Multi-Scale Domain Classification Based Heart Sound Compression

Ravi M. Potdar, Manoj K. Kowar, Amit Biswas, Mayur Amtey

***Abstract*—In recent days, fractal compression has gained a wide popularity due to its inherent features and efficiency in compressing data. In the present communication, fractal compression technique has been applied on heart sound signals for effective compression. Fractal heart sound coding based on the representation of a heart sound signal (1D or vector) by a contractive transform, on the sound data, for which the fixed point (reconstructed heart sound) is close to the original heart sound. The work is intended to provide an approach on this process by introducing the idea of multi-scale Domain pool classification using Variance Fractal Dimension (VFD) based on complexity of the heart sound data. A pre-processing analysis of the heart sound data by VFD to identify the complexity of each sound data samples block for classification has been undertaken. The performance result of the present work has focused in terms of good fidelity signal reconstruction versus encoding time and amount of compression.**

***Index Terms*— Phonocardiogram, Fractal Compression, Variance Fractal Dimension, Domain Classification**

I. INTRODUCTION

The heart is considered to be an analogous of electromechanical system, and as such its health can be characterized by both its electrical behavior (reflected in ECG) and its mechanical functioning. The mechanical functioning of heart generates sounds. Such sounds can be detected by using stethoscope, echocardiography, catheterization, or computerized analysis of the sounds emitted by the heart. Normal hearts generate at least two heart sounds – the first heart sound (S1) and the second heart sound (S2). S_1 and S_2 are produced due to the closure of the mitral and aortic valves at the beginning and end of the systolic process [1] respectively [2]. Depending upon an individual's health, additional S_3 and S_4 sounds may occur (Fig. 1). S_3 is produced due to the filling of the ventricle in the early stage of diastole. If blood enters in a relatively "non-compliant" ventricle late in diastole, it generates S_4. The four heart sounds are illustrated in Figure 1. The presence of an S_3, for example, is a strong indication of Congestive Heart Failure (CHF) [3]. Historically, heart sounds have been observed through auscultation of the heart wherein the cardiologist listens to the heart through a stethoscope.

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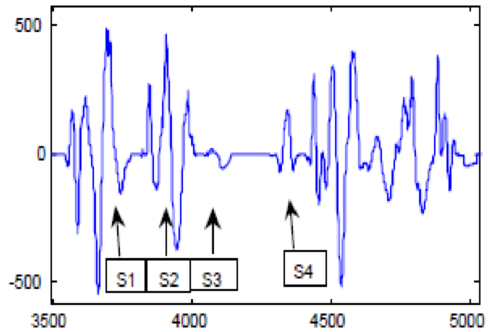
Forming a diagnosis based on sounds heard through the stethoscope is subjective, varies from person to person and also depends on
a cardiologist's past experience. In recent years, there has been an effort automate the heart sound diagnosis operation in order to remove subjectivity and to make heart sound auscultation more quantitative

Fig. 1 Heart Sound Signals with S_1, S_2, S_3 and S_4



Heart sounds analysis can provide lots of information about heart condition whether it is normal or abnormal. Heart sounds signals are time-varying signals where they exhibit some degree of non-stationary. Compression of the heart sounds (or phonocardiogram) is very convenient to reduce bandwidth in tele-diagnosis systems that aid the physician in the evaluation of the cardiovascular state. Data compression plays an important role in computational practices especially in signal analysis and transmission due to heavy memory requirement for storing and retrieving such complex data. The goal of signal compression is to reduce storage space and to save transmission complexities. The availability of the sound file and the variance fractal dimension technique helped in the development of the software and thus allows the biomedical environment to switch from old system to new system. The fractal compression is mostly based on fractal system's ability to approximate discontinuous functions, where audio signals exhibits greater smoothness.

The mechanical function of heart, blood flow and valve movements produce the heart sound during contraction and relaxation phases of the heart. The heart sound signal is important clinical information in the diagnostic process of heart malfunctions [4]. The objective of this work is to compress the sound file (heart sound) using the technique of variance fractal dimension [5, 6] and the major motive of this work is to develop a software through which heart sound data can be compressed efficiently to make it suitable for transmission over the networks. The techniques for recording and analysis have been changing as new electronic devices and signal processing techniques have become available.

Fractal compression can be considered as one of this new form [5]. A more detailed description of fractal encoding method, the notation used and a description of basic implementations can be found in [3]. Fractal dimension values indicate the complexity of a pattern, or the quantity of information embodied in a pattern in terms of morphology, entropy, spectra or variance. The VFD analysis does not create windowing artifacts in the Fourier sense, which is often introduced in fractal spectral analysis. Therefore, the VFD is an excellent tool for investigating the time series signals by calculating the variance fractal dimension trajectories. The main objective of this work is to develop a MATLAB program based on variance fractal dimension algorithm for the compression of heart sound and to facilitate the evaluation of result of simulation for parameters like PSNR, encoding time and compression ratio. Section 2 explains the mathematical foundation of the VFD coding for the proposed method. Section 3 demonstrates experimental results. Section 4 provides the conclusion.

II. ENCODING USING FD

The encoding process follows Fisher's conventional fractal encoding algorithm [7]. Fixed partition has been used based on the representation of heart sound data as a sample sequence in 1-D space. Simple classification has been applied to improve encoding efficiency.

The range and domain blocks in fractal audio coding is based on heart sound samples. The size of the block is the number of audio samples in the block. The domain block size has been restricted to twice of the range blocks, and each range block to contain ten samples. The range blocks are selected based on the fixed partition and the affine mapping. Then the resulting range blocks are compared with all potential domain blocks from the domain pool to find the optimal domain blocks achieving the minimum_{rms} distance through affine transforms. In the latter situation, we store the transform with the minimum_{rms} distance. Once a mapping w_i has been identified, the transform coefficients (i.e., scaling and offset) and the domain block locations have been stored in the output file. The final output file stores parameters of the map $W = \bigcup = Uw_i$ from the encoding process.

Four kinds of domain pools (D_1, D_2, D_3, D_4) have been allowed in the present work. VFD approach has been used to classify the domain blocks with a fixed spacing l=1/2.The idea is that it is more important to find a good domain-range fit for larger range blocks, because the encoding will require a fewer number of transforms [8], which also means a higher compression ratio.

The domain-range comparison step of fractal encoding is very computationally intensive. Different classification schemes have been invented to minimize the number of domain-range comparisons starting from Jacquin's original work on fractal image compression. The basic idea is to categorize the domain blocks under certain criteria before the encoding actually takes place. During the encoding, one range block is classified using the same scheme, and only needs to be compared with the domain blocks in the same category. By reducing the number of domain-range comparisons, the classification improves fractal encoding efficiency. VFD has been used here to subdivide the domain blocks into classes of complexity (Fig. 2). For fast VFD evaluation a simple and efficient algorithm has been used [9].

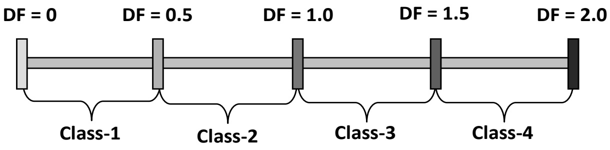


Fig. 2: Relating the Domain Pool and VFD with the Domain Class

For the present experimental needs, some user definable encoding parameters have been allowed in the algorithm. The parameters are based on the previous studies on fractal image encoding [2], which explores some main properties of fractal encoding. Such parameters are listed below:

a) e_{rms} : The maximum rms mean error for a domain-range mapping.
b) D_{pos} : The position of the domain block.
c) s_{max} : The maximum allowable scaling factor of an affine transform.

In the algorithm developed, e_{rms} is 0.5, s_{max} is usually 1 to guarantee absolute contractiveness. The parameters s_i and o_i which are not related with the algorithm are used to store scaling and offset coefficients of a transform.

III. PERFORMANCE EVALUATION

In order to test the proposed compression scheme and parameter settings, nine different heart sounds (PCG) signals in wave format from different references have been utilized. The wave files are Aortic Insufficiency, Aortic Stenosis, Coarctation of the Aorta, Ebsteins Disease, and Functional or innocent murmurs hum.

The compression level and fidelity of various heart sounds are normally specified by the measure called Peak Signal to Noise Ratio (PSNR) which represents a measure of the peak error and is expressed in decibels (dB).The higher the PSNR, the better the quality of the compressed or reconstructed sound files. Typical values for lossy compression of a signal are between 30 and 50 dB. For evaluating performance of the method both objective and subjective tests are conducted. In objective test, PSNR of signal after reconstruction is computed.

Another important measure is the compression ratio. General compression ratio common in the data compression field can be defined as

$$compression\ ratio = \frac{raw\ size}{compressed\ size} \quad (1)$$

Where raw size is the size of the original is file and compressed size is the size of the file after encoding. The file size is measured in bytes. Another measure as the number of transforms stored from the encoding process has also been considered. This measure generally depicts the performance of the fractal schemes on certain heart sound sequences. Less number of transforms means a higher compression ratio, or more patterns are identified in some big size blocks.

Sometimes the number of transforms gives more accurate information of the performance than the compression ratio does, because the latter has limited precision as a floating point number.

Finally, it is obvious that the comparison of this measure only makes sense to be applied to the same size audio sequences (i.e., containing the same number of samples).

The domain-range mapping error tolerance e_{rms} is a very critical parameter dominating the compression ratio as well as the encoding quality in our fractal scheme. Intuitively, the bigger the e_{rms} value, the lower the PSNR that can be achieved, because more mappings can be accepted for the range blocks. On the other hand, a smaller e_{rms} value gives a better encoding quality. In this experiment e_{rms} value is taken as 0.5 for fractal coding of heart sound data. The bit allocation for scaling factor s_i=8 bits and offset factor, o_i=8 bits has been fixed.

In order to compare the performances of the developed model, parameters of the encoding process, viz., Encoding time and the Peak Signal to Noise Ratio (PSNR) and the Compression Ratio on heart sound data have been depicted in the Table 1 as well as the comparative graphical analysis have been shown in fig. 3 to 7.

Table -1 Various Heart Sounds with their Encoding Time and PSNR

Name of heart sound	Number of samples	Encoding Time (sec)	PSNR (dB)
Aortic insufficiency	29670	86.063	40.6533
Aortic stenosis	28509	89.563	45.7341
Coarctation of the aorta	28810	105.532	48.8783
Ebsteins disease	29369	99.375	51.1592
Normal heart sound	30745	117.578	51.2744

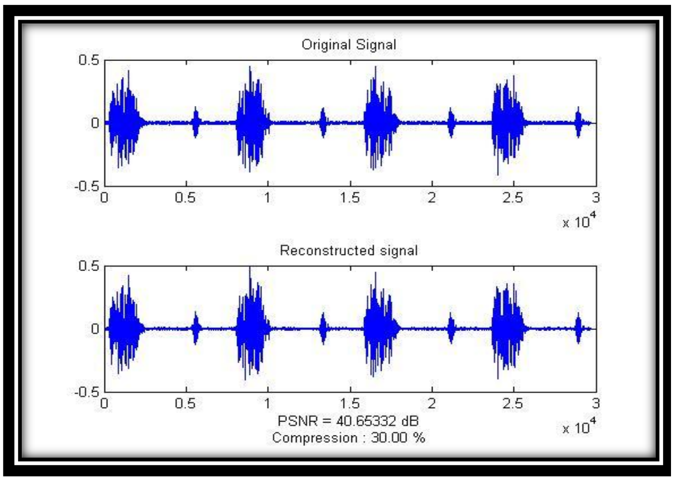


Fig.3 Aortic Insufficiency

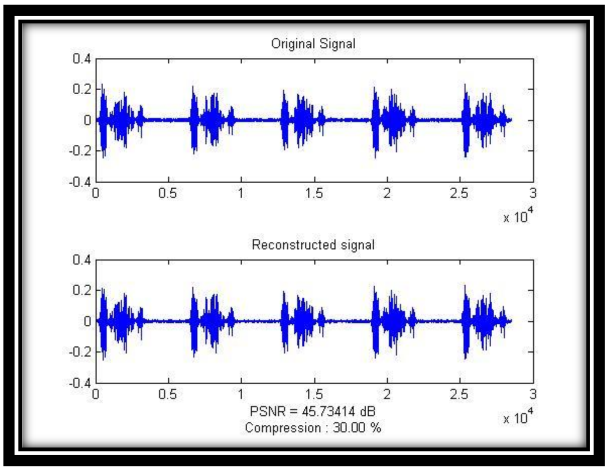


Fig.4 Aortic Valve Stenosis

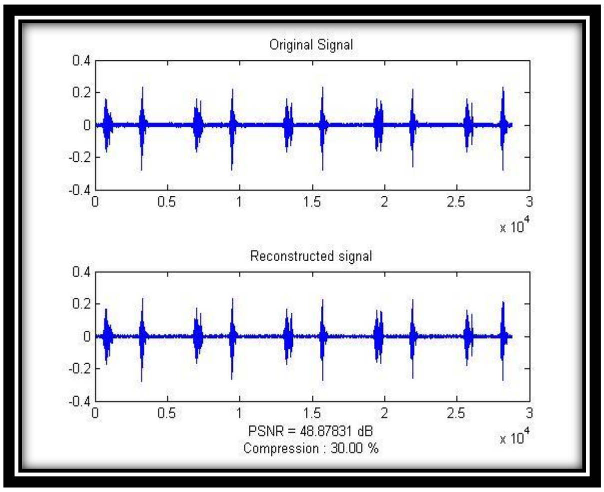


Fig 5 Coarctation of Aorta

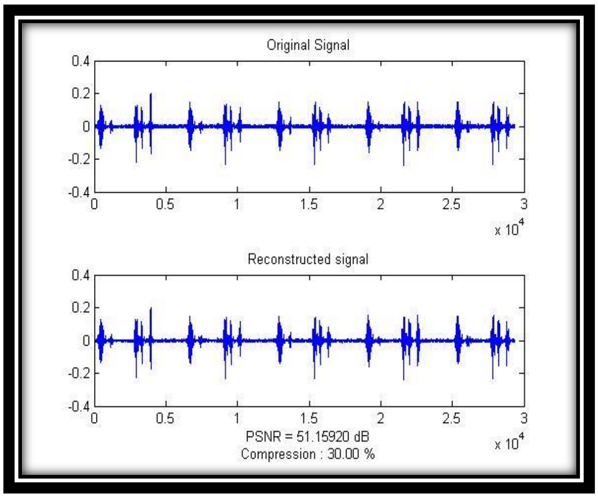


Fig.6 Ebsteins disease or Ebsteins anomaly

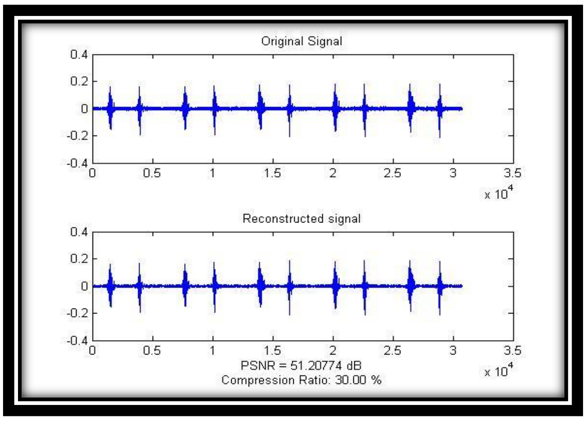


Fig.7 Normal Heart Sound

IV. CONCLUSION

The results have shown that the fractal compression schemes are efficient to provide a high compression ratio. Present research on fractal coding is an attempt to extend fractal coding to heart sound data and to explore how the fractal system performs with heart sound signals. The outcomes of the present research work will definitely contribute to future fractal heart sound coding research. Through a large amount of studies in fractal image coding, it has been demonstrated that fractal coding can be nicely integrated into the framework of conventional compression techniques such as vector quantization and transform methods. Built upon the success of applying wavelet in image compression, fractal wavelet compressions of heart sounds signals were introduced [10]. This idea has also been practiced in fractal heart sound coding too. This direction may also lead to a potential success on applying fractal audio coding in practice. In terms of improvement, preprocessing may be promising. A preprocessing step can be used to break the continuity in heart sound sequences, which may provide a way of compensating for the shortcomings of fractal systems. Specific to heart sound a proper classification shall help the compression scheme greatly. Fractal heart sound coding can clearly take the advantage of the classification to improve the mapping quality as well as searching efficiency. However, the classification of heart sound is probably a more general topic to be concerned which may be addressed from other different perspectives.

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