New Generation Mic Recording System

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ABSTRACT

— Multichannel audio systems have become ubiquitous with the advent of new and effective audio compression, multimedia storage, and delivery method. Voice recording and reproduction is an electrical or mechanical inscription and re-creation of sound waves, the technical performance of audio systems is necessary to choose the microphone system that best meets operational requirements. Client A system is attached one mic which is record sound file1 .Similarly Client B system is attached second mic which is record sound file 2 .Server will store backup Client A and Client B system file 1 and File 2 in one Folder. This paper outlines procedures developed to non-traditionally measure the frequency response of audio systems using recorded data files. Record Sound using two different mic with two different system and Store in Server.

Keyword:- *Human Computer Interaction, Audio processing, Gesture Recognition, Embedded Systems, Computer Vision, Morphology*

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1. INTRODUCTION

Voice recording and reproduction is an electrical or mechanical inscription and re-creation of sound waves, such as spoken voice, singing, instrumental music, or sound effects. The two main classes of sound recording technology are analog recording and digital recording.There are a variety of systems used to record and transmit audio signals. Each system has its own advantages and disadvantages, typically trading of cost, size, and power consumption for audio fidelity. Client A system is attached one mic which is record sound file1 .Similarly Client B system is attached second mic which is record sound file 2 .Server will store backup Client A and Client B system file 1 and File 2 in one Folder. In order to select the right system for a given situation, it is desirable to use unbiased indicators of performance to determine which system has the best audio quality for a given set of circumstances. Therefore, it is necessary to develop a simple method to quantify and compare systems that operate in vastly different ways. One of the best ways of quantifying audio quality is to measure a systems frequency response gauge of a systems output spectrum given a specific input. The frequency response is used to graphically display how the recording system changes or distorts the input audio signal. The most desirable result is that the plot of the frequency response is a nearly flat line, meaning that at any given frequency the systems response to the input signal is constant. Historically, an audio systems frequency response was determined by placing the microphone in an anechoic chamber and sweeping through the audio spectrum with a well defined noise source, i.e. a loudspeaker. The system output would then be connected to a network analyzer which would display its response across the desired audio spectrum. This technique is not available to digital systems that record audio for later review there is no connection to allow audio to be fed into an analyzer. The lack of a real time analog audio stream prevents the use of traditional network analyzers and associated procedures. This complicates extracting the frequency response of the system and must be taken into account when developing a test methodology

2. BLOCK DIAGRAM

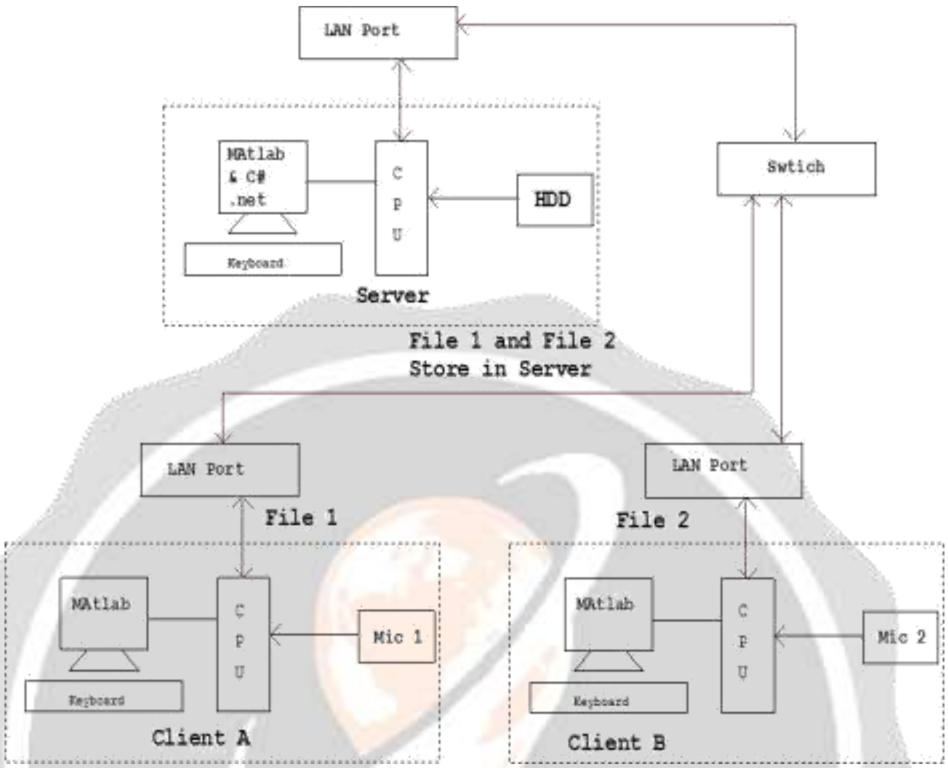

Fig no 2.1: Hardware System Design

2.1 Audio compression

The idea of audio compression is to encode audio data to take up less storage space and less bandwidth for transmission. To meet this goal different methods for compression have been designed. Just like every other digital data compression, it is possible to classify them into two categories: lossless compression and lossy compression.

2.2 Lossless compression

Lossless compression in audio is usually performed by waveform coding techniques. These coders attempt to copy the actual shape of the analog signal, quantizing each sample using different types of quantization. These techniques attempt to approximate the waveform, and, if a large enough bit rate is available they get arbitrary close to it. A popular waveform coding technique, that is considered uncompressed audio format, is the pulse code modulation (PCM), which is used by the Compact Disc Digital Audio (or simply CD). The quality of CD audio signals is referred to as a standard for hi-fidelity. CD audio signals are sampled at 44.1 kHz and quantized using 16 bits/sample Pulse Code Modulation (PCM) resulting in a very high bit rate of 705 kbps. As mentioned before, human perception of sound is affected by SNR, because adding noise to a signal is not as noticeable if the signal energy is large enough. When digitalize an audio signal, ideally SNR could to be constant for al quantization levels, which requires a step size proportional to the signal value. This kind of quantization can be done using a logarithmic compander (compressor-expander). Using this technique it is possible to reduce the dynamic range of the signal, thus increasing the coding efficiency, by using fewer bits. The two most common standards are the -law and the A-law, widely used in telephony. Other lossless techniques have been used to compress audio signals, mainly by finding redundancy and removing it or by optimizing the quantization process. Among those techniques it is possible to find Adaptative PCM and Differential quantization. Other lossless techniques such as Huffman coding and LZW have been directly applied to audio compression without obtaining significant compression ratio.

2.3 Lossy compression

Opposed to lossless compression, lossy compression reduces perceptual redundancy; i.e. sounds which are considered perceptually irrelevant are coded with decreased accuracy or not coded at all. In order to do this, it is better to have scalar frequency domains coders, because the perceptual effects of masking can be more easily implemented in frequency domain by using subband coding. Using the properties of the auditory system we can eliminate frequencies that cannot be perceived by the human ear, i.e. frequencies that are too low or too high are eliminated, as well as soft sounds that are drowned out by loud sounds. In order to determine what information in an audio signal is perceptual irrelevant, most lossy compression algorithms use transforms such as the Modified Discrete Cosine Transform (MDCT) to convert time domain sampled waveforms into a frequency domain. Once transformed into the frequency domain, frequencies component can be digitally allocated according to how audible they are (i.e. the number of bits can be determined by the SNR). Audibility of spectral components is determined by first calculating a masking threshold, below which it is estimated that sounds will be beyond the limits of human perception Briefly, the modified discrete cosine transform (MDCT) is a Fourier-related transform with the additional property of being lapped. It is designed to be performed on consecutive blocks of a larger data set, where subsequent blocks are overlapped so that the last half of one block coincides with the first half of the next block. This overlapping, in addition to the energy-compaction qualities of the DCT, makes the MDCT especially attractive for signal compression applications, since it helps to avoid artifacts stemming from the block boundaries.
cable or power it with an AC-to-DC adapter or battery to get started. The Uno diers from all preceding boards in that it does not use the FTDI USB-to-serial driver chip. Instead, it features the Atmega8U2 programmed as a USB-to-serial converter. Uno means one in Italian and is named to mark the upcoming release of Arduino 1.0. The Uno and version 1.0 will be the reference versions of Arduino, moving forward. The Uno is the latest in a series of USB Arduino boards, and the reference model for the Arduino platform; for a comparison with previous versionArduino UNO is used to control the operations of RFID reader at the doors as well as the transmitters and receivers. Arduino UNO is a microcontroller board which is based on the ATMEGA 328P [5]. It has 14 digital Input /Output pins, 6 Analog Input/ Output pins, a 16 MHz quartz crystal, a USB connection, a power jack, an ICSP header and a reset button. It also includes: Flash Memory 32 KB (ATmega328) of which 0.5 KB used by boot loader, SRAM 2 KB (ATmega328) EEPROM 1 KB (ATmega328).

2.4 Speech compression

Speech signals has unique properties that differ from a general audio/music signals. First, speech is a signal that is more structured and band-limited around 4kHz. These two facts can be exploited through different models and approaches and at the end, make it easier to compress. Many speech compression techniques have been efficiently applied. Today, applications of speech compression (and coding) involve real time processing in mobile satellite communications, cellular telephony, internet telephony, audio for videophones or video teleconferencing systems, among others. Other applications include also storage and synthesis systems used, for example, in voice mail systems, voice memo wristwatches, voice logging recorders and interactive PC software.

Basically speech coders can be classified into two categories: waveform coders and analysis by synthesis encoders. The first was explained before and are not very used for speech compression, because they do not provide considerable low bit rates. They are mostly focused to broadband audio signals. On the other hand, encoders use an entirely different approach to speech coding, known as parametric coding, or analysis by synthesis coding where no attempt is made at reproducing the exact speech waveform at the receiver, but to create perceptually equivalent to the signal. These systems provide much lower data rates by using a functional model of the human speaking mechanism at the receiver. Among those, perhaps one of the most popular techniques is called Linear Predictive Coding (LPC) encoder

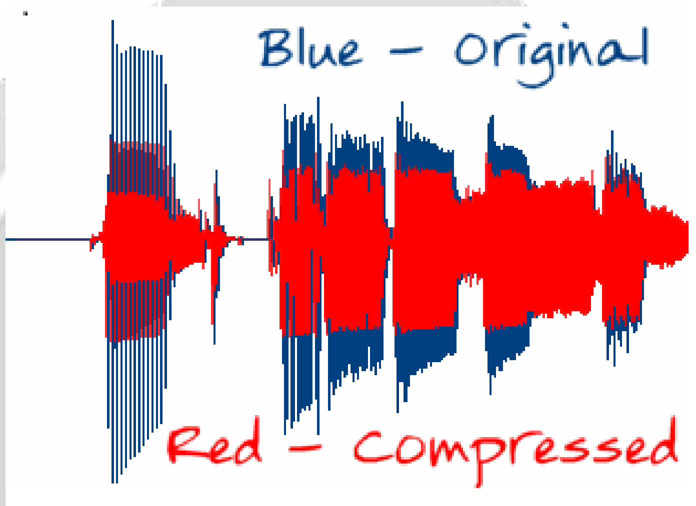

Fig no 2.2: Audio Compression

3. FLOWCHART

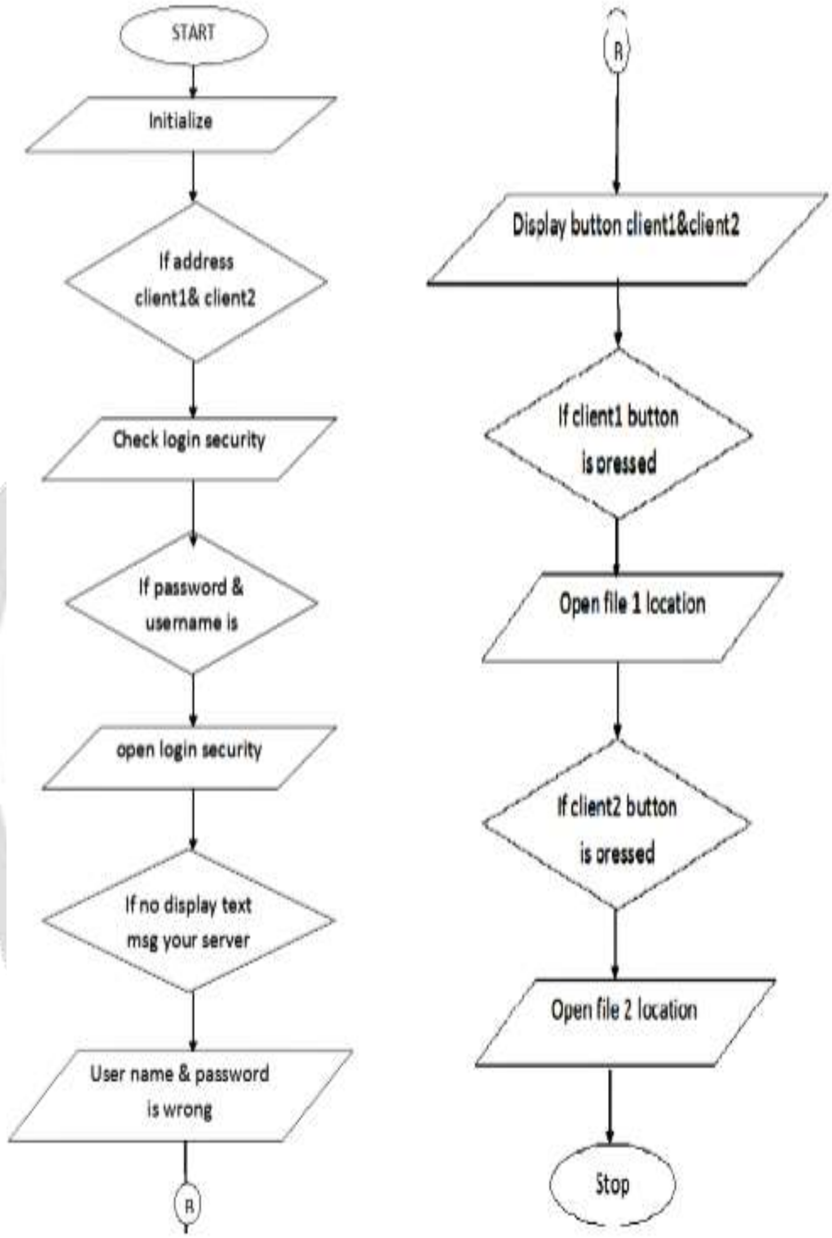

Fig no 3.1: Flow Chart of Server

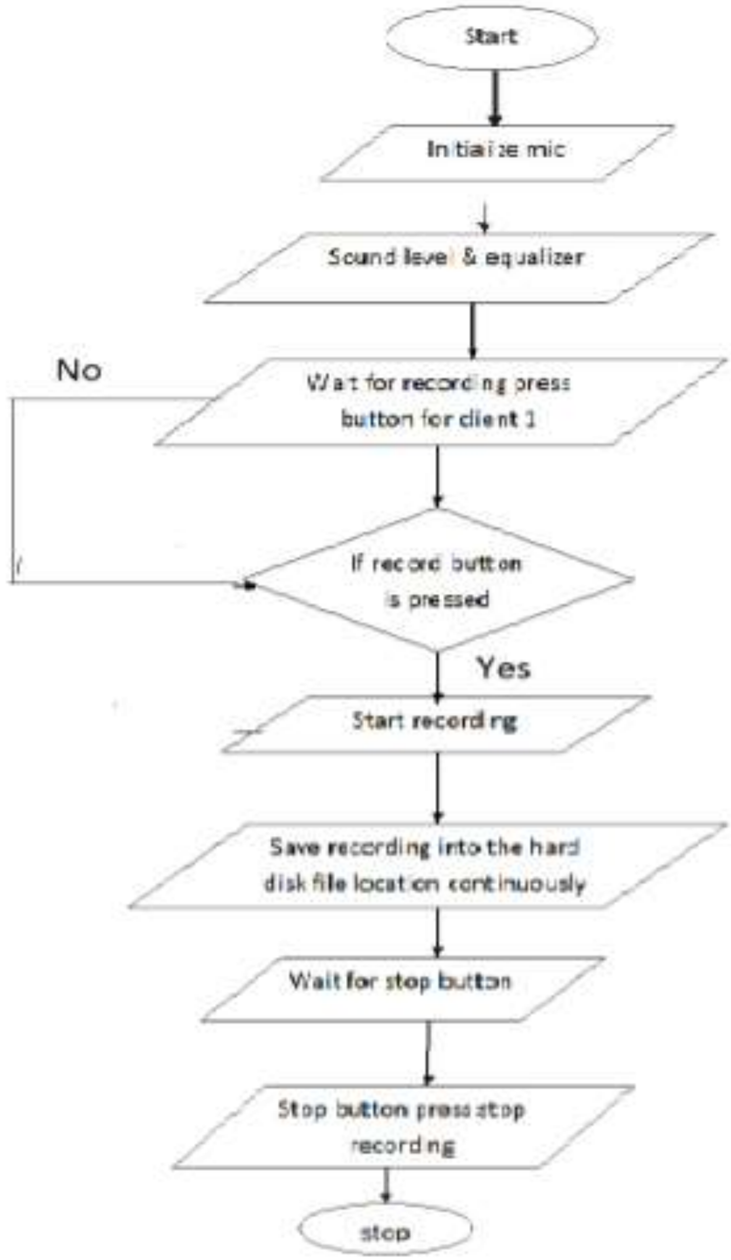

Fig no 3.2: Flow Chart of Client A

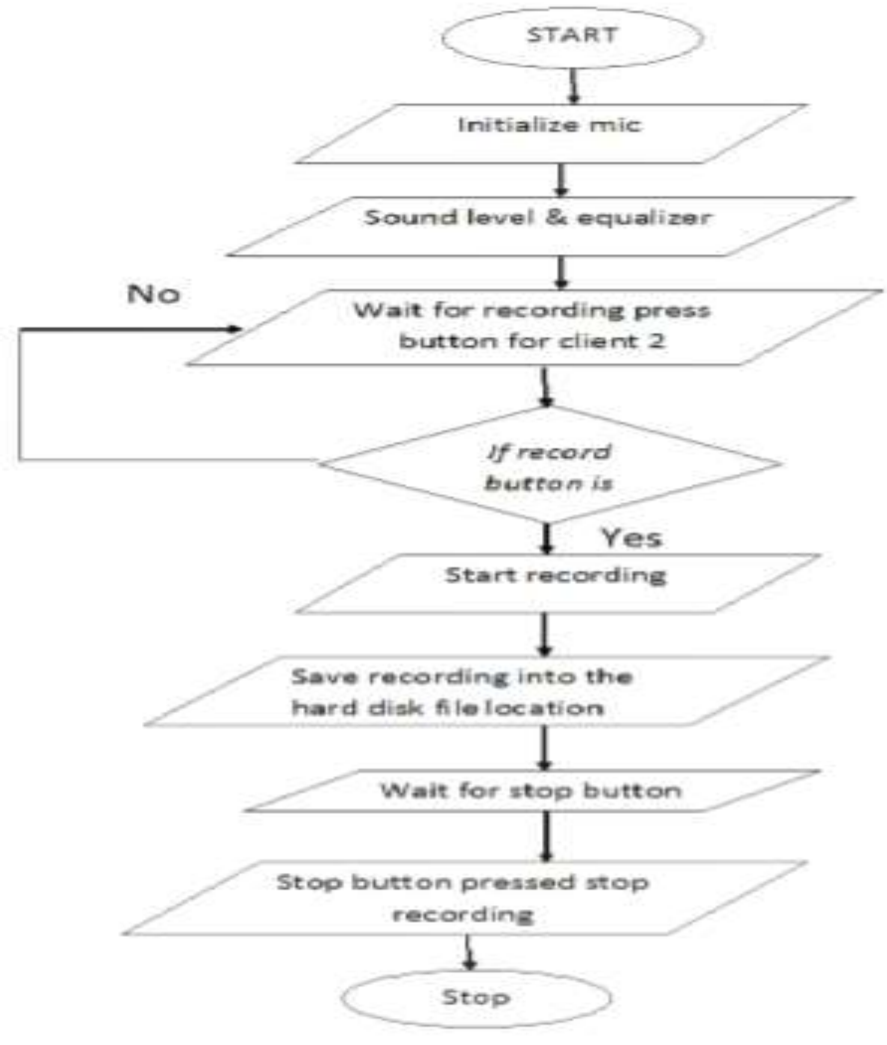

Fig no 3.3: Flow Chart of Client B

The main goal of the algorithm presented in project is to compress high quality audio maintaining transparent quality at low bit rates. Several steps are considered to achieve this goal:

• Design a wavelet representation for audio signals.

• Design a psychoacoustic model to perform perceptual coding and adapt it to the wavelet representation.

• Reduce the number of the non-zero coefficients of the wavelet representation and perform quantization over those coefficients.

• Perform extra compression to reduce redundancy over that representation

• Transmit or store the steam of data. Decode and reconstruct. Evaluate the quality of the compressed signal.

Exemplar based In painting technique is used for in painting of text regions, which takes structure synthesis and texture synthesis together. The in painting is done in such a manner, that it fills the damaged region or holes in an image, with surrounding color and texture. The algorithm is based on patch based fling procedure. First find target region using mask image and then find boundary of target region. For all the boundary points it defined patch and find the priority of these patches. It starts fling the target region from the highest priority patch by finding the best match patch. This procedure is repeated until entire target region is in painted. The algorithm automatically generates mask image without user interaction that contains only text regions to be in painted.

4. CONCLUSION

In this paper scheme is proposed for finding out the frequency of a recorded signal such as speech signal. Using data acquisition toolbox in MATLAB, the scheme can be further modified for finding the frequency of a Real Time Signal

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