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Audio Compression using a Modified Vector Quantization algorithm for Mastering Applications

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Abstract—**Audio data compression is used to reduce the transmission bandwidth and storage requirements of audio data. It is the second stage in the audio mastering process with audio equalization being the first stage. Compression algorithms such as BSAC, MP3 and AAC are used as standards in this paper. The challenge faced in audio compression is compressing the signal at low bit rates. The previous algorithms which work well at low bit rates cannot be dominant at higher bit rates and vice-versa. This paper proposes an altered form of vector quantization algorithm which produces a scalable bit stream which has a number of fine layers of audio fidelity. This modified form of the vector quantization algorithm is used to generate a perceptually audio coder which is scalable and uses the quantization and encoding stages which are responsible for the psychoacoustic and arithmetical terminations that are actually detached as practically all the data detached during the prediction phases at the encoder side is supplemented towards the audio signal at decoder stage. Therefore, clearly the quantization phase which is modified to produce a bit stream which is scalable. This modified algorithm works well at both lower and higher bit rates. Subjective evaluations were done by audio professionals using the MUSHRA test and the mean normalized scores at various bit rates was noted and compared with the previous algorithms.**

Keywords—**vector quantization; scalable; perceptual coder; audio mastering; bit stream**

I. INTRODUCTION

AUDIO compression is classified as lossy and lossless, wherein the lossy compression detriments eminence for movement by eliminating the unimportant or some of the perceptible data. The Lossy compression remains perfect for space restraint and data communication is a matter of concern. Subject to the circumstances, the lossy compression could attain more compression ratio that demands to portable gadgets that prioritizes compactness than anything else. Nevertheless, the persistent development of storage capacity and price permits for the audio files which are stored to be huger. Lossless audio compression does not reject any information and the fidelity of recording is well-maintained. Numerous lossless audio codecs [1] are available in the recent times and each has their own advantages compared to other. For example: ALAC, FLAC and Wavpac.

Lossless compression functions by sensing as well as removing numerical redundancy from an audio signal so as to lessen bit-rate. This approach is generally attained using linear estimate of redundancy elimination and entropy for coding estimate. This technique is utilized by the lossless coders with differences in algorithms in both entropy as well as predictor coder.

While the MPEG layer 4 audio coder generates exceptionally scalable values of bit streams by means of BSAC (Bit Scaled Arithmetic Coding), and its perceptual fidelity at lower bit rates is deprived. In the contrary, the non - scalable vector quantization commonly called as VQ achieves well at lower bit rates. Vector quantization confirms the finest conceivable perceptual eminence at lower bit rates. The proposed research is an altered form of vector quantization algorithm which produces a scalable bit stream which has a number of fine layers of audio fidelity.

II. RELATED WORK

A. Lossless and Lossy audio compression

A novel technique for compressing audio without any loss in information by separating the phase and amplitude components and then encoding the individual components into a suitable pattern was presented by Mondal [2]. Techniques like Burrows–Wheeler transform and Huffman coding were used for encoding the components. This technique finds a better ratio of compression and less encoding/decoding ratio. A form of lossless audio compression which adopts pre-processing was developed by Huang [3]. The pre-processing is used to smoothly flatten the spectrum envelope. The compressor also utilizes a coder that adopts the scalable property. This is an IEEE compressor and found to be better than the MPEG layer 4 coding and the advanced audio coding. The limitation of the literature is that the compression does not work well at lower bit rates and also works only with single channel audio.

B. Quantization based Compression

Streaming of audio with high quality is challenging as it involves compression. Scalability plays an important role in streaming of compressed audio. A novel combined channel and source coder which delivers packet-loss recovery and constant

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bitrate scalability was presented by Sandler and Black [4]. This research seemed to be suiting good for audio streaming over future generation broadband networks.

An audio coder that produced a bit-stream and has encoded by adding one of the layers of fidelity was proposed by Srivastsan Kandadai and Charles D Creusere [5]. The proposed encoder quantized each critical band individually. The best possible perceptual quality with low bit rates was developed using this algorithm. A novel method to integrate psychoacoustic models into an adaptive wavelet packet system was presented by Pramila and Leah [6] to attain compression of high- at around 45 kb/s. This method resulted in developing an algorithm that was helpful in transfer of quality audio for streaming and also in storage applications.

C. MPEG Audio Coding

MPEG is one of the common standards for audio and video compression for eliminating the noise from the transmitted signals that comes through the satellite. It also makes easy to store and send data through internet with good quality [7-8]. The MPEG coder, compression is done without making conventions about the nature of source but it adventures the perceptual boundaries of the human aural structure.

D. DCT based Compression

An audio signal compression was proposed by the application of compressive sampling (CS) and discrete cosine transform (DCT). In this process, DCT is used as a pre-processor to characterize a signal in frequency domain along with CS characterization for gain of the signal [9]. The quantizer discarded the coefficients with moderately lesser amplitude with no audio distortion in a reconstructed signal. In multimedia transmissions, the media appearance will remain erroneous when the audio synchronization fails. The transform coding methodology considered the signal quality lessening and impact of noise as a function of the nested compression [10].

E. LPC based Compression

A new proposal of stereo decorrelation and optimal predictor techniques for lossless compression was completed by Florin [11], estimating around 1.5% greater compression than previous algorithms. The algorithms were implemented in C++ as OFR. The slow convergence problem and removal of spectral tilt of LMS algorithm was resolved by the low-order RLS predictor cascading prior to the LMS predictor for an audio signal [12].

High quality and multichannel audio have developed more relevant in various fields. The IEEE 1857.2 standard was reviewed and analysed by Teddy [13]. Linear Predictive Coding was utilized as the predictor. By increasing the predictor order doesn't improve the overall compressibility of the predictor possibly for the reason of the effect on the pre-processing output performance whereby entropy flatness performance decreases with the predictor output.

A Compressed Sensing [14-16] application towards audio signals has analyzed its audio perceptual quality through PEAQ. They proposed a lossy Audio CS system that used the encoder. For decoding, l_1 minimization was done and inverse transform to obtain time domain signal. Compression based on Trellis-Coded Quantization and Entropy-constrained polar quantization was marked with great significance in this research [17-18].

Finally, Compression is done for better sound quality. This quality should be good in all the speakers, from portable to car stereo. Generally, sound is heard good in hi-fi speakers but not in the portable ones. Therefore, a good amount of compression without any loss should be done so that the stereo sound is clear and good in all kinds of speakers. This process is called as Mastering. Most of the research mentioned is applied to mono signals only. An audio algorithms architecture for stereo portable devices was proposed based on a spatializer in the context of small and very close loudspeakers that has a poor spatial sound image [19]. The results showed that the spatializer was proficient of enhancing the spatial impression.

From the literature it is very clear that the limitations of the previous algorithms are that they were not able to compress the stereo signals in most cases. This paper proposes a modified version of the vector quantization algorithm considering the property of scalability [20] to perform compression on stereo audio as well as it can perform well at both lower and higher bit rates. Thus, it is useful in the mastering process where sound is clear in all kinds of speakers.

III. TRANSFORM FIELD WEIGHTED INTERWEAVED VECTOR QUANTIZATION

The prime conception of Vector Quantization (VQ) algorithm is analogous to the scalar quantization technique but is prolonged to numerous facets. The values are represented within a limited interval in a 1D space using a reconstruction quantity in the scalar quantization technique In VQ algorithm, the *n*-component *code vector* signifies the vectors which are positioned inside the region in an h - dimensional space. Consequently, a *codebook* is constructed with the collection of these code vectors for the vector quantizer.

Transform field Weighted Interweaved-VQ (TWIN-VQ) is a methodology designed for the compression of audio signals and have been modified to be efficient at lower bit rates compared with the previous algorithms. Fig 1 and Fig 2 displays blocks describing the encoder and decoder of the TWIN-VQ for 1 sequential frame of audio input signal.

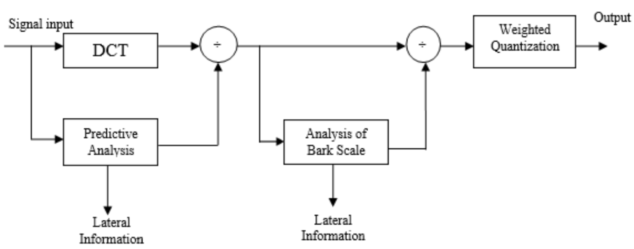


Fig.1. Encoder Block of Vector Quantization

The audio input is initially converted into a frequency domain by means of a 512- point Discrete Fourier Transform, and simultaneously Linear Predictive Coding (LPC) spectrum is well computed straight following the input signal. The LPC spectrum band produces a fairly irregular depiction of a spectral envelope for modified DCT coefficients, and then splits the coefficients so as to smooth the LPC spectrum roughly. This is correspondent for the execution of the LPC filtering and deduction within the time domain to attain an enduring signal. As a consequence, the normalization of a modified DCT spectrum was performed by means of the envelope of bark scale. The Bark-scale regularization processes primarily to enhance the quantization of the signals with more of harmonic structures and well-defined linear prediction coding coefficients along with the boosted envelope as from the subsequent smoothing process are quantized in scalar and conveyed to next stage, decoder to be the lateral information.

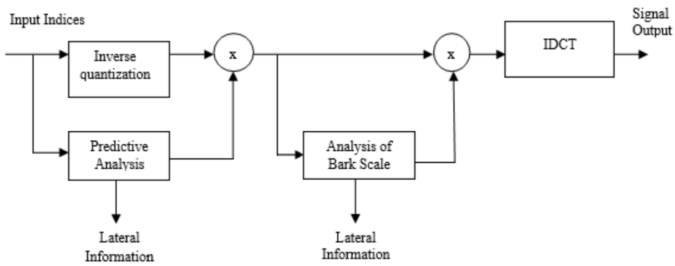


Fig.2. Decoder Block of Vector Quantization

The decoder just reverses the encoder phase. The smoothly flattened spectrum is rebuilt from vector quantization indices. The recreated bark scale coefficients, as well as the LPC coefficient will increase the spectrum which produces an estimate of the valid spectrum band. The audio wave signal which is in the time domain is now replicated by smearing the inverse discrete cosine transform to the rebuilt DCT coefficients. This research modifies the VQ algorithm to generate a perceptual audio coder which is scalable. The quantization along with the encoding phase is responsible for the psychoacoustic and arithmetical terminations which are actually detached as virtually all the data detached as a result of the prediction phases in the encoder end is supplemented for the audio signal during the decoder stage. Therefore evidently, the quantization stage is modified to produce bit- stream which is scalable.

A. *Discrete Cosine Transform.*

Figures should always be cited in consecutive numerical order. (Figure 3) Parts in a figure can be identified by a, b, c etc. and cited as Fig. 3a, Fig. 3b, Fig. 3c. The improved version of the Discrete Cosine Transform (DCT) is created from type-IV discrete cosine transform (DCT-IV), by adding the *lapped* process: it is intended to be executed on successive blocks of relatively bigger dataset, wherein the succeeding blocks remain overlapped hence the latter half of the block overlaps with the initial part of the following block. The overlapping feature, along with the energy-compaction properties of the transform, makes the modified version particularly desirable for the compression of audio signal applications, as this facilitates in avoiding the artifacts halting from the boundaries of the block. Because of these improvements, the MDCT (Modified DCT) was exercised in most of the innovative lossy audio formats which includes Windows Media Audio, MP3, AC-3, Vorbis, ATRAC, Opus, AAC, LDAC and Cook.

B. *Linear Prediction Coder.*

Linear Prediction Coder (LPC) techniques are used generally in audio and speech signal processing to represent a spectral envelope of the speech signal in a digital compressed form, handling a data of the linear predictive prototype. This is one of the highly influential speech analysis tools, also the most valuable approaches for encoding speech with good quality at a lower bit rate then delivers very precise estimations of speech and audio parameters.

LPC begins using the postulation that, the speech signal is formed with a buzzer at the very end of the tube (voiced sounds), along with occasionally enhanced popping as well as hissing sounds (plosive as well as sibilants sound). Though evidently crude, the model is really a precise estimate of reality of the audio and speech production. The glottis which is the space between the vocal folds forms a buzz, which is illustrated by its loudness and pitch (intensity and frequency respectively). The vocal tract creates as a tube, that is depicted as resonances and it raises to formants, otherwise improved frequency bands in the produced sound. Pops as well as hisses were generated through the action of tongue, throat as well as lips through plosives and sibilants.

LPC analyses the audio and speech waves by approximating the formants, eliminating the effects from the speech signal and determining the frequency as well as the intensity of outstanding buzz. This course of elimination of the formants is known as inverse filtering; the residual signal following the subtraction of filtered modelled signal is known as residue. The residue signal and the formant data which represents the frequency and the intensity of the buzz can be conveyed or stored elsewhere. The LPC recreates the speech wave by the inverse process: i) use the residue and the buzz parameters which generate a source signal ii) create a filter representing the tube by using the formants iii) run the source by means of a filter ensuing the speech.

C. *Quantizer.*

The quantizer used in Transform Field Weighted Interweaved Vector Quantization is based on three key concepts:

1) Weighted vector quantization,
2) Interleaving
3) Dual-channel conjugate

1) *Weighted Vector Quantization*

A vector quantized correspondent of adaptive bit allocation is known as Weighted Vector Quantization (WVQ). The optimized bit allocation arrangement is regulated for a specified DCT coefficient (a group of arbitrary variables), based on the discrepancy of each arbitrary variable. The SNR enhancement or the coding values gained by means of WVQ is specified by

$$G = \frac{\sum_{i=1}^{k} \sigma_i^2}{k \prod_{i=1}^{k} (\sigma_i^2)^{\frac{1}{k}}} \tag{1}$$

where σ_i^2 is the variance of the ith variable. Optimum bit allocation is performed with assigning of the bits to the arbitrary variables proportionate to the variances. For the optimum bit allocation, grounded on the masking parameters of a human aural system, the variances were altered.

Henceforth, the WVQ structure could obtain a similar performance gain as the adaptive bit allocation using a weighted distortion measure. Considering x as the input flattened signal, y as codebook vector and z as the perceptual weight, the squared distortion is given by

$$d^2 = \sum_i w^2 (x_i - y_i)^2 \quad (2)$$

where i denotes the element of each vector.

Obviously, using (2) to enhance a vector quantizer efficiently smooths the quantization error, driving as a higher value for some essentials as well as lower one for others. An additional improvement by using the WVQ algorithm is that it works finer compared to the adaptive bit allocation technique as the side information from the Encoder end explaining the bit allocation gets misplaced due to the transmission errors occurred. In the proposed VQ outline, Bark-scale analysis, the LPC spectrum and the perceptually weighting coefficients help to attain the weighting vector.

2) Interweaving

Table tools in Microsoft Word are recommended for inserting a table. The whole group of the DCT coefficients sent towards the vector quantization phase formulates a huge vector set, naturally within the full range of 256-1024 elements. The full set of DCT coefficients being quantized as a single vector was impossible for the reason that mutually the high computation complication in the VQ index search with the struggle of attaining a decent pattern by means of an iterative process identical to LBG which converges only towards the locally optimum solutions. The proposed vector quantization system strongly affects these problems by separating the lengthier DCT coefficient into a small sub- vector so as to conform the geometric mean of the premium coefficients where each sub- vector relics the same. This is implemented with the decimation of the DCT spectrum through the numerous sub-vectors that are to be produced. For example, assume $[a1, a2, a3, a4 \ldots\ldots an\text{-}1, an]^T$ as an n-dimensional vector, interweaving as two sub-vectors that will produce vectors $[a1, a3 \ldots\ldots an\text{-}1]^T$ and $[a2; a4 \ldots\ldots\ldots an]^T$.

3) Dual-Channel Conjugate Quantization

The last quantization process of the proposed VQ algorithm was done with a method called Dual-Channel conjugate VQ. This method employs two dissimilar codebooks that are "conjugates" for each other. This term "conjugate" was first devised by Moria in his research to demonstrate the corresponding features of two codebooks. This Dual-Channel conjugate encoder chooses the codebook vector from every two codebooks, computes its average, and compares the average with the specified input vector by means of the perceptually weighted mean square noise measure.

IV. SCALABILITY ON STEREO AUDIO SIGNALS

To design the novel, scalable codec which can be used for stereo audio, transform field-VQ is used as the initial point as realized in natural audio coding of MPEG 4 and to integrate an estimation of primary training set from dual-channel conjugate VQ codebook is applied. The training set created were used to create a new set of VQ codebooks that function within the distinctive critical bands and that, subsequently, the scalable codec allows supporting perceptual layering.

The VQ techniques could be observed as a non-uniform histogram; a codebook vector which creates the bin centres as well as the codebook index likelihood with quantization bin frequency. The provided impression could be manipulated to integrate a training set with the known codebook so that this has values same as the prime training set that are sourced to enterprise a VQ codebook.

V. RESULTS AND DISCUSSIONS

Six stereo audio sequences which represent an extensive variation of musical genres are applied to assess the encoding system as depicted in the Table I. These audio samples have been sampled at a 48 kHz sampling frequency with a resolution of 16 bits for a sample. For testing and verification, firstly encoding of the sequences at bit-rate of 128 kb/s using the VQ encoder is done, and then the audio samples are reconstructed at 8, 16, 32, and 64 kb/s. At lower bitrate of 8 kb/s, it is found that just the first 12 critical bands are being decoded. Least raises of 6 bits per frame are being added in to the bit-stream later.

TABLE I
SEQUENCES IN SUBJECTIVE EVALUATION

Sequence	Length (sec)	Genre [a]
Guitar	20	Jazz
Piano	12	Classical
Trumpet	15	Swing
Duet	25	Test
Trio	20	Test
Flute	9	Waltz

For comparison, the audio samples using the various audio compression standards, specifically AAC, BSAC, MP3 and the proposed algorithm were generated. At the bit-rate of 128 Kb/s, the proposed algorithm is compared only with the MP3 standard since it is the only algorithm that operates well at higher bitrates. Similarly, at 8 Kb/s, the proposed algorithm is compared with the VQ algorithm since it operates well at lower bit rates. Likewise, BSAC for 16 Kb/s and AAC for 64 Kb/s. Subjective evaluation was done using 5 audio professionals. The original signal and two compressed signals (one of them is the proposed algorithm) were given to the audio professionals in different order (the first one is the original) and the order was not told to them. The professionals were told to give a score of five to the second and third signal on the basis of which is close to the original signal. Evaluation was done using the following subjective grades as shown in the table II.

TABLE II
SUBJECTIVE EVALUATION GRADES

Subjective Grade	Nature of Signal [a]
5	Very Good
4	Good, but not Distorting
3	Slightly Distorting
2	Distorting
1	Very Distorting

After the tests, a normalized mean was calculated for each evaluation set for six different musical instruments. The following figures depicts the mean normalized scores at various bit rates. It is very evident from Fig 5a, Fig 5b, Fig 5c and Fig 5d that the proposed algorithm performs very well at low bitrates and reasonably well at higher bit rates.

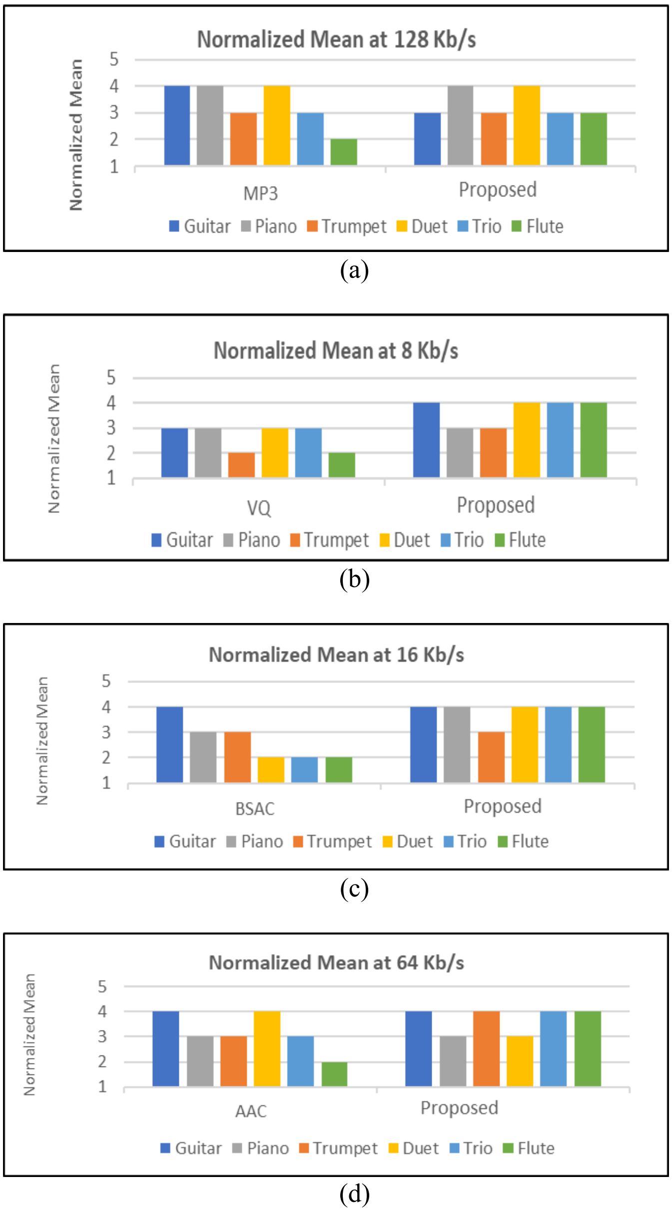
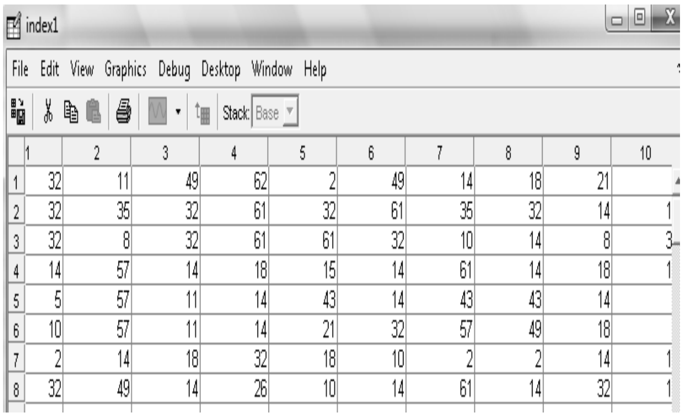


Fig.5. Mean Scores (a) MP3 vs Proposed (b) VQ vs Proposed (c) BSAC vs Proposed (d) AAC vs Proposed

The scalable algorithm accepts mono as well as stereo audio which is required for mastering applications. The encoder output is the side information which is given as input to the decoder. The side information is stored as a mat file. The side information parameters are inde1, index 2, sign 1 and sign 2. Index are the code book vectors and Sign represents the sign on vector as shown in Fig 6a and 6b. This side information is given as input to the decoder and the compressed audio signal was obtained. The decoded stereo output representation is shown in Fig7.


index1
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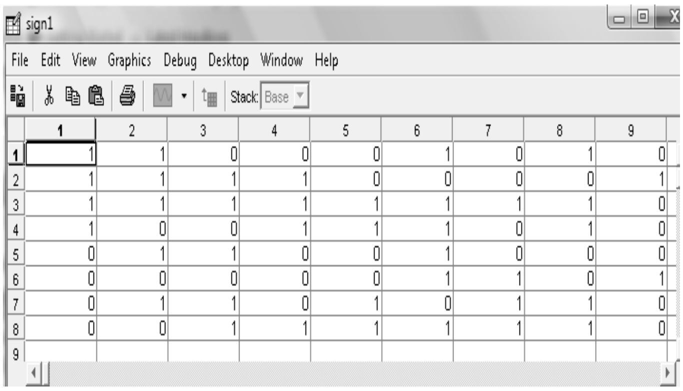
Stack: Base

	1	2	3	4	5	6	7	8	9	10
1	32	11	49	62	2	49	14	18	21	
2	32	35	32	61	32	61	35	32	14	1
3	32	8	32	61	61	32	10	14	8	3
4	14	57	14	18	15	14	61	14	18	1
5	5	57	11	14	43	14	43	43	14	
6	10	57	11	14	21	32	57	49	18	
7	2	14	18	32	18	10	2	2	14	1
8	32	49	14	26	10	14	61	14	32	1

(a)


sign1
File Edit View Graphics Debug Desktop Window Help
Stack: Base

	1	2	3	4	5	6	7	8	9
1	1	1	0	0	0	1	0	1	0
2	1	1	1	1	0	0	0	0	1
3	1	1	1	1	1	1	1	1	0
4	1	0	0	1	1	1	0	1	0
5	0	1	1	0	0	1	0	0	0
6	0	0	0	0	0	1	1	0	1
7	0	1	1	0	1	0	1	1	0
8	0	0	1	1	1	1	1	1	0
9									

(b)

Fig. 6. (a) Code Book Vectors (b) Sign on Vectors

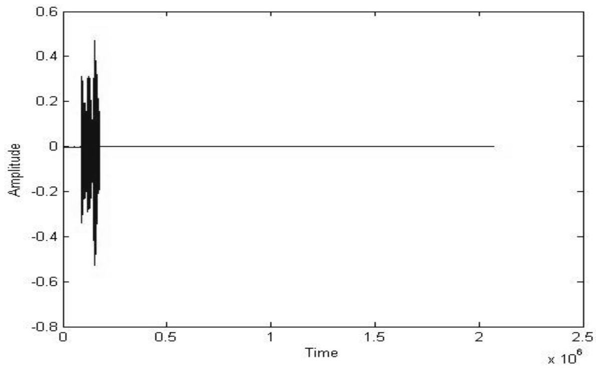


Fig7. Decoded Stereo Output

CONCLUSION

The audio signals were compressed using the weighted interleaved vector quantization method. Results were compared with the conventional compression algorithms for a stereo audio. The quantization and encoding phase were responsible for the psychoacoustic and arithmetical terminations which are actually detached because virtually all of the data detached by the prediction phases in the encoder side is supplemented to the audio signal in the decoder stage. Therefore, it was clearly the quantization stage which was modified to produce a bit stream which is scalable. The proposed algorithm is very efficient at 8 kb/s and quite reasonable at high bit rates. Deep learning approaches will be applied as future research.

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