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Audio Compression using DCT & CS

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***Abstract*—This paper describes the technique to apply DCT and CS techniques to the compression of audio signals. We present a study on compressed sensing of real, non-sparse, audio signals. With the help of spectral analysis and properties of the DCT, we can treat audio signals as sparse signals in the frequency domain. CS has been traditionally used to compress certain sparse images. DCT used as a signal preprocessor in order to obtain sparse representation in the frequency domain, we show that the subsequent application of CS represent our signals with less information than the well-known sampling theorem. Compressed sampling is an attractive compression scheme due to its universality and lack of complexity. This means that our results could be the basis for a new compression method for audio and speech signals.**

Key Words: Audio signal, Compressive sampling, DCT, Sparse signal reconstruction.

I. INTRODUCTION

Internet facility has become medium for file sharing. If size of the file is large then large time are consumed for downloading and large space is also required for its storage. For avoiding this condition we used compression.

The Shannon/Nyquist sampling theorem specifies that to avoid losing information when capturing a signal, one must sample at least two times faster than the signal bandwidth. In many applications, including digital image and video cameras, the Nyquist rate is so high that too many samples result, making compression a necessity prior to storage or transmission. In other applications, including imaging systems (medical scanners and radars) and high-speed analog- to-digital converters, increasing the sampling rate is very expensive.

There are different signal processing techniques are used for compression of audio signals. Signal processing techniques are DCT & CS. Compress sampling is the new framework for sampling and compressing certain signals. In CS, the band limited model (i.e. the Nyquist sampling theorem) is replaced by a sparse model, assuming that a signal can be efficiently represented using only a few significant coefficients in some transform domain.

This is the new method to capture and represent compressible signals at a rate significantly below the Nyquist rate. This method, called compressive sensing, employs non adaptive linear projections that preserve the structure of the signal; the signal is then reconstructed from these projections using an optimization process

CS requires that the signal is very sparse in some basis—in the sense that it is a linear combination of a small number of the basic functions—in order to correctly reconstruct the original signal. However, the CS measurements made are usually not dependent on the basis used in reconstruction, and thus the measurement process is universal as it does not need to change as different types of signals are sensed. In particular, it is still unknown how to construct a sparse audio signal, especially when CS relies on two principles: sparsity, and incoherence. Sparcity is pertains to the signal of interest and incoherence is pertains to the sensing modality.

For the problem of making a sparse representation of an audio signal, we introduce the DCT which is at present, the most widely used transform for image and video compression systems. Its popularity is due mainly to the fact that it achieves a good data compaction, because it concentrates the information content in relatively few coefficients. This means that we can obtain a compressed version of an audio signal by first obtaining a sparse representation in the frequency domain, and later processing the result with a CS algorithm.

The remaining part of this paper is organized as follows. In the next section, Section 2, we discussed about system overview. Then, in Section 3, we discussed about block diagram of audio compression system. Finally, we provide simulation result for system before concluding on our work in Section 5.

II. RESEARCH METHODOLOGIES

Figure.1 shows the basic diagram of Audio compression using different transform technique.

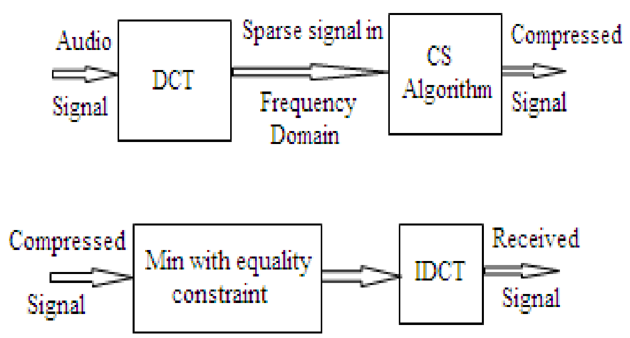

Fig. 1: System Overview of Audio Compression Using DCT and CS.

2.1 Compressive sampling

The central results state that a sparse vector $x^0 \in R^N$ be recovered from a small number of linear measurements $h = Ax^0 \in R^N, K \ll N or b = Ax_0 + e$ when e is the measurement noise by solving a convex program.

Consider a length N, real valued signal x and suppose that the basis ψ provides a K sparse representation of x. In terms of matrix notation, we have $x = \psi f$ in which f can be well approximated using only $K << N$ nonzero entries and ψ is called as the sparse basis matrix. The CS theory states that such a signal x can be reconstructed by taking only $M = O(K\log N)$ linear, non adaptive measurements as follows

$$y = \phi x = \phi\psi f \cdots (1)$$

Where y represents an $M \times 1$ sampled vector and Φ is an $M \times N$ measurement matrix that is incoherent with ψ i.e., the maximum magnitude of the element in $\Phi\Psi$ is small. Finally, with this information we decide to recover the signal by ℓ_1 norm. When f is sufficiently sparse, the recovery via ℓ_1 minimization is probably exact.

2.2 Properties of DCT

Decorrelation - The main advantage of signal transformation is the removal of redundancy between neighboring values. This leads to uncorrelated transform coefficients which can be encoded independently. Energy Compaction - Efficacy of a transformation scheme can be directly gauged by its ability to pack input data into as few coefficients as possible. This allows quantizes to discard coefficients with relatively small amplitudes without introducing visual distortion in the reconstructed image. DCT exhibits excellent energy compaction for highly correlated signals.

FFT and DWT are other alternatives for getting sparse signal but we prefer DCT. If we use FFT or DWT for getting sparse signal then this representation has real and complex parts, which result in a difficult reconstruction due to the phase angle changes

III. BLOCK DIAGRAM OF AUDIO COMPRESSION SYSTEM

Figure.2 shows that Block Diagram of Audio Compression using DCT and CS. In this section, we introduce our proposed techniques applied to an audio signal, and describe the technique for representing it in a sparse way. We then analyze its application to a compressive sampling algorithm.

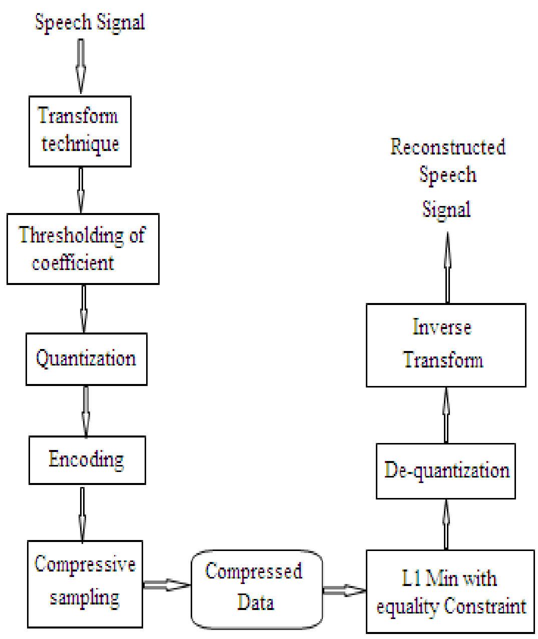

Fig. 2: Block Diagram of Audio Compression Using DCT and CS.

We used a special case of the FFT called the DCT. As mentioned above, one of the properties is that it attempts to de-correlate the data. After de-correlation, each transform coefficient can be encoded independently without losing compression efficiency.

IV. SIMULATION RESULTS

There are number of panel are running together concurrently. At transmitter side we have used DCT block to obtain sparse audio signal. Hence, in order to recover the original audio signal, sparse audio signal is then given to IDCT block. There are three input parameter are required for the audio processing enter the name of wav file, block size, compression factor. Figure.3 shows that startup GUI. This GUI is designed in MATLAB.

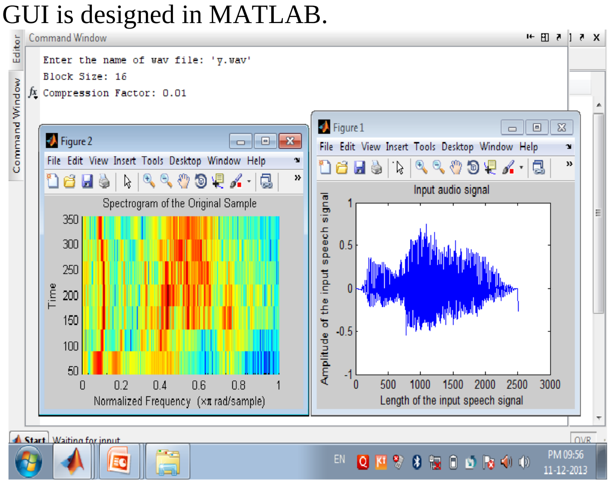

Fig.3 Startup GUI

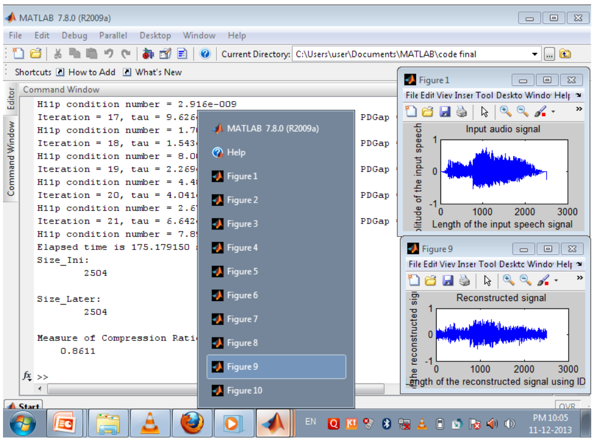

Fig.4 Different Signal

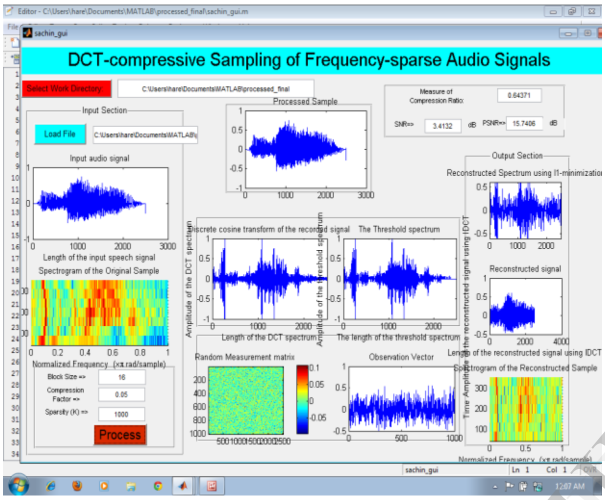

Fig.5 DCT-Compressive Sampling of Frequency-Sparse Audio Signals

Figure.5 shows DCT-Compressive Sampling of Frequency-sparse Audio Signals. It includes representation of different type of signal. Different Tables gives different readings. We are also plot different graphs for different ratios.

Table.1: Measure of Compression Ratio for Various Values of Block Size

Sr. No	Block Size	Measure of Compression Ratio
1	8	0.6877
2	16	0.64371
3	32	0.60878
4	64	0.58672

Table.2: Ratio for Various Values of Block Size

SR. No.	Block Size	SNR in dB	PSNR in dB
1	16	3.1055	15.4329
2	64	3.4062	15.7336
3	128	3.5541	15.7331
4	512	4.0212	16.0433

Table.3: Ratio for Various Values of Sparsity

SR. No.	Sparsity	SNR in dB	PSNR in dB
1	128	-0.56777	11.6113
2	300	-0.07818	12.1008
3	600	1.6737	13.8527
4	800	2.3328	14.5118

Table.4: Measure of Compression Ratio for Various Values of Compression Factor

SR. No.	Compression Factor	Measure of Compression Ratio
1	0.01	0.86133
2	0.03	0.68516
3	0.05	0.55898
4	0.08	0.4375

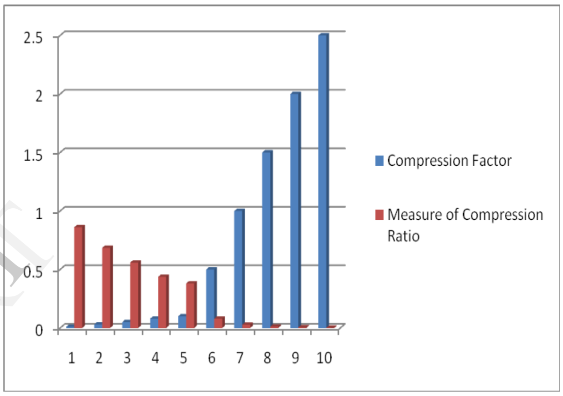

Fig.6 Graph of Compression Ratio for Various Values of Compression

Table.5: Ratio for Various Values of Compression Factor

SR. No.	Compression Factor	SNR in dB	PSNR in dB
1	0.01	3.395	15.7545
2	0.1	3.7526	15.9674
3	0.5	3.6443	16.7103
4	1	3.9149	17.0527

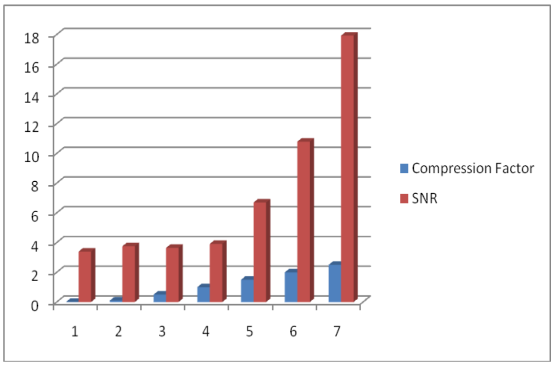

Fig.7: Graph of SNR for Various Values of Compression Factor

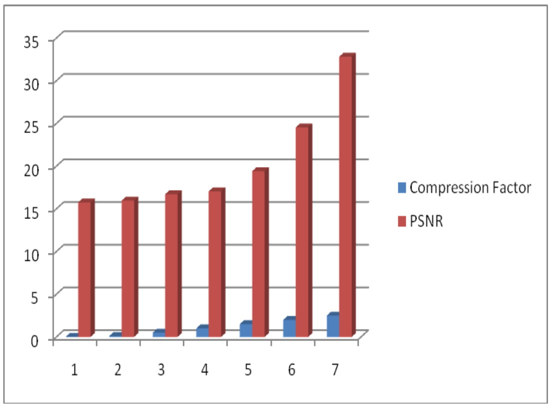

Fig.8: Graph of PSNR for Various Values of Compression Factor

V. CONCLUSION

Audio compression using DCT and CS was designed and implemented. It was tested with different values. There are different applications such as Audio Conferencing, Broadcast Gateway, iTunes, Computers, Embedded Systems, and You Tube etc. This study represents a DCT speech signal representation has the ability to pack input data into as few coefficients as possible. This allows quantizes to discard coefficients with relatively small amplitudes without introducing audio distortion in the reconstructed signal. Although the compressive sampling technique is used primarily for compression sample images, we can achieve reasonable results due to the preprocessing of the audio signal.

This technique can achieves a significant reduction in number of samples required to represent certain audio Signal and it reduces required number of bytes for encoding. There are some drawbacks in our implementation such as huge gap between CS Theory and application to audio signal and more time computation for signal processing on large audio files.
To reduce the amount of time required for the entire signal processes, hardware implementation using FPGA or CPLD. Further improvements are possible with advanced coding techniques like Wavelet or DWT.

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