
Female Speech	66	19
Glockenspiel	63	26
Male Speech	65	19
Piano	65	38
Triangle	65	27
Violin	65	25
Flute	64	34
Gong	63	31

Table 1. Nominal Encoding Bitrate and Segmental SNR for each test signal.