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Hi-Fi Audio Coding Technique for Wireless Communication based on Packet Transformation

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ABSTRACT

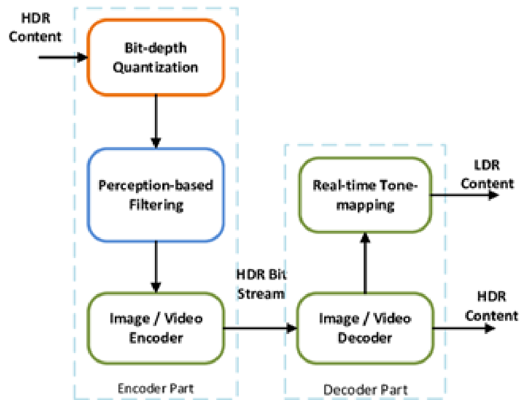
Hi–Fi is a fairly vague term used to differentiate a customized, high-end surround sound system from built-in TV speakers. The paper presents a satisfactorily low bit rate Hi – Fi audio coding technique with low totaling designed for transmit real-time high-quality audio signal over wireless channel. In the past few decades, quality audio recording tools has become rather easy to obtain, so just about any music or video you'll want to pay attention to will possibly sound pretty good. The programmed audio data is frame with some critical field is secluded by channel coding to improve noise protection when frames are transmit wirelessly. Latest results show that clear CD-audio quality can be achieved at 80 kbps encoding bit rate. Many audio philes focus on head-phones because unlike a speaker system, the sound isn't artificial by the acoustic of the room, leading to a much more steady listening experience. This method apply wavelet packet convert to decompose audio signal into subband to abolish unnecessary data using spectral and temporal masking properties. These cheering results clearly exhibit the better features of our technique compared to others such as Ogg / Vorbis and MP3, which are universally working now a days.

KEYWORDS:

HI-FI, Wireless Communication, CODEC, Sub-band.

I. INTRODUCTION

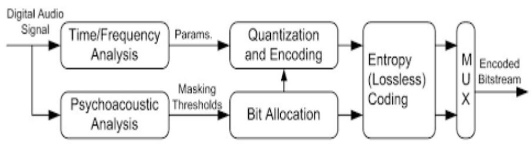
Whereas wireless communication has played a great role in our existence but to convey high reliability audio signal wirelessly at a sensible cost is still. Hi-Fi audio CODEC utilize entropy coding such as run-length and Huffman code where important parameter required demanding. Presently available audio coding techniques aim to trim down bit rate and put less concern on complication and capable wireless broadcast. Such audio CODEC like ISO / MPEG and Ogg / Vorbis are right for non real time application and audio documents. On the other hand, this research focuses on a Hi-Fi audio coding procedure that provides technically low bit rate and difficulty In addition, the technique must endure the noise in wireless channel at a hum degree. Modern to decode are assumed to be miscalculation. The structure will be unnecessary making it disposed to deafening wireless channel. As a outcome, we recognize such parameter and protect them using channel coding that can correct up to 29 bits (one of each seven bits). Perceptual coding technique is used reduce the bit rate based on "human earshot mask" property. In general, perceptual codecs consists of five module which are filer bank, psychoacoustic analyzer, bit allowance, quantizing, programming, and framing, The result from filter bank (frequency analysis / time) is quantized according to masking threshold planned by the psychoacoustic analyzers. In the use polyphase filter bank which require 512 co-efficient to represent each filter. This can take major time to encode the signal. In dissimilarity, we use wavelet based filter bank to convert signal into wavelet domain that analyzes both time and incidence concurrently.



The filter bank uses fewer coefficient and can characterize variable sized sub-band that more accurately match the characteristic of non-stationary audio signal human can only detect frequency variation at low or medium frequency. This hypothesis leads us to consider that the proposed procedure should be simpler, In addition, MATLAB experiment confirm that, at 80kbps, our wavelet packet audio codec yields equal audio quality to that of the 64kbps, MP3 and Ogg / Vorbis. It is value nothing that the higher bit rate of wavelet codec moderately accounts for channel coding. This makes it wireless broadcast ready.

II. WAVELET-PACKET AUDIO CODEC

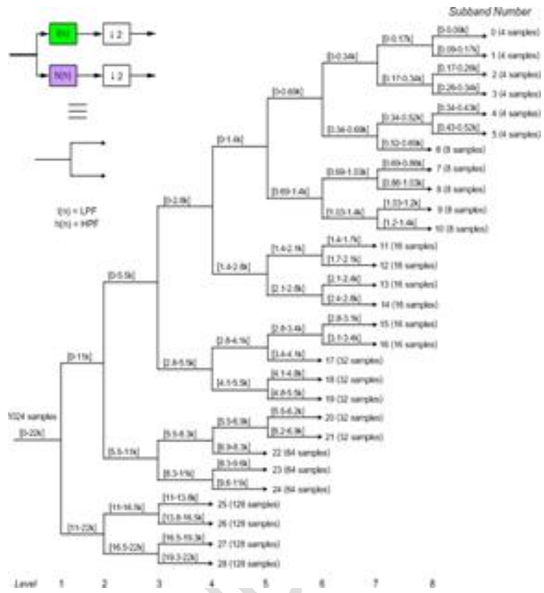
This collection and function of the proposed wavelet proposed wavelet packet codecs which takes audio CD sample (44.1ksps @ 16bps) as its inputs. Audio sample are framed at 1024 samples. Less sample would affect sample would affect the coding efficiency, whereas more sample would require long coding delay, violate more sample would require long coding delay, violate the real time requisite. Subsequent frame share 16 overlapping samples to reduce discontinuity between reconstructed frames at the decoder. Each frame is windowed by raise cosine filter to avoid swift change of the signal then proceeds to the encoder



Structure of Wavelet-packet Audio CODEC

Wavelet Packet Decomposition

This part decompose and transform the audio frame into wavelet domain. The decomposition tree consists of 29 sub-bands chosen to differ the critical band of human hear as depicted. Each sub-band will be quantized differently according to the signal to mask ratio (SMR) calculated by the Psychoacoustic Model.



Decomposition Tree for 44.1kbps Audio Frame

To avoid cases where wavelet coefficients are more than the number of samples in time domain, each frame is viewed as periodic is used to decompose frame and down sampling by two at each level of the tree Primary concept of the perceptual encoder is to reduce the audio signals that human cannot recognize because of signal masking or uncertainty. In fact, we need not encode nor be attracted in signal components that are below the hearing threshold. Three border patterns, which are the absolute threshold of inquiry, regularity masking, and temporal masking are therefore used to find the appropriate quantizing bits for each sub-band while minimize the quantization sound.

Algorithm to find masking threshold is shown below:

1) Each frame is windowed by Henning filter to decrease spectrum spread and then changed using FFT. Power spectrum is calculate for each point (X_k) which is used to find the spectral flatness measure— SFM in the next step. SFM is used to determine the noise like or tone like nature of the signal.

2) Calculate the masking energy offset for each sub- band and subtract it from the energy of that sub-band ($E_{sb}(i)$). This offset depends on distinguishing of the signal if it is similar to tone or vice versa. In [10], tone masking noise and noise masking tone are lower than $E_{sb}(i)$ about 14.5+i dB and 5.5dB. therefore, the offset

O_i in dB can be calculated by

$$O_i \square \square (14.5 \square i) \square (1 \square \square) 5.5$$

Where, Tonality co-efficient (□) is

Where $SFMdB_{MAX}$ =-60dB, if SFMdB = -60dB then signal is totally tone like else if SFMdB = 0dB then signal is completely noise like.

3) The spreading of growl energy $SP_F(i)$ is the difficulty of energy in the sub-band that is considered tone in (2) using dispersal function. The result is averaged over all co-efficient within the critical band.

4) secular masking is based on temporal spreading energy in the critical band, which is the complexity between the temporal energy series (C^2_{ij}) and the resemble version of the linear temporal spreading function within the critical band. The in order spreading energy complies $SP_T(i,j)$-C^2_{ij}, so, we can define the chronological masking factor as

$$\square_{ij} = \frac{SP_T(i,j)}{C^2} \square 1$$

Now $\square_{ij}$=1 means that C_{ij} is not masked by adjoining coefficients whereas $\square_{ij}$>1 means or else and the temporal frequency masking (M_{ij}) can be expected by adding $SP_F(i)$ with $\square_{ij}$.

5) The absolute threshold of examination ($ATH_{SPL}(f)$) is the average sound pressure sound pressure level (SPL) below which the human ear do not detect any motivation. This threshold is represent in dB as

6) $$ATH_{SPL}(f) \square 3.64 f^{\square 0.8} \square 6.5e^{\square 0.6(f \square 3.3)^2} \square 0.001 f^4$$

Where f is frequency in kHz.

7) Compare M_{ij} with $ATH_{SPL}(f)$ for each sub-band and select the higher value to correspond to the masking threshold at that regularity.

8) Find the least masking threshold for each sub-band that will be used to calculated the signal to mask ratio ($SMR_{sb}(i)$) of that sub-band.

Bit share for Wavelet Co-efficient of the Sub-band

To allocate bits for the sub-band, we find the mask to noise ratio (in dB) MNR=SNR-SMR. The sub-band with minimum MNR is allocated bits first because lower MNR value reflect lower noise masking. This reduce quantizing error and minimize noise that might be heard in the reconstructed audio signal.

Quantizing the Wavelet Co-efficients

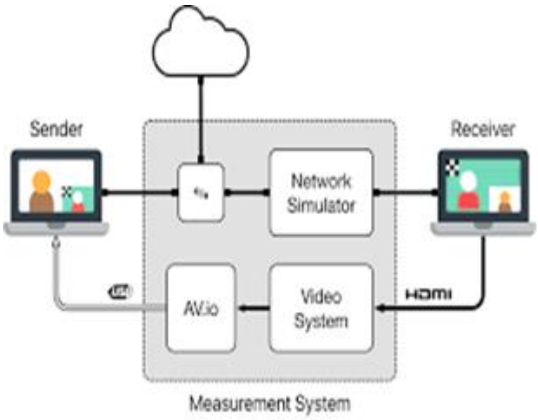
Wavelet coefficients are normalized before quantizing to advance accuracy. We define 64 scale factor to be selected as appropriate for each sub-band. The scaled coefficients are equivalently quantized according to the number of bits received from bit allocation algorithm. Experimental results suggest that the quantized values should be threshold familiar to be all positive because they give up recovered reconstructed audio signal.

Data Framing

Frame structure is illustrate. The 24 bit sync field (01010101) is used for frame organization. The 203 bit field signify the number of bits allocated to each sub-band. The following (Maximum) 174 bit field stores scaling factors for each sub-band. The last variable bits field contains the quantized wavelet co-efficients.

It is clear that bit errors during broadcast through wireless channel would considerably affect the decoding performance. though, within the frame, the 203 bit field that provide bit allocation is the most perceptive to error because the decoder would not know the right bit allocation of the subbands. This may lead to frame skip or incomplete decoding. As a result, we protect this field using BCH(7,4) code which can correct one bit out of 7 bits and up to 29 bits can be corrected.

The scaling factor is 6 bit data for each sub-band and is sent only for the sub-band with non-zero bit allocation. The length of quantized wavelet coefficient data depends on binary encoding of all sub-bands. We protection bit 0's to the frame to keep the bit rate stable and to ease the decode progression at the decoder



Frame Structure of Packet Audio Coder

III. EXPERIMENTAL RESULTS

The proposed wavelet packet audio codecs is realized as m-files and simulated using MAT LAB software. We adjust parameter such as structure of the disintegration tree, frame size, number of wavelet co-efficients, etc. The suitable set of parameter is selected to optimize between decoded audio quality, encoded bit rate, and computation complication.

To measure audio quality, we input mono audio-CD signal (705.6kbps) to the proposed encoder / decoder. The signal to noise ratio (SNR) is calculated and used as a beginning audio quality index. We later on play the decoded audio signal in the control room among 29 attendes to measure the hearing pleasure and bit rate, compared to the original audio-CD and audio signals decode by the mp3 and Ogg / Vorbis software decoder.

To measure noise presentation, the encoded data is sent through AWGN channel at various BERs and is reconstruct at the decoder. We execute the same for both mp3 and Ogg / Vorbis CODEC and use investigation satisfaction to compare noise resistance.

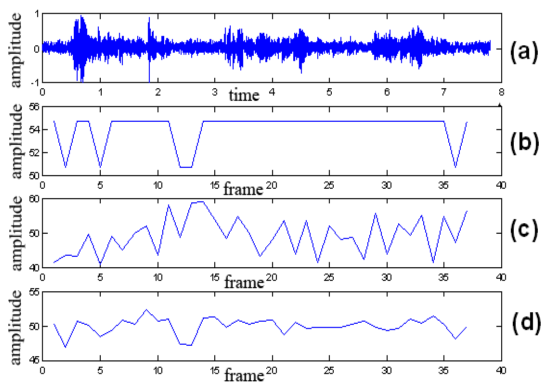
It is widely accepted that SNR cannot truly stand for audio quality under perceptual codecs and our results confirm this concept. As shown in Table 1 is the SNR association among wavelet packet audio CODECs at bit rate of 96 kbps, mp3 and Ogg / Vorbis at bit rate of 64 kbps measured through MAT LAB simulation. Low SNR yield low audio value but high SNR may or may not signify high audio quality. therefore, hearing satisfaction is used as an alternative. We select ten audio-CD samples 8 songs and 2 human voice and input them to the encoder and try to reduce the encoding bit rate. The lowest feasible bit rate is 80 kbps which provide comparable audio quality to that of 64 kbps mp3 and Ogg / Vorbis CODECs. Experiment perform at NECTEC concludes that more than 66% of the attendees cannot between the original audio CD sample and the decoded wavelet packet audio sample.

SNR Comparison of the proposed CODEC and other schemes

Music samples	wavelet	Mp3	Ogg/vorbis
Alone.wav	21.65	22.9	21
Morning.wav	20.05	18.93	21.97
piano	36.81	25.87	26.65
News.man.wav	26.31	24.45	21

Noise presentation of the considered audio CODECs is summarize in Table 2. visibly, Mp3 and Ogg / Vorbis are planned for storage of which low encoded bit rate is great consequence. The Mp3 decoder can stand BER of less than 10^{-6} which is not easily achieved in wireless statement. The Ogg / Vorbis decoder is even inferior, it cannot decode without frame drop at BER of 10^{-7}. On the added hand, the wavelet-packet decoder can stand BER up to 10^{-5} with suitable audio quality. At BER of 10^{-4} or more, it still can decode the audio signal without frame drops but one may attend to sound like water dropping every once in a while. This better noise resistance clearly benefits from BCH coding in the bit distribution field.

Noise Performance Comparison Testing



IV. CONCLUSION AND FUTURE WORK

This paper present wavelet packet exchange method used to program high quality audio signal for wireless communication. now, the

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encoded bit rate of 80 kbps, description included, yields obvious audio CD quality. This technique is potentially less calculation compared to the everywhere Mp3 and Ogg / Vorbis which give similar audio quality at 64 kbps. therefore, the wavelet packet audio codec should easily be realized in hardware and meet the real time prerequisite of high quality audio communication. In combination with high noise resistance, we believe that the proposed technique is suitable for wireless application such as wireless micro phone, wireless speaker, and so on. Last but not least, it is design and developed by Thai investigate group which we can use with no royalty fee

We at current implement the codec in hardware using FPGA and expect the prototype to be prepared in 2Q05. The result should confirm its real time performance and all other claims made here. We are also working towards lower bit rate (64kbps) wavelet packet audio codec that would gain more thrust because it can match the current PCM bit rate at much enhanced audio quality. obviously, the proposed scheme takes less than half of that requisite by the Mp3, 37962 against 81920!

V. ACKNOWLEDGEMENTS

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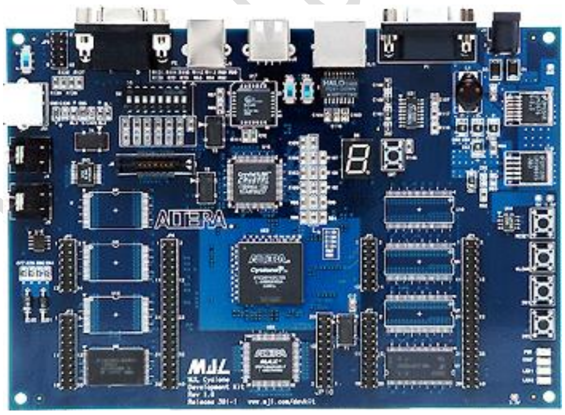

An FPGA Prototype of the CODEC

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